A Unified Mobility Management Architecture for Interworked Heterogeneous Mobile Networks

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Declaration

This is to certify that to the best of my knowledge and belief, the work presented in this thesis is original, unless specified otherwise, and that no part of this material has been submitted, either in full or in part, at this or any other institution.

Kumudu S. Munasinghe

September 2008

to
Savithri, Amma, and Thaththa
with love

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Abstract

The buzzword of this decade has been convergence: the convergence of telecommunications, Internet, entertainment, and information technologies for the seamless provisioning of multimedia services across different network types. Thus the future Next Generation Mobile Network (NGMN) can be envisioned as a group of co-existing heterogeneous mobile data networking technologies sharing a common Internet Protocol (IP) based backbone. In such all-IP based heterogeneous networking environments, ongoing sessions from roaming users are subjected to frequent vertical handoffs across network boundaries.

Therefore, ensuring uninterrupted service continuity during session handoffs requires successful mobility and session management mechanisms to be implemented in these participating access networks. Therefore, it is essential for a common interworking framework to be in place for ensuring seamless service continuity over dissimilar networks to enable a potential user to freely roam from one network to another. For the best of our knowledge, the need for a suitable unified mobility and session management framework for the NGMN has not been successfully addressed as yet. This can be seen as the primary motivation of this research.

Therefore, the key objectives of this thesis can be stated as:

- To propose a mobility-aware novel architecture for interworking between heterogeneous mobile data networks
- To propose a framework for facilitating unified real-time session management (inclusive of session establishment and seamless session handoff) across these different networks.

In order to achieve the above goals, an interworking architecture is designed by incorporating the IP Multimedia Subsystem (IMS) as the coupling mediator between dissipate mobile data networking technologies. Subsequently, two different mobility management frameworks are proposed and implemented over the initial interworking architectural design. The first mobility management framework is fully handled by the IMS at the Application Layer. This framework is primarily dependant on the IMS's default session management protocol, which is the Session Initiation Protocol (SIP). The second framework is a combined

method based on SIP and the Mobile IP (MIP) protocols, which is essentially operated at the Network Layer.

An analytical model is derived for evaluating the proposed scheme for analyzing the network Quality of Service (QoS) metrics and measures involved in session mobility management for the proposed mobility management frameworks. More precisely, these analyzed QoS metrics include vertical handoff delay, transient packet loss, jitter, and signaling overhead/cost. The results of the QoS analysis indicates that a MIP-SIP based mobility management framework performs better than its predecessor, the Pure-SIP based mobility management method. Also, the analysis results indicate that the QoS performances for the investigated parameters are within acceptable levels for real-time VoIP conversations. An OPNET based simulation platform is also used for modeling the proposed mobility management frameworks. All simulated scenarios prove to be capable of performing successful VoIP session handoffs between dissimilar networks whilst maintaining acceptable QoS levels.

Lastly, based on the findings, the contributions made by this thesis can be summarized as:

- The development of a novel framework for interworked heterogeneous mobile data networks in a NGMN environment.
- The final design conveniently enables 3G cellular technologies (such as the Universal Mobile Telecommunications Systems (UMTS) or Code Division Multiple Access 2000 (CDMA2000) type systems), Wireless Local Area Networking (WLAN) technologies, and Wireless Metropolitan Area Networking (WMAN) technologies (e.g., Broadband Wireless Access (BWA) systems such as WiMAX) to interwork under a common signaling platform.
- The introduction of a novel unified/centralized mobility and session management platform by exploiting the IMS as a universal coupling mediator for real-time session negotiation and management.
- This enables a roaming user to seamlessly handoff sessions between different heterogeneous networks.
- As secondary outcomes of this thesis, an analytical framework and an OPNET simulation framework are developed for analyzing vertical handoff performance. This OPNET simulation platform is suitable for commercial use.

Publications Arising from the Thesis

Book Chapters

[1] K. S. Munasinghe and A. Jamalipour, "Mobility Management QoS Measures Evaluation for Next Generation Mobile Networks," in Wireless Communications: CNIT Thyrrenian Symposium Series: Signals and Communications Technology, Ed. Pupolin Silvano, USA: Springer, pp. 265-280, 2008.

Journal Articles

[1] K. S. Munasinghe and A. Jamalipour, An Analytical Evaluation of Mobility Management in Integrated WLAN-UMTS Networks, *Elsevier International Journal on Computers and Electrical Engineering SI "Signal Processing and Communication Systems"*, (Early View).

[2] K. S. Munasinghe and A. Jamalipour, Interworked WiMAX-3G Cellular Networks: An Architecture for Mobility Management and Performance Evaluation, *IEEE Transactions on Wireless Communications*, (Early View).

[3] K. S. Munasinghe, M. Rubaiyat Kibria, and A. Jamalipour, Designing VoIP Session Management over Interworked WLAN-3G Cellular Networks, in *IEEE Wireless Communications*, vol. 15, no. 4, pp. 86-94, August 2008.

[4] K. S. Munasinghe and A. Jamalipour, Interworking of WLAN-UMTS Networks: An IMS-based Platform for Session Mobility, in *IEEE Communications*, vol. 46, no.9, pp. 184-191 , September 2008.

Conference Proceedings

[1] K. S. Munasinghe and A. Jamalipour, Evaluation of Session Handoffs in a Heterogeneous Mobile Network for Pareto based Call Arrivals, Accepted for publication in *Proceedings of the IEEE Wireless and Communications Conference (WCNC 2009)*, Budapest, Hungary, April 2009 (in press).

[2] K. S. Munasinghe and A. Jamalipour, Analysis of Vertical Session Handoff for Self-Similar Traffic in a Heterogeneous Mobile Data Network, in *Proceedings of the*

International Conference on Signal Processing and Communication Systems (ICSPCS 2008), Gold Coast, Australia, December 2008.

[3] K. S. Munasinghe and A. Jamalipour, Analysis of Signaling Cost for a Roaming User in a Heterogeneous Mobile Data Network, in *Proceedings of the IEEE Global Communications Conference (GLOBECOM 2008)*, New Orleans, USA, November 2008.

[4] K. S. Munasinghe and A. Jamalipour, An Architecture for Mobility Management in Interworked 3G Cellular and WiMAX Networks, in *Proceedings of the IEEE Wireless Telecommunications Symposium (WTS 2008)*, Pomona, California, USA, April 2008.

[5] K. S. Munasinghe and A. Jamalipour, Analytical Modeling of IMS based Interworking in Heterogeneous Mobile Data Networks, in *Proceedings of the International Conference on Signal Processing and Communication Systems (ICSPCS 2007)*, Gold Coast, Australia, December 2007.

[6] K. S. Munasinghe and A. Jamalipour, A Unified Mobility and Session Management Platform for Next Generation Mobile Networks, in *Proceedings of the IEEE Global Communications Conference (GLOBECOM 2007)*, Washington DC, USA, November 2007.

[7] K. S. Munasinghe and A. Jamalipour, Mobility Management QoS Measures Evaluation for Next Generation Mobile Networks, in *Proceedings of the Tyrrhenian Workshop on Digital Communications (TIWDC 2007)*, Naples, Italy, September 2007.

[8] K. S. Munasinghe and A. Jamalipour, A 3GPP-IMS based Approach for Converging Next Generation Mobile Data Networks, in *Proceedings of the IEEE International Conference on Communications (ICC 2007)*, Glasgow, Scotland, United Kingdom, June 2007.

[9] K. S. Munasinghe and A. Jamalipour, Interworking between WLAN and 3G Cellular Networks: An IMS based Architecture, in *Proceedings of the IEEE International Conference on Wireless Broadband and Ultra Wideband Communications (AusWireless 2006)*, Sydney, Australia, March 2006.

Technical Reports

[1] K. S. Munasinghe and A. Jamalipour, "WLAN-UMTS-CDMA2000 Interworking Architectures," *Technical Report for Singtel Optus Networks*, April 2007.

[2] K. S. Munasinghe and A. Jamalipour, "WLAN and 3G Cellular Network Interworking," *Technical Report for SingTel Optus Networks*, April 2006.

[3] K. S. Munasinghe and A. Jamalipour, "A Novel Architecture for WLAN and 3G Cellular Network Interworking," *Technical Report for SingTel Optus Networks*, November 2005.

[4] K. S. Munasinghe and A. Jamalipour, "Wireless LAN and Cellular Network Interworking Architectures: Open issues and Research Themes," *Technical Report for SingTel Optus Networks*, August 2005.

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Glossary of Acronyms and Abbreviations

3G	3^{rd} Generation
3GPP	3rd Generation Partnership Project
3GPP2	3rd Generation Partnership Project 2
3PCC	Third-Party Call Control
4G	4^{th} Generation
AAA	Authentication, Authorization, and Accounting
AEN	Access Edge Node
ASN	Access Service Network
B2BUA	Back-to-Back User Agent
BEN	Border Edge Node
BS	Base Stations
BSS	Basic Service Set
BU	Binding Update
BWA	Broadband Wireless Access
CDMA	Code Division Multiple Access
CMR	Call-to-Mobility Ratio
CoA	Care of Address
CS	Circuit-Switched
CSCF	Call State Control Functions
CSMA/CA	Carrier Sense Multiple Access with Collision Avoidance
CSN	Connectivity Services Network
DFS	Dynamic Frequency Selection
DHCP	Domain Host Configuration Protocol
DSSS	Direct Sequence Spread Spectrum
ETSI	European Telecommunications Standards Institute
FA	Foreign Agent
FCC	Federal Communications Commission
FHSS	Frequency Hopping Spread Spectrum
FMIPv6	Fast Handover for Mobile IPv6

FTP	File Transfer Protocol
GGSN	Gateway GPRS Support Node
GIF	GPRS Interworking Function
GPRS	General Packet Radio Service
GSM	Global System for Mobile communications
HA	Home Agent
HiperMAN	High Performance Radio Metropolitan Area Network
HLR	Home Location Register
HSDPA	High-Speed Downlink Packet Access
HTTP	Hypertext Transfer Protocol
I-CSCF	Interrogating-CSCF
IEEE	Institute of Electrical and Electronic Engineers
IISA	Integrated InterSystem Architecture
IMS	IP Multimedia Subsystem
IMS-MGW	IMS Media Gateway
IMT-2000	International Mobile Telecommunications 2000
IP	Internet Protocol
ISM	Industrial, Scientific, and Medical
ITU	International Telecommunications Union
LLC	Logical Link Control
LoS	Line-of-Sight
MAC	Media Access Control
MAP	Mobility Anchor Point
MGCF	Media Gateway Control Function
MICS	Media Independent Command Service
MIES	Media Independent Event Service
MIH	Media Independent Handover
MIHF	Media Independent Handover Function
MIIS	Media Independent Information Service
MIMO	Multiple-Input Multiple-Output
MIP	Mobile IP

MMD	Multi Media Domain
MMS	Multimedia Message Service
MN	Mobile Node
MRFC	Media Resource Function Controller
MRFP	Media Resource Function Processor
MSS	Mobile Subscriber Station
MVNO	Mobile Virtual Network Operator
NAT	Network Address Translation
NGMN	Next Generation Mobile Network
NLOS	Non-Line-of-Sight
NWG	Network Working Group
OFDM	Orthogonal Frequency Division Multiplexing
OFDMA	Orthogonal Frequency Division Multiple Access
PCF	Packet Control Function
P-CSCF	Proxy-CSCF
PDCP	Packet Data Convergence Protocol
PDF	Probability Density Function
PDG	Packet Data Gateway
PDN	Packet Data Network
PDP	Packet Data Protocol
PDS	Packet Data Subsystem
PDSN	Packet Data Serving Node
PDU	Protocol Date Unit
PHY	Physical
PLMN	Public Land Mobile Network
PRACK	Provisional Response ACKnowledgement
PS	Packet-Switched
PSTN	Public Switched Telephony Network
QoS	Quality of Service
RA	Routing Area
RTCP	Real Time Control Protocol

RTP	Real Time Protocol
SC	Single Carrier
SCC	Seamless Connection Control
SCP	Service Control Point
S-CSCF	Serving-CSCF
SDP	Session Description Protocol
SGSN	Serving GPRS Support Node
SIP	Session Initiation Protocol
SLA	Service Level Agreements
SMC	Short Message Centre
SMS	Short Message Service
TCP	Transmission Control Protocol
TIA	Telecommunications Industry Standard
TMM	Transform Matching Method
TPC	Transmit Power Control
UAC	User Agent Client
UAS	User Agent Servers
UDP	User Datagram Protocol
UMTS	Universal Mobile Telecommunications System
UTRAN	UMTS Terrestrial Radio Access Network
VoIP	Voice over Internet Protocol
VGSN	Virtual GPRS Support Node
WAF	WLAN Adaptation Function
W-APN	WLAN Access Point Name
W-CDMA	Wideband Code Division Multiple Access
WIG	WLAN Interworking Gateway
WiMAX	Worldwide Interoperability for Microwave Access
WLAN	Wireless Local Area Networks
WMAN	Wireless Metropolitan Area Networks
WPAN	Wireless Personal Area Networks
WWAN	Wireless Wide Area Networks

Chapter 1

An Introduction to Interworking Heterogeneous Wireless Networks

The emergence of various radio access technologies and wireless data communication networks over the past decade has revolutionized the entire telecommunications industry. By and large, these wireless networking technologies can be categorized as Wireless Personal Area Networks (WPAN), Wireless Local Area Networks (WLAN), Wireless Metropolitan Area Networks (WMAN), and Wireless Wide Area Networks (WWAN), which have been widely accepted as a convenient alternative to the conventional wired networks. The increasing demand for ubiquitous high speed data access has resulted in widespread deployment of heterogeneous wireless networking domains. However, in order to ensure anywhere anytime wireless data access, roaming facilities between dissimilar networks must be in place, which is also the primary requirement of an NGMN. Hence, there is growing demand for efficient architectures and platforms for interworking heterogeneous wireless networks, which is the primary motivation of this research. Therefore the aim of this Chapter is to establish the objective and scope of this thesis. The remainder of this Chapter is organized as follows. Firstly, an overview on wireless data networks is presented. Secondly, the basic concepts of interworking and trends are introduced. Followed by this are the sections on the objectives, approach, and contributions of the thesis. The Chapter concludes by presenting an outline of the thesis, which briefly presents a summary of the forthcoming Chapters.

1.1 Wireless Data Networks: An Overview

The development of new wireless technologies and the increased user demand for ubiquitous high speed data access has given rise to the rapid deployment of wireless networks such as WWAN, WMAN, WLAN, and WPAN over the last decade. Despite the rapid growth of the above mentioned networks, physical characteristics such as underlying radio access technologies, data rates, geographical coverage, and mobility support of each of these technologies are highly diverse in nature. For example, a modern third generation (3G) cellular network (considered under a WWAN) is capable of providing high speed mobility and relatively large coverage, but has relatively lower data rates. On the other hand, a WLAN supports relatively high data rates, but relatively smaller area of coverage with limited mobility. Therefore, this section presents a brief overview about various wireless technologies comprised in a typical NGMN.

1.1.1 Wireless Local Area Network (WLAN)

The Institute of Electrical and Electronic Engineers (IEEE) formed the 802.11 Work Group in September 1990 with the objective of developing a standard for wireless LANs to operate on a low-power unlicensed frequency range. The selected frequency range was the Industrial, Scientific, and Medical (ISM) bands, either the 2.4 GHz band or the 5 GHz band, which was set aside by the Federal Communications Commission (FCC). As a result, the first IEEE's 802.11 standard was released in 1997 [1].

This standard addresses the Media Access Control (MAC) and Physical (PHY) standards separately. The original PHY standard provides data rates of 1-2 Mbps and three fundamentally different mechanisms of operation. They are namely: Infrared, 2.4 GHz Frequency Hopping Spread Spectrum (FHSS), and 2.4 GHz Direct Sequence Spread Spectrum (DSSS). The task assigned to the 802.11 MAC is to coordinate an access mechanism, which allows fair access to the medium. Since wireless stations do not have the capability of detecting collisions, the IEEE 802.11 WLANs have an access method that made every effort to avoid collisions, which is known as Carrier Sense Multiple Access with Collision Avoidance (CSMA/CA).

In 1999, the IEEE released a new PHY standard named the IEEE 802.11b [2]. This standard is capable of providing higher bit rates up to 11 Mbps using DSSS within the 2.4 GHz range. About the same time, IEEE released another PHY standard named IEEE 802.11a. It provides bit rates up to 54 Mbps and operates on the 5 GHz range [3]. However, instead of using DSSS as in the previous cases, IEEE 802.11a uses a new modulation method called Orthogonal Frequency Division Multiplexing (OFDM). A later contribution (June 2003) to the family of IEEE 802.11 PHY standards was the IEEE 802.11g standard, operating in the 2.4 GHz range and using OFDM [4]. This works in the 2.4 GHz band (like IEEE 802.11b), but uses the same

OFDM based transmission scheme as IEEE 802.11a and capable of providing maximum data rates up to 54 Mbps.

The IEEE 802.11n is a much recently proposed amendment to the IEEE 802.11 wireless networking standard to significantly improve network throughput over previous standards with a significant increase in the raw data rate from 54 Mbps to a maximum of 600 Mbps [5]. The IEEE 802.11n is built on previous IEEE 802.11 standards by adding Multiple-Input Multiple-Output (MIMO) and channel-bonding operation to the PHY layer, and frame aggregation to the MAC layer. Although the work on the IEEE 802.11n standard dates back to 2004, the draft is yet to be finalized in March 2009 with possible publication in December 2009 [6].

Another amendment added to the IEEE 802.11 is the standard for Spectrum and Transmit Power Management Extensions, which is known as IEEE 802.11h-2003 (or 802.11h) [7]. It solves problems like interference with satellites and radar using the same 5 GHz frequency band. The standard provides Dynamic Frequency Selection (DFS) and Transmit Power Control (TPC) to the 802.11a MAC [8]. It has been integrated into the full IEEE 802.11-2007 standard.

1.1.2 Wireless Metropolitan Area Network (WMAN)

A WMAN provides network access to buildings through exterior antennas communicating with central radio Base Stations (BS). Because wireless systems have the capacity to address broad geographic areas without the costly infrastructure development required in deploying cable links to individual sites, the technology may prove less expensive to deploy and may lead to more ubiquitous broadband access [9].

The IEEE 802.16 Working Group is the IEEE group working on WMANs, in particular the air interface for fixed broadband wireless access systems. The working group develops standards and recommended practices to support the development and deployment of fixed broadband wireless access systems. The first IEEE 802.16 standard was approved in December 2001. It delivered a standard for point to multipoint broadband wireless transmission in the 10-66 GHz band, with only a Line-of-Sight (LoS) capability. It uses a Single Carrier (SC) PHY standard [10].

The IEEE 802.16a was an amendment to the IEEE 802.16 and delivered a point to multipoint capability in the 2-11 GHz band [11]. For this to be of use, it also required a non-line-of-sight (NLOS) capability, and the PHY standard was therefore extended to include OFDM and Orthogonal Frequency Division Multiple Access (OFDMA). The IEEE 802.16a was ratified in January 2003 and was intended to provide "last mile" fixed broadband access. The IEEE 802.16c, a further amendment to the IEEE 802.16, delivered a system profile for the 10-66 GHz IEEE 802.16 standard [12].

In September 2003, a revision project called the IEEE 802.16d commenced with the aim to align the standard with aspects of the European Telecommunications Standards Institute's (ETSI) High Performance Radio Metropolitan Area Network (HiperMAN) standard [13]. This project concluded in 2004 with the release of IEEE 802.16-2004 which superseded the earlier IEEE 802.16 documents, including the a/b/c amendments [14]. The IEEE 802.16e-2005 is an amendment to the IEEE 802.16-2004 standard and is often referred to in shortened form as the IEEE 802.16e. This new amendment introduced support for mobility, amongst other things and called as Mobile Worldwide Interoperability for Microwave Access, which is better known as "Mobile WiMAX" [15].

1.1.3 Wireless Wide Area Network (WWAN)

A WWAN essentially comprises of an umbrella of 3G cellular networking technologies such as UMTS [16], General Packet Radio Service (GPRS) [17], CDMA2000 (i.e., CDMA2000 1xRTT [18], CDMA2000 EV-DO [19], and CDMA2000 EV-DV [20]).

Currently, the most common form of UMTS uses Wideband Code Division Multiple Access (W-CDMA) as the underlying air interface. It is standardized by the 3rd Generation Partnership Project (3GPP), and addresses the International Telecommunications Union's (ITU) International Mobile Telecommunications 2000 (IMT-2000) [21] requirements for 3G cellular radio systems. UMTS supports up to 14.0 Mbps data transfer rates (theoretically) with High-Speed Downlink Packet Access (HSDPA) [22], despite the fact that at the moment users in deployed networks can expect a transfer rates of up to 384 kbps for UMTS Release 99 handsets, and 7.2 Mbps for HSDPA handsets in the downlink connection.

The 3rd Generation Partnership Project 2's (3GPP2) CDMA2000 can be deployed in several phases. The first phase, CDMA2000 1x, supports an average of 144 kbps packet data in a mobile environment. The second release of 1x, called 1x-EV-DO supports data rates up to 2

Mbps on a dedicated data carrier. Finally, 1x-EV-DV (which probably will rarely be deployed) will support even higher peak rates, simultaneous voice and high-speed data, as well as improved QoS mechanisms.

1.2 Interworking Trends and Issues

As mentioned earlier, the future NGMN can essentially be seen as a group of overlapping heterogeneous mobile data networks that are interworked together [23],[24]. Therefore, the integration of these dissimilar technologies using a common framework can enable a potential user to freely roam from one network to another. Furthermore, seamless session handoff from one network to another will also become a possibility.

The main benefits of interworking can be summarized as:

- Ability for catering bandwidth demands for high performance applications such as multimedia, video, and teleconferencing,
- Provisioning of global mobility and service portability across heterogeneous networking boundaries,
- Realization of an all-IP based packet switched telecommunications system with converged voice and data capability,
- Realization of the future NGMN or 4G networking platform which supports ubiquitous data services and very high data rates across heterogeneous networks, and
- Provisioning of always best connectivity to the subscriber (in an NGMN).

In the recent years, much research has been done in the area of interworking between various wireless data networks such as WLAN, 3G Cellular (i.e., UMTS and CDMA2000), WiMAX [25]. By and large, these internetworking architectures can be categorized as tight coupling, loose coupling, and peer-to-peer networking (also referred as no-coupling) [26], [27]. The design concepts behind the categorization of these architectures can be summarized as follows. In the tight coupling architecture, the WLAN is directly connected to the UMTS core network. Thus the WLAN data traffic gets routed via the UMTS core network before reaching the external Packet Data Networks (PDNs). Therefore, UMTS mobility management techniques may be directly applied in this method. On the other hand, the loosely coupled architecture exchanges signaling between the WLAN and the UMTS core network while the data flows via independent IP based networks. There are also other variants to this internetworking architecture, which may require the data traffic to be routed via the UMTS core network [28], [29]. Since the data traffic is routed directly via an IP network this method may help avoiding a potential traffic bottleneck. Nevertheless, in this method, the handoffs are less efficient and therefore real-time session mobility may not always be guaranteed [27]. Lastly, the peer-to-peer coupling framework can be described as coupling the 3G cellular network and the WLAN as peers, which may also be argued to be a variant of the loose coupling architecture [26]. In this case, a higher layer mobility management protocol such as Mobile IP (MIP) could be used for provisioning mobility management [30]. However, due to known deficiencies of the MIP protocol itself (i.e., the issue of triangular routing, conflict with security frameworks of cellular systems, and so on), this may not be the best solution for frequently roaming users.

Despite these recent attempts, many open and unresolved issues still exist in this area. The first of which is the issue of session mobility across WLAN and UMTS networks. Efficient ways to provide/enable seamless continuity of service across WLAN and 3G cellular networks can be ranked as a top issue. Another important issue is to define a mechanism for data routing in heterogeneous networks. Additionally, matching the QoS requirements and service provisioning in such environments are other related issues. Therefore, there exists a need for the development of an architecture capable of overcoming these challenges, which motivates this work.

1.3 Objectives

The objective of this thesis is to propose a mobility-aware novel architecture for interworking heterogeneous mobile data networks, which facilitates real-time session management including session establishment and seamless session handoff across dissipate networks. This framework must conveniently enable a 3G cellular technology (such as UMTS or CDMA2000 system), a WLAN technology, and a WMAN technology (i.e., a WiMAX system) to interwork under a common signaling platform. Therefore, a roaming user in such a heterogeneous network will be controlled via a centralized common mobility management platform where terminal mobility and session mobility is managed in a real-time environment. This framework will essentially exploit the IP Multimedia Subsystem (IMS) as a universal coupling mediator for real-time session negotiation and management [31]. This thesis also analyses and simulates vertical handoff performance measures such as delay, transient packet loss, jitter, and signaling overhead/cost.

1.4 Approach

The approach used for achieving the above objectives can be explained in three main stages as follows. Initially an in-depth review of the current literature published in the area was carried out. This was essential for identifying the trends and issues, formulating the problem, and defining the motivations.

The first stage of the research was to design and develop an initial framework for interworking UMTS and WLAN systems (with limited mobility). This framework used the IMS as a universal coupling mediator for real-time session negotiation and management. Since the initial goal was merely to interwork between UMTS and WLAN systems and the design was somewhat close to a loose coupling model, the 3GPP's IMS became the natural candidate for a coupling mediator. Along with this, the Session Initiation Protocol (SIP) was used as the protocol for mobility and session management (since it is already used within the IMS for session management). As a result, the initial architecture inherited a pure SIP-based mobility management platform, which was fully controlled at the Application Layer.

The second stage of the research further extended the existing framework to be capable of interworking between different 3G cellular technologies (i.e., UMTS and CDMA2000) and WLAN. However, this required major architectural changes within the initial design. As a result, the mobility management concepts used in the 3GPP2's core network and the 3GPP2 IMS framework were adopted. However, at this stage, it became obvious that re-designing of the entire mobility management framework was necessary. Therefore, instead of having a pure SIP-based mobility management framework, a MIP and SIP combined mobility management framework was designed. Within this new framework, terminal/IP mobility is managed at the network layer by MIP and session mobility is handled by SIP at the application layer.

The third and the last stage of the research further extended the model to interwork between 3G cellular, WLAN, and BWA (WiMAX) networks. Similar to the second stage, terminal/IP mobility was managed at the network layer by MIP and session mobility was handled by SIP at the application layer. Since the default mobility management protocol for mobile WiMAX networks was MIP, major modifications were not required for the initial mobility management framework. Therefore, it can now be said that the final design is capable of interworking with a variety of heterogeneous mobile networks, and thus proposes a suitable platform for a NGMN or a 4G network [32].

An analytical model was derived for evaluating the proposed scheme for analyzing the performance of QoS metrics and measures involved in call or session mobility management. Furthermore, an OPNET based simulation platform was also used for modeling the proposed interworking architectures.

1.5 Contribution

This thesis proposes two novel interworking architectures and mobility management frameworks capable of providing session and terminal mobility for a roaming user in a NGMN. The thesis also provides a reliable analytical model for evaluating the performance of such heterogeneous networks. The introduction of the OPNET based interworking platform can also be a considered as a commercially useful tool for simulating similar interworked scenarios under controlled environments. The presented performance results (i.e., vertical handoff delay, transient packet loss, and jitter) can be used by network designers as a performance guideline or a bench mark for evaluating similar scenarios. Furthermore, these results can also be used by application designers/providers and Telco's for fine tuning their application parameters for achieving optimal QoS levels and reliability over a heterogeneous wireless networking environment.

1.6 Outline of the Thesis

This thesis is organized in such a way that it initially introduces different wireless data communication networks (i.e., WLAN, WMAN, and WWAN) to the reader and explains the growing possibility of experiencing the presence of more than one wireless network at a given geographical location. Thus the reader is gradually exposed to the concept of the NGMN and the benefits of roaming between such dissimilar wireless networks. Once the motivations are clarified for this research the objectives, approach, and contribution are clearly specified.

This section intends to provide a general overview of the forthcoming Chapters of this thesis with the relevant corresponding publications. Each Chapter starts with a brief introduction to its contents and ends with a summary and conclusion outlining the main topics discussed with additional ending remarks. A complete set of references is cited at the end of the thesis with the most relevant literature for the reader's easy access.

Chapter 2 provides an in-depth discussion on the current architectures, future trends, and research issues in relation to interworked heterogeneous mobile data networks. The discussion begins by introducing the current and the most notable interworking architectures. The next section presents the open issues and research trends. This Chapter concludes by introducing the IMS and establishing its importance as a coupling media for the newly proposed interworking architectures. The contents presented in this Chapter are partly generated from [33], [34].

Chapter 3 introduces an initial interworking architecture with the IMS as a universal coupling mediator for real-time session negotiation and management. This initial model was aimed for interworking UMTS and WLAN systems with limited mobility. The mobility management framework proposed in this Chapter is fully dependant on the IMS, and its session management protocol. Subsequently, a Queuing Theory based analytical model is introduced for analyzing its vertical handoff performance and an OPNET based test bed is introduced for simulating the architectural design. The interworking architecture and the OPNET simulation platform presented in this Chapter is generated from [35], the basic introduction to the analytical model is obtained from [36] and [37], the extended analysis using the Queuing Theory is generated from [38], and lastly the signaling cost analysis method is generated form [39].

Chapter 4 focuses on extending the proposed WLAN-UMTS interworking architecture presented in Chapter 3 to a unified framework for interworking dissimilar 3G cellular networking standards. As a result, global roaming and interoperability beyond one cellular system (say, UMTS) to another cellular system (say, CDMA2000) becomes possible. Subsequently the sections on analytical modeling and performance evaluations are presented. Finally an OPNET based simulation model is introduced for validation. The interworking architecture and the OPNET simulation platform presented in this Chapter is generated from [40] and the OPNET simulation results are partly obtained from [41] and [42].

Chapter 5 further extends the proposed interworking architecture in to a unified mobility and session management framework for interworking heterogeneous mobile and wireless networks. More specifically, this Chapter introduces interworking BWA networks (such as WiMAX) with the existing platform. Therefore, the proposed platform will not only be capable of providing global roaming between multiple cellular systems, but also be able to successfully interwork with WLAN and WiMAX networks, thus creating a truly seamless inter-network roaming experience for the user. The interworking architecture and the analytical model presented in this Chapter is generated from [43] and the OPNET simulation results are partly obtained from [44]. Finally, Chapter 6 presents the concluding remarks for this thesis by summarizing the important conclusions based on the overall results of the research.

1.7 Summary and Conclusions

The purpose of this Chapter was to expose the reader to the concept of the NGMN or the 4G Networks. Subsequently the reader was introduced to different wireless data communication networks (i.e., WLAN, WMAN, and WWAN) and established the growing possibility of experiencing the presence of more than one wireless network at a given time and a geographical location. Followed by this, the importance of roaming between such dissimilar wireless networks was explained. Once the motivations were clarified for this research the objectives, approach, and contribution was clearly specified. Finally, an outline of the thesis is briefly presented, which was an outline of the forthcoming Chapters.

Chapter 2

Interworking of Heterogeneous Networks: Architectures, Issues, and Trends

This Chapter provides an in-depth discussion on the current architectures, research issues, and future trends on interworking heterogeneous mobile data networks. The discussion begins by introducing the current and the most notable interworking architectures and relating design issues. This Chapter also discusses the open issues and research trends in interworking and introduces the IMS as a potential coupling media for interworking dissimilar networks. The organization of this Chapter can be outlined as follows. Firstly the reader is introduced to the concept and benefits of interworking heterogeneous networks. Then the main flavours and standards/specifications of interworking architectures are introduced. Subsequently, the sections on the IEEE 802.21 Media Independent Handover and the IMS follow prior to the concluding remarks. The contents presented in this Chapter have contributed to [33] and [34].

2.1 Introduction

Modern cellular networks are capable of providing better mobility, whereas WLANs are known for their relatively higher bandwidth. Ubiquitous data services and very high data rates across heterogeneous networks may be achieved by the use of a WLAN as a complementary technology to cellular data networks. Hence there is a strong need for efficient interworking mechanisms between WLANs and cellular data networks [45]. These interworking mechanisms, are expected to be equipped with integrated authentication, integrated billing, roaming, terminal mobility, and service mobility [25],[46]. A variety of interworking architectures have been proposed by numerous researchers and groups. By and large, these proposed integration architectures can be categorized as tight coupling, loose coupling, and peer-to-peer networking (also referred as no-coupling) [26],[27],[47]. As described in [27], the definitions for these coupling mechanisms can be given as follows.

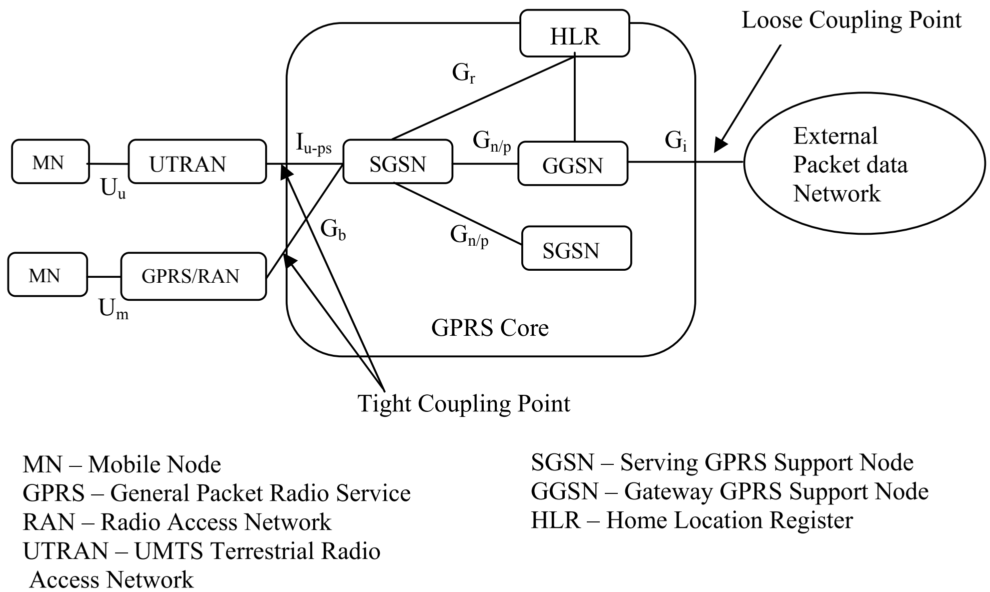


Fig. 2.1. A Reference Diagram Showing Tight and Loose Coupling Points [25].

In a tight coupling scenario, an IEEE 802.11 WLAN is connected to the 3G cellular core network via a Serving GPRS Support Node (SGSN) emulator. Both data and UMTS signaling are transported by the IEEE 802.11 WLAN to the 3G core network via an SGSN emulator. Thus, the IEEE 802.11 Basic Service Set (BSS) acts as another SGSN coverage area to the UMTS core network. On the other hand, a loosely coupled architecture transports UMTS signaling over the IEEE 802.11 WLAN to the 3G core network, while data flows directly to the IP based network. Fig. 2.1 is a reference diagram showing these tight and loose coupling points.

In tight and loose coupling methods, GPRS/UMTS signaling is carried over WLAN, where the two networks look like one from the Network Layer and above. In both of these cases the 3G core network acts as the “master” network and the IEEE 802.11 WLAN behaves as the “slave” network. Although the last coupling mechanism, which is peer-to-peer networking, may be seen as an extension to the loose coupling architecture, it treats the two networks as peers [26]. MIP is used to provide a framework for mobility among these peers. Last but not least, there are also other various proposals of hybrid coupling schemes [48]. Such methods are capable of differentiating the data and signaling paths according to the type of traffic. For example, in the case of [48], a tightly coupled architecture is used for real-time traffic, and a loosely coupled network architecture is used for non-real time and bulky traffic.

2.2 The Tight Coupling Architecture

As mentioned previously, in tight coupling, the WLAN is directly connected to the 3G cellular core network [49]. Fig. 2.2 illustrates a basic system configuration diagram for a tight coupling architecture. Thus the WLAN data traffic passes through the GPRS core network before reaching the external PDNs.

The key functional element in the system is the GPRS Interworking Function (GIF), which interconnects an IEEE 802.11 Extended Service Set (ESS) to a SGSN via the standard G_b interface [25]. This is also referred to as an SGSN emulator [27]. The GIF is the function that makes the SGSN consider the WLAN as a typical GPRS Routing Area (RA) composed of only one cell.

The handover between the WLAN and the GPRS can be considered as a handover between two individual cells. It is also worth noting that the GIF and all interconnected WLAN terminals use a 48-bit IEEE 802 MAC address. The WLAN Adaptation Function (WAF) is the main component, which helps the Mobile Node (MN) to identify the MAC address of the GIF (Fig. 2.3). Hence, there is a WAF implemented in every dual mode MN as well as the GIF for 3G signaling and data exchange over the IEEE 802.11 WLAN. The WAF also provides the following functions:

- Signaling the activation of WLAN interface as the MS enters a WLAN area,
- Discovering the MAC address of the GIF,
- Helping the SGSN page a mobile station over the G_b interface,
- Transferring Logical Link Control (LLC) Protocol Data Units (PDU) from mobile station to the GIF and vice-versa, and
- Supporting QoS by implementing transmission scheduling in the GFS and the MN.

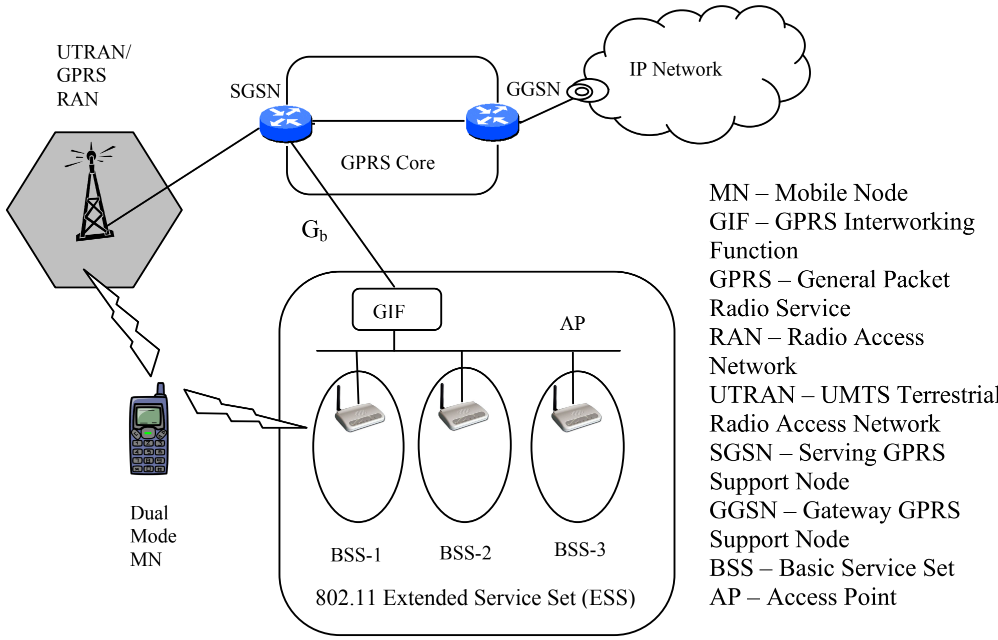


Fig. 2.2. WLAN–3G Integration with Tight Coupling: System Overview [25].

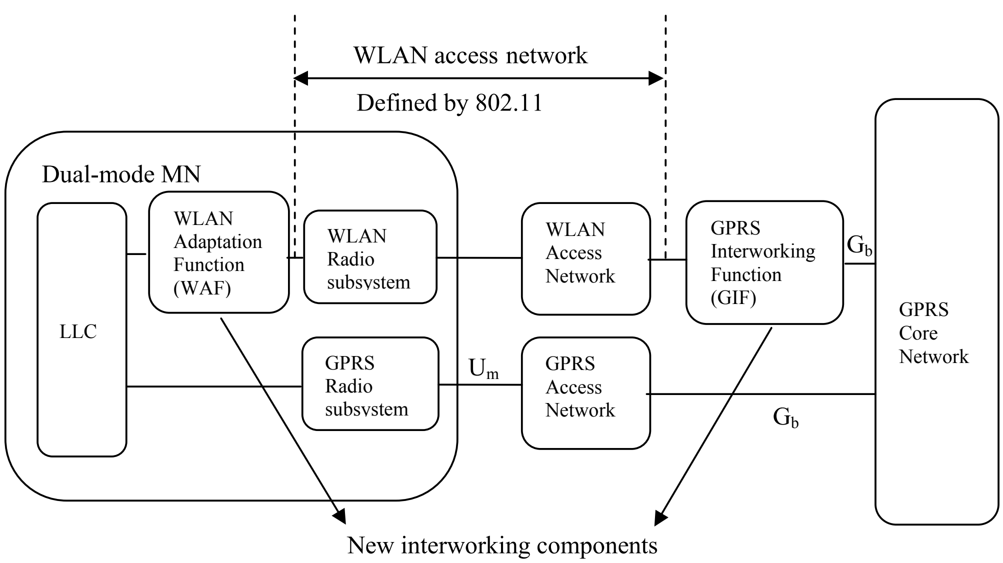


Fig.2.3. Tight Coupling Architecture: The New Interworking Components [25].

Additionally, the tight coupling architecture can be conveniently extended by adding new functionalities such as location based service support by enabling efficient support for location aware secure fast roaming with location privacy control functions and a location based policy authority [50].

Since the WLAN and GPRS networks connect to the same Gateway GPRS Support Node (GGSN), IP addresses are assigned by the same pool. Hence, the mobility across the two networks do not require a change of an IP address [27]. Lastly, a tight coupling architecture provides the following benefits [25]:

- Seamless service continuation across WLAN and 3G networks,
- Less complicated mobility management mechanisms (since it follows GPRS/UMTS mobility management mechanisms),
- Ability to use the GPRS/UMTS Authentication, Authorization, and Accounting (AAA) system,
- Ability to use the GPRS/UMTS infrastructure for routing (e.g., core network resources, subscriber database, billing systems),
- Increased security (GPRS/UMTS security can be applied on top of WLAN security),
- Common provisioning and customer care, and
- Access to core GPRS/UMTS services (Short Message Service (SMS), location-based service, Multimedia Message Service (MMS), etc.)

However, it is important to note that, the tight coupling architecture is primarily designed for WLANs owned by cellular operators [51]. Thus it lacks implementation capability for third-party WLANs. Further, there are cost and capacity concerns associated with the connection of a WLAN to an SGSN.

2.3 The Loose Coupling Architecture

Similar to the previous coupling method, loose coupling architecture is also a master/slave framework. As shown in Fig. 2.4, the GPRS network acts as a master and the IEEE 802.11 WLAN acts as a slave network (or a visiting network) where only the AAA traffic is routed through the 3G core network (not the user data traffic) [28] [52]. In this scenario, only AAA signaling is exchanged between the WLAN and the 3G home Public Land Mobile Network

(PLMN) (via the 3G visited PLMN) to provide authentication, authorization and accounting (charging).

Two alternative authentication models (or flavors) for loose coupling can been identified in the literature [53], [54]. These are described as the "IETF flavor" and the "UMTS flavor" [53]. The primary difference between these two is essentially the authentication server it self [54].

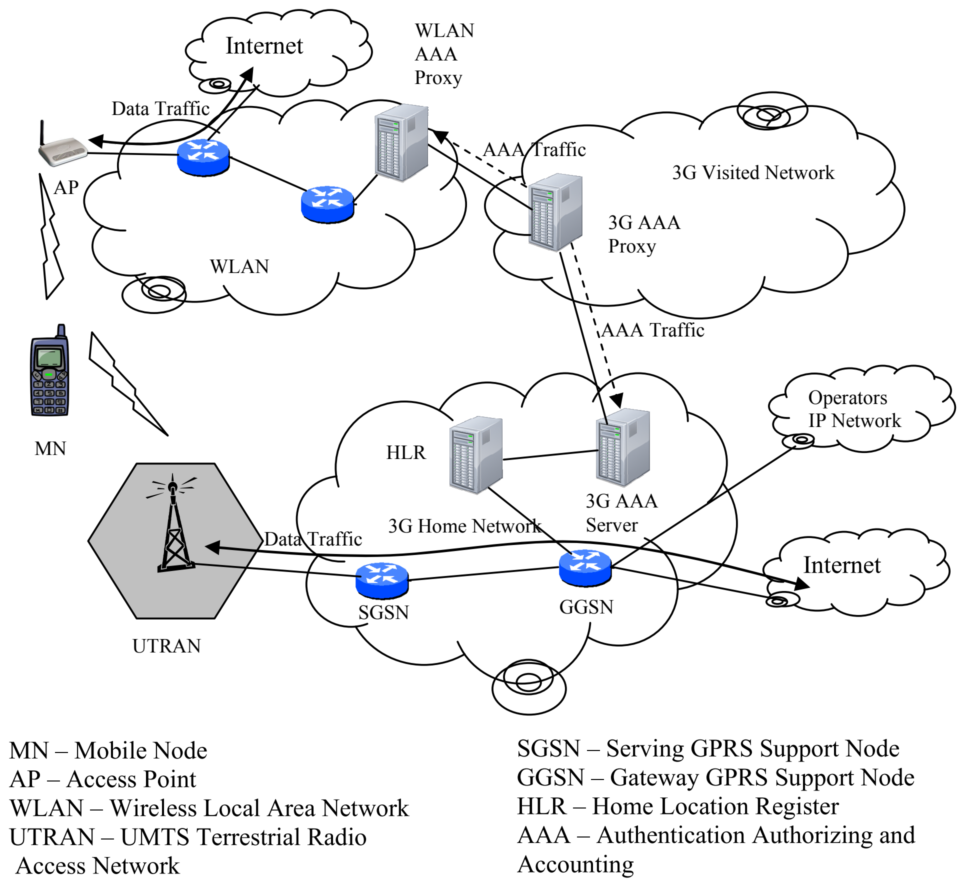


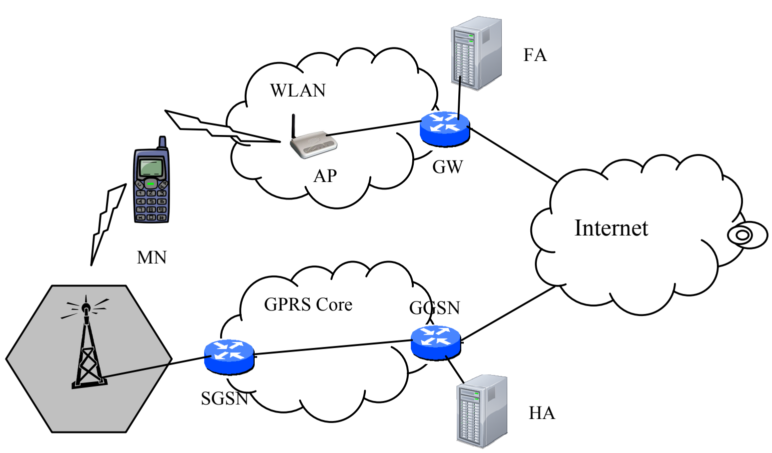
Fig. 2.4. WLAN–3G Integration with Loose Coupling: With 3G Based Access Control and Charging [28].

During the authorization phase, the 3G AAA server can establish policies for the user data traffic. It can also serve as a proxy to route AAA signaling to/from other 3G PLMNs (Fig. 2.4). Thus the 3G AAA server is an important component introduced in the loosely coupled interworking architectures. However, more advanced types of interworking scenarios (as discussed later) may also require the user data traffic to be routed to the UMTS core network [28], [29].

A clear advantage of the above method is that, since the data traffic is routed directly to and from the IP network (Internet) without having to route through the 3G network, a potential traffic bottleneck can be avoided. Since the 3G network and the WLAN are likely to be in different IP address domains, the MN will be allocated an IP address from the pool of addresses of the connected network. The changing of an IP address may result in loss of connectivity. Therefore, in the loose coupling architecture, handoffs are less efficient and mobility management is generally more complicated when the user is in an active session [27].

2.4 Peer-to-Peer Networking Architecture

Unlike the previously discussed master/slave coupling methods, this approach treats the two networks as peers. In a peer-to-peer networking architecture, 3G access and WLAN access can be provided by the same or different operators. Further, MIP and AAA servers are used for providing a framework for mobility [30]. MIP is used to restructure connections when an MN roams from one peer network to another and AAA servers provide AAA functionality.



MN – Mobile Node
AP – Access Point
WLAN – Wireless Local Area Network
UTRAN – UMTS Terrestrial Radio Access Network

SGSN – Serving GPRS Support Node
GGSN – Gateway GPRS Support Node
HA – Home Agent
FA – Foreign Agent
GW – Gateway

Fig. 2.5. WLAN–3G Integration with Peer-to-Peer Coupling [55].

The MIP framework consists of a MIP client (i.e., the MN), a MIP Foreign Agent (FA), and a MIP Home Agent (HA) as illustrated in Fig. 2.5. Outside of the home network the MN is identified by a care-of-address associated with its point of attachment. The MN registers its care-of-address with its HA. The HA resides in the home network of the MN and is responsible for intercepting datagrams addressed to the MN's home address and tunnel them to the associated care-of-address.

When a 3G network and a WLAN are accessing a public IP network, MIP can be used in the following manner [56]. The HA function could be implemented in the GGSN of the 3G network. As the MN (whose home network is 3G) moves to a foreign network (the WLAN), it registers with the HA (the GGSN) its current care-of-address through the FA at the foreign network. When the GGSN (the HA) receives packets whose destination is the MN, which tunnels these IP packets to the FA, and eventually reaches the MN. Likewise, the FA functionality may be implemented at the GGSN or SGSN of the 3G network.

Nevertheless, implementation of the HA does not necessarily have to be at the GGSN. The HA could also be implemented at an external IP network [25]. This architecture implements the FA functionality at the 3G and WLAN networks, which is integrated with the HA located on an external IP network. However, both 3G and WLAN networks may need to subscribe to this IP network. Although the peer-to-peer networking architecture is identified as a separate interworking architecture, some literature merely considers it as a variation of the previously discussed loose coupling mechanism [26].

Although MIP is recognized as an acceptable solution for mobility in general, it suffers from several drawbacks [57]. Firstly, since it is a purely Network Layer oriented solution, it is unable to solve any link layer handoff problems. Secondly, triangular routing may also result in extended delays [58], [59]. Thirdly, it may not deal with vertical handoffs due to discrepancies in the mobility management techniques of a heterogeneous networking environment (e.g., UMTS core network does not use MIP) [60].

Additionally, during the time frame between the MN leaving its current subnet to a new subnet, associating with a new FA, and updating the HA with the new care-of-address, a packet loss (or disruption of service) can be noticed. This is due to the fact that during this time interval the HA is still tunneling packets to the old FA's care-of-address. Further, as the distance between the MN and HA increases the service, the latency and packet loss increases.

Therefore the, MIP framework is not be the best solution for frequently roaming users between two networks [61], [56].

Another variation to the peer-to-peer networking architecture, which is better known as a gateway (or proxy based) approach, have also been proposed in related literature [28], [30], [55], [61], [62]. However, due to the above mentioned deficiencies, not all rely on the MIP framework. Finally, despite mobility and seamless handover still exist as open issues, the use of a Virtual GPRS Support Node (VGSN) has also been proposed as an interim solution in [30], [61].

2.5 3GPP's Approach to WLAN and Cellular Interworking

The 3GPP has made some recent attempts in standardizing and developing a WLAN-Cellular interworking architecture [63], [64], [65], [66]. The 3GPP proposes a specific WLAN-3G cellular interworking architecture capable of supporting common AAA, WLAN sharing, consistent service provisioning, and several access control schemes.

Under this proposed framework, six common interworking scenarios have been discussed [28], [65], [67]. The first scenario, which is the simplest, provides only a common bill and customer care to the subscriber but no real interworking between the WLAN and the 3G PLMN.

Scenario two provides the 3G cellular subscriber with a basic IP network connection via the WLAN and includes no other 3G services. The goal of scenario three is to extend the access to the 3G Packet-Switched (PS) services to subscribers in the WLAN environment. In addition, an IP service selection scheme is used for selecting the PS based service to which to connect. Such services may include IP multimedia, location-based services, instant messaging, presence-based services, and MMS.

However, scenario three does not address service continuity across these access networks. Therefore scenario four addresses this issue and helps maintain service continuity across 3G cellular and WLAN radio access technologies.

Scenario five takes it one step further by introducing seamless service continuity between the 3G and WLAN systems. That is, PS-based services should be utilized across the 3G cellular and WLAN radio access technologies in a seamless manner, without the user noticing any significant differences. Finally, scenario six describes access to 3G Circuit-Switched (CS) services from the WLAN system including seamless mobility for these services.

The frameworks proposed by 3GPP include a non-roaming (Fig. 2.6) and a roaming (Fig. 2.7) interworking architecture [63]. The roaming architecture is further divided into two categories. That is, when the 3GPP PS services are provided via the home (Fig. 2.7a) or visited 3G network (Fig. 2.7b).

The W_n interface connects the WLAN access network and WLAN Access Gateway (WAG). W_a is the reference point between the WLAN access network and the 3GPP AAA server/proxy, which is used for charging and control signaling. W_i is the interface between the Packet Data Gateway (PDG) and an external IP network (or the Internet), and W_p is the interface between WAG and PDG. Further, W_u is the reference point between WLAN interface and a PDG, and W_d the reference point between the 3GPP AAA proxy and the 3GPP AAA server. Detailed explanations on the above and the remaining reference points can be found in [63].

The functioning of the interworking mechanism can be best summarized as follows [68]. The WLAN MN uses a WLAN Access Point Name (W-APN) to indicate the network the service or the set of services, which it requires to access. Therefore, a service request is firstly delivered to the 3GPP AAA server to obtain a service authorisation to the PDG. After receiving the service authorisation form 3GPP AAA server, the PDG returns a return response to the 3GPP AAA server containing filtering attributes, IP configuration and other information.

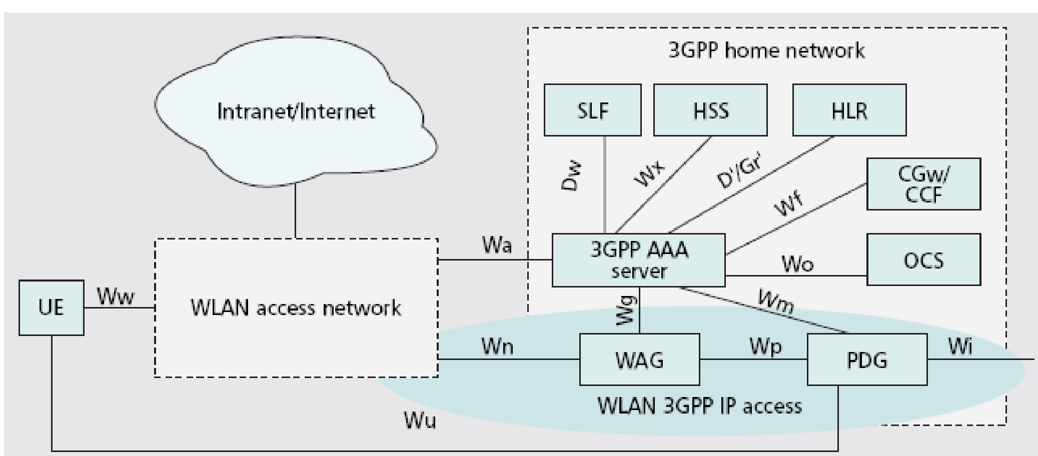


Fig. 2.6. 3GPP's Non-Roaming Reference Architecture [64].

Then the 3GPP AAA server responds to the WLAN Access Network with service authorisation information, which is relayed to the WLAN UE (or MN). Finally, the WLAN UE initiates a tunnel establishment request to the PDG. Once the tunnel is established between PDG and the IP network, WLAN UE can access the service and user data can be routed through the tunnel.

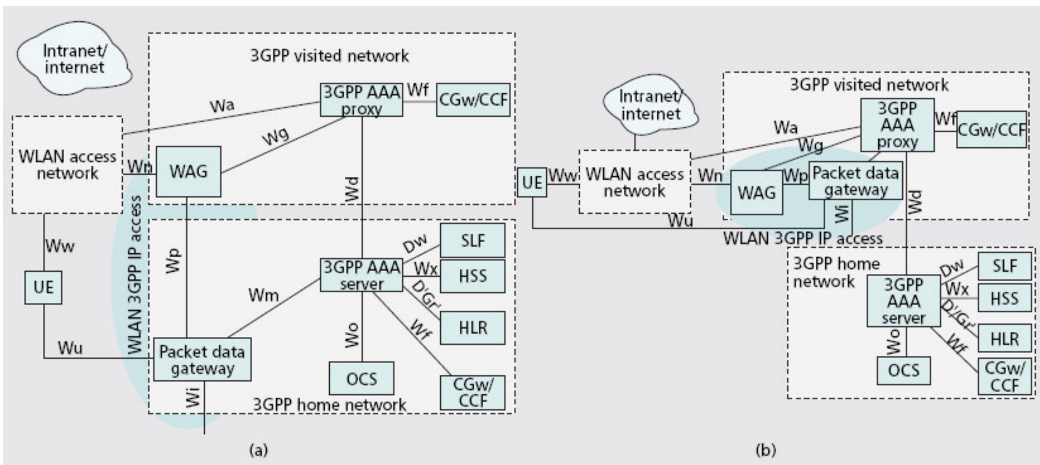


Fig. 2.7. 3GPP's Roaming Reference Architecture: PS Based Services Provided via the a) 3GPP Home Network and b) 3GPP Visiting Network [64].

2.6 3GPP2's Approach to WLAN and Cellular Interworking

Similar to the previously mentioned case of 3GPP, 3GPP2 has also identified the benefits of interworking the existing CDMA2000 network with the WLAN. The intention of 3GPP2–WLAN interworking is to extend the 3GPP2 packet data services and/or capabilities to the WLAN environment. The following briefly outlines the specifications for a proposal for interworking between 3GPP2 systems and WLANs as specified by [69].

A logical interworking model depicting the interworking relationship scenarios is shown in Fig. 2.8. The logical model in Fig. 2.8 shows the 3GPP2-WLAN interworking relationships where the relationships are either direct or indirect (i.e. through a broker system). In the case of a direct 3GPP2-WLAN interworking relationship, the interworking relationship is defined between the 3GPP2 system and the WLAN system. In the case of an indirect 3GPP2-WLAN interworking relationship, the interworking relationship is defined between a broker system and a WLAN system, and between a broker system and a 3GPP2 System. An interworking relationship between a 3GPP2 system and a WLAN system can be considered as a many-to-many relationship. As an extension to the above report [69], descriptions of requirements, scenarios and networking architectures based on the IPv6 environment has been presented in [70]. Furthermore, the authors have suggested specific cases or scenarios, which the interworking model should carefully comply in order to comply with IPv6 mobility management.

Although it is of lesser importance, there are other contributions made in this area of study. For example, [71] proposes a solution for smart seamless handoff with QoS guarantee for CDMA2000 and WLAN systems. According to this proposal, the MN conducts the QoS negotiation with the new network during the handoff. Similar ideas with respect to end-to-end QoS over CDMA2000 and WLANs have been expressed in [72]. Also, another proposal that presents a Seamless Connection Control (SCC) scheme for interworking between CDMA2000 and WLAN systems by a method where the MN drives handover is available in [73].

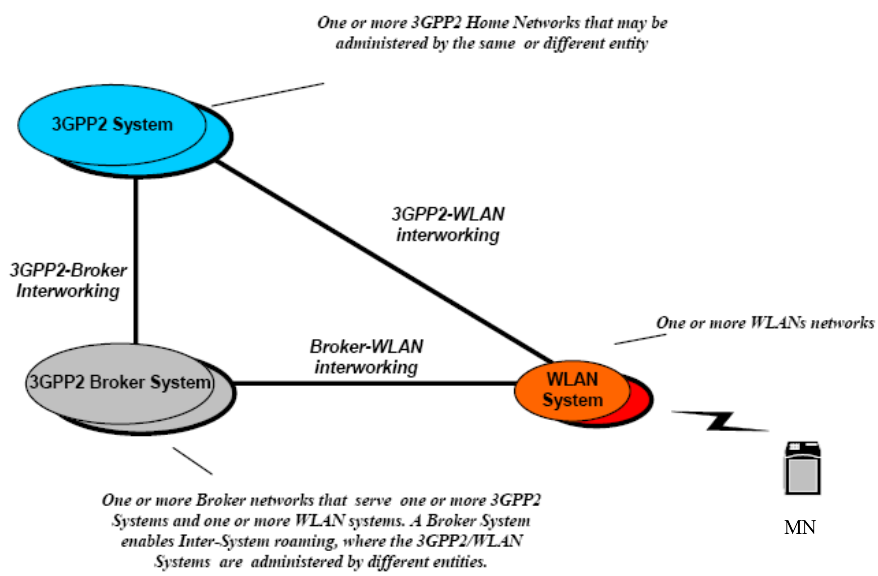


Fig. 2.8. 3GPP2's Proposed CDMA2000-WLAN Interworking Architecture [69].

2.7 Interworking between other Disparate Networks

Although the discussion so far has been concentrated around various interworking techniques between WLAN and 3G cellular (UMTS in particular) networks, there is growing interest in interworking with other disparate networks. Furthermore, interworking of heterogeneous networks has become an important area of research with the emergence of the concept of 4G networks. In this regards, there are some (although not many) works on interworking between UMTS and CDMA2000 networks, interworking between UMTS and WiMAX networks, and interworking between WLAN and WiMAX networks. This section provides a brief overview of some of the currant works that are being done in this area.

2.7.1 Interworking between UMTS and CDMA2000 system

One of the primary motivations of the IMT-2000 project was to provide global access capability with a harmonized radio access network and a unified core network [74], [75]. Despite these motivations, the current 3G systems were developed as extensions of the platforms of the second generation predecessors.

For example, the Global System for Mobile communications (GSM) was extended to UMTS and the cdmaOne system was extended to CDMA2000 by the 3GPP and 3GPP2, respectively. Thus the goal of having a unified communication system was not achieved, and global roaming and interoperability beyond one system and carrier still remains a challenge.

Therefore, it is essential that in the development of the NGMN that there exists a unified platform for these systems to interwork with each other. This has given rise to much research activity aiming towards providing interoperability and global roaming among different cellular networking standards and technologies [76], [60]. Due to architectural and technical differences in these two core networking technologies many aspects need to be addressed prior to achieving fully seamless interworking between these networks.

Figure 2.9 shows a currently proposed architecture that interconnects UMTS and CDMA2000 systems [60]. This method shares much similarity with the previously stated peer-to-peer interworking mechanism and uses MIPv4 as the preferred mobility management mechanism [30]. In this figure, the UMTS network is connected to the IP network through the

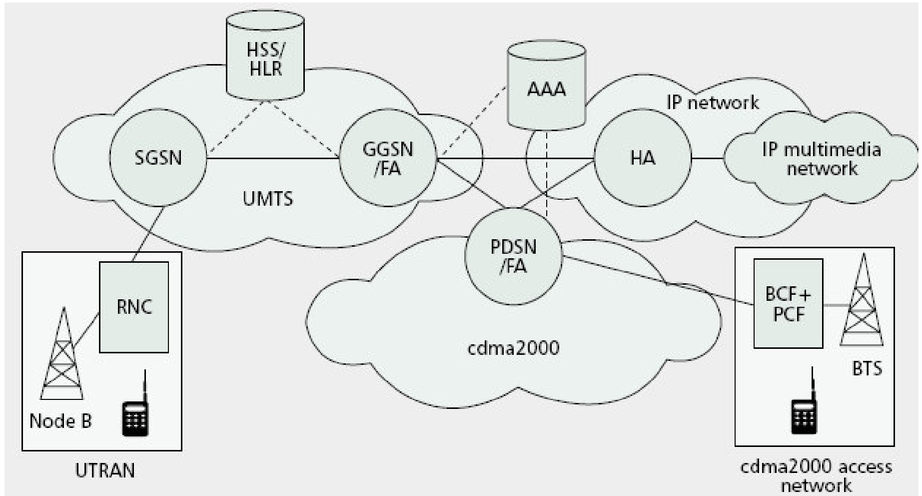


Fig. 2.9.UMTS and CDMA2000 Interworking: MIPv4 based Approach.

GGSN that acts as the MIP FA. Following the GPRS attach and Packet Data Protocol (PDP) context activation procedures, the MN performs the MIP registration to its HA via the GGSN (i.e., MIP FA).

The HA maintains the MN's location information (e.g., the GGSN, which is the MIP FA address) and tunnels the IP datagrams to the MN. In the CDMA2000 network, the PDSN is the MIP FA, which communicates with the HA through the IP network as previously described. Both the UMTS and CDMA2000 networks connect to the IP multimedia network through the MIP HA. That is when a terminal in the IP multimedia network originates a multimedia call to the UMTS or CDMA2000 MN by using protocols such as H.323 [21] and SIP [22], and the voice packets are first delivered to the MIP HA.

Then the MIP HA forwards these packets to the MN through the MIP FA (GGSN or the Packet Data Serving Node (PDSN)). Despite the successful interworking framework used for addressing IP mobility management, this architecture raises concerns such as service continuation, seamlessness, and session mobility management. Thus this interworking architecture cannot be successfully adopted in a heterogeneous networking environment such as the future 4G network.

Another conceptually similar proposal to the above mentioned is illustrated in Fig. 2.10 [77], [78]. This proposed architecture, called Integrated InterSystem Architecture (IISA), which adapts an integrated loose coupling model and implements MIPv6 [79] and Hierarchical

MIPv6 (HMIPv6) [80] functionalities. Despite the fact that this model overcomes the usual drawbacks of MIPv4, it requires substantial modifications within some functional entities of the UMTS and the CDMA 2000 core network.

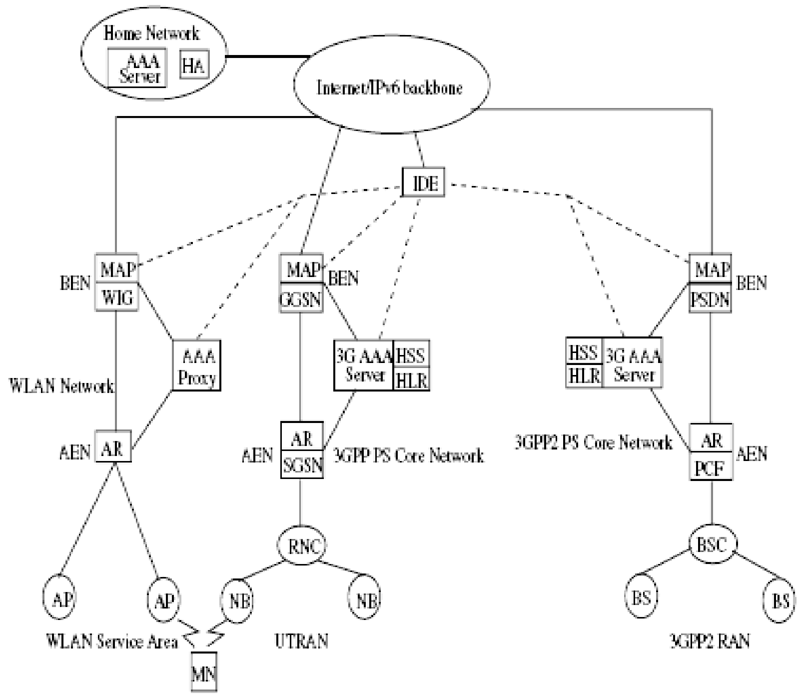


Fig. 2.10. Interworking IISA Architecture for 4G Networks.

Therefore, SGSN/Packet Control Function (PCF) is enhanced with the functionalities of an access router and is called an Access Edge Node (AEN) and GGSN/PDSN is extended as a Mobility Anchor Point (MAP). Similarly, GGSN/PDSN is extended with MAP and interworking functionalities (to enable message formats conversion, QoS requirements mapping, etc.) and is called the Border Edge Node (BEN). The BEN has the information of ARs such as IP address, subnet prefix, link address within its domain. The WLAN Interworking Gateway (WIG) acts as a route policy element, ensuring message format conversion. Extended functionalities can be integrated into existing network entities or implemented separately.

However, incorporation of such modifications require major overhaul within the UMTS and CDMA2000 core networks, thus the current 3GPP and 3GPP2 standards need to be revised.

Therefore, it is not deemed as a feasible approach for interworking. Another reason for making this impracticable from the network operators' point of view is that in order for MIPv6 to be implemented an IPv6 based networking platform must exist.

Two other possibly noteworthy proposals for interworking UMTS-CDMA2000 networks are found in [76]. The first is a network configuration based on the gateway solution. The gateway solution can be considered as the simplest way of implementing interworking and interoperability between two dissimilar technologies, without any modification to the existing CDMA2000 systems and the new UMTS systems. Fig. 2.11 depicts a network configuration of the gateway solution, which requires the introduction of a gateway system between UMTS and CDMA2000 networks.

According to the authors of [76], this solution is capable of addressing the global roaming problem between different technologies, where the basic concept and functions were standardized as per the TIA/EIA/Interim Standard (IS)-129 [81] and improved as in the 3GPP2's N.S0028 [82]. The gateway solution may be of interest to new UMTS operators that do not own a legacy CDMA2000 network, but directly adopts the UMTS system and only require support for global roaming among other international CDMA2000 service providers.

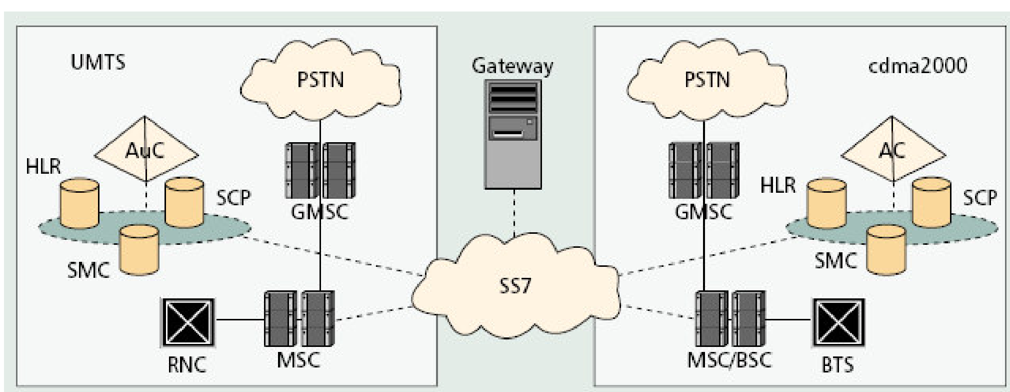


Fig. 2.11. An Interworking Architecture based on the Gateway Solution.

The lack of total seamlessness for every function may not affect the global roaming market, especially where roaming subscribers are accustomed to differences in the capabilities supported by the home and visited networks. Another shortcoming of this solution is that the subscriber database must be managed in three different places, which is not a technical problem but requires another interface from the central database to the gateway system.

Secondly, Fig. 2.12 depicts a network configuration of a dual-stack solution, which requires the Home Location Register (HLR), Service Control Point (SCP), and Short Message Centre (SMC) to be modified and upgraded to support both protocol stacks required by the CDMA2000 and UMTS technologies. A simple implementation of the two protocol stacks in these systems is not sufficient to achieve integration of the two networks. Also, each protocol stack performs and operates its own mobility management, call delivery, supplementary service management functionalities.

However, coordination of mobility management, call delivery, and supplementary service management between the two networks is kept through the dual-stack HLR, SCP, and SMC. In summary, it can be said that the dual-stack solution best suits a network service provider that wants to provide its customers both number portability and service transparency across its CDMA2000 and UMTS systems. However, this solution does not sufficiently address the issue of IP/terminal mobility or the session mobility for a potential user roaming between these two networks.

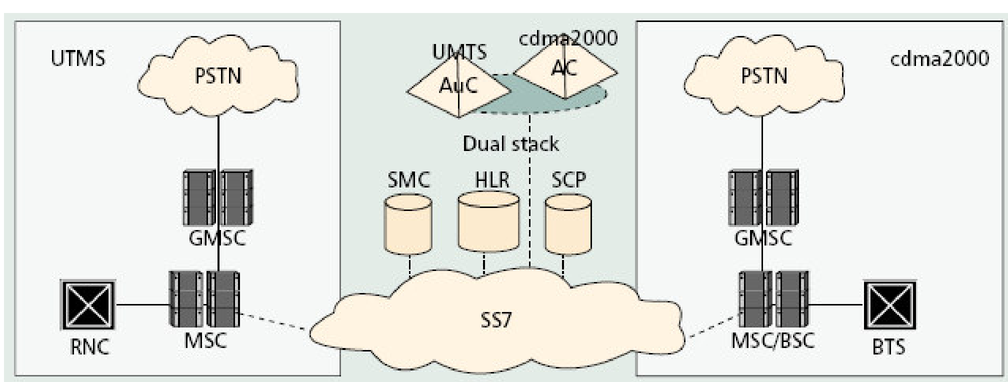


Fig. 2.12. An Interworking Architecture based on the Dual-Stack Solution.

2.7.2 Interworking between UMTS and WiMAX system

With the above motivations, similar attempts have also been made for interworking WiMAX systems with 3G cellular technologies [83]. The initial works concentrated on different flavours of the commonly known Loose and Tight Coupling architectures where a series of proposals about different mobility management mechanisms have been presented in [84]. In the case of interworking WiMAX with 3G cellular systems, MIP based approaches for mobility management have been the most widely proposed [85]. Furthermore, despite its well known drawbacks, MIP has become the popular choice for mobility management in mobile WiMAX networks [57].

2.7.3 Interworking between WLAN and WiMAX system

Similar to the above two sections, there are also some notable contributions towards interworking WLAN and WMAN systems in the current literature. A novel scheme for achieving the above is given in [86], where a generic framework based on the IEEE 802.21 [87] has been proposed. Firstly this framework aims to provide continuity of service by introducing a mechanism of QoS adaptation by which applications can adjust their QoS in order to cope with the QoS provisioned in the visited network. Secondly, it points towards making an intelligent handoff decision according to policies predefined in the system. Another simple and straight forward proposal for interworking WLAN and WiMAX system is given in [88], which also uses the IEEE 802.21 [87] standard as a coupling mediator.

2.8 IEEE 802.21 Media Independent Handover Services

The current IEEE 802 standards do not support vertical handover between different types of networks. They also do not provide triggers or other services (such as methods for selecting the optimal network) to facilitate vertical handovers in a heterogeneous networking environment. Therefore a new standard called IEEE 802.21 for supporting algorithms for enabling seamless handover between different network types has been proposed for providing information to allow handing over to and from cellular, GSM, GPRS, Wi-Fi, Bluetooth, IEEE 802.11, and IEEE 802.16 networks through different handover mechanisms.. This standard is also called the Media Independent Handover (MIH) or vertical handover function [89].

The objectives of the IEEE 802.21 standard can be summarized as [90], [91]:

- Enable roaming and seamless handover between heterogeneous wireless networks:
    - To be available for use by multiple vendors and users,
    - To be compatible with other IEEE 802 standards (e.g. WiMAX and WLAN),
- Include definitions for managed objects that are compatible with management standards like the Simple Network Management Protocol (SNMP) [92],
- Include definition of a new link layer Service Access Point to provide a technology independent common Link layer interface,
- Define a set of handoff enabling functions to coordinate with the upper layers (e.g., MIP based mobility management at the Network Layer) to perform efficient vertical handoffs,
- Provide support for authentication, authorization, and network detection and selection (although security algorithms and security protocols are defined in the standard).

When a session is handed off from one access point to another access point using the same technology, the handover can usually be performed within that wireless technology itself without involving the Media Independent Handover Function (MIHF) or the IP. For instance a Voice over IP (VoIP) call from a Wireless Fidelity (Wi-Fi) handset to a Wi-Fi access point can be handed over to another Wi-Fi access point within the same network. However if the handover is from a Wi-Fi access point in a corporate network to a public Wi-Fi hotspot, then the MIH is required, since the two access points cannot communicate with each other at the link layer, and are on different IP subnets [93].

The key functionalities provided by the MIH are communication among various wireless layers, and between them and the IP layer. This handover procedure uses the information from both MN and the network infrastructures. The IEEE 802.21 framework informs the available network nearby to the MN and helps the MN to detect and select the network. This information includes Link Layer information. The MIH may communicate with various IP protocols including SIP for signaling, MIP for mobility management and DiffServ and IntServ for QoS.

The required messages are relayed by the MIHF that is located in the protocol stack between the Layer 2 wireless technologies and IP at Layer 3 [89]. The MIHF provides homogeneous interfaces independent from access technologies [94]. This interface handles communications between the upper layer and lower layer as illustrated in Fig. 2.13.

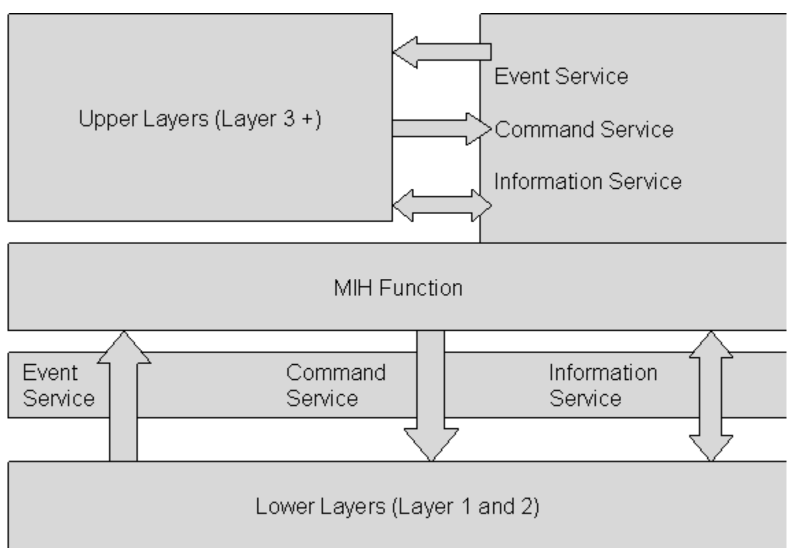


Fig. 2.13. IEEE 802.21 Media Independent Handover Architecture.

The MIHF provides three services as:

- Media Independent Event Service (MIES)
- Media Independent Command Service (MICS)
- Media Independent Information Service (MIIS)

The MIES supports the transfer, filtering, and classification of dynamic changes from Layer 2 to Layer 3. This delivers triggers on events such as link up, link down, and new link available. The MICS offers the functions for managing and controlling Layer 2 from Layer 3. If the MIH application wants handover and mobility, it can control the MAC layer by using MICS. A set of standard commands for handover control would be switch link, configure link, and initiate handover.

The MIIS offers information that is needed to perform the handover. It defines a service that provides information for faster handovers such as a list of available networks, IP version, and network operators. Using this information, the mobile terminal is able to make a decision on the handover.

The transportation of the MICS, MIES, and MIIS messages over Layer 2 or Layer 3 is facilitated via the MIHF transport protocol. IEEE 802.11u defines transport for IEEE 802.21 messages over IEEE 802.11. This is a proposed amendment to the IEEE 802.11-2007 standard to add features that improve interworking with external networks [95].

As a current hot topic in research, many contributions have been made to the draft IEEE 802.21 standard [88],[86]. More specifically, there is growing interest towards investigating how the emerging IEEE 802.21 standard could enable seamless, inter-technology handover [96],[97].

One such proposal is for dynamically predicting and adjusting the buffer size to enhance the functionality of the IEEE 802.21 [98]. The mechanism of the proposed scheme includes a service specific layer, which decides the opportunity of handover and adjusts the buffer, which stores the temporal streaming data for applications between handover.

There is also another proposal of a multi-interface scheme for the IEEE 802.21 MIH [93]. This proposal aims for having multi-interface mobile nodes to work with the standard Transmission Control Protocol (TCP) and MIP protocols. Similarly, the problems and how to efficiently use these multi-interfaced devices to achieve best data throughput, reduced packet loss, and optimized handover is presented in [99].

There are also other contributions on how the IEEE 802.21 standard may be applied to fast moving vehicular networks [100]. In this, the authors' have used the IEEE 802.21 MIH services for optimizing the handover procedure in the Fast Handover for Mobile IPv6 (FMIPv6) protocol.

2.9 Open Issues and Research Themes

Despite the fact that there is a vast range (or flavors) of interworking architectures that have recently been proposed, there exist many open issues. This section attempts to summarize some of these issues, which may be considered as potential research themes under the current project work.

The first of which is the issue of session mobility across WLAN and 3G cellular networks. For instance, when a MN with an active IP-based session (File Transfer Protocol (FTP) [101] or Hypertext Transfer Protocol (HTTP) [102], say) moves from a WLAN to a 3G cellular network (or vice-versa), the session must be seamlessly handed over from one network to another. Possible research questions that may arise from such a situation are: how to detect the entering and leaving from a hotspot; which part of the network to handoff; and which mechanisms to perform authenticated handoff. Thus, how to provide/enable seamless continuity of service across WLAN and 3G cellular networks can be ranked as a top issue.

Naturally, the solution/s to the above would depend on the type of integration architecture used. For instance, mobility management for a tight coupling architecture can be seen as a moderately complex issue since 3G mobility management schemes could be directly applied [1]. However, this becomes an extremely complex and a rather challenging issue for a loosely coupled architecture. It requires emulators for 3G signaling, possible protocol stack modifications, and multiple mobility management schemes [3]. Such mobility management schemes may comprise of 3G cellular and IEEE 802.11 mobility management schemes, IP and SIP registration schemes, and Domain Name System (DNS) [103] update schemes.

Lastly, the peer-to-peer interworking architecture, where each network behaves independently, is considered to be the least complex. The most commonly used mobility management scheme under this approach is the MIP approach [3]. However, other schemes may apply depending on the type and the level of integration. A similar concern, which is very much related to the above is the problem of IP address allocation and distribution for a MN in a WLAN-3G heterogeneous networking environment.

Another important issue is to define a mechanism for user data routing via heterogeneous networks. How an optimal routing path can be decided is still an open issue, which is currently being investigated by the 3GPP. Matching the QoS requirements and service provisioning between WLAN and the 3G cellular networks can also be pointed out as another related issue.

Provisioning of unified/integrated mechanisms for AAA and security in general is another high priority concern. Among the related work done by the 3GPP in this area, one such recommendation is to eventually adopt the IEEE 802.11i standard [TS 33.234]. Although not much, [50] and [104] are some works that deals with this security aspect in regards to interworking heterogeneous networks. A mechanism for location aware authentication for interworked 3G-WLAN systems is proposed in [50] and a handover architecture for seamless mobility and an extended authentication mechanism is proposed in [104].

Another related question is: whether AAA is managed at the WLAN, 3G core network or independently? Furthermore, charging options such as post-paid, pre-paid charging and IP-flow based charging must also be considered. Last but not least, terminal capabilities such as support of high data rate applications, screen size, computational power, and network selection mechanisms must also be addressed.

Out of the above mentioned research challenges, this thesis aims towards proposing a mobility-aware novel architecture for interworking heterogeneous mobile data networks. This is expected to facilitate real-time session management including session establishment and seamless session handoff across dissipate networks. Further, by considering the complications of the existing network coupling mechanisms, a universal coupling mediator for real-time session negotiation and management is proposed. This mechanism is further capable of resolving the issue of IP mobility with mechanisms in place for the issue of IP address allocation and distribution in such environments. Finally, although it is out of the scope of this thesis, the proposed framework has adequate provisioning to comply with the current IEEE 802.11i AAA standards as outlined in the future works section.

2.10 IMS: IP Multimedia Subsystem

As stated above, the aim of this thesis is to propose an interworking model, which is capable of providing a MN the highest possible level of access over different heterogeneous networks, where fully seamless access is considered to be the optimal accomplishment. A novel approach for solving this problem is to explore the possibility of introducing one single centralized entity for handling mobility management between different networks.

The quest for such a centralized mobility management entity came across a somewhat similar idea expressed in [105] and [106]. According to [105], it was possible to use the 3GPP's IMS [31] for supporting real-time session negotiation and management between heterogeneous networks. This was a positive direction since this could be the ideal candidate for session mobility management. However, apart from the concept of having the IMS as a session manager and a service delivery framework [107] and [108] do not address any issues pertaining to mobility management.

Furthermore, complementing to the ideas expressed in [105], there is also another paper expressing how the SIP protocol (i.e., the main signaling protocol used in the IMS), is capable of handling session management in a UMTS-WLAN networking environment [109]. However, the issue of mobility management was not addressed in this case either. Therefore, [105], [109] can be seen as the main inspirations to our works.

A similar idea of using the IMS as a coupling medium (only for session mobility management) has also been recently published at a much later stage in [110], it mainly focuses on the QoS guarantee and AAA aspects provided by the IMS. Most importantly the architecture fails to sufficiently describe a mechanism for seamless vertical handoff and mobility management. Additionally there are also similar works carried out in relation to using the 3GPP2 IMS [111] as a mediator for coupling in the process of interworking CDMA2000 networks with a WLANs as per the Telecommunications Industry Standard (TIA) report [112].

This thesis proposes a solution capable of overcoming the above limitations by adopting a unified coupling framework for interworking between heterogeneous networks [35]. This framework provides a MN the highest possible level of internetworking, with fully seamless continuity of service (mobility) across heterogeneous networks. As an arbitrator for internetworking between the WLAN and the WiMAX systems with the 3G Cellular network (either UMTS or CDMA2000), the IMS is deployed.

A clear advantage of using the IMS is its ability for real-time session negotiation and management using the SIP [113]. Nevertheless, MIP also plays an important role in the IP (terminal) mobility management in the proposed approach. In order to set the ground for introducing our proposed interworking mechanisms the next sections provide a comprehensive overview of the 3GPP and 3GPP2's IMS architectures.

2.10.1 3GPP IMS Architecture

In UMTS, prior to Release 4, IP connectivity was a service provided by the core network [114]. However, this was only limited for standardization of basic services such as initiation, modification, and termination of multimedia sessions. Even when the wireless provider offered (non-standard) enhanced IP services, end-to-end service provisioning across network boundaries was infeasible. Hence, subscribers had to use a variety of third-party applications and use third-party providers of IP multimedia services [115].

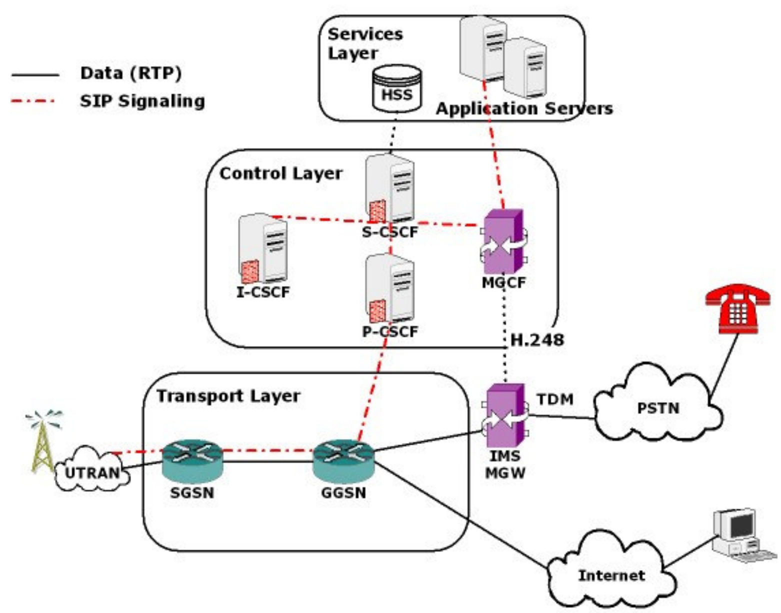


Fig. 2.14. The IP Multimedia Subsystem (IMS) Architecture.

The 3GPP's UMTS Release 5 [116] and 6 [117] introduced the IMS to the UMTS core network to overcome the above drawbacks [31]. The IMS comprised the required characteristics for controlling of multimedia sessions, and thus essential for the provisioning of IP multimedia services in UMTS. It also provides for an entry point for third-party multimedia applications and services in a controlled and secure manner. In order to facilitate these services, the IMS is featured with a number of key mechanisms such as session negotiation and management, QoS, mobility management and provisioning for AAA.

Figure 2.14 provides a general overview of the IMS architecture, outlining the essential network elements used for providing real-time IP multimedia services [31]. Since the IMS architecture specifically relies on the packet-switching domain for transport and local mobility management, the IMS operates independently to the circuit-switched domain. Thus circuit-switching elements such as mobile switching centers have been excluded from Fig. 2.14.

2.10.1.1 Call Session Control Functions

The Call State Control Functions (CSCF) and the Home Subscriber Server (HSS) are the key elements of the IMS, which play a vital role in call/session control. By and large, the role of a CSCF can be seen analogous to the SIP Server in the IETF architecture [118]. They are essentially involved in processing signaling messages for controlling multimedia/call sessions for users. Apart from this, the CSCFs are also involved in the address translation, performing service switching and vocoder negotiation, and handling of the subscriber profile. Depending on various configurations and scenarios, the roles for the CSCFs are categorized as Proxy CSCF, Interrogating CSCF, and Serving CSCF.

2.10.1.2 Proxy CSCF

The Proxy-CSCF (P-CSCF) is the first contact point of the IMS, which is located in the same network as the GGSN. This could either be in the home network or the visiting network. The P-CSCF has two main functions. Its primary function is to be the QoS policy enforcement point within the IMS. Its secondary responsibility is to provide the local control for emergency services. It also provides the local numbering plans directory assistance. The P-CSCF forwards the SIP registration messages and session establishment messages to the home network of the user. The P-CSCF can be seen as equivalent to a proxy server in an IMS architecture.

2.10.1.3 Interrogating CSCF

The I-CSCF is the “main entrance” to the home network from a visited network. With the assistance of the HSS, the I-CSCF selects the appropriate Serving-CSCF (S-CSCF). This is an optional node in the IMS architecture as the P-CSCF may also be configured to contact the S-CSCF directly. Nevertheless, the I-CSCF has a number of functionalities. It performs load balancing between S-CSCFs with the support of the HSS. The I-CSCF can hide the home network form other network operators by providing a single point of entry into the network. It is also capable of performing some forms of billing. Lastly, since the I-CSCF acts as the gateway into the home network, it also supports the firewall function.

2.10.1.4 Serving CSCF

The S-CSCF is the node that eventually performs the actual user registration and session control for the IMS network. The S-CSCF may also be capable of provisioning a set of specialized services. There can be several S-CSCFs in the network. They can be added as needed based on the capabilities of the nodes or the capacity requirements of the network. One key advantage of this architecture is that the home network provides the service features to the MN. Thus the capabilities of the visited network do not restrict the functionality of the MN.

2.10.1.5 Home Subscriber Server

The HLR has evolved into the Home Subscriber Server (HSS). Interfacing with the I-CSCF and the S-CSCF, the HSS can be regarded as a master database, which acts as a repository for subscription and location information. Since multiple HSS functions may be available in a network, the subscription location function is queried by CSCFs during registration or session initiation. The HSS uses a non IETF based protocol, the C_x interface, which is an IP based protocol [119].

2.10.1.6 Media Gateway Control Function and Media Gateway

In an all-IP environment, there would be no need for anything other than the CSCFs and the HSS. However, this is not the case in reality. Hence the IMS must have facility to interwork with the legacy Public Switched Telephony Network (PSTN) and mobile networks. Therefore, the Media Gateway Control Function (MGCF) interconnects with circuit switched networks via the corresponding IMS Media Gateway (IMS-MGW). This feature supports for scenarios where sessions are established between the packet switching domain and the PSTN. The media translation from signals encoded in one format to another is performed by the IMS-MGW. Also there is the Media Resource Function Controller (MRFC), which performs processing of media streams through the corresponding Media Resource Function Processor (MRFP). Further discussion on the IMS-MGW and MGCF are not included since it is beyond the scope of this section.

2.10.1.7 IMS Related Protocols

The protocols which have been defined within the IMS architecture can be classified under thee broad categories. The first category, which is also the most important, comprises of the protocols used in the signaling and session control plane. The second and third categories consist of protocols used in the media plane and protocols used for authentication and authorisation.

2.10.1.8 The Session Control Protocol: SIP

The SIP is the core protocol chosen by the 3GPP to perform signaling and session management within the IMS [118]. It is essentially a standard application layer protocol designed to facilitate call information to be carried across network boundaries and control the attributes of end-to-end sessions in IP based networks. Therefore, SIP is not a vertically integrated communication protocol. It is rather a component that needs to be used with other IETF protocols to build a complete multimedia architecture.

The capabilities of SIP can be listed as:

- Determine the location of the target end point: SIP supports address resolution, name mapping, and call redirection,
- Determine the media capabilities of the target end point: SIP determines the lowest level of common services between end points using the Session Description Protocol (SDP),
- Determine the availability of the target endpoint: If the call cannot be completed, SIP determines why the target end point was unavailable,
- Establish a session between the originating and target end point: If the end point is located, SIP establishes a session. It also supports mid-call changes, and
- Handle transfer and termination of calls: SIP supports the transfer of calls form one end point to another.

It is also an extensible protocol, which enabled the IETF to introduce new methods, define new headers and easily integrate in to the core protocol. This extensible nature of the protocol was used for incorporating additional features to suit the needs of the 3GPP's proposed IMS architecture. These new 3GPP SIP extensions can be broadly classified as general, session operation, QoS, AAA, and security. A detailed description and a comparison of the IETF SIP versus the 3GPP SIP is available from [105].

2.10.1.9 Other Protocols

As previously mentioned, the remaining protocols used within the IMS architecture can be broadly categorized as protocols used in the media plane and the AAA protocols. The IMS uses the Real Time Protocol (RTP) for transporting real time audio and video over the User Datagram Protocol (UDP) as the transport protocol [120]. It further uses the Real Time Control Protocol (RTCP) for providing the RTP with QoS statistics and information to provide inter-media synchronization on the media stream [120]. Lastly, the authentication protocol used by the IMS is DIAMETER [121]. Further details on the above protocols (RTP, RTCP and DIAMETER) are not included as it is beyond the scope of this thesis.

2.10.2 3GPP2 IMS Architecture

The 3GPP2's Multi Media Domain (MMD) is a collection of core network entities providing 3G capabilities, which is based on IP protocols, elements, and principles. The MMD of the all-IP Network comprises of multimedia session capabilities built on top of the general packet data supporting capabilities. The general packet data support portion of the MMD is known as the Packet Data Subsystem (PDS) and the entities that facilitate the multimedia session capabilities in an all-IP network is collectively known as the IMS [111]. The initial release of the 3GPP2-IMS was based on the IMS specified in Release 5 of the 3GPP specifications [122], [123]. As per the illustration in Fig. 2.15, the key elements of the 3GPP2-IMS architecture are much similar to the descriptions provided in Fig. 2.14.

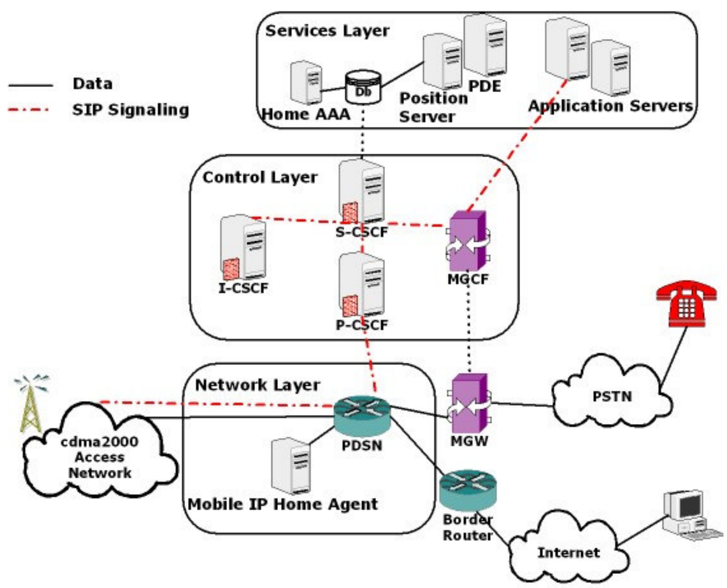


Fig. 2.15. The 3GPP2-IMS Architecture.

Despite the fact that 3GPP2-IMS specifications closely follow their 3GPP-IMS counterparts, substantial differences exist between them [124]. Firstly the issue of mobility must be looked since it is handled in two different ways. The 3GPP2-IMS uses MIP for IP mobility (i.e., terminal mobility) management and SIP for session mobility management where as 3GPP IMS exclusively uses SIP for all types of IP mobility management. Secondly, attention can be drawn to the use of IPv4 vs. IPv6. Although 3GPP mandates the IMS over IPv6, the 3GPP2 IMS specification does not make such distinction.

As a common interworking platform for mobility management in NGMN is considered, resolving the interaction of 3GPP-IMS with MIP becomes a new challenge. Therefore, the framework proposed in this thesis uses MIP for providing terminal mobility and limits SIP for session mobility management in a generalized IMS architecture.

As previously mentioned, although 3GPP mandates IMS over IPv6, 3GPP2 does not make such a specification. Therefore, for 3GPP2, either MIPv4 [125] over IPv4 or MIPv6 [126] over IPv6 can be used, which also gives more freedom for network operators. As per the specifications of 3GPP2, if MIPv4 is deployed, the PDSN of the home network may act as the MIP HA. As the MN moves form one PDSN to another, the new PDSN may act as the MIP FA, which becomes the new point of contact for the MN.

Therefore when the MN is attached to a different PDSN the HA becomes an anchor point for the MN's traffic flow, particularly when reverse tunneling and triangular routing are taking place. In the event when MIPv6 is being implemented, a direct peer-to-peer communication can be established through its route optimization operation [126]. Since 3GPP2's standard for CDMA2000 does not fully support inter-PDSN mobility for IPv6, the proposed design will primarily be based on MIPv4 [127]. As a result the route optimization option will not be available in its current form. However, transition of this model to MIPv6 can be easily done by addressing user authentication, address allocation and enabling the MN to perform the MIPv6 update procedures [128].

Although it is of lesser importance, other differences between 3GPP and 3GPP2 IMS can be briefly stated as:

- 3GPP2's Home AAA and its database being equivalent to 3GPP's HSS. 3GPP2-IMS is not using PDP context activation for P-CSCF discovery,
- Unlike in 3GPP, the P-CSCF and PDSN do not need to reside in the same network,
- 3GPP2-IMS defines a new position server and a position determining entity for providing positioning information, and
- The other differences are codecs, QoS procedures, S-CSCF/P-CSCF/HSS interfaces, authentication procedures to name a few.

Detailed discussions on the above are not provided as these issues are beyond the scope of this thesis.

2.11 Summary and Conclusions

The purpose of this Chapter was to provide an in-depth discussion on the current architectures, research issues, and future trends on interworking heterogeneous mobile data networks in a NGMN. The discussion was initiated by introducing the current and the most notable interworking architectures and related design issues. Amongst the discussed interworking architectures were the three coupling methods (i.e., Tight Coupling, Loose Coupling, and Pee-to-Peer Networking), 3GPP's proposed approach for interworking, 3GPP2's proposed approach for interworking, the IEEE's proposed approach for interworking (i.e., IEEE 802.21 MIH), and other approaches adopted for interworking dissimilar networks such as different 3G cellular networking technologies and BWA networks. This Chapter also discussed the open issues and research trends in interworking and introduced the IMS as a potential candidate for a coupling media for interworking dissimilar networks. Finally the Chapter concludes by explaining the 3GPP's and 3GPP2's IMS architectures, which facilitates the necessary groundwork for the introduction of a novel interworking framework in the next Chapter.

Chapter 3

A Basic Framework for Interworking WLAN and UMTS Networks

This Chapter proposes a novel architecture for interworking WLAN-UMTS by using the IMS as a unified coupling media for real-time session negotiation and management. As a result, mobile users will be able to use WLANs as a complementary technology for 3G cellular data networks, thus experience ubiquitous data services and very high data rates across heterogeneous networks while providing the end user with a seamless experience. The remainder of the Chapter is organized as follows: Firstly the introduction section reiterates the motivations for this work and the importance of an interworking framework for heterogeneous networks. The next section describes the proposed architectural framework followed by a discussion on various interworking scenarios of the design. Subsequently OPNET based simulation model is introduced for validation. Finally the section on analytical modeling and performance evaluations are presented. The contents provided under this Chapter have contributed to the following publications: The interworking architecture and the OPNET simulation platform presented in this Chapter has contributed to [35], the basic analytical model has contributed to [36] and [37], the extended analysis using the Queuing Theory has contributed to [38], and lastly the signaling cost analysis method has contributed to [39].

3.1 Introduction

As modern high-speed data applications tend to impose a challenge on the bandwidth limitations of existing 3G cellular networks, a strong urge for the development of efficient mechanisms for interworking these with WLAN technologies have been raised. In the event of successful interworking of theses two technologies, mobile users will be able to experience ubiquitous data services and very high data rates across heterogeneous networks by using

WLANs as a complementary technology for next generation cellular data networks, while providing the end user with a seamless experience [25]. This will enable a user to access 3G cellular services via a WLAN, while roaming within the range of a hotspot.

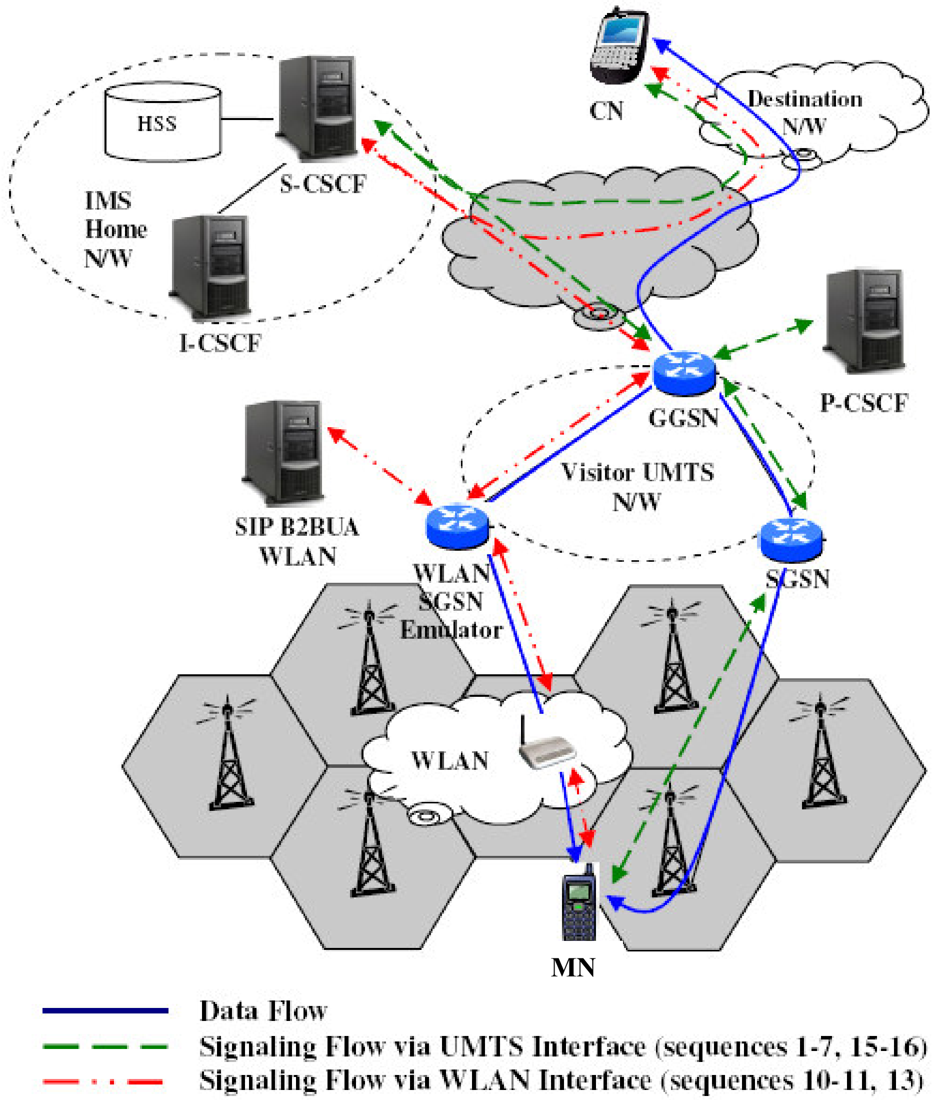
With the aim for addressing this need, a variety of internetworking architectures for 3G Cellular and WLANs have been proposed [31]. By and large, these internetworking architectures may be categorized as tight coupling, loose coupling, and peer-to-peer networking (also referred as no-coupling) [26], [27]. However, these approaches seem to provide limited internetworking capability as none of these designs have successfully addressed the issue of seamless continuation of services. A detailed and comprehensive discussion on the current research activities and trends in relation to interworking of dissimilar networks was presented in the previous Chapter.

Having realized the importance of such an interworking mechanism, our work was motivated towards developing a solution that is capable of achieving the highest possible level of service continuation during the vertical handoff of sessions between such heterogeneous networks. Therefore, this Chapter presents the proposed architecture for WLAN-UMTS interworking, which is capable of meeting these challenges as initially published in [33] and [34]. The novelty of this proposed solution for WLAN-UMTS interworking is that it uses the 3GPP's IMS as an arbitrator for real-time session negotiation and management.

3.2 Architectural Framework

The recommended framework for internetworking is illustrated in Fig. 3.1. The flow of data originates from the source MN, through the SGSN and the GGSN of the visiting UMTS network, and reaches the destination network. This model uses the Visitor-GGSN approach to avoid the inter-PLMN backbone and to make data routing simpler for the network operator [129]. In whichever the approach, the data flow bypasses the IMS network. Thus the IMS is said to follow the philosophy of having different paths for user data and signaling through the network.

The SIP signaling messages originate from the MN via the UMTS Terrestrial Radio Access Network (UTRAN), through the SGSN and GGSN, out to the CSCFs and finally to the



CN – Corresponding Node
GGSN – Gateway GPRS Support Node
SGSN – Serving GPRS Support Node
B2BUA – Back to Back User Agent
IMS – IP Multimedia Subsystem
P-CSCF – Proxy Call Session Control Function
I-CSCF – Interrogating Call Session Control Function
S-CSCF – Serving Call Session Control Function
HSS – Home Subscriber Server

Fig 3.1. Architecture for WLAN-3G Cellular Internetworking.

destination network. It is important to note that when the MN requires establishing a session, this request is always sent to the (Home) S-CSCF via the (Visiting) P-CSCF. During the exchange of the SIP signaling, both the SGSN and the GGSN act as routers by merely forwarding SIP messages. The data originating from the WLAN gets routed via the SGSN emulator through the GGSN. Hence, it essentially emulates the WLAN as another SGSN belonging to the same UMTS network. Thus mobility can be managed by the UMTS network. Some of the functionalities of the BSS are bypassed in this approach and the load on the UMTS network, created by the high volume of the WLAN data traffic, may also be sufficiently reduced. Furthermore, the MN does not require any change of IP addressing between the WLAN and the UMTS network as long as the two networks are connected via the same GGSN.

3.2.1 Session Establishment

The establishment of a SIP session within an UMTS IMS environment is involved with several functions. The key steps required for obtaining access to SIP services can be outlined as follows. The first step involves with the MN powering on and locking to the UMTS network. It is assumed that this step is already performed by the MN, and will not be discussed in detail. Once the above mentioned system acquisition is done, the next step is to establish a data connection, or set up a data pipeline, for the SIP and other services. In order to perform the SIP session establishment the MN is initially unaware of the IP address of the P-CSCF. Thus the data connection must be completed in two-steps by using the Attach [130] and PDP context activation [131] message sequences. The activation of a PDP context assigns an IP address for the MN. With the activation of the PDP context the MN is able to identify the P-CSCF for the registration with the UMTS SIP network.

Prior to establishing a SIP session, the MN requires performing a service registration function to let the IMS know of its location. The MN acts as a SIP client and sends a SIP registration message to its home system through the P-CSCF. The basic steps for a SIP service registration can be summarized as follows. Firstly, the HSS for the MN is notified of its current location for the HSS to update the subscriber profile accordingly. Next the HSS checks if the MN is allowed to register in the network based on the subscriber profile and operator limitations, and grants authorization. Once authorized, a suitable S-CSCF for the MN is assigned and its subscriber profile is sent to the designated S-CSCF.

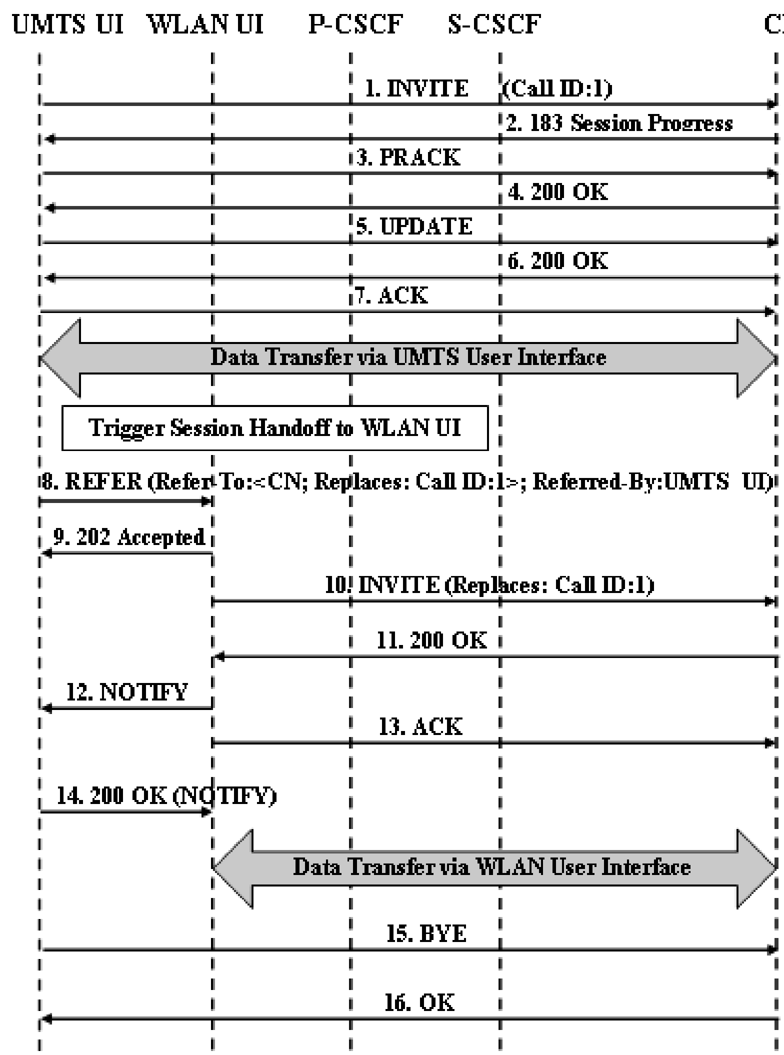


Fig. 3.2. IMS-SIP Based Session Handoff.

After the activation of the PDP context and the service registration, the MN is ready to establish a media/data/call session. As illustrated in Fig. 3.2, the sequence of the SIP session origination procedures can be described as follows. The mobile origination procedure is initiated by a SIP INVITE message sent form the UMTS interface of the source MN (step 1).

This initial message is forwarded from the P-CSCF to the S-CSCF of the originating network, via the CSCFs of the terminating network, and finally to the Corresponding Node of the destination. This SIP INVITE request carries a Session Description Protocol (SDP) [132] body indicating the IP address and port numbers where the source wants to receive the media streams. Furthermore, the INVITE also contains a request to follow the precondition call flow model. This is important because some clients require certain preconditions (that is, QoS levels) to be met before establishing a session. The requirement for using the preconditions call flow model in IMS is mainly because in cellular networks radio resources for the media plane are not always available. If the preconditions extension was not used, when the called party accepts the call, the source and destination may not be ready and consequently the first few packets may be lost.

Next, this model requires that the destination responds with a 183 Session Progress containing a SDP answer (step 2). The SDP answer contains the media streams and codecs that the destination is able to accept for this session [132]. The acknowledgement for the reception of this provisional response by a Provisional Response ACKnowledgement (PRACK) [133] request follows afterwards (step 3). If the destination does not receive a PRACK response within a determined time, it will transmit the provisional response. When the PRACK request successfully reaches the destination a 200 OK response is generated by the destination with an SDP answer (step 4). Next an UPDATE request [134] is sent by the source containing another SDP offer, in which the source indicates that the resources are reserved at his local segment (step 5). Once the destination receives the UPDATE request, it generates a 200 OK response (step 6). Once this is done, the MN can start the media/data flow and the session will be in progress (via the UMTS interface).

3.2.2 Session Handoff: UMTS to WLAN

When the MN, which is currently connected to the UMTS network, detects the presence of a WLAN around its vicinity, vertical handoff procedures may be initiated. This can specifically be seen as a make-before-break instance of vertical handoff where there exists an overlapping coverage. As per the illustration in of the overlapping coverage scenario in Fig. 3.3, the WLAN signal strength is first observed over time t_0 to t_1 (say). As the intensity of the WLAN signal starts growing from *L* (say) and exceeds a certain threshold level *H* (say), the network selection algorithm will start making a decision for handoff. Apart from the WLAN signal strength, the network selection procedure will also be considering other conditions such as available/required bandwidth, delay, and user preferences. Since network selection criteria are beyond the scope of our current research, a detailed discussion will not be included.

However, once such a decision has been made, the next step is to activate the existing WLAN interface and initiate the IMS-SIP based handoff mechanism. As the WLAN interface becomes active, the ongoing media sessions may be handed-over any to its newly activated WLAN interface. This is where the need of a mechanism for a pure SIP (or Application Layer) based session handoff arises. This is because the IMS performs pure SIP based signaling at the Application Layer and IPv4 or IPv6 mobility can not directly support such session mobility. Current works point out that at least three such mechanisms for achieving the above exist [135]. These are namely; the SIP ReINVITE method with new location information [118], the third-party call control (3pcc) mechanism [136], and the SIP REFER method [137]. The SIP Re-INVTE method is the simplest mechanism for an ongoing session handoff with the least overheads. This method can successfully be implemented in an interworking environment where the IP address remains static or belongs to the same pool of addresses as in [35].

However, in a situation where the IP address allocation does not come from the same pool of addresses, it lacks the ability for handling terminal mobility at the Application Layer, thus, is incapable of providing seamless session handoff. Therefore, SIP ReINVITE method has been ruled out. On the other hand, the second mechanism, 3pcc method has the disadvantage of requiring the original session participant (interface in this case) to always be contacted as a proxy for all future session modifications. That is, if the WLAN interface was the original session participant, availability of a WLAN must always be there for it to be contacted. This is infeasible since it cannot be always guaranteed that a particular network service (WLAN in this case) will always be available when the original session participant interface needs to be contacted.

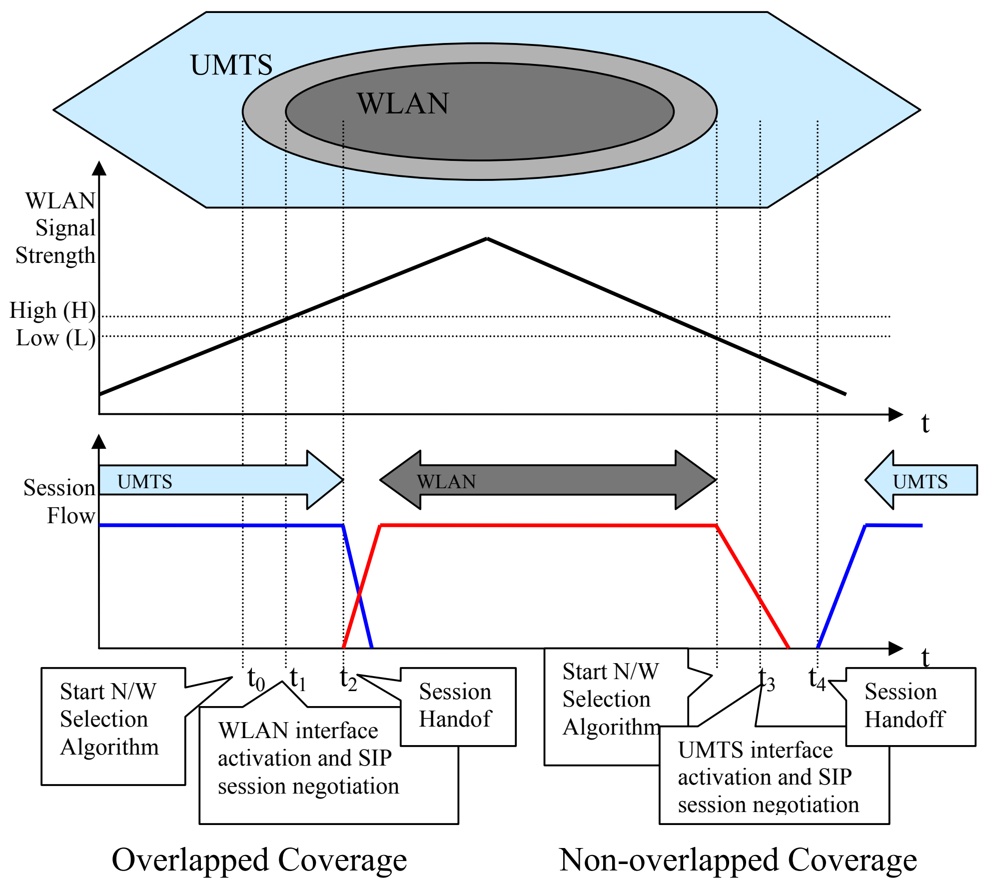


Fig. 3.3. Vertical Handoff Scenarios for Overlapped and Non-Overlapped Coverage.

Hence the 3pcc method has also been ruled out. However, the SIP REFER method is capable of explicitly transferring the session to the new interface. Furthermore, it is capable of ensuring terminal mobility as well as session mobility at the Application Layer, and hence becomes the obvious choice for the given task. Also, under realistic conditions, vertical handoff decision must ideally be triggered by a network selection mechanism. Since the network selection criteria are beyond the scope of this thesis, a manual triggering for handoff is considered. It is also worth noting that all activated interfaces (WLAN, say) need to perform a SIP registration function with the S-CSCF of the originating home network.

The basic steps for an IMS-SIP based session handoff is illustrated in Fig. 3.2 and can be described as follows. The UMTS interface notifies the WLAN interface with a SIP REFER request (step 8). The REFER request contains a "Refer-To" header line containing the destination SIP Uniform Resource Identifier (URI) and a "Replaces" header line identifying the existing session to be replaced by the new session. Next the WLAN interface sends the corresponding node an SIP INVITE message with the "Replaces" header received from the previously received REFER request (step 10). Also the new IP address and port numbers are also included in the SDP body of this INVITE message. The receipt of the "Replaces" header is what indicates that the initial session is to be replaced by the incoming INVITE request and hence be terminated. Now the WLAN interface has successfully established a direct signaling relationship with the corresponding node. Once the WLAN interface has successfully established a session with the corresponding node, it sends a NOTIFY request to the UMTS interface updating the final status of the REFER transaction (step 12). This NOTIFY message contains the session information of the newly established session allowing the UMTS interface to subsequently retrieve the session (if so desired). Once the data flow is established between the WLAN and the corresponding node, the UMTS interface tears down its session with the corresponding node (steps 15-16). Also note that in the event that the provided information in the replaced header does not match an existing session the triggered SIP INVITE does not replace the initial session and will be processed normally. Thus any failed session handoff attempt can not destroy the initial session.

3.2.3 Session Retrieval: WLAN to UMTS

When the MN roams out of the WLAN, as illustrated in Fig. 3.3, the situation relates to a non-overlapped coverage. That is, in the event of a fast roaming user, the WLAN link may be lost before MN reverts back to the UMTS interface. Thus there is a higher likelihood that a break-before-make instance of vertical handoff may take place. (It is also worth noting that as a result of a sudden drop in signal strength, a break-before-make handoff may result in situations where coverage is overlapping). In the non-overlapped coverage scenario depicted in Fig. 3.3, the WLAN is the preferred interface. As the WLAN signal strength starts dropping rapidly the network selection algorithm will be activated at time t_3 (say). Next, at time t_4 the UMTS interface activation and the IMS-SIP based session handoff mechanism will take into effect. During this time frame the WLAN signal strength will further deteriorate and consequently coverage disruption may result between t_4 and t_5. The typically long UMTS call setup delay may further worsen the handoff delay for such a non overlapped break-before-make scenario. Nevertheless, due to various network conditions there could be other delays in the order of SIP requests reaching the destination.

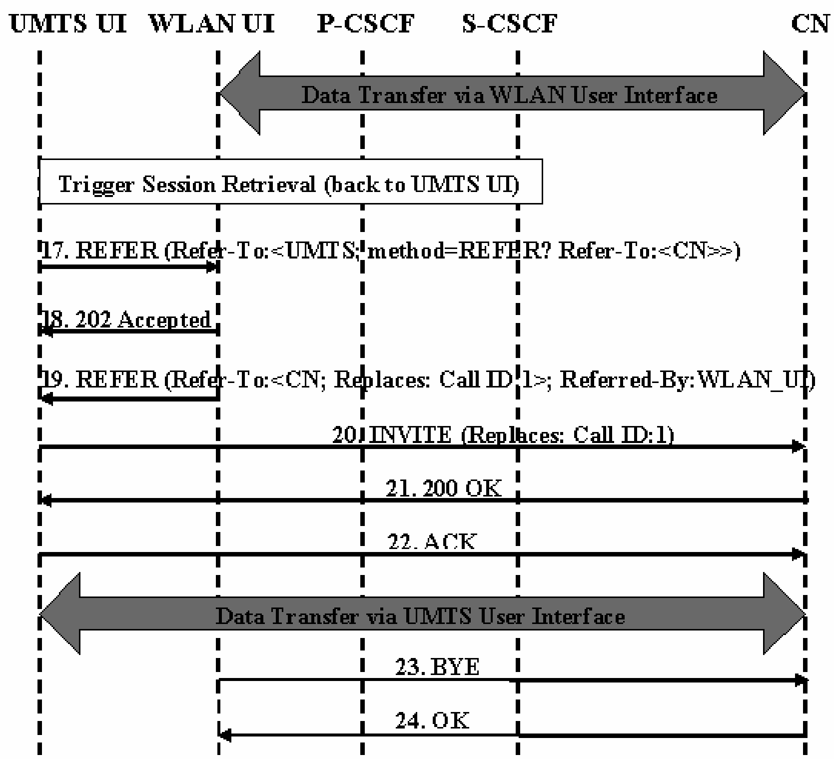


Fig. 3.4. *IMS-SIP Based Session Retrieval.*

Therefore, in the event when the UMTS interface wishes to retrieve a session which has been previously handed-off to a WLAN interface, the message flow illustrated in Fig. 3.4 takes place. As previously discussed, the UMTS interface must receive a SIP REFER request from the WLAN interface with its URI included in the "Referred-By" header (step 19). However, in order to prompt a REFER request from the WLAN interface is for the UMTS interface to send a "nested REFER", which is a REFER request for another REFER according to RFC 3892 (step 17). The "Refer-To" header of a nested REFER request specifies the UMTS interface as the refer target and that the referral be in the form of a REFER request. Next the UMTS interface is ready for initiating a session with the corresponding node by sending a SIP INVITE with "Replaces" and "Referred-By" headers, which will replace the corresponding node's existing session (steps 20-22). When the corresponding node accepts the INVITE request from the UMTS interface and once the session is established, the session with the WLAN interface will be terminated (steps 23-24).

3.3 Common Scenarios of Interworking

As a benchmark for identifying the capability of interworking, the suggested architecture is assessed against the scenarios defined by the 3GPP TR 22.934 [65]. As discussed under Section 2.5, this framework identifies and discusses six common interworking scenarios [34]. Prior to discussing the compliance of the proposed architecture, the six common interworking scenarios will be briefly revisited.

The first scenario is only capable of providing common billing and customer care to the subscriber, thus provides no real interworking between WLAN and UMTS PLMN. The second scenario limits the UMTS subscriber with a basic IP network connection through a WLAN. Scenario three extends the access to UMTS PS services for subscribers in a WLAN environment.

However, it does not guarantee service continuity across these access networks. Scenario four helps maintain service continuity across UMTS and WLAN radio access technologies. Scenario five takes it another step further by introducing seamless service continuity between the UMTS and WLAN. Finally, scenario six ensures access to UMTS CS services via a WLAN system including seamless mobility. The following sections discuss how different scenarios of interworking may be achieved form the presented architecture.

3.3.1 Scenario 1: Common Billing and Customer Care

The first scenario includes common billing and common customer care. As the WLAN routes the data traffic via the visiting network, an acceptable charging model based on the quantity of traffic exchanged can be incorporated into the UMTS billing system.

3.3.2 Scenario 2: UMTS based Access Control and Charging

The second scenario uses the UMTS access control procedures for WLAN users within the UMTS domain. It also includes features of scenario 1. Facilities exist for both WLAN and UMTS networks for accessing the HSS, and the charging and billing systems. However, interworking at this level does not require any negotiation of SIP sessions between the elements of the IMS. Since the sole purpose of the new architecture is to use the IMS based services, the next level of interworking must be considered.

3.3.3 Scenario 3: Access to UMTS IMS-based Services

This level extends the services of the IMS to a MN connecting via a WLAN. However, this scenario lacks the service continuity, which may require the MN to re-establish the session in the new access network. As previously described in Section 3.2, to facilitate SIP signaling between the IMS and the WLAN, the IETF SIP elements of the WLAN must be translated into the 3GPP SIP. Therefore, interworking between the CSCFs and the WLAN SIP proxy becomes a challenging task with a top priority. The authentication and access control for IMS services takes place during the SIP registration. This also facilitates mutual authentication between the MN and the UMTS network. It is also important to manage billing and accounting procedures of the UMTS network for the interworking scenario.

3.3.4 Scenario 4: Service Continuity

Service continuity requires the levels of interworking as defined in the previous two scenarios (two and three). However, it does not guarantee seamless continuity of service, which means that some applications may require re-establishing of sessions. Therefore, mobility management (roaming and hand-off) need to be considered under this level. The 3GPP's enhanced SIP services, used in the IMS, are capable of providing service continuity. However, the problems to be solved at this level mainly exist in the transport network. Thus mechanisms for supporting efficient ways for data routing in the WLAN and the UMTS interworked environments need to be identified. The suggested solution for efficient data routing is to have the WLAN emulate a SGSN, which connects to the UMTS network at the G_n interface. Thus data traffic from the WLAN is routed via the SGSN emulator to the UMTS network. The SGSN emulator approach may also be substantially reduced.

3.3.5 Scenario 5: Seamless Continuity of Service

The scenario 5 is very much similar to the Scenario 4. However, it includes seamless service continuity with fully transparent services to the end-user. This is an issue which is not addressed so far by the UMTS Release 6, and is currently being investigated by the 3GPP. Regardless of the numerous obstacles and complexities in achieving this goal, it is important to note how the suggested interworking model makes an important contribution towards reaching a step closer to the objective of seamlessness. The mobility management plays a vital role in achieving seamless continuity of service. The IMS architecture can also be used as a key mediator for reducing or eliminating potential issues such as resource allocation in interworking. Furthermore, this interworking model addresses the current issues in IP address allocation and distribution. Therefore, a MN does not require any change of IP addressing between the WLAN and UMTS network as long as the two networks are connected to the same GGSN.

3.4 Network Modeling

For validating the potential for interworking of the presented architecture, a network model is constructed and simulated using the OPNET Modeler 11.5 [35]. Since OPNET's standard SIP model components do not address the specifications of the 3GPP's IMS, substantial modifications are required. Some of the main drawbacks of the OPNET's existing SIP model can be summarized as:

- Incapable of interacting between multiple SIP proxies and user terminals,
- Unable to make SIP proxies to signal between CSCFs needed to establish an IMS session (especially in multi-domain and roaming scenarios),
- Incapable of considering the full message exchange between CSCFs in the IMS, and
- Incapable of allowing the control of intermediaries' process delays (e.g. the delay incurred as a result of HSS lookup).

Therefore, modifications were made for:

- SIP Proxy Servers (User Agent Servers (UAS)) to function as different CSCFs,
- UAC processes to communicate with modified UASs,
- IMS-SIP based messaging and flow between the CSCFs,
- Roaming facility enabled between multiple domains, and
- Process delay controls introduced (i.e. for messages sent between CSCFs and the HSS queries).

There are new attributes in both the SIP Proxy Server model and the User Agent Client (UAC) model needed to configure the scenario. Fig. 3.5 illustrates the newly added attributes to a sample SIP proxy server configured as an S-CSCF in the example. Similarly, Fig. 3.6 shows the new attributes of the SIP UAC. Subsequently, a fully functional SIP-IMS model for OPNET was constructed and integrated to the OPNET's existing UMTS Special Module. The newly developed SIP-IMS model is an enhanced version of the basic IMS-SIP signaling model, which is currently available under the contributed models library of the OPNET University Program [138].

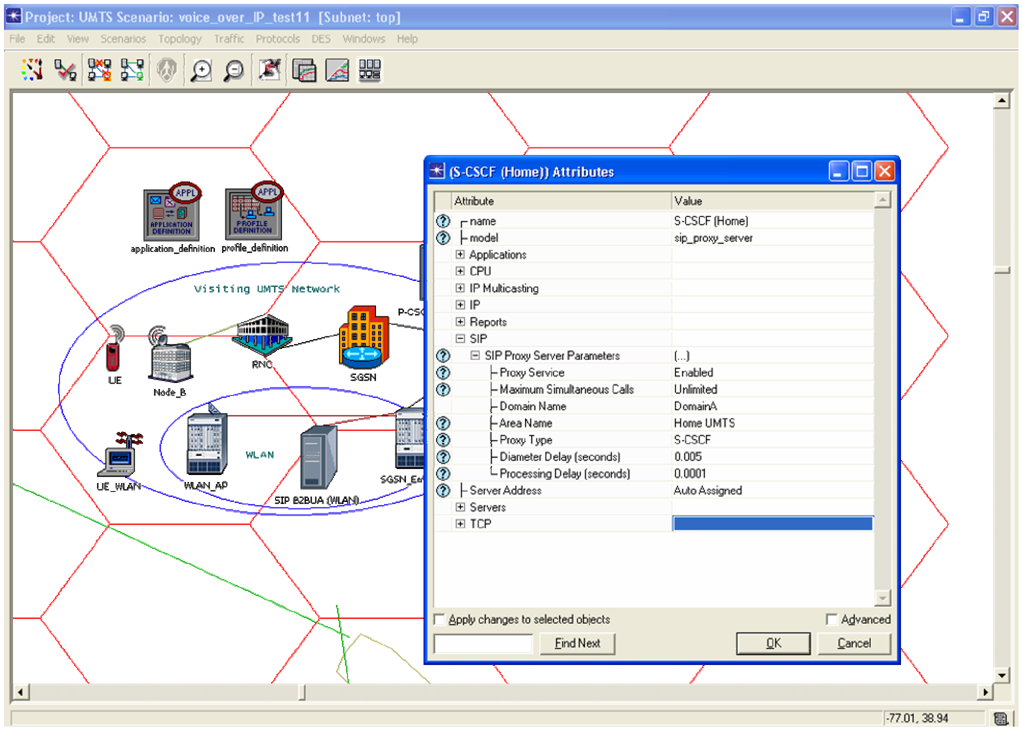


Fig. 3.5. The New Attributes of the IMS CSCF (SIP Proxy Server).

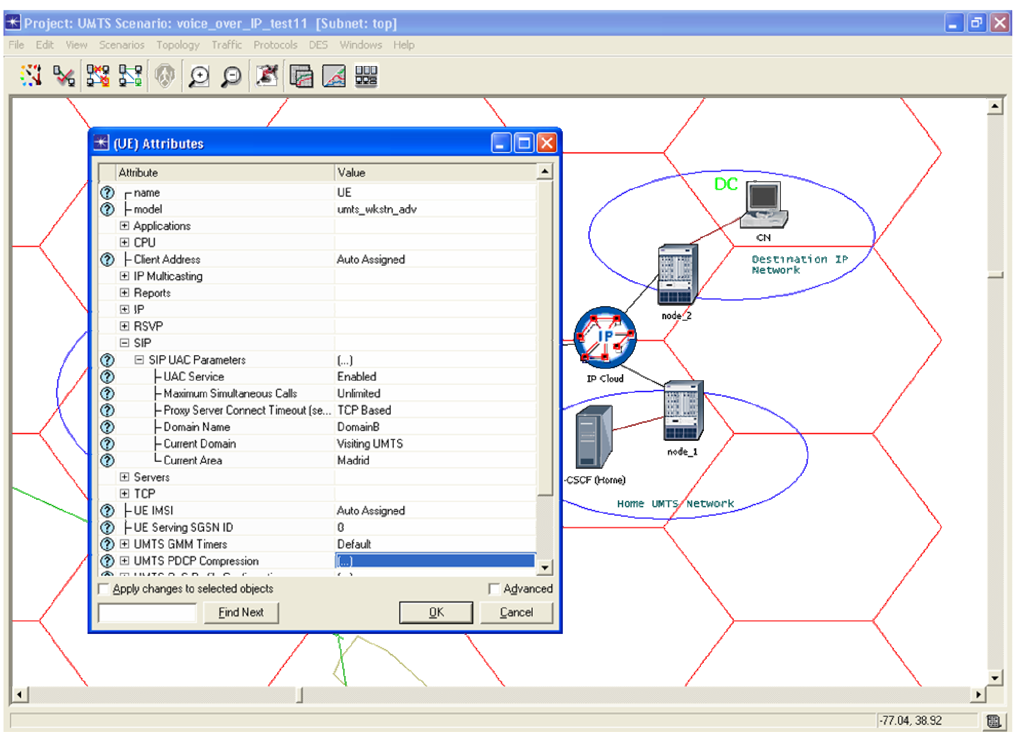


Fig. 3.6. The New Attributes of the SIP User Agent Client.

As a result, a UMTS network that is fully capable of using IMS based SIP signaling for session management was developed. Next a simple WLAN is connected via an SGSN emulator to the GGSN of the Visiting UMTS Network. The P-CSCF (WLAN) can be seen as a SIP Back-to-Back User Agent (B2BUA), which is capable of interworking with the IMS-SIP and capable of forwarding the SIP requests. The S-CSCF is the only IMS node implemented in the Home UMTS Network. This is since the I-CSCF is mainly used for the SIP Registration process and it is assumed that both the UMTS and the WLAN interfaces of the MN have already been registered. The corresponding node, which is an IMS-SIP UAC, is connected to a destination IP network via a public IP network (IP Cloud). The IMS-SIP message flow basically follows the sequence described in Fig. 3.2. Fig. 3.7 illustrates the constructed simulation scenario.

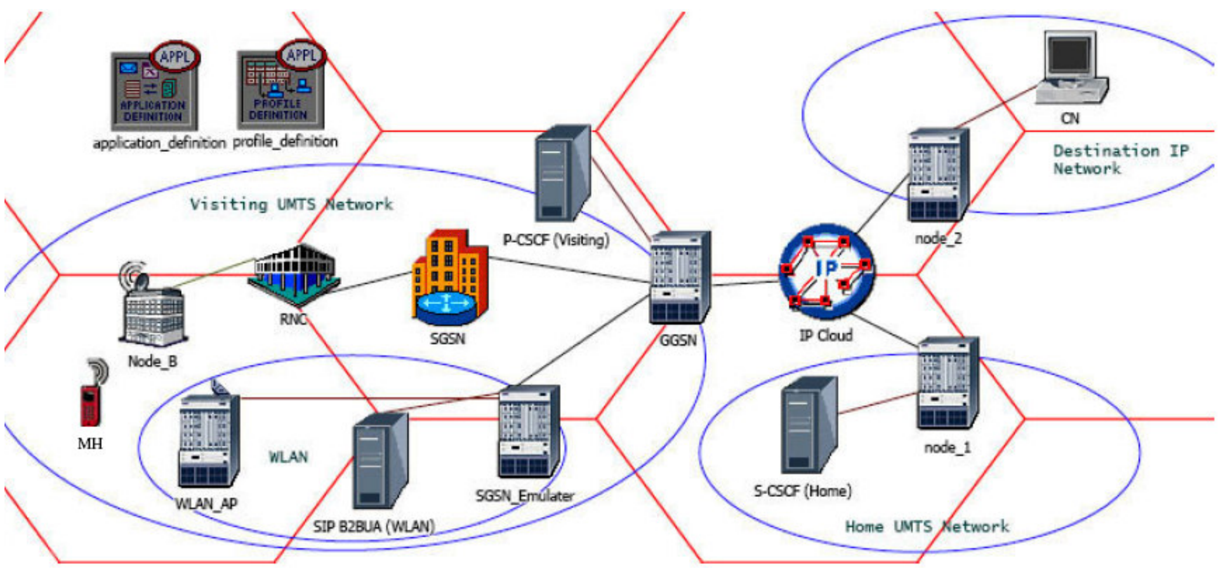


Fig. 3.7. The OPNET Simulation Model.

3.4.1 Simulation of Vertical Handoff

Using the newly developed IMS-SIP based platform a series of simulations are performed for evaluating vertical handoff for the previously described scenarios [35]. That is, firstly a make-before-break handoff (from UMTS to WLAN) and secondly a break-before-make handoff (WLAN to UMTS) were simulated. It is also important to note the assumptions made when obtaining these results. Both the UMTS network and the WLAN belong to different IP subnets where IP addressing and routing were statically assigned. Since there were no multiple networks available (except for one UMTS and one WLAN), the need for a network selection algorithm could be eliminated. Therefore, the handoff decisions are individually based on the signal strength of the WLAN, to which the MN either roams in to or roams out of.

Figure 3.8 indicates the numbers of active IMS-SIP sessions (above) and the corresponding application traffic flow (below) during a make-before-break handoff from UMTS to WLAN. This indicates that the proposed architecture is capable of providing acceptable levels of service continuity during a make-before-break handoff where overlapped coverage is present. The graphs also indicate the instantaneous presence of dual SIP sessions during the make-before-break handoff from UMTS to WLAN and a flow of seamless data flow from the UMTS interface to the WLAN interface. However, there is some indication of data duplication taking place (when both interfaces are active). This can be easily addressed at higher layers and therefore lies beyond the scope of this thesis.

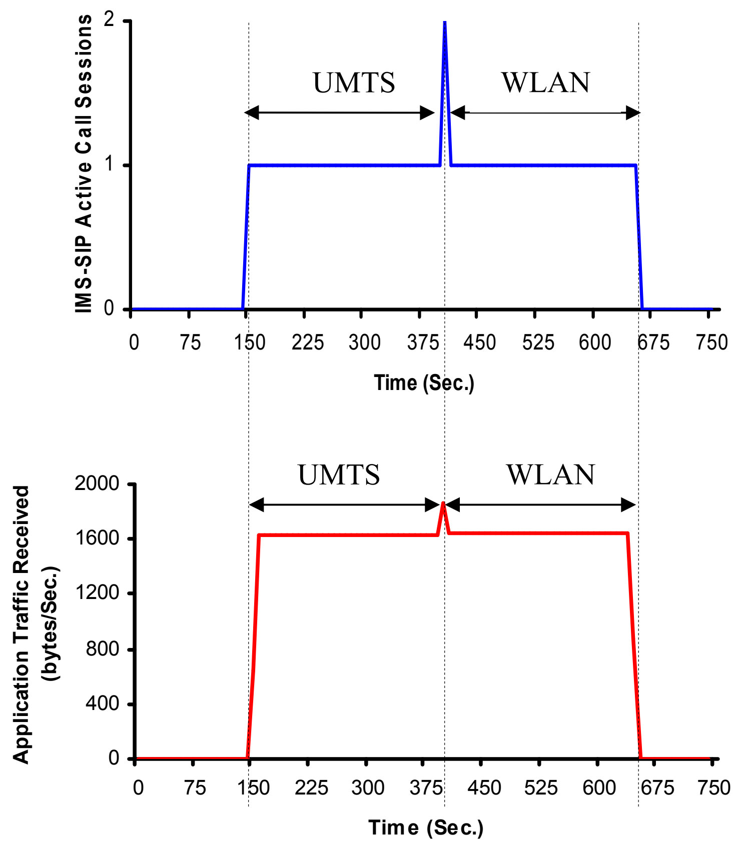


Fig. 3.8 Numbers of Active IMS-SIP Sessions (above) and Corresponding Application Traffic Flow (below) during a make-before-break Handoff from UMTS to WLAN.

On the other hand, Fig. 3.9 shows the numbers of active IMS-SIP sessions (above) and the corresponding application traffic flow (below) during a *break-before-make* type handoff from WLAN to UMTS during a non-overlapped coverage. The graphs indicate a brief service disruption delay as the handoff takes place for a non-overlapped coverage environment. Due to the resultant transient packet loss and delay, seamless continuity of service cannot be guaranteed, which means that some applications may require re-establishing of sessions.

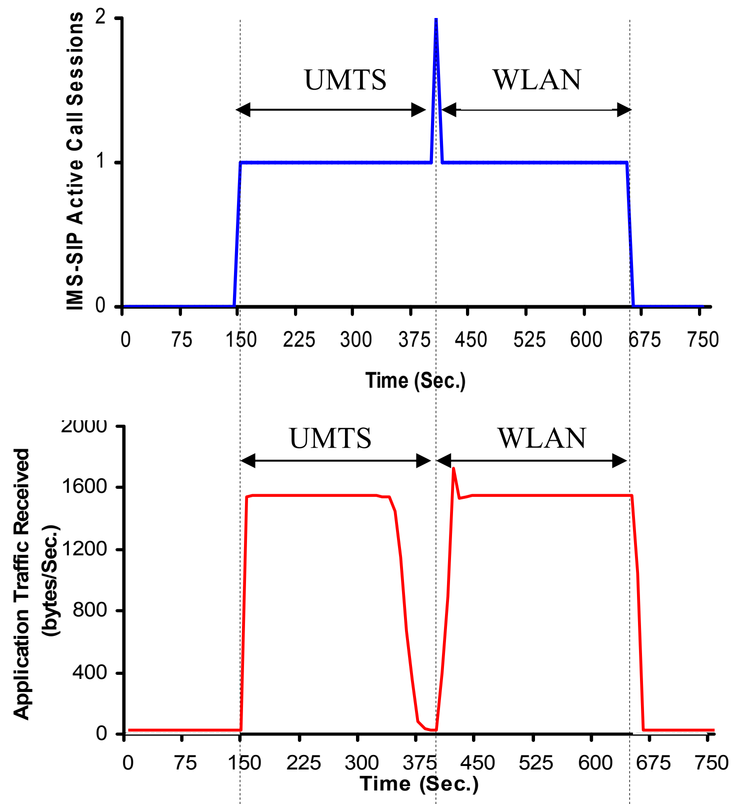


Fig. 3.9. Numbers of Active IMS-SIP Sessions (above) and Corresponding Application Traffic Flow (below) during a Break-before-make type Handoff from WLAN to UMTS.

3.4.2 VoIP Session Management

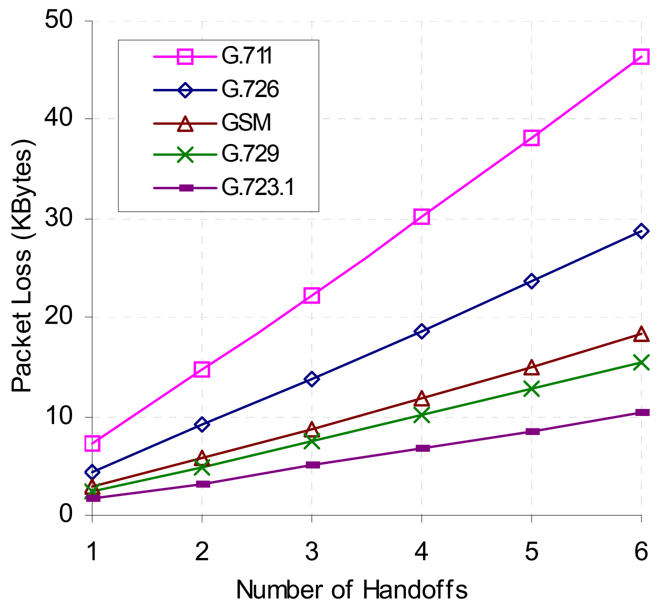
Using the same OPNET based simulation platform, the performance of VoIP session behavior in a similar interworking framework is further investigated and reported [139]. A series of VoIP connections are established for simulating VoIP session establishment and vertical handoff delays. Five different types of voice codecs are used for voice traffic generation. These are namely: G.711 (data rate of 64 kbps), G.726 (data rate of 32 kbps), GSM (data rate of 13 kbps), G.729 (data rate of 8 kbps), and G.723.1 (data rate of 5.3 kbps). Out of the above five codecs, G.723.1, GSM, and G.729 are currently the most widely used in GPRS and UMTS systems. Besides these, the generated voice packets are considered to have fixed IP header of 40 bytes, which includes a 12 byte RTP header, an 8 byte UDP header, and a 20 byte IP header.

Beyond this, depending on the transmission medium, an additional overhead of 34 bytes are added if the IEEE 802.11 Media MAC layer is used and a minimum overhead of 6 bytes are added if the UTRAN is used. Furthermore, no header compression option is considered at UTRAN Packet Data Convergence Protocol (PDCP) layer, no silence suppression is used, and no play-out buffer has been used to compensate the jitter.

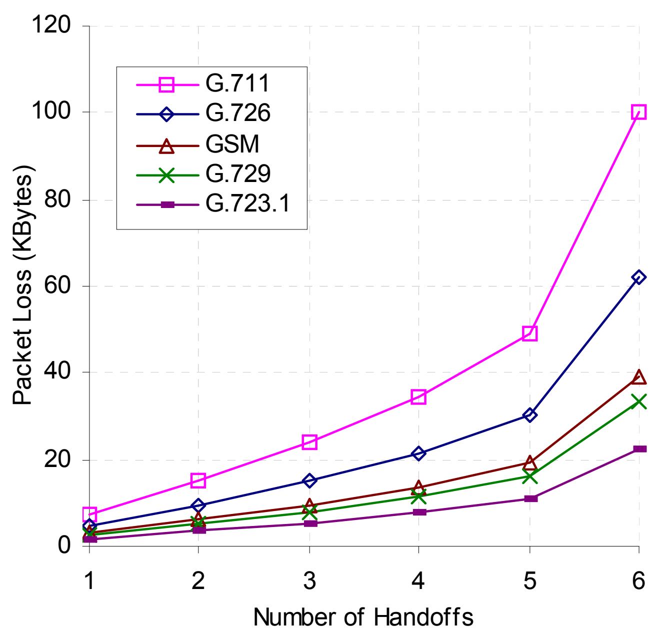
3.4.3 Simulation Results

The results for average session establishment and vertical handoff delays for the above two scenarios can be stated as follows. In the case of the SIP REFER (i.e., Pure-SIP) method, the average IMS based VoIP session establishment delay over the UMTS and the WLAN interfaces are 197 ms and 182 ms respectively, whereas the average vertical handoff delays for WLAN-to-UMTS and UMTS-to-WLAN are 245 ms and 215 ms respectively. The large 3GPP SIP message sizes and heavy application layer based IMS latencies (e.g., HSS look-up, SIP REFER method message exchange, and routing all SIP signaling via the home network) are the major contributors towards the simulated vertical handoff delay.

Secondly, the transient packet loss during the vertical handoff was investigated for different codecs. This transient packet loss was observed by simulating a break-before-make type handoff as described in [35]. Fig. 3.10 (a) and Fig. 3.10 (b) illustrate the transient packet loss scenarios for UMTS-to-WLAN and WLAN-to-UMTS scenarios respectively. The complicated structure of the UTRAN tends to increase the session setup and vertical handoff delays at the UMTS interface in contrast to the rather simple WLAN, which eventually contributs to a relatively higher transient packet loss.



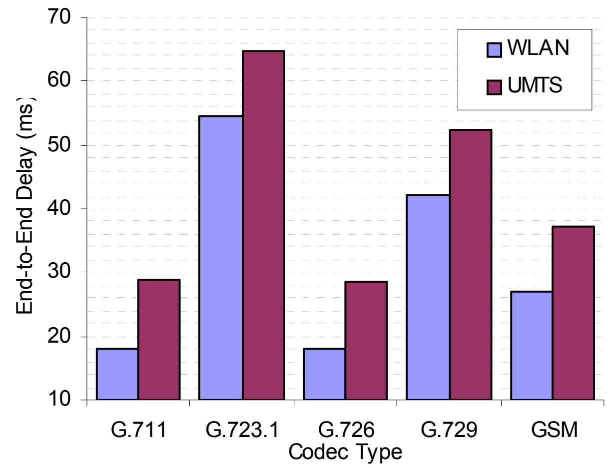
(a). UMTS-to-WLAN Handoff.



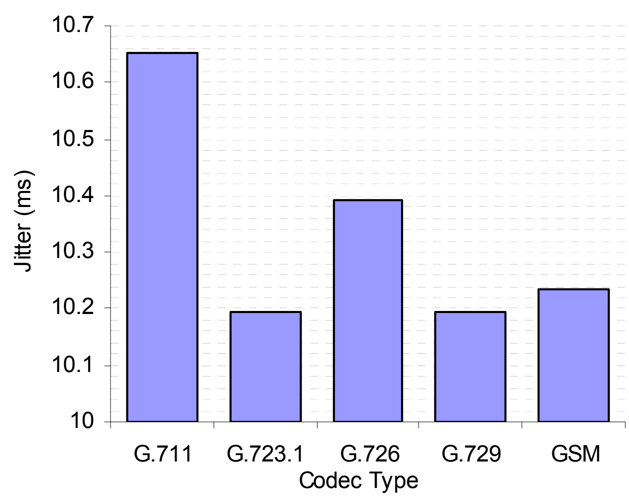
(b). WLAN-to-UMTS Handoff.

Fig. 3.10. Transient Packet Loss during Vertical Handoff.

Next, Fig. 3.11 (a) illustrates the end-to-end delay metric investigated for the WLAN and the UMTS interfaces. This is dependent on the end-to-end VoIP signaling and data paths, the codec, and the payload size of the packets. In fact, the delays from the end points to the codecs at both ends, the encoder (processing) delay, the algorithmic delay, the packetization delay, serialization delay, queuing and buffering delay, and the fixed portion of the network delay yields the end-to-end delay for the connection (i.e., WLAN or UMTS). Note that SIP REFER method does not have an impact on this metric. Furthermore, longer packetization intervals (sampling periods) result in creating relatively larger voice payloads. These relatively large payloads with proportionately smaller IP header overheads lead to better bandwidth utilization. However, as the period of time required for constructing a single packet increases, the amount of time it takes for this packet to reach the other end and be decoded increases as well. Thus it can be said that longer packetization intervals may lead to higher latencies. This phenomenon is demonstrated in the case of codec G.723.1 (sampling period 30 ms) in Fig. 3.11 (a). Additionally, the fact that G.723.1 is a codec that requires very high processor powers to provide high quality audio compression has also contributed to its increase in end-to-end delay. Jitter, the variation of delay, is another factor which affects delay, especially during a vertical handoff (when the link capacity changes and so on). The jitter is illustrated by Fig. 3.11 (b).



(a). Average End-to-End Delay over WLAN and UMTS.



(b). Average Jitter -Variation of Delay at Vertical Handoff.

Fig. 3.11. End-to-End Delay and Jitter.

It is evident from these simulation results that the voice capacity over a heterogeneous network is a function of the system parameters, transmission rate, voice packet payload length (depending on the codec used), and sampling period. The G.723.1 and G.729 codecs appears to have the worst end-to-end delays. However, this can be easily overlooked since they are way below the ITU's recommendation for one-way end-to-end delay threshold of 150ms [140].

Furthermore, G.723.1 and G.729 also have the lowest jitter values. On the other hand, G.711 and G.726 result in relatively high transient packet loss and packet delay variation during vertical handoffs. Therefore, since G.723.1 shows the lowest transient packet loss and jitter, it can be selected as the most appropriate scheme for the considered heterogeneous networking framework.

3.5 Analytical Modeling

Next an analytical model is derived for evaluating the proposed scheme for analyzing the QoS metrics and measures involved in call or session mobility management [38]. More precisely, QoS metrics that are analyzed are handoff delay, average packet loss, jitter and signaling overhead/cost. The primary assumption made for this analysis is that, VoIP call/session arrivals follow a Poisson process. However, despite the closeness and the attractiveness of the Poisson model, the validity of a Poisson process for modeling VoIP call arrivals over a packet switching IP backbone has often been questioned [141], [142]. The reason being that there is a wealth of evidence proving that Internet traffic is bursty in nature, which could be best characterized as self-similar with long range dependence [143], [144] , [145].

However, the reasons for our assumption (i.e., VoIP call/session arrivals follow a Poisson process) can be justified as follows. Although streaming flows (e.g., voice or video for real-time play) and aggregates of elastic flows (e.g., email, web browsing, MP3 or MP4 track) typically exhibit properties of long-range dependence and self-similarity [145], it has been argued in [146] that modeling VoIP session arrivals to be Poisson may be an adequate assumption according to the conclusions of [147], especially when the system utilization is relatively low. This is because all user initiated VoIP sessions display the essential defining characteristic of being mutually independent [148], [149], [150]. Therefore, as shown in [151] and [152], for large populations, where each user independently contributes to a small portion of the overall traffic, user sessions (i.e., human initiated call/data sessions) can be assumed to follow a Poisson arrival process [153]. Furthermore, based on the traces of wide-area traffic, there is further evidence that Poisson arrivals appear to be suitable for traffic at the session level when sessions are human initiated [144].

On the other hand, for analysis and design, using bursty background traffic is not practical as a network element with a non-Poisson arrival (say, Pareto) rate makes it difficult to approximate the delay leading to an intractable analytical solution [146].

3.5.1 Handoff Delay

In this section, a formula is derived for analyzing the vertical session handoff delay (at the core network level regardless of the underlying radio network and link layer delays) for the architecture in Fig. 3.1. The derived method will be successfully applicable for analyzing vertical session handoff delay from WLAN-to-UMTS and vice-versa for a considered VoIP data session.

A standard end-to-end vertical handoff delay (*D*, say) during mid-session mobility consists of the following sub-procedures (or delays); D_1 = Link Layer handoff delay, D_2 = movement detection delay, D_3 = address allocation delay, D_4 = session re-configuration delay, and D_5 = packet re-transmission delay [154]. The vertical handoff delay/s at the Network Layer (and above) are calculated independent of the Link Layer delay D_1 and mainly consist of D_3 and D_4. According to our proposed architecture for IMS based vertical handoff in Fig. 3.1, there is no Domain Host Configuration Protocol (DHCP) related address allocation. This is due to relatively high latency involved in DHCP address negotiation and allocation in an interworked WLAN-UMTS configuration scenario (approx. between 1-2 seconds) [155], which may lead to a falsely inflated result for the IMS based session re-configuration delay. Hence it can be argued that D_4 is the main contributor for network layer based vertical handoff delay, *D*. The session re-configuration delay, D_4 mainly consists of the previously mentioned IMS based session negotiation and handoff and HSS related message exchange delays.

In order to derive an expression for *D*, we must first derive an expression for analyzing the end-to-end transmission delay. Hence, let us assume that the end-to-end transmission delay for a packet size *S* sent from network *A* to network *B* over a number of hops via a wireless and wired links to be expressed as:

$$D(S, H_{a-b}) = D_{wl} + D_w + L_{wl} + L_w \qquad (3.1)$$

where, D_{wl} is the total delay at the wireless interface (say, BS), D_w is the total delay at the wired link, L_{wl} is the latency of the wireless link, and L_w is the latency of the wired link. In order to derive D_{wl} and D_w, as clarified in the beginning of Section 3.5, a *M/M/1* queuing model has been applied to the packet flow of the data session at the wireless BS and other networking elements of the IMS on the path of signaling and data routing of Fig. 3.1. It is important to note that to apply the results of *M/M/1* analysis, several assumptions must be made. The most important of them is that the service times that a packet experiences at different nodes are independent of each other. It has also been found that this independence assumption can be used in large networks [156]. Using the results from the Queuing Theory, expressions for D_{wl} and D_w can be expressed as follows [157]:

$$D_{wl} = \frac{1}{(\mu_{wl} - \lambda_{wl})} \tag{3.2}$$

where, μ_{wl} is the service rate at the wireless interface and λ_{wl} is the packet arrival rate at the wireless interface. For clarity and convenience sake, the units for μ_{wl} are changed from packets/sec to bits/sec. If the probability density function of for packet size, x, in bits be $\mu e^{-\mu x}$ with a mean packet length of $1/\mu$ bits/packet, the capacity of communication channel i be C_i bits/sec, and the arrival rate for channel i be λ_{wl} packets/sec, then the product μC_i becomes the service rate in packets/sec. Therefore, for channel i, we get

$$D_{wl} = \frac{1}{(\mu C_i - \lambda_{wl})} \tag{3.3}$$

where, D_{wl} includes both queuing and transmission delays. Also note that the mean packet size does not depend on the channel as the capacity and the input rates.

On the other hand, when D_w (i.e., total delay at the wired link) is considered, it can be expressed as a collection of delays experienced at each individual node/multiplexer, which can be considered as a collection of multiple *M/M/1* queues. In such an environment, if the output of several *M/M/1* nodes feed into the input queue of another node, the resulting input process also shows the properties of a Poisson process, with mean equal to the sum of the means of the feeding process [158], [159].

Therefore, the total wired network delay experienced by a packet can be expressed as:

$$D_w = \frac{1}{\lambda_w} \sum_j \lambda_j \left(\frac{1}{\mu C_j - \lambda_j} \right) \tag{3.4}$$

where, λ_w is the total packet arrival rate to the network, λ_j is the packet arrival rate at j^{th} node, and μC_j is the service rate in packets/sec at the j^{th} node (Refer Appendix A for proof). Thus by combining equations (3.1), (3.3) and (3.4) we get:

$$D(S, H_{a-b}) = \left(\frac{1}{\mu C_i - \lambda_{wl}} \right) + \left\{ \frac{1}{\lambda_w} \sum_j \lambda_j \left(\frac{1}{\mu C_j - \lambda_j} \right) \right\} + L_{wl} + L_w \tag{3.5}$$

Now, an expression for the vertical handoff delay D can be expressed by applying (3.5) to the entire IMS signaling flow involved in the vertical handoff mechanism as illustrated in Fig. 3.2. Thus the final expression for D is a combination of the following end-to-end delay components as indicated in equation (3.6).

$$\begin{aligned} D \quad = \; & D(S_{Refer}, H_{UMTS\text{-}WLAN}) + D(S_{202Accept}, H_{UMTS\text{-}WLAN}) \\ & + D(S_{INVITE}, H_{WLAN\text{-}CN}) + D(S_{183\text{-}SP}, H_{WLAN\text{-}CN}) \\ & + D(S_{Notify}, H_{UMTS\text{-}WLAN}) + D(S_{PRACK}, H_{WLAN\text{-}CN}) \\ & + D(S_{OK}, H_{WLAN\text{-}CN}) + D(S_{UPDATE}, H_{WLAN\text{-}CN}) \\ & + D(S_{OK}, H_{WLAN\text{-}CN}) + D(S_{ACK}, H_{WLAN\text{-}CN}) \\ & + D(S_{OKNotify}, H_{UMTS\text{-}WLAN}) + \Delta \end{aligned} \tag{3.6}$$

where, Δ is additional IMS (application layer) latency due to the HSS lookup process. Thus equation (3.6) is capable of providing the total vertical session handoff delay from UMTS-to-WLAN in this case. Similarly, vertical session handoff delay from the WLAN-to-UMTS can also be calculated with appropriate substitutions to equation (3.5).

The important point to note here is that the derivation of equation (3.5) has not taken into account the errors that may cause various messages to be damaged or lost. This is since for successful session establishment the entire message flow must take place and if any message is damaged or lost the vertical handoff process will fail. Hence it has been assumed that the channel is error free during the process of the vertical handoff taking place. It is also worth reminding that make-before-break handoff is applied in the proposed handoff scenarios, which helps compensate for large handoff delays. However, for purpose of a complete analysis of the vertical session handoff delay, the standard straight forward case of break-before-make handoff scenario is used.

3.5.2 Packet Loss

The total packet loss (*Loss*) during a vertical session handoff can be defined as the sum of all lost packets during the vertical handoff while the MN is receiving the downlink data packets [160]. It is assumed that the packet loss begins when the Layer 2 handoff is detected and all in-flight packets are lost during the vertical handoff time [161]. Thus, it can be expressed as:

$$Loss = \left[\frac{1}{2T_{ad}} + D \right] \times \lambda_{wl} \times N_m \tag{3.7}$$

where, T_{ad} is the time interval between *P-CSCF* discovery times, λ_{wl} is the downlink packet arrival rate at the wireless interface, and N_m is the average number of vertical handoffs during a single session [154]. N_m plays a major role in the calculation of packet loss since the packet loss due to vertical handoff is directly proportionate to the number of handoffs it is subjected within a given session.

3.5.3 Signaling Overhead/Cost

This section presents the signaling cost analysis for the described vertical handoff scenario. The resultant signaling cost of mobility management during vertical handoff can be analyzed as follows [39]. The signaling cost or overhead is the accumulative traffic load on exchanging signaling messages during the MN's communication session [162]. Therefore the signaling cost incurred by a message can be defined as:

$$Cost = P \times S_{message} \times H_{A-B} \tag{3.8}$$

where, P defines the probability associated with the occurrence of a particular event, $S_{message}$ is the average size of a signaling message/packet related to this event, H_{A-B} is the average number of hops/distance the signaling packet traverses between the source node A to the destination node B.

Next we shall be applying the generalized expression given in (3.8) to a scenario where a call session is handed off from UMTS-to-WLAN as illustrated in Fig. 3.12. When a MN moves from one network (say, from UMTS) to another (say, to WLAN), the following conditions must be satisfied for a successful vertical session handoff to take place [163].

The first condition is that, a data session that is initiated from the MN's current network (say, UMTS) must remain active (or long enough) until it has moved into the region of the second network (say, into the WLAN), as shown in Fig. 3.12.

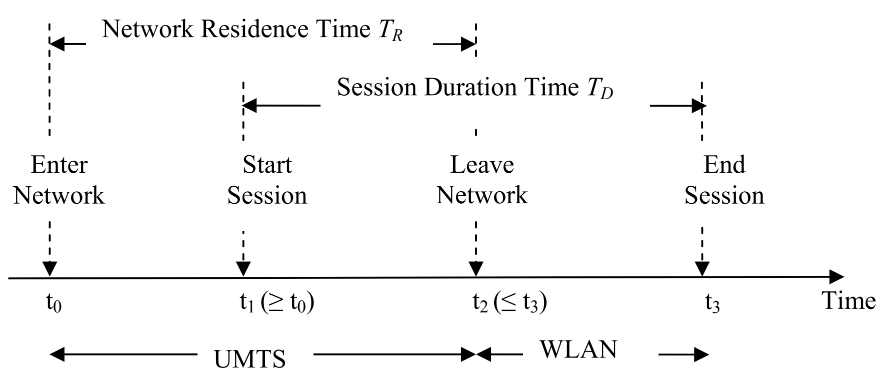


Fig.3.12. Timing Diagram for a UMTS-to-WLAN Session Handoff.

Based on this condition, $P_1(t)$, the arrival probability of a session that is likely to be subjected to a vertical handoff for an inter-network roaming MN can be derived. Let's assume that session arrivals follow Poisson process with the average session arrival rate λ *calls/min*, thus the probability that there is one session arrival in a time period t becomes $\lambda t e^{-\lambda t}$ [156]. Hence, P_1, the session arrival probability for an inter-network roaming MN when it is resident in the UMTS network (i.e., according to Fig. 3.12), can be expressed as:

$$P_1(t) = \int_0^{t_2} \lambda t e^{-\lambda t} f_{T_R}(t)dt \tag{3.9}$$

where, $f_{T_R}(t)$ is the probability density function (*pdf*) of the network residence time T_R. It is assumed that the residence time of the MN in a given network, T_R, is exponentially distributed with a mean $1/\eta$, where η is the inter-network mobility rate. Hence $f_{T_R}(t)$ is expressed as:

$$f_{T_R}(t) = \eta e^{-\eta t} \tag{3.10}$$

Further, as per the justification used in [163], since the maximum limit for t_2 can be extended until the end of the session, in order to obtain a closed form value for P_1, the upper limit of the integral in (3.9) is changed to ∞. By substituting $f_{T_R}(t)$ in (3.9) and solving the equation P_1 can be derived as:

$$P_1(t) = \int_0^{\infty} \lambda t e^{-\lambda t} \eta e^{-\eta t} dt$$

$$P_1(t) = \lambda\eta \int_0^{\infty} t e^{-(\lambda+\eta)t} dt$$

$$= \lambda\eta \left\{ \left[\frac{te^{-(\lambda+\eta)t}}{-(\lambda+\mu)} \right]_0^{\infty} - \int_0^{\infty} \frac{e^{-(\lambda+\eta)t}}{-(\lambda+\mu)}.1.dt \right\}$$

$$= \lambda\eta \left\{ [0] + \frac{1}{(\lambda+\mu)} + \int_0^{\infty} e^{-(\lambda+\eta)t} dt \right\}$$

$$= \frac{\lambda\eta}{(\lambda+\mu)}\left[\frac{e^{-(\lambda+\eta)t}}{-(\lambda+\mu)}\right]_0^\infty$$

$$= \frac{\lambda\eta}{(\lambda+\eta)^2} \tag{3.11}$$

The second condition is that, for a vertical handoff to take place (say, from UMTS to WLAN), the session duration time T_D must be greater than network residence time T_R. It is assumed that the session duration time, T_D, is exponentially distributed with a mean $1/\mu$, where μ is the mean message processing/service rate of the session initiating wireless link (i.e., UMTS in this case). Hence the *pdf* of T_D, $f_{T_D}(t)$ can be expressed as:

$$f_{T_D}(t) = \mu e^{-\mu t} \tag{3.12}$$

Therefore the probability P_2(t) for this condition, which is vertical session handoff probability, can be expressed as:

$$P_2(T_D \rangle T_R) = \int_0^\infty\int_t^\infty \mu e^{-\mu y} f_{T_R}(t)dydt$$

$$P_2(T_D \rangle T_R) = \int_0^\infty\int_t^\infty \mu e^{-\mu y}\eta e^{-\eta t}dydt$$

$$P_2(T_D \rangle T_R) = \mu\eta\int_0^\infty\int_t^\infty e^{-(\mu y+\eta t)}dydt$$

$$P_2(T_D \rangle T_R) = \mu\eta\int_0^\infty\int_t^\infty e^{-(\mu y+\eta t)}dydt$$

$$P_2(T_D \rangle T_R) = \mu\eta \int_0^{\infty} \left[\frac{e^{-(\mu y+\eta t)}}{-\mu} \right]_t^{\infty} dt$$

$$P_2(T_D \rangle T_R) = \eta \int_0^{\infty} e^{-(\mu+\eta)t} dt$$

$$P_2(T_D \rangle T_R) = \eta \left[\frac{e^{-(\mu+\eta)t}}{-(\mu+\eta)} \right]_0^{\infty}$$

$$P_2 = \frac{\eta}{(\mu+\eta)} \tag{3.13}$$

For the sake of clarity and convenience, the units for μ (i.e., the mean message processing/service rate of the session initiating wireless link) are changed from packets/sec to bits/sec as in equation (3.3), where C_i is the capacity of the communication channel i in bits/sec.

$$P_2 = \frac{\eta}{(\mu_{wl} C_i + \eta)} \tag{3.14}$$

Therefore, the total signaling cost for the scenario in Fig.3.2 can be expressed as the sum of the two individual signaling costs associated with the above two conditions. Thus, P_1 and P_2 are substituted for P in the generalized expression given in (3.8) for calculating these individual signaling costs. Additionally, since the signaling cost is calculated for a roaming MN, the average network mobility rate *(η)* and the average session arrival rate *(λ)* components are also included in the final equation. The SIP INVITE message sequence (steps 1-7 from Fig. 3.2) is associated with P_1 and session arrival rate. Similarly, the SIP REFER message sequence (steps 8-14 from Fig. 3.2) is associated with P_2 and inter-network mobility rate.

Hence, the total signaling overhead incurred by vertical handoffs during a given data session can be expressed as:

$$Cost = P_1 \lambda \sum_{i=1}^{n_1} \left(S_{INVITE_i} \times H_{(A-B)_i} \right) + P_2 \eta \sum_{i=1}^{n_2} \left(S_{REFER_i} \times H_{(A-B)_i} \right) \tag{3.15}$$

where n_1 and n_2 represent the number of messages involved in each handoff/message exchange sequence as illustrated in Fig. 3.2. If η is the average network mobility rate of a MN and λ is the average session arrival rate, λ/η may be defined as the Call-to-Mobility Ratio (*CMR*) [164], [165]. Thus (3.15) can be re-arranged as:

$$Cost = \left[P_1 \eta \sum_{i=1}^{n_1} \left(S_{INVITE_i} \times H_{(A-B)_i} \right) \right] \frac{\lambda}{\eta} + P_2 \eta \sum_{i=1}^{n_2} \left(S_{REFER_i} \times H_{(A-B)_i} \right) \tag{3.16}$$

$$Cost = \left[P_1 \eta \sum_{i=1}^{n_1} \left(S_{INVITE_i} \times H_{(A-B)_i} \right) \right] CMR + P_2 \eta \sum_{i=1}^{n_2} \left(S_{REFER_i} \times H_{(A-B)_i} \right) \tag{3.17}$$

3.6 Performance Analysis

The following numerical results are generated using 3GPP-SIP messages. Table 3.1 shows the typical SIP message sizes and other related parameters. IMS-SIP values are based on [166]. Other related parameters have been partly obtained from [154] and [164] to maintain consistency. The relative distances in hops are illustrated in Fig. 3.13 (the concept of relative distances in hops has been partly inspired by the ideas from [167] and [168]). Based on these assumptions, the following analytical results are derived for equations (3.6), (3.7), (3.15) and (3.17) for the scenario of an IMS based vertical handoff.

Figure 3.14 illustrates how the vertical handoff delay behaves against increasing handoffs per session. The graphs indicate higher transient handoff delays for lower data rates (i.e., 2 Mbps). The graphs also indicate that handoff delays for relatively high data rates (i.e., 11 Mbps and 54 Mbps) are much closer to each other. This indicates that as the link bandwidth increases the vertical handoff delay exponentially drops and stabilizes. Therefore, the results also indicate that beyond a certain level, vertical handoff delays cannot be simply reduced by purely increasing the wireless link bandwidth.

The next important point to note is that as the number of handoffs is increased, the 2 Mbps link shows an extremely sharp exponential increase in delay. What happens here is that due to multiple handoffs being processed by the link, the utilization of the system $\rho=\lambda_{wl} / \mu_{wl} \rightarrow 1$. If the handoffs are further increased, a point where $\lambda_{wl} > \mu_{wl}$ is reached, the arrival rate increases the maximum capacity of the link, which eventually fails the link [169], [170]. Furthermore, according to the presented analytical model, the average session establishment delay over the UMTS and WLAN interfaces are 170 ms and 155 ms, respectively. Next, the average vertical handoff delays for WLAN-to-UMTS and UMTS-to-WLAN are 210 ms and 186 ms, respectively.

TABLE III.1

IMS MESSAGE SIZES AND PARAMETER VALUES USED FOR ANALYSIS.

Message	**Size (Bytes)**	**Parameter**	**Value/s**
INVITE	736	L_{wl}	2 ms
183 Ses. Pro.	847	L_w	0.5 ms
PRACK	571	Δ	100 ms
200 OK	558	T_{ad}	1 sec
UPDATE	546	λ_d	Voice Codecs used
ACK	314	C_i	2 – 11 Mbps
ReINVITE	731		
REFER	750		
200Accepted	550		
NOTIFY	550		
OKNOTIFY	550		

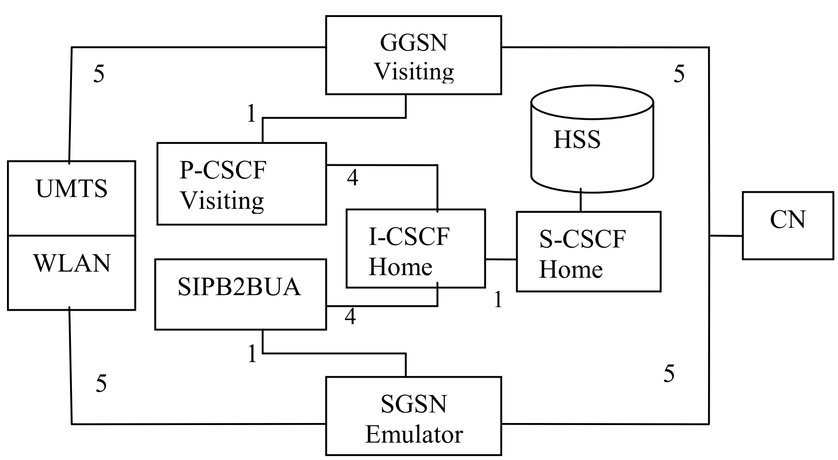


Fig. 3.13. Relative Distances in Hops.

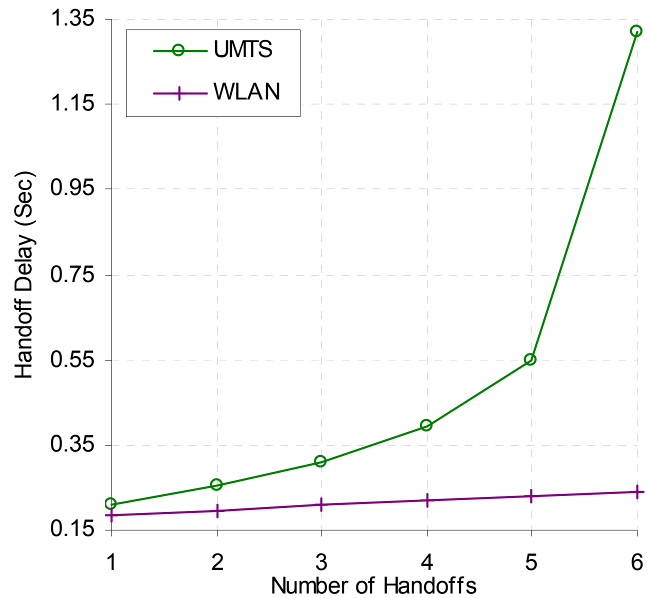


Fig. 3.14. Vertical Handoff Delay vs. Number of Handoffs per Session.

Figure 3.15 and Fig. 3.16 demonstrate the normalized packet loss during vertical handoffs for the GSM voice codec as the number of handoffs per session increases (in the case of a break-before-make handoff scenario). The reason for using a popular voice codec is the need for a more realistic VoIP session vertical handoff to be simulated.

According to equation (3.7), the packet loss during a vertical handoff is directly proportional to the vertical handoff delay. Therefore, high vertical handoff delays related to the 2 Mbps link in the graph in Fig. 3.14 directly contributes to the exponential behavior of packet loss for WLAN-to-UMTS handoff as shown in Fig. 3.16. Similarly the packet loss is relatively low in Fig. 3.15 for a UMTS-to-WLAN handoff, which is in line with the corresponding handoff delay shown in Fig. 3.14. The reason for the graphs to show such behavior is the application layer based additional IMS related latencies. These additional latencies substantially contribute towards increasing handoff delays as the number of handoffs per session increases, which eventually results in packet loss. However, it is important to note that the proposed model uses a make-before-break handoff technique to avoid such transient packet losses.

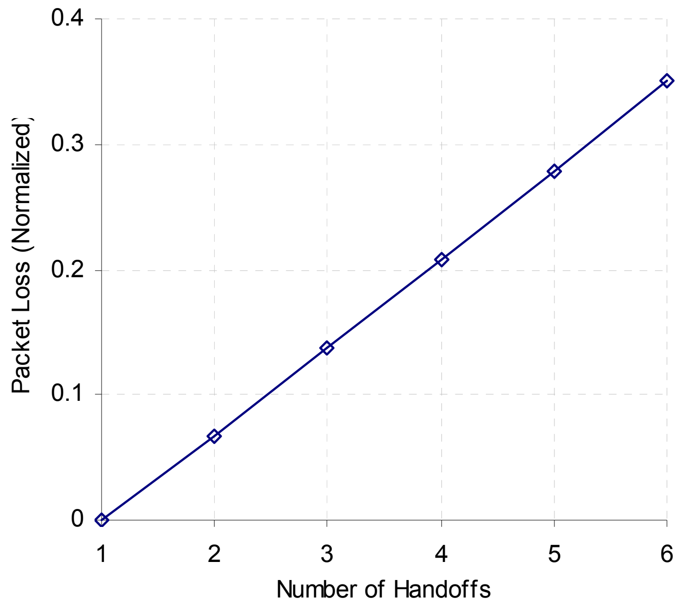


Fig. 3.15. Packet loss for a UMTS-to-WLAN Handoffs per Session.

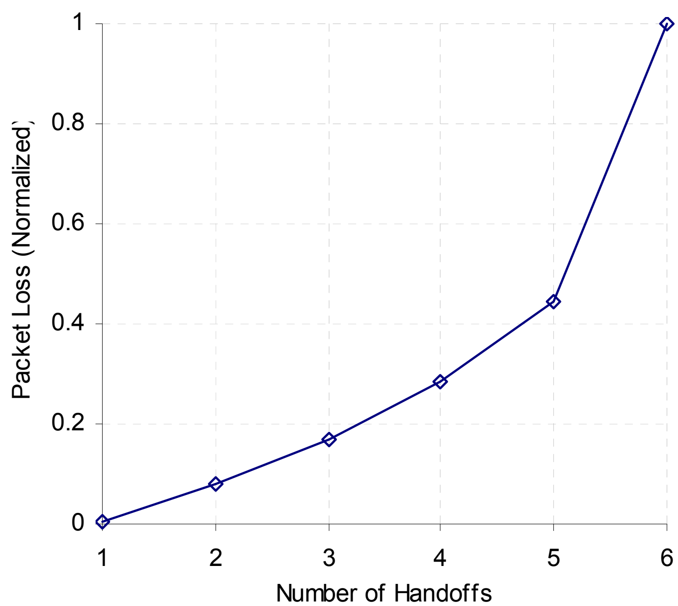


Fig. 3.16. Packet loss for a WLAN-to-UMTS Handoffs per Session.

Figure 3.17 illustrates the behavior of the jitter during vertical handoffs, which is the variation of delay, for the same voice codec. The jitter is simulated by using (3.5) for calculating the end-to-end delay for a given codec over the given link and plotting the variation of delay when a session is handed off from UMTS to WLAN. According to the graphs in Fig. 3.17, the jitter levels are within acceptable limits for a VoIP session as specified in many Service Level Agreements (SLA) [171], [172], [173] .

However, as the number of vertical handoffs increases, the interesting fact is that these jitter graph tend to indicate rather exponential curves. This could well be confirmed from Fig. 3.18, which illustrates the behavior of jitter for an extended number of session handoffs. This phenomenon can also be easily explained as in the previous case. That is, when the number of VoIP sessions keep increasing there comes a point where $\rho=\lambda_{wl} / \mu_{wl} \rightarrow 1$, which increases delay exponentially and so does jitter.

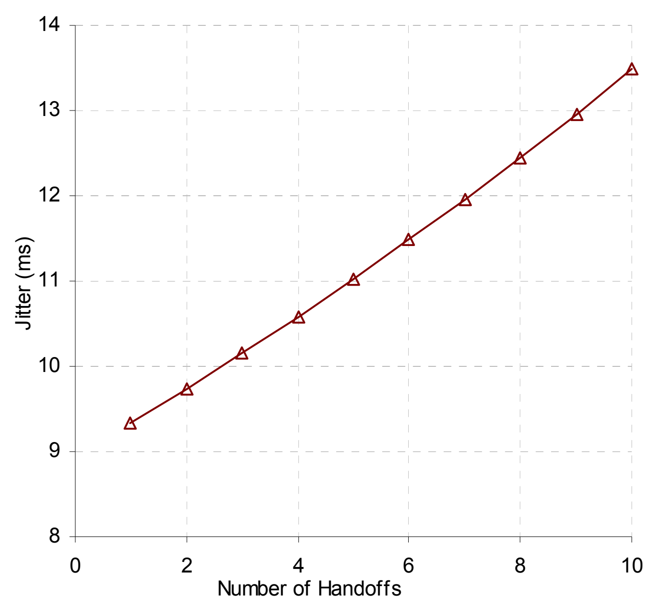


Fig. 3.17. Variation of End-to-End Delay (Jitter).

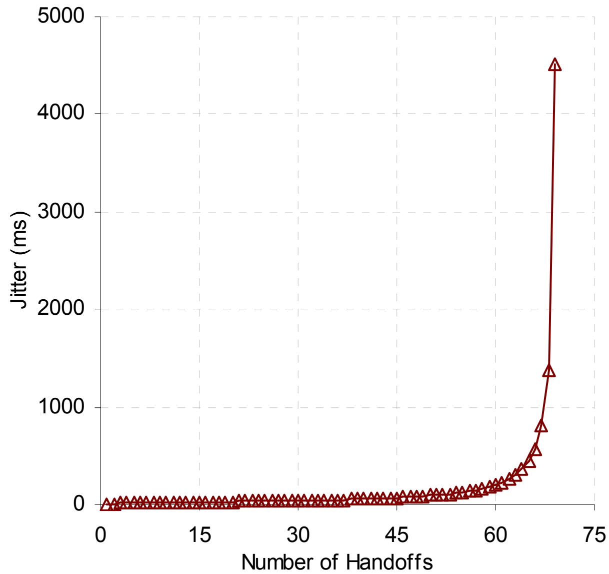


Fig. 3.18. Behavior of Jitter vs. Number of Handoffs.

Figure 3.19 onwards illustrates the behavior of signaling cost. According to the graphs in Fig. 3.19, it is clear that the signaling cost increases against average session arrival rate (λ *calls/min)* for increasing values of average network mobility rate (η min^{-1}). In general, this is due to the increase of P_1, the arrival probability of sessions that are likely to be subjected to a vertical handoff, with the increase of λ *calls/min*. Nevertheless, it is interesting to observe the behavior of P_1 against λ for different η values (Fig. 3.20) to exactly find out the reasons for different gradient levels of the three graphs in Fig. 3.19.

According to Fig. 3.20, for λ ranging from 0.01-0.1 *calls/min*, when η = 0.01 min^{-1}, a negative slope is observed. This negative gradient has contributed towards slowing down the increase of signaling cost against increasing λ within the considered range. Similarly, in Fig. 3.20, for the same λ range, when η = 0.1 min^{-1} a positive gradient is observed. This contributes to a rather rapidly increasing signaling cost for the graph corresponding to η = 0.1 min^{-1} in Fig. 3.19.

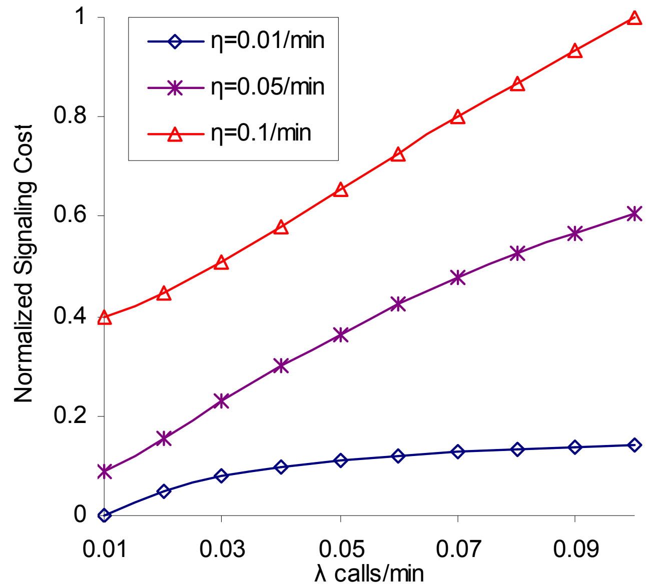


Fig. 3.19. Normalized Signaling Cost vs. Average Session Arrival Rate (λ calls/min).

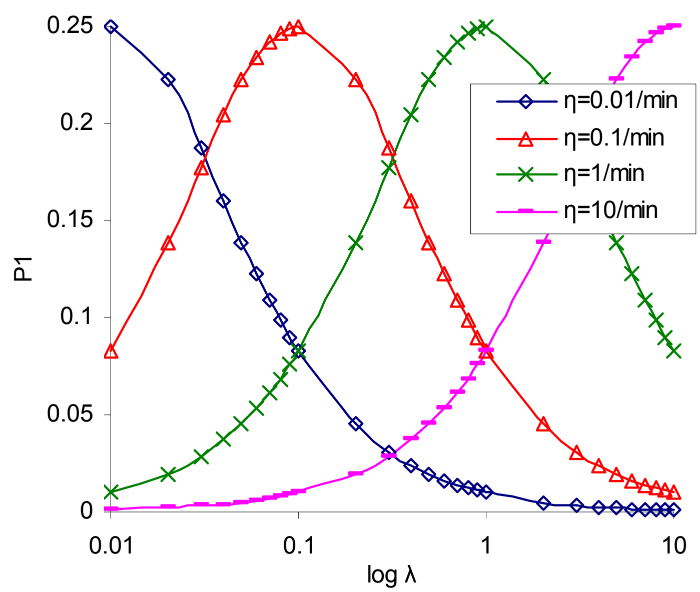


Fig. 3.20. P_1 vs. Average Session Arrival Rate (λ calls/min).

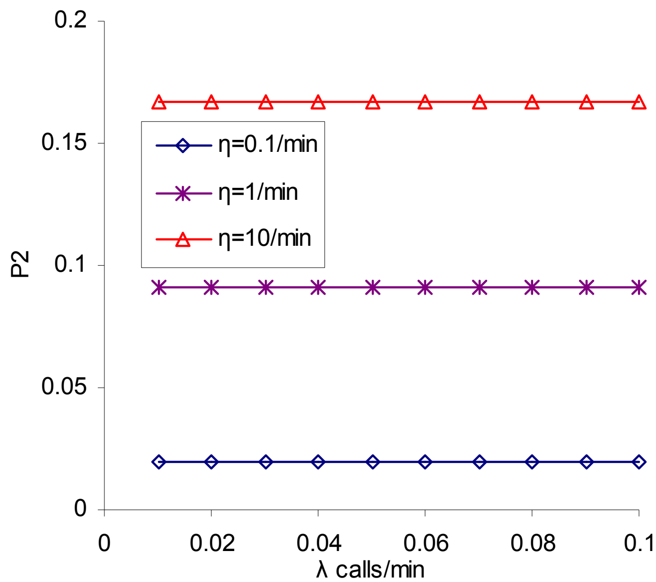


Fig. 3.21. P_2 vs. Average Session Arrival Rate (λ calls/min).

According to Fig. 3.21, since P_2 remains constant for a given η, it does not impose a dramatic effect in this case. It is also important to note that the range of λ is kept below the service rate ($\mu_{wl}C_i$). This is because, in the event that λ reaches the service rate ($\mu_{wl}C_i$), the utilization of the system will rapidly reach 100%. Thus the graphs in Fig. 3.19 are only plotted up to $\lambda = 0.1$ *calls/min*.

Figure 3.22 illustrates the behavior of normalized signaling cost against average network mobility rate (η min^{-1}) for different average session arrival rate (λ *calls/min*) values and a constant service rate. In this case, normalized signaling cost generally increases as (η min^{-1}) increases. Also this increase becomes rapid for values of (η min^{-1}) ranging from 1-10 min^{-1}. The reason for such behavior is that as the network mobility rate (η min^{-1}) increases, more sessions can be subjected to vertical handoff, which eventually increases the session handoff probability (P_2) giving rise to the signaling cost. For comparison purposes, P_1 and P_2 curves against increasing network mobility rate is illustrated in Fig. 3.23 and Fig. 3.24 respectively. According to Fig. 3.24, it is clear how P_2 approaches 1 with the increase of η min^{-1}. The effect of P_1 in this case is relatively minimal since P_1 does not exceed 0.25, much similarly to the

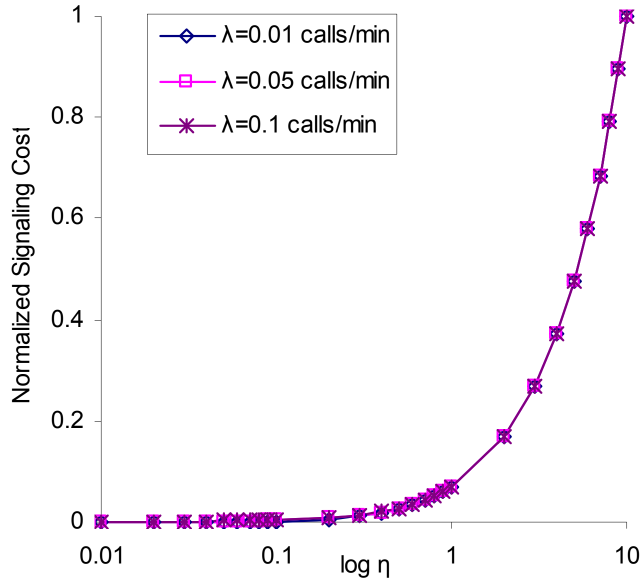


Fig. 3.22. Normalized Signaling Cost vs. Average Network Mobility Rate (η min^{-1}).

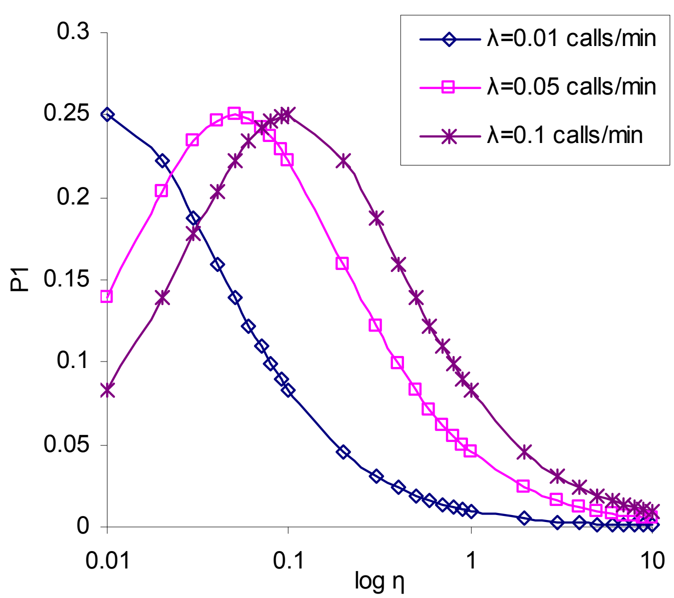


Fig. 3.23. P1 vs. Average Network Mobility Rate (η min^{-1}).

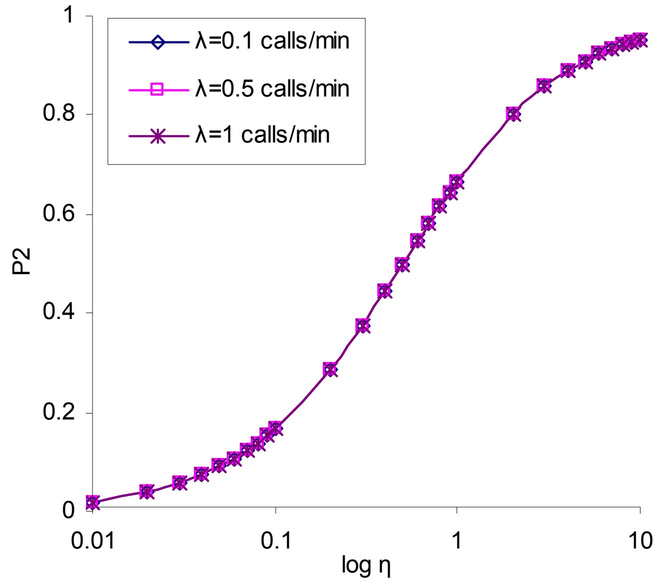


Fig. 3.24. P2 vs. Average Network Mobility Rate (η min^{-1}).

pattern indicated in Fig. 3.20 (i.e., since P1 curves behaves the same despite λ calls/min and η min-1 being interchanged).

The next investigation is the behavior of signaling cost against the CMR. Fig. 3.25 illustrates normalized signaling cost against CMR by having λ *calls/min* as a constant. As per the illustration of the graphs in Fig. 3.25, the normalized signaling cost reduces exponentially as CMR increases. As the CMR increases by keeping λ *calls/min* as a constant, η min^{-1} tends to decrease rapidly, which has a direct impact on P_2 that reduces it exponentially against increasing CMR. However, the impact of decreasing η min^{-1} does not tend to have a drastic effect on P_1. P_1 shows a closely similar pattern as in Fig. 3.20 with a maximum peak of 0.25 for CMR=1. Hence, in this case, the signaling cost curve is shaped according to the behavior pattern of P_2 as CMR increases. Also, it is important to note that the signaling cost is higher for larger values of λ.

Last but not least, Fig. 3.26 illustrates normalized signaling cost against CMR by having η min^{-1} as a constant. As per the illustration of the graphs in Fig. 3.26, the normalized signaling cost increases as CMR increases and eventually reaches a saturation point. As the CMR increases by keeping η min^{-1} as a constant, λ *calls/min* tends to increase rapidly, which

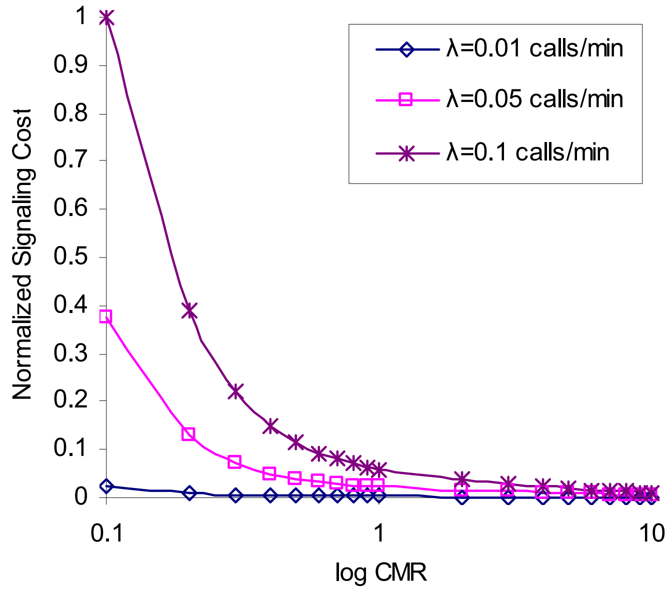


Fig. 3.25. Normalized Signaling Cost vs. CMR (λ constant).

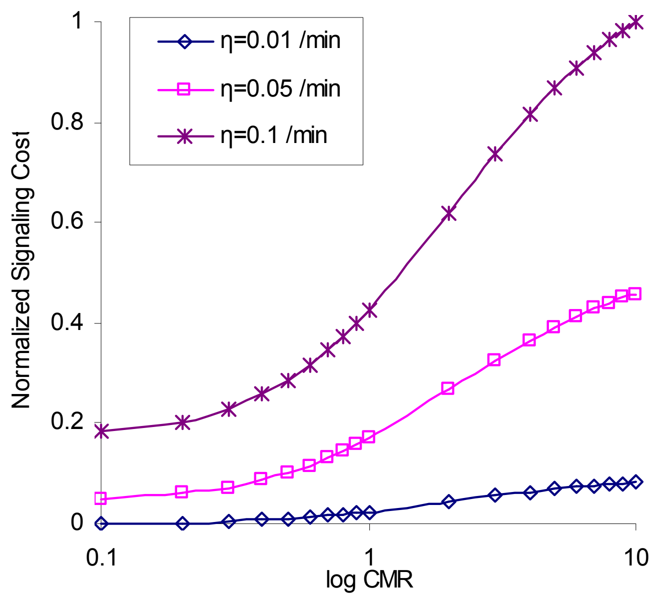


Fig. 3.26. Normalized Signaling Cost vs. CMR (η constant).

eventually results in increasing the signaling cost. As in the above case, P_1 behaves in a similar manner with a maximum peak of 0.25 for CMR=1. However, the notable point is that P_2 remains constant for a given value of η, which eventually contributes towards shaping the signaling cost curves by reaching a saturation level. Furthermore, signaling cost is also higher for larger values of η.

It must also be noted that the above analysis and results have been obtained for users moving with randomized patterns independently to each other. However, there may be instances where additional correlation exists amongst users, which leads to aggregation of their movements, namely the group mobility [174], [175]. This leads to the exploration of advanced mobility models for realistically characterizing group mobility. In group mobility, each group has a logical center, which generally defines the group trajectory. Usually, users are considered to have random mobility with respect to a logical center and be randomly distributed within the geographic scope of the group [176]. The movement of the logical center could be defined by a predefined motion path or an individual mobility model [177]. If it is a predefined motion path, the movement of the centers will not be random [178]. However, as a special case scenario of group mobility, if the movements of these logical centers behave according to a random walk mobility model [179], it could be argued that such a scenario may be closely approximated to our analyzed mobility model [180].

3.7 Summary and Conclusions

The purpose of this Chapter was to introduce an initial interworking architecture with the IMS acting as a universal coupling mediator for real-time session negotiation and management. This initial model was aimed for interworking UMTS and WLAN systems with limited mobility. The mobility management framework proposed in this Chapter is fully dependant on the IMS, and its session management protocols. Subsequently, a queuing theory based analytical model was introduced for analyzing its vertical handoff performance and an OPNET based test bed was introduced for simulating the architectural design. However, the framework proposed in this Chapter is limited to interworking between a UMTS cellular network and a WLAN system. Therefore, the next Chapter discusses how this architecture could be extended for interworking dissimilar cellular networking technologies with WLAN systems.

Chapter 4

An Extended Framework for Interworking WLAN, UMTS and CDMA2000 Networks

This Chapter focuses on extending the proposed WLAN-UMTS interworking architecture presented in Chapter 3 to a unified framework for interworking dissimilar 3G cellular networking standards. As a result, global roaming and interoperability beyond one cellular system (say, UMTS) to another cellular system (say, CDMA2000) will become possible. The Chapter is organized as follows. Firstly the introduction section outlines the motivation for this work and the importance of a heterogeneous interworking framework. Followed by this comes the sections on IMS based mobility management in 3G cellular networks and related architectural considerations. The next section outlines the proposed extended architecture. Subsequently the sections on analytical modeling and performance evaluations are presented. Finally an OPNET based simulation model is introduced for validation of the performance. The contents provided under this Chapter have contributed to the following publications. The interworking architecture and the OPNET simulation platform presented in this Chapter has contributed to [40] and the presented OPNET simulation results have partly contributed to [41] and [42].

4.1 Introduction

One of the main motivations behind the development process of the NGMN is to have a common platform for different cellular technologies to interwork with each other [181]. In order to address this requirement, previously proposed WLAN-UMTS interworking framework has been extended where a common platform is proposed for coupling a CDMA2000 network at the core network. As elaborated in Chapter 3, the novelty of this solution for WLAN-UMTS interworking is its adaptation of the 3GPP's IMS as an arbitrator for real-time session negotiation and management [31]. Since 3GPP2 also adopts a similar concept as 3GPP by

introducing an IMS within the CDMA2000 core network, the IMS can be named as a top candidate for a universal coupling mediator [111].
However, as mentioned in Section 2.10, there are substantial differences between the 3GPP and the 3GPP2 core networks in regards to the level of contribution of the IMS in relation to how terminal and session mobility are managed [182]. Particularly the fact that the 3GPP-IMS exclusively handles both terminal and session mobility at the Application Layer in contrast to the 3GPP2-IMS, which only handles session mobility at the Application Layer (leaving terminal mobility to be handled at the Network Layer). Therefore, prior to the introduction of the new framework, the challenges of having the IMS as a universal coupling mediator and the cross layer interactions of the IMS with the 3GPP2's Network Layer based mobility management protocols will be presented.

4.2 IMS based Mobility Management in 3G Cellular Networks

As described under Section 2.11, the UMTS Release 5 was the first to introduce the IMS within its core network, for controlling multimedia sessions. The key elements of the IMS are the CSCFs, which can be generalized as a SIP proxy server and a user profile database called the HSS. Amongst three CSCFs (i.e., P-CSCF, I-CSCF, and S-CSCF), the S-CSCF is the actual SIP server, which performs user registration and handles session management in the IMS. The 3GPP-IMS exclusively uses SIP as the default protocol for session signaling (i.e., session establishment and management) and mobility management at the Application Layer. For the purpose of achieving a fully SIP based mobility management, as mentioned in Section 3.2.2, a range of SIP methods exist. These methods are namely; the SIP ReINVITE method, the SIP 3pcc mechanism, and the SIP REFER method [183].

Similarly, the 3GPP2 has also introduced the IMS for multimedia session handling within its PDS. The initial release of the 3GPP2-IMS was influenced by the 3GPP's original IMS specification. Although the 3GPP2-IMS uses SIP as the default protocol for session signaling (i.e., session establishment and management), its mobility management framework is not solely based at the Application Layer. Furthermore, as summarized below, many considerable differences exist:

- In the 3GPP-IMS:
    - SIP is used for session mobility,
    - IPv6 based core network,
    - HSS manages subscriber user data,
    - PDP context activation used for P-CSCF discovery, and
    - P-CSCF and associated visiting GGSN must reside in the same network.
- In the 3GPP2-IMS:
    - MIP (at Network Layer) is used for IP / terminal mobility management and SIP (at Application Layer) is used for session mobility management,
    - Flexibility in using IPv4 or IPv6,
    - Home AAA server and databases handle subscriber user information,
    - PDP context activation not used for P-CSCF discovery,
    - P-CSCF and PDSN do not need to reside in the same network, and
    - A new position server and a position determining entity has been defined.

4.3 Architectural Considerations

As a common interworking platform for mobility management for the NGMN is considered, resolving the interactions of 3GPP-IMS with MIP becomes a new challenge. Therefore, the framework proposed in this section uses MIP for providing terminal mobility and limits SIP for session mobility management in a generalized IMS architecture.

The next relevant point is making the decision for choosing IPv4 or IPv6. Although the 3GPP mandates the IMS over IPv6, the 3GPP2 does not make such a distinction in its specification. Therefore, either MIPv4 [125] over IPv4 or MIPv6 [126] over IPv6 can be used, which also gives more freedom for network operators. As defined by the 3GPP2, if MIPv4 is deployed, the PDSN of the home network may act as the MIP HA. As the MN moves form one

PDSN to another, the new PDSN may act as the MIP FA, which becomes the new point of contact for the MN.

Therefore when the MN is attached to a different PDSN the HA becomes an anchor point for the MN traffic flow, particularly when reverse tunneling and triangular routing takes place. In the event when MIPv6 is being implemented, direct peer-to-peer communication can be established through its route optimization operation [126]. Since the 3GPP2's standard for CDMA2000 does not fully support inter-PDSN mobility for IPv6, the proposed design will primarily be based on MIPv4 [127]. As a result, the route optimization option will not be available in its current form. However, transition of this model to MIPv6 can be easily done by addressing user authentication, address allocation, and enabling the MN to perform the MIPv6 update procedures [128].

Therefore as per the above mentioned design considerations, the newly proposed extended architecture uses the IMS as an arbitrator for session management over a MIP based IP mobility management framework. By and large this can be seen as a universally applicable, combined MIP-SIP based approach for mobility and session management in NGMNs. Nevertheless, given the fact that it incorporates a joint MIP-SIP based mobility management framework, it must be noted that, the aim of this approach is not to develop a hybrid version of the IMS by integrating MIP features (i.e., a S-CSCF with an integrated MIP-HA) as suggested in [184]. Depending on the version of MIP used (i.e., MIPv4 or MIPv6) few variations to the signaling framework of this architecture exist. Therefore, prior to looking at the proposed architecture in detail some of these complexities will be discussed.

If MIPv4 is used for handling IP mobility for 3GPP's (IPv6 based) IMS signaling framework, the first concern should be Network Address Translation (NAT). With SIP, a NAT is required for mapping external IP headers and port numbers as well as certain fields in the packet payload (SIP header fields and SDP body). Possible solutions of how this may be achieved are studied in detail in [185]. The second issue in having IMS-SIP signaling over MIPv4 is that depending on the location of the P-CSCF and the S-CSCF with respect to the FA and HA, there may be a possibility for triangular routing between the IMS elements. Therefore, SIP signaling that originates from the P-CSCF located in a visited/foreign network could be routed through the S-CSCF in the home network. Since the 3GPP IMS standard recommends SIP signaling to be routed through the S-CSCF, triangular routing does not cause any extra overheads [31]. However, the actual performance degradation takes place when the data flow gets routed via the home network (GGSN/PDSN).

On the other hand, if MIPv6 is used for handling IP mobility for the 3GPP's (IPv6 based) IMS signaling framework a different set of addressing concerns may arise. Under the MIPv6 mobility management framework, the MN is expected to have two IP addresses: namely the static Home Address and Care of Address (CoA). The CoA address is updated as the MN moves from one network to another. Hence, the question arises as to which address must be used for SIP signaling at the IMS level. As mentioned in the previous case for MIPv4, the Home Address is used by SIP for initial registration and session establishment. Therefore, one may argue that SIP could simply use the MN's Home Address. However, as the MN changes its point of attachment from the Home Network and connects to a foreign network it gets a new CoA. Hence it can also be counter-argued that the CoA could be a better candidate for SIP signaling. Additionally it must also be taken into consideration that the design of the IMS includes a security mechanism for verifying the source IP address to be the same as the IP address mentioned in the SIP header and SDP body. Hence the same IP address (Home Address or CoA) must be used as the source address and the address at SIP message level throughout a given session.

Based on the above aspects, when MIPv6 is used for handling IP mobility for 3GPP's (IPv6 based) IMS signaling framework the following SIP-MIP interworking choices could be assumed. The first choice is to have both the Home Address and CoA to be used by IMS-SIP. However, the current standard does not support the use of two multiple IP addresses and therefore requires changes to the current IMS-SIP standards. Therefore, the above will be directly ruled out.

Secondly, if the CoA is fully utilized for all IMS-SIP based signaling (SIP header and SDP body), a fully SIP based mobility and session management framework could be constructed. The main drawback in this approach is that each time the MN gets a new CoA, re-registration of the CoA and re-establishment of a new (or ongoing) SIP session is required. As a result of continuous SIP session re-negotiations, in a real-time communication scenario, service interruptions may take place. Above all, MIP's advantages of IP mobility management can not be fully exploited in this method. A similar type of approach (i.e., by exclusively using SIP) has been presented in [35].

The third approach would be pretty much the opposite of the second approach. That is, to use the Home Address for all IMS-SIP based signaling, which uses MIP for handling IP mobility. The main advantage of having MIP for managing IP mobility is that the MN does not need to re-register every time it changes its CoA. In an IPv6 network, the MN may uses MIPv6 signaling for updating other nodes as the CoA keeps changing. Further, with the help of route optimization, changing of CoA will be fully transparent to the IMS.

The fourth and final approach, which has also been used in our design, although similar in concept to the third method, works on an IPv4 platform. Main reasons for using MIPv4 is that it eliminates the complexity of managing two IP addresses (as in the above scenarios), enables IP mobility management (by transporting single/static IP address) in such a way that node mobility is transparent to the layers above. As a result, the IMS is able to provide session mobility for a roaming MN without any disruption of IP connectivity thus, enabling seamless service continuity.

Since the 3GPP2's IMS standard specifies that the IMS signaling could also be implemented on IPv4, previously discussed NAT related complications (IPv4 to IPv6 translation) can be avoided in this method. Nevertheless, since the IMS is primarily a framework for SIP based session control, even though the specification mentions otherwise, 3GPP's IMS could also be successfully implemented over IPv4. However, it is worth noting that since route optimization is not available, depending on the type of implementation, the IPv4 may contribute for triangular routing and reverse tunneling between the P-CSCF (located in a foreign network) and S-CSCF (located in a home network) of the IMS. Nevertheless, as previously stated, the 3GPP IMS standard recommends all SIP signaling to be routed through the S-CSCF, therefore MIPv4's triangular routing effect could be easily overlooked. Last but not least, another important motive for using an all IPv4 based platform is the fact that network operators are gradually realizing that despite of the initial hype, the migration towards IPv6 is slowly becoming more and more practically unrealistic.

4.4 The Interworking Architecture

The proposed extended internetworking architecture, with signaling and data routes, is illustrated in Fig. 4.1 [40]. Based on the existing UMTS and CDMA2000 mobility management mechanisms, this architecture proposes a framework for the NGMN's for providing real-time IP multimedia services.

As per the illustration on Fig. 4.1, the UMTS core network is connected to the IP network through the GGSN, which also acts as its MIPv4 FA. Prior to acquiring the IP address, the MN goes through the following steps. The first step involves with the MN powering on and locking to the UMTS network. Once the appropriate cell is selected the MN can be considered ready for the establishment of a data session. It is assumed that this function is already performed by the MN, and will not be discussed in detail.

Once the above mentioned system acquisition is done, the next step is to establish a data connection, or set up a data pipeline. This must be completed in two steps by using the Attach and PDP context activation message sequences. The activation of the MN's PDP context does not allocate an IP address and the IP address field of the PDP context is not filled at this point. The actual IP address allocation for the MN is initiated by sending the MIPv4 registration requests to it's HA via the GGSN (MIP-FA). This mechanism is based on the specifications given under 3GPP's [186].

This is triggered when the MN receives a periodically broadcasted agent advertisement sent by the GGSN (MIP-FA). When the MN receives this message it sends a MIP Registration Request message to the GGSN (MIP-FA). The GGSN (MIP-FA) forwards this request to the MIP-HA and the HA assigns the home IP address to the MN via a MIP Registration Reply message. Further, the HA maintains the MS's location information such as the GGSN (MIP-FA) address. When the GGSN (MIP-FA) receives this information the MN gets its home IP address assigned.

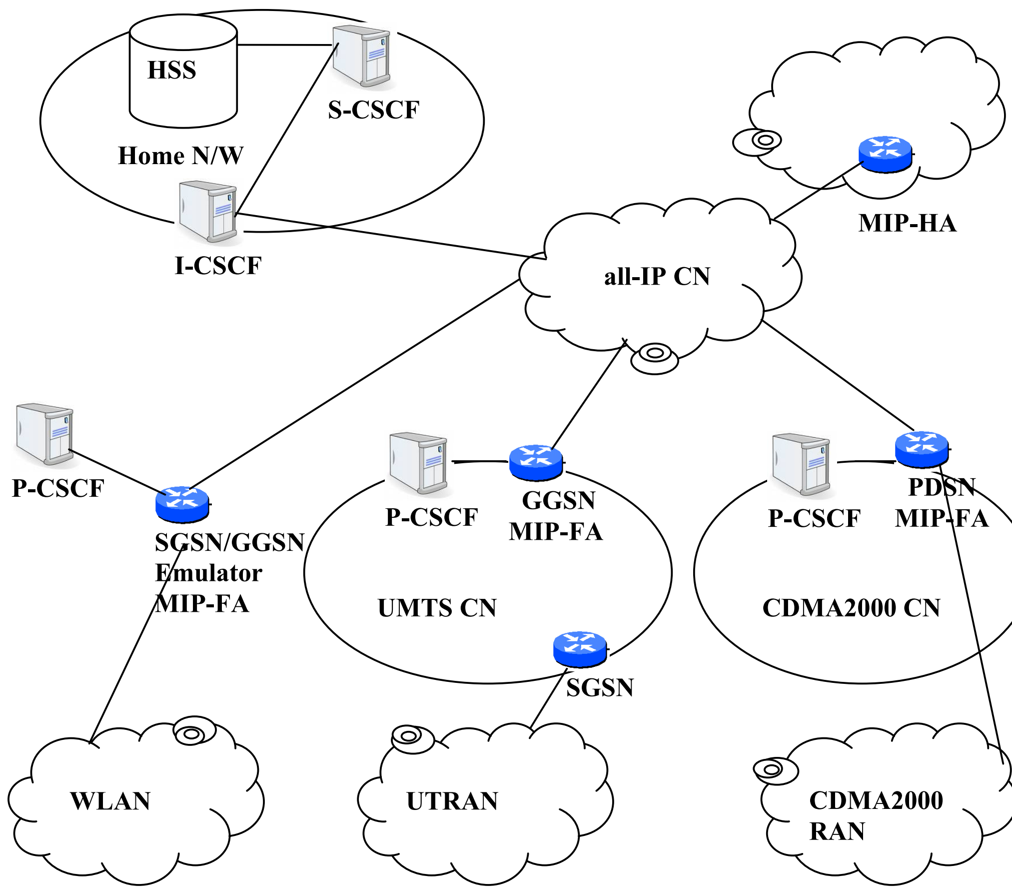


Fig. 4.1. Interworking Architecture with MIP-SIP based Mobility Management.

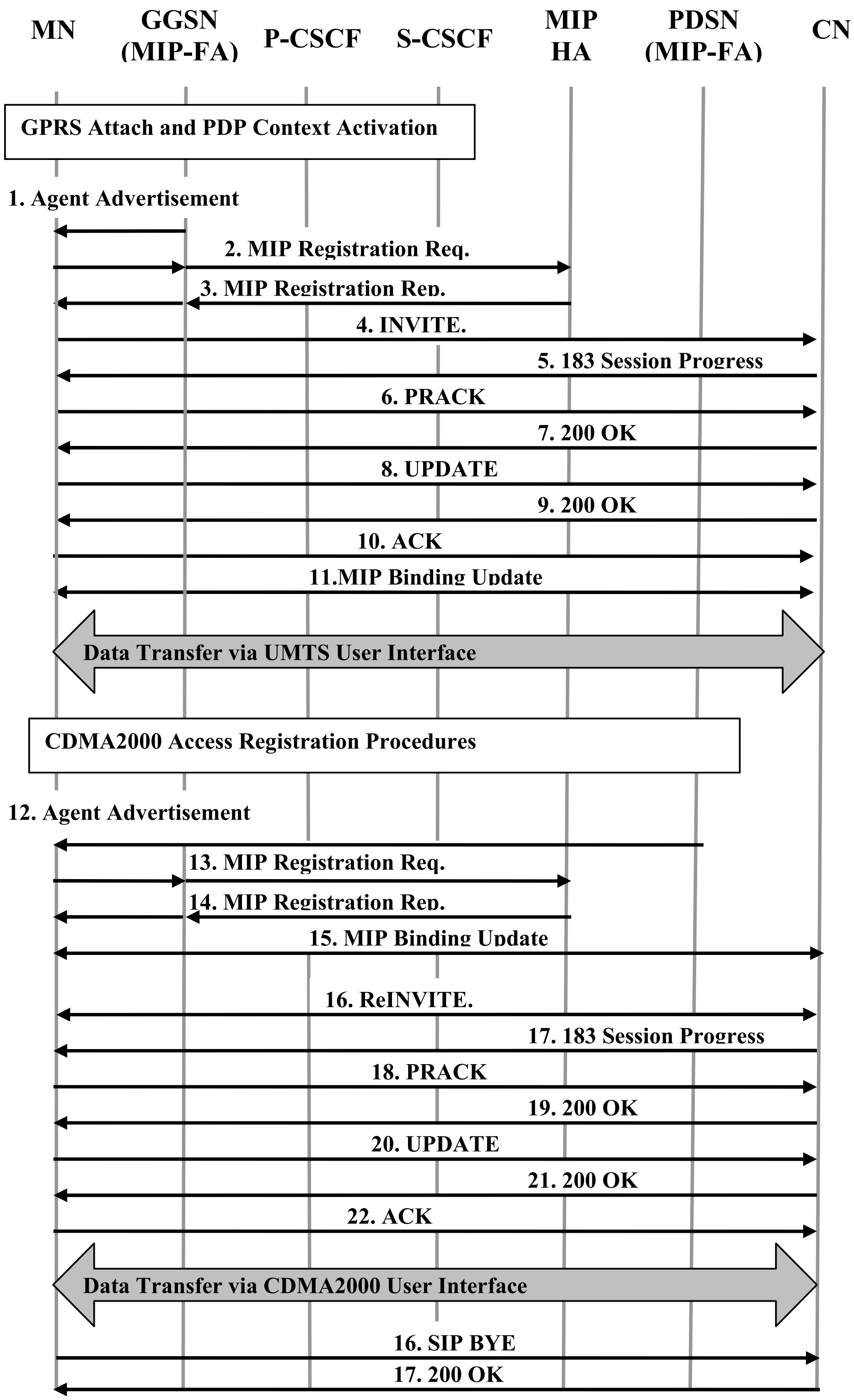


Fig. 4.2. MIP-SIP Signaling Framework.

Now the MN is able to identify the P-CSCF for the registration with the IMS. Prior to establishing a SIP session, the MN requires performing a service registration function to let the IMS know of its location. The MN acts as a SIP client and sends a SIP registration message to its home system through the P-CSCF. The basic steps for a SIP service registration can be summarized as follows. Firstly, the HSS for the MN is notified of its current location for the HSS to update the subscriber profile accordingly. Next the HSS checks if the MN is allowed to register in the network based on the subscriber profile and operator limitations, and grants authorization. Once authorized, a suitable S-CSCF for the MN is assigned and its subscriber profile is sent to the designated S-CSCF.

After the activation of the PDP context and the service registration, the MN is ready to establish a media/data/call session. As illustrated in Fig. 4.2, the sequence of the SIP session origination procedure can be described as follows. The mobile origination procedure is initiated by a SIP INVITE message sent form the UMTS interface of the source MN. This initial message is forwarded from the P-CSCF to the S-CSCF of the originating network, via the CSCFs of the terminating network, and finally to the destination. This SIP INVITE request carries a SDP body indicating the IP address and port numbers where the source wants to receive the media streams. Furthermore, the INVITE also contains a request to follow the precondition call flow model. This is important because some clients require certain preconditions (that is, QoS levels) to be met before establishing a session. The requirement for using the preconditions call flow model in the IMS is mainly because in cellular networks radio resources for the media plane are not always available. If the preconditions extension was not used, when the called party accepts the call, the source and destination may not be ready and consequently the first few packets may be lost.

Next, this model requires that the destination responds with a 183 Session Progress message containing a SDP answer. The SDP answer contains the media streams and codecs that the destination is able to accept for this session. The acknowledgement for the reception of this provisional response by a PRACK request follows afterwards. If the destination does not receive a PRACK response within a determined time, it will transmit the provisional response. When the PRACK request successfully reaches the destination a 200 OK response is generated by the destination with an SDP answer. Next an UPDATE request is sent by the source containing another SDP offer, in which the source indicates that the resources are reserved at its local segment. Once the destination receives the UPDATE request, it generates a 200 OK response. Once this is done, the MN can start the media/data flow and the session will be in progress (via the UMTS interface).

When this MN roams between CDMA2000 and UMTS systems inter-system roaming takes place. The message flow for an inter-system roaming from UMTS to CDMA2000 can be described as follows. Firstly the standard CDMA2000 link layer access registration procedures are performed. Next the CDMA2000 interface performs the MIP registration procedures with the PDSN (i.e., MIP FA) as explained previously. This is when the PDSN forwards this request to the MIP-HA and the HA assigns the home IP address to the new CDMA2000 interface. Lastly the exchanging of a MIP Binding Update (BU) message between the MN and the corresponding node for avoiding triangular routing [125].

The next stage is the taking place of the IMS-SIP session handoff procedures. This requires sending a SIP ReINVITE (with the same Call-ID and other identifiers corresponding to the ongoing session) to the destination SIP UAC. Followed by this is a resource/preconditions reservation for the CDMA2000 interface. Once this is successfully done the new session flow can be initiated. It is important to note that until such time that the new data flow is initiated via the CDMA2000 interface, the data flow via the UMTS interface remains active. Thus the model follows the make-before-break handoff mechanism as proposed in our previous works [35]. Finally, the UMTS interface performs the implicit detach procedures with the SGSN and the PDP context is deactivated. Inter-system roaming form CDMA2000 to UMTS can also take place in a similar manner.

Furthermore, since this design is an extension to the UMTS-WLAN interworking platform proposed in Chapter 3, this extended architecture could be considered as a generalized framework for roaming between WLAN-UMTS-CDMA2000 networks. The introduction of MIP, unlike the Pure-SIP based approach, helps to reduce extra application layer based signaling, thus enabling a smoother vertical handoff. The next sections will be essentially investigating and reporting the performance comparisons between these two vertical handoff approaches.

4.5 Analytical Modeling

This section provides an analytically modeled approach for evaluating the newly proposed extended version of the interworking architecture. The primary aim of this analytical model is to help compare the benefit of introducing MIP for terminal mobility management (below an IMS based session mobility management framework) in contrast to having terminal and session mobility management purely handled by the IMS.

This analysis uses a similar approach, as explained in Section 3.5, for evaluating QoS metrics and measures involved in the above MIP and SIP combined mobility management approach. QoS metrics such as reconfiguration and vertical handoff delay (from the network layer and above), total packet loss, jitter, and signaling overhead are analyzed under this section and compared against the results obtained for a Pure-SIP based scenario. Since the aim of this thesis is to develop a universal mobility management platform by having the IMS as a common arbitrator, delay components relating to the Physical and Link Layers of various wireless networks/links (i.e., UMTS, CDMA2000, WLAN, and so on) has been overlooked. Furthermore, for the purpose of easy understanding and comparison against the results of Chapter 3, a MIP-SIP based UMTS-WLAN interworking scenario has been considered (rather than having a UMTS-CDMA2000 interworking scenario).

This analysis is also based on a similar assumption as in Section 3.5, that the call/session arrivals follow a Poisson process. Furthermore, it can be stated that the arguments provided in Section 3.5 for justifying VoIP session arrivals to be modeled as a Poisson process will also be similarly applicable.

4.5.1 Handoff Delay

In this section, the derived formula under Section 3.5.1 will be appropriately modified for a comparative analysis of the vertical session handoff delay for the newly proposed MIP-SIP based architecture illustrated in Fig. 4.1. A modified analysis method is used for comparing the efficiency of the new MIP-SIP based vertical session handoff mechanism (i.e., delay, in this case) against the previously presented Pure-SIP based session handoff mechanism, for a data session establishment.

Similar to the approach used under Section 3.5.1, the vertical handoff sub-procedural delays such as the Link Layer handoff delay, movement detection delay, and packet re-transmission delay have not been taken into consideration. Additionally, according to the proposed enhancements to the architecture for an SIP-MIP based vertical handoff (as illustrated in Fig. 4.2), there is no DHCP based address allocation delay. Thus the main contributor to the Network Layer based vertical handoff delay is the session re-configuration delay. Therefore, it can be expressed that the new vertical session handoff delay (at the core network level regardless of the underlying radio network and Link Layer delays) for the newly considered scenario mainly consists of the (previously considered) IMS based session handoff delay at the Application Layer and the newly introduced MIP related delay at the Network Layer. Another reason for the exclusion of the above mentioned delay components is that, these delay components may significantly vary depending on the radio network technology used, and thus not being able to provide a fair evaluation of the performance of the interworking architecture.

In order to derive an expression for the vertical handoff delay for a MIP-SIP combined method, equation (3.5) will be used. As mentioned under Section 3.1, equation 3.5 expresses the end-to-end transmission delay for a packet size S, sent from network A to network B over a number of hops via a wireless and wired links. This has been directly applied to the entire MIP and IMS SIP signaling flow involved in the vertical handoff mechanism illustrated in Fig. 4.2. Therefore, the final expression in this case (i.e., a MIP and SIP combined handoff approach), can be expressed as a combination of the following end-to-end delay components as expressed in equation (4.1).

$$\begin{aligned}D_{MIP\text{-}SIP} = {} & D(S_{MIPReq}, H_{UMTS\text{-}MIP\text{-}HA}) + D(S_{MIPRep}, H_{UMTS\text{-}MIP\text{-}HA}) + D(S_{MIP\text{-}BU}, H_{UMTS\text{-}CN}) \\ & + D(S_{ReINVITE}, H_{WLAN\text{-}CN}) + D(S_{183\text{-}SP}, H_{WLAN\text{-}CN}) + D(S_{PRACK}, H_{WLAN\text{-}CN}) \\ & + D(S_{OK}, H_{WLAN\text{-}CN}) + D(S_{UPDATE}, H_{WLAN\text{-}CN}) + D(S_{OK}, H_{WLAN\text{-}CN}) \\ & + D(S_{ACK}, H_{WLAN\text{-}CN}) + D(S_{BYE}, H_{UMTS\text{-}CN}) + D(S_{OK}, H_{CN\text{-}UMTS}) + \Delta \qquad (4.1)\end{aligned}$$

where, Δ is additional IMS (application layer) latency. Thus equation (4.1) is capable of providing the total vertical session handoff delay for a MIP and SIP combined scenario. Although this particular scenario concentrates on a vertical handoff from UMTS-to-WLAN, other vertical handoff scenarios relating to different radio interfaces such as WLAN-to-UMTS, UMTS-to-CDMA2000, CDMA2000-to-UMTS could also be constructed with appropriate substitutions using equation (3.5). As mentioned under Section, 3.5.1, the important point to note here is that since it has been assumed that the channel is error free during the vertical handoff, the likelihood of the occurrence of errors has been overlooked. This is since for successful session establishment the entire message flow must take place and if any message is damaged or lost the vertical handoff process will fail.

4.5.2 Packet Loss

The total packet loss during a MIP-SIP based vertical session handoff is also analyzed similarly to Section 3.5.2. Thus, the total packet loss ($Loss_{MIP\text{-}SIP}$) during a MIP-SIP combined vertical session handoff can be similarly defined as the sum of all lost packets during the vertical handoff while the MN is receiving the downlink data packets. It is assumed that the packet loss begins when the Layer 2 handoff is detected and all in-flight packets are lost during the vertical handoff time. Thus, it can be expressed as:

$$Loss_{MIP-SIP} = \left[\frac{1}{2T_{ad}} + D_{MIP-SIP} \right] \times \lambda_d \times N_m \tag{4.2}$$

where, T_{ad} is the time interval between P-CSCF discovery times, λ_d is the downlink packet arrival rate at the wireless interface (this depends on the used voice codec), and N_m is the average number of vertical handoffs during a single session [154].

4.5.3 Signaling Overhead/Cost

This section presents a comparative analysis of the signaling cost between a Pure SIP based vertical handoff scenario and a MIP-SIP based vertical handoff scenario. Similar to the description used under Section 3.5.3, signaling cost or overhead is defined as the accumulative traffic load on exchanging signaling messages during the MN's communication session (i.e., session establishment, session handoff, and so on). Therefore, similar to equation (3.8), the above definition can be expressed as:

$$Cost = P \times S_{message} \times H_{A-B} \tag{4.3}$$

where, P defines the probability associated with the occurrence of a particular event, $S_{message}$ is the average size of a signaling message/packet related to this event, H_{A-B} is the average number of hops/distance the signaling packet traverses between the source node A to the destination node B. Using the above definition and a mobility scenario similar to the illustration in Fig. 3.10, an expression for signaling cost is derived for a MIP-SIP based scenario.

Now, the signaling cost for the described MIP-SIP based vertical handoff scenario illustrated in Fig. 4.2, can be expressed as the sum of the following two signaling components. The first component of signaling cost computation is associated with the entire MIP and SIP INVITE (session establishment) message sequence. As mentioned under Section 3.3, the computation of this signaling cost component is associated with P_1, which is the arrival probability of a session to a MN that is likely to be subjected to a vertical handoff. The second component of signaling cost computation is associated with the MIP and SIP ReINVITE (vertical handoff) message sequence. Similar to what has been mentioned under Section 3.3, the MIP and SIP ReINVITE sequence is associated with P_2, which is the vertical handoff probability and the inter-network mobility rate. Detailed steps for the derivation for P_1 and P_2 have been omitted from this section and the reader is referred to Section 3.5.3 for the same. The total signaling overhead incurred by vertical handoffs during a given data session can be expressed as:

$$Cost = P_1 \lambda \sum_{i=1}^{n_1} \left(S_{INVITE_i} \times H_{(A-B)_i} \right) + P_2 \eta \sum_{i=1}^{n_2} \left(S_{\mathrm{Re}\,INVITE_i} \times H_{(A-B)_i} \right) \tag{4.4}$$

where, λ *calls/min* is the average session arrival rate, η min^{-1} is the average network mobility rate associated with a roaming MN, n_1 and n_2 represent the number of messages involved in each handoff/message exchange sequence as illustrated in Fig. 4.2. If η is the average network mobility rate of a MN and λ is the average session arrival rate, λ/η may be defined as *CMR* [164]. Thus equation (4.4) can be re-arranged as:

$$Cost = \left[P_1 \eta \sum_{i=1}^{n_1} \left(S_{INVITE_i} \times H_{(A-B)_i} \right) \right] \frac{\lambda}{\eta} + P_2 \eta \sum_{i=1}^{n_2} \left(S_{\mathrm{Re}\,INVITE_i} \times H_{(A-B)_i} \right) \tag{4.5}$$

$$Cost = \left[P_1 \eta \sum_{i=1}^{n_1} \left(S_{INVITE_i} \times H_{(A-B)_i} \right) \right] CMR + P_2 \eta \sum_{i=1}^{n_2} \left(S_{\mathrm{Re}\,INVITE_i} \times H_{(A-B)_i} \right) \tag{4.6}$$

4.6 Performance Analysis

The following numerical results are generated using MIPv4 and 3GPP-SIP messages. Table 4.1 shows the typical SIP message sizes and other related parameters. IMS-SIP values are based on [166]. Parameters relating to MIP and other values have been partially obtained from [154] and [164] to maintain the consistency. The relative distances in hops are illustrated in Fig. 4.3. Although Fig. 4.3 has a close resemblance and similar hop counts to Fig. 3.11, the inclusion of the MIP HA and the MIP FAs at the visiting UMTS GGSN and the WLAN's SGSN Emulator makes it possible to account for the hop counts related to MIP related signaling.

Based on the values of Table IV.I and Fig. 4.3, the following analytical results are derived for equations (4.1), (4.2), (4.4) and (4.6) for the scenario of a MIP-SIP based vertical handoff. Fig. 4.4 and 4.5 illustrate the behavior of vertical handoff delay against increasing session handoffs for UMTS-to-WLAN and WLAN-to-UMTS interfaces respectively. The graphs in Fig. 4.5 indicate relatively higher vertical handoff delays for the WLAN-to-UMTS case. On the other hand, the two graphs in Fig. 4.4 indicate a relatively lower handoff delay for the UMTS-to-WLAN case. This indicates that when a session is transferred to a network with relatively lower link bandwidth, a relatively higher vertical handoff delay can be expected.

TABLE IV.I
IMS-SIP MESSAGE SIZES AND PARAMETER VALUES USED FOR ANALYSIS.

Message	Size (Bytes)	Parameter	Value/s
INVITE	736	L_{wl}	2 ms
ReINVITE	731	L_w	0.5 ms
183 Ses. Pro.	847	Δ	100 ms
PRACK	571	T_{ad}	1 sec
200 OK	558	λ_d	GSM Codec
UPDATE	546	C_i	2 – 11 Mbps
ACK	314		
MIP Reg. Req.	60		
MIP Reg. Reply	56		
MIP Binding Updat.	66		
MIP Binding Ack.	66		
MIP Agent Solicit.	67		
MIP Agent Advert.	28		

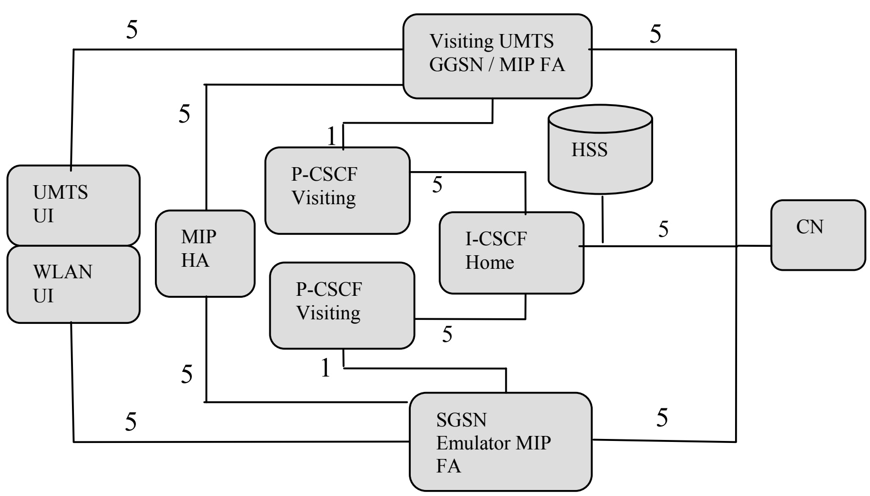


Fig. 4.3 Relative Distances in Hops.

According to the results of the presented analytical model, for a MIP-SIP combined mechanism, the average session establishment delay over the UMTS and WLAN interfaces are 192 ms and 176 ms, respectively. On the other hand, according to Section 3.6, for a Pure-SIP based mechanism (purely based on IMS related mobility management), the average session establishment delay over the UMTS and WLAN interfaces are 170 ms and 155 ms, respectively. The reason behind the relatively higher session establishment delay for the MIP-SIP combined approach is the additional MIP related messages such as MIP Registrations and MIP BUs (refer Fig. 4.2). However, the effects of the IMS (i.e., message flows and additional latencies) could be treated as equivalent in these scenarios.

Next, the average vertical handoff delays based on the MIP-SIP mechanism for WLAN-to-UMTS and UMTS-to-WLAN are 192 ms and 176 ms, respectively. Similarly, according to Section 3.6, the corresponding values for the average vertical handoff delays based on the Pure-SIP mechanism for WLAN-to-UMTS and UMTS-to-WLAN are 210 ms and 186 ms, respectively. On the contrary to the observation relating to session establishment, the vertical handoff delay for a MIP-SIP combined mechanism is relatively lower than for a Pure-SIP based method. This could be easily clarified by referring to Fig. 3.2 and Fig. 4.2. According to these figures, it is clear that the Pure-SIP method is fully involved with heavy IMS based vertical handoff signaling in comparison to the MIP-SIP combined vertical handoff method, which is partly based on the IMS and partly based on MIP. Therefore, the additional latency involved with the IMS based SIP REFER method contributes to the increase in the vertical handoff delay in this case.

Another important observation is that both the session establishment delay and session handoff delay in the case of the MIP-SIP combined method are the same. According to the MIP-SIP signaling framework illustrated in Fig. 4.2, it can be said that all MIP session establishment procedures outside its home network (i.e., the MIP HA in this case) are treated as a case of potential MIP handoff. Thus similar signaling (i.e., MIP Registration and MIP BU) messages are used. Similarly, for the case of the IMS based session handoff procedures, the SIP ReINVITE method is used, which also has a similar message flow to the SIP INVITE method. Hence identical session establishment and handoff delays are experienced under a MIP-SIP combined method.

The next important point to note for the WLAN-to-UMTS case (illustrated in Fig. 4.5) is that as the number of handoffs increase from 5 to 6, an extremely sharp exponential increase in delay is noted. As explained under Section 3.6, this is clearly a sign of the link reaching its maximum utilization limit due to multiple handoffs being processed at the (a relatively low bandwidth) link. However, this observation is not there in the case of UMTS-to-WLAN. This is since the session is handed-off to a network with relatively high bandwidth and the system is far from reaching a state of saturation.

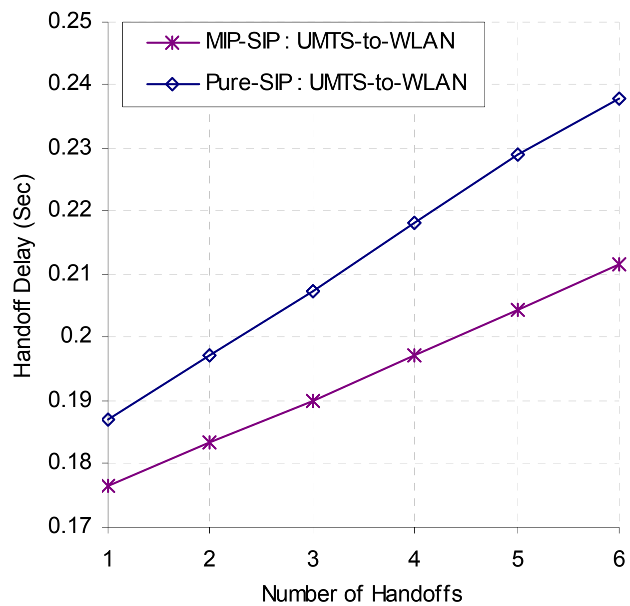


Fig. 4.4. Vertical handoff delay from UMTS-to-WLAN.

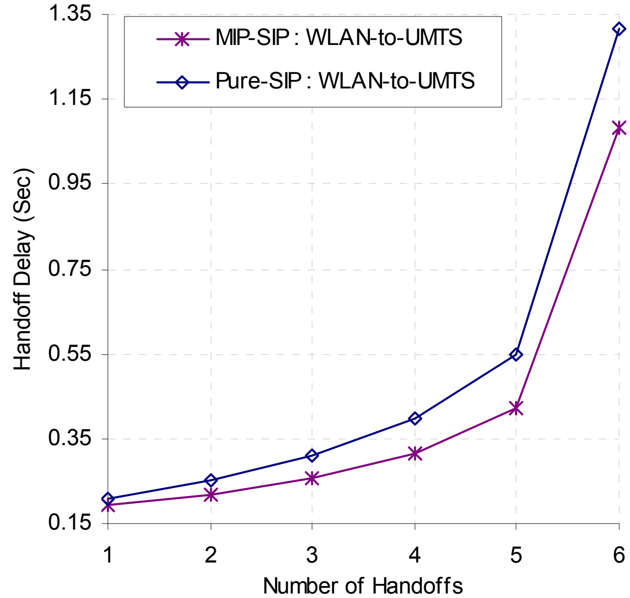


Fig. 4.5. Vertical handoff delay from WLAN-to-UMTS.

Figure 4.6 illustrates the normalized transient packet loss during vertical handoffs as the number of handoffs increase (in the case of a break-before-make handoff scenario). The voice codec considered for the downlink packet transmission in this case is the GSM codec. According to equation (4.2), the packet loss during a vertical handoff is directly proportional to the vertical handoff delay. Therefore, relatively high vertical handoff delays indicated by the WLAN-to-UMTS graphs in Fig. 4.5 directly contribute to the exponential behavior of packet loss as shown in Fig. 4.7. Similarly, the packet loss is relatively low in Fig. 4.6 for a UMTS-to-WLAN handoff, which is also in line with the corresponding handoff delay shown in Fig. 4.4.

Furthermore, as expected, Pure-SIP approach shows a relatively higher packet loss in all considered scenarios in comparison to the graphs representing the MIP-SIP method. Another notable point (although not very important) from the plots in Fig. 4.5 and Fig. 4.7 is that the graphs diverge from one another as the packet transmission rates and the numbers of handoffs per session increases. This is similar to the phenomenon observed in the graphs in Fig. 3.14. As explained under Section 3.6, the reason for the divergence between the graphs is the

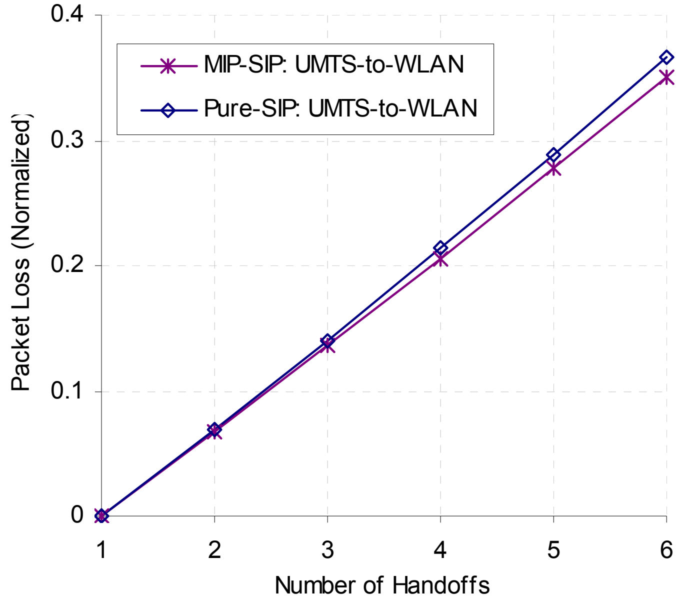


Fig. 4.6. Packet loss for a UMTS-to-WLAN handoff.

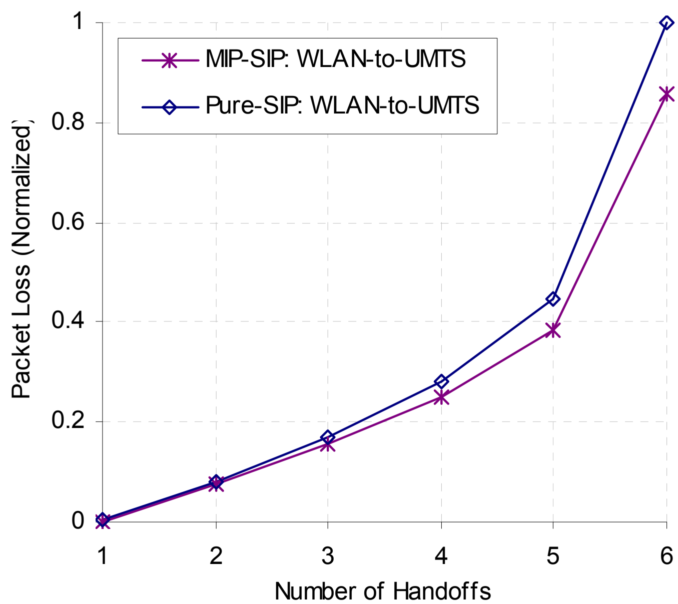


Fig. 4.7. Packet loss for a WLAN-to-UMTS handoff.

accumulating application layer based IMS related latencies involved with the increasing number of handoffs in the case of the Pure-SIP approach.

Figure 4.8 to Fig. 4.11 illustrates the behavior of normalized signaling cost between MIP-SIP and Pure-SIP vertical handoff scenarios. The graphs in Fig. 4.8 indicate how the signaling cost increases against the increasing session arrival rate (*λ calls/min*), when network mobility rate (*η* min^{-1}) and service rate (μC_i) are constant. Within the context of our discussion, the most important observation from Fig. 4.8 is the difference of signaling cost between the two vertical handoff methods, i.e., MIP-SIP and Pure-SIP. The graphs indicate that the MIP-SIP method has a significantly low signaling cost in contrast to the Pure-SIP method. A similar observation can be noted in Fig. 4.9 for the graph plotted for the increasing signaling cost against the network mobility rate (*η* min^{-1}), when the session arrival rate (*λ calls/min*) and service rate (μC_i) are constant. A clear difference between the two signaling cost curves for the MIP-SIP based and Pure-SIP based handoff mechanism could be observed. The effects of P_1 and P_2 on the above graphs have not been included in this discussion since a comprehensive description on the effects of these two parameters has been included under Section 3.6. Nevertheless, it is important to note that since the effect of P_1 and P_2 are equal for both these scenarios (i.e., MIP-SIP and Pure-SIP), their effects may be overlooked in this instance.

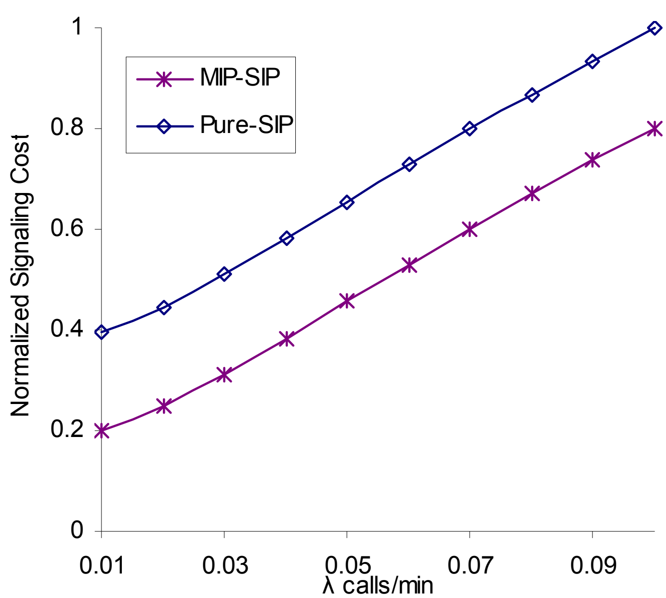


Fig. 4.8. Normalized Signaling Cost vs. Average Session Arrival Rate (λ calls/min).

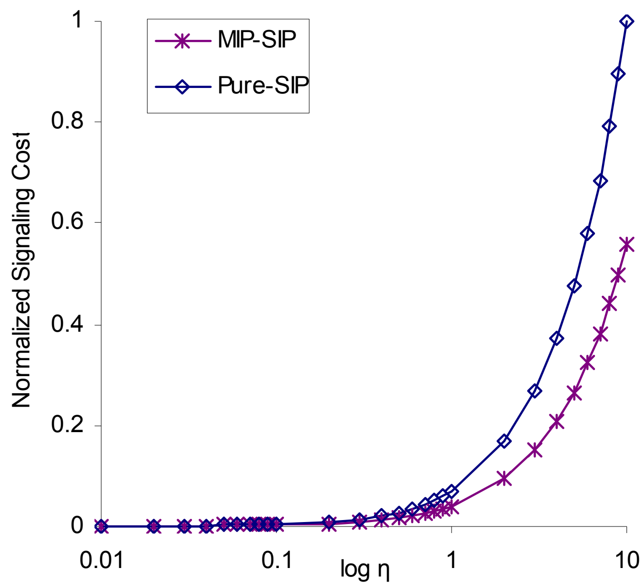


Fig. 4.9. Normalized Signaling Cost vs. Average Network Mobility Rate (η min^{-1}).

The next investigation illustrates the behavior of signaling cost against the CMR for the two vertical handoff scenarios considered. Fig. 4.10 illustrates the normalized signaling cost graphs for MIP-SIP and Pure-SIP when CMR is increased and λ *calls/min* is constant. As per the illustration in the graphs in Fig. 4.10, the normalized signaling cost for both the scenarios reduces exponentially as CMR increases. Finally, Fig. 4.11 illustrates normalized signaling cost against CMR by having η min^{-1} as a constant for the two considered scenarios. As per the illustration of the graphs in Fig. 4.11, the normalized signaling cost increases as CMR increases and eventually reaching a saturation point. Fig. 4.11 also indicates that the cost curve for the Pure-SIP method reaches this point much earlier since it incurs relatively higher signaling overhead. Similar to the above case, the effects of P_1 and P_2 on the following graphs have not been included in this discussion since its effects are equal for both these scenarios.

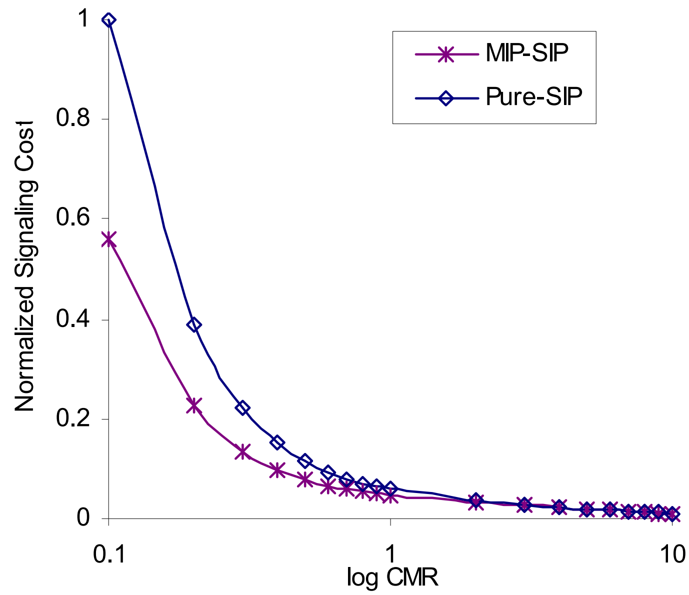


Fig. 4.10. Normalized Signaling Cost vs. CMR (λ calls/min constant).

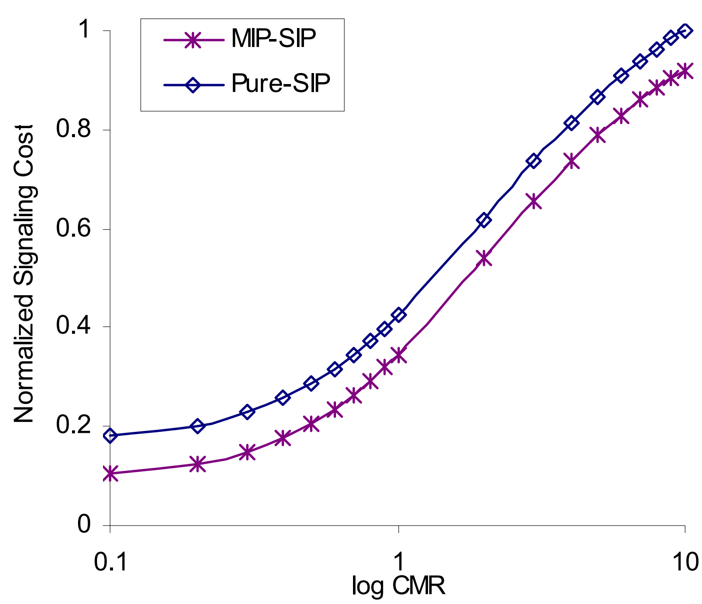


Fig. 4.11. Normalized Signaling Cost vs. CMR (η min^{-1}constant).

4.7 Network Modeling and Results

In order to investigate the validity of the above described architecture, a network simulation scenario is modeled using OPNET Modeler 11.5. Since OPNET's standard SIP components do not address the specifications of the IMS, substantial modifications are required for the development of a fully functional SIP-IMS model. Our newly developed SIP-IMS model is an enhanced version of the basic IMS-SIP signaling model, which is currently available under the contributed models library of the OPNET University Program [138]. These modifications can be summarized as follows: SIP UASs modified as CSCFs, IMS-SIP based messaging and flow generation between CSCFs, introduce roaming facility between multiple domains, and facility for introducing process delay controls (i.e. for messages sent between CSCFs and the HSS queries). As a result, an IMS that is capable of basic SIP signaling for session management is developed.

Next, underneath the existing IMS architecture a MIP v4 framework is constructed. The MIP framework consists of a common HA and multiple FAs. The intention of having MIP is for handling IP mobility at the Network Layer (and limiting the IMS exclusively for session mobility management). Followed by this is the task of creating an all-IP based, heterogeneous networking environment with MIP and SIP signaling as illustrated in Fig. 4.2. The simplest way to create such an environment (within the scope of OPNET 11.5) is by interworking between a WLAN and a UMTS core network. Since OPNET does not currently have a specialized module for CDMA2000 networks, it has been excluded from our simulation test-bed. Nevertheless, it can be argued that since the aim of these simulations are for validating the potential of combining the IMS with MIP for achieving a unified mobility platform at the core network level for an all-IP heterogeneous network, a WLAN-UMTS interworked test-bed should essentially be sufficient.

Details of the WLAN-UMTS interworking model are as follows. As per Fig. 4.1, the WLAN is connected via an SGSN/GGSN emulator to the all-IP core network. There are two types of signaling taking place. The first type is the IMS-SIP signaling, which takes place via the P-CSCFs (i.e., a SIP B2BUA) connected to each of these networks. These P-CSCFs are then connected to the S-CSCF, the IMS node implemented in the all-IP core network. It must also be noted that the I-CSCF is excluded in this model since it is mainly used at the time of

SIP Registration process and it is assumed that SIP registration has already taken place with the S-CSCF. The second type of signaling is the MIP signaling. In order to enable MIP signaling in UMTS and WLAN networks, FAs are setup at the GGSN and SGSN/GGSN emulator respectively. These FAs will be connected to the HA in the all-IP core network for MIP signaling.

Using the above described platform, measurements are collected for investigating a joint SIP-MIP vertical hand-off (say, from UMTS to WLAN in this case). At this instance, the measurements for vertical handoff are observed at the core network level and the radio network related delays and DHCP address distribution delays are not taken into consideration. The only delays considered are MIP registration requests, MIP binding update requests, and SIP related signaling and processing delays. The argument for measuring vertical handoff delay at the core network is needed to evaluate the joint SIP-MIP session and mobility management mechanism against our previously published works (i.e., a pure IMS-SIP based session and mobility management approach) [35]. Therefore these results may be applicable for an all-IP core network (under similar conditions).

The simulation results for vertical handoff delay for a VoIP session obtained for the two mobility mechanisms are as follows. The average session establishment delay for a VoIP session based on a MIP-SIP based joint mechanism for a UMTS and a WLAN interface are 240 ms and 200 ms respectively. Also, according to Section 3.4.3, the average session establishment delay for a VoIP session based on a Pure-SIP based joint mechanism for a UMTS and a WLAN interface are 197 ms and 182 ms respectively. Next, the average vertical handoff delay for a VoIP session based on a MIP-SIP based joint mechanism for a WLAN-to-UMTS handoff and a UMTS-to-WLAN handoff are 220 ms and 196 ms, respectively. Then again, according to Section 3.4.3, the average vertical handoff delay for a VoIP session based on a Pure-SIP based joint mechanism for a WLAN-to-UMTS handoff and a UMTS-to-WLAN handoff are 245 ms and 215 ms respectively.

Similar to our analytical results, the handoff delay for the MIP-SIP based architecture reveals lower delay as compared to the MIP-SIP combined approach. Furthermore, the above results show rather close resemblance to the analytical results presented in Section 4.6. However, the simulation results obtained for the average session establishment delay and the average handoff delay are approximately 12-20% higher of the analytical results. Such a degree of variation in the simulated result is unfortunately inevitable since the system initialization delays may affect the final outcome. The interesting point about our simulation results is that, to a certain extent, they are inline with results published for the case of horizontal handoff delays [164], [187]. As mentioned in the previous Chapter, a make-before-break handoff technique must be implemented for ensuring seamless vertical handoffs as described in [35].

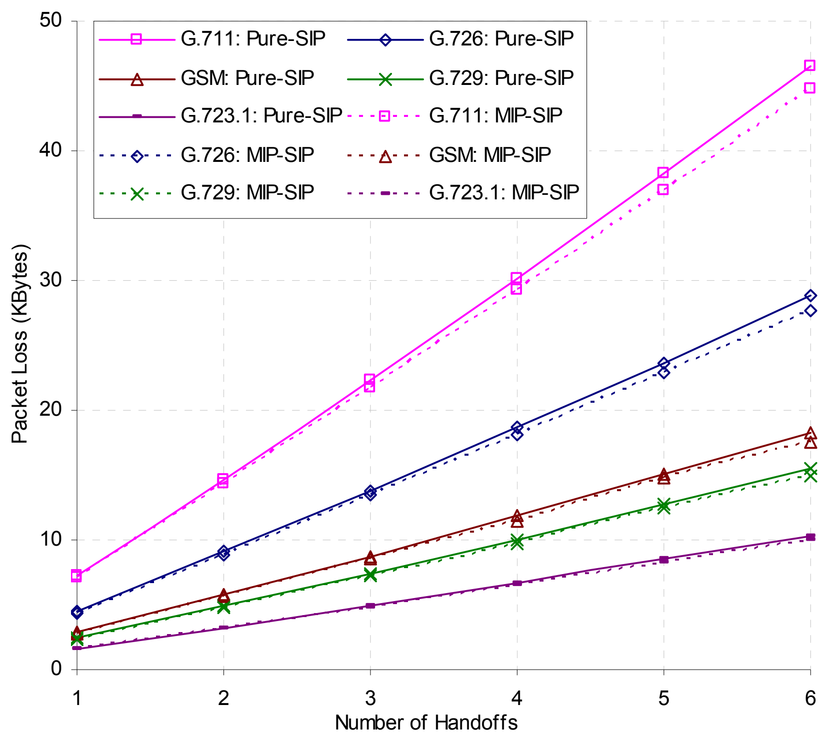


Fig. 4.12. Transient packet loss comparison for UMTS-to-WLAN vertical handoff for MIP-SIP and Pure-SIP mechanisms.

Figures 4.12 and 4.13 illustrate the transient packet loss scenarios for UMTS-to-WLAN and WLAN-to-UMTS handoff over different codecs respectively. These results can also be seen as an extension to the results presented in Fig. 3.8, where the performance of each codec behaves similarly. The packet loss is observed by simulating a break-before-make type handoff as described in detail in the previous section. According to the previous explanations, since the Pure-SIP method incurs relatively higher vertical handoff delay, it also incurs relatively higher transient packet loss in both the considered scenarios.

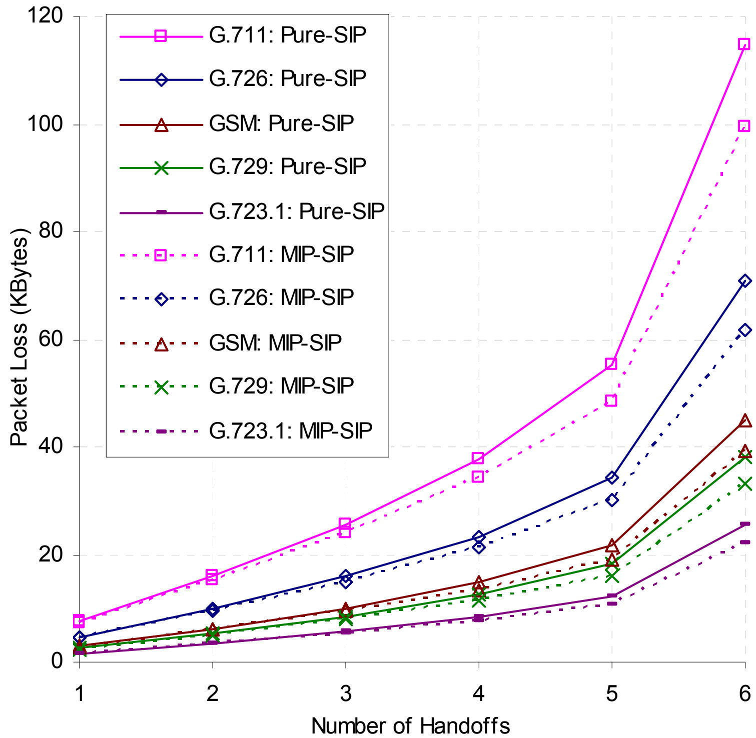


Fig. 4.13. Transient packet loss comparison for WLAN-to-UMTS vertical handoff for MIP-SIP and Pure-SIP mechanisms.

4.8 Summary and Conclusions

This Chapter focused on extending the proposed WLAN-UMTS interworking architecture to a unified framework for interworking dissimilar 3G cellular networking standards. As a result, global roaming and interoperability beyond one cellular system (say, UMTS) to another cellular system (say, CDMA2000) has become a possible reality. However, this required major architectural changes to the initial design. As such, a coupling framework similar to the 3GPP2's IMS was chosen. It also required re-designing of the entire mobility management framework. Instead of having a Pure-SIP based mobility management framework, a MIP and SIP combined mobility management framework was designed. Within this new framework, terminal/IP mobility was managed at the network layer by MIP and session mobility was handled by SIP at the Application Layer. The QoS analysis as well as the OPNET simulation model indicated that a MIP-SIP based mobility management framework performed better than its predecessor, the Pure-SIP based mobility management method. Also, the analysis results indicated that the QoS performance for the investigated parameters were within acceptable levels for real-time VoIP conversations. In order to claim that the proposed architecture is fully capable of interworking with all wireless heterogeneous networks, the possibility of interworking with BWA technologies such as a WiMAX network must also be investigated. Therefore, the next Chapter discusses how this architecture can be further extended for interworking with mobile WiMAX networks.

Chapter 5

A Unified Framework for Interworking Heterogeneous Mobile Wireless Networks

This Chapter further extends the proposed interworking architecture into a unified mobility and session management framework for interworking heterogeneous mobile and wireless networks. More specifically, this Chapter introduces interworking BWA networks (such as WiMAX) with the (previously proposed) existing platform. Therefore, the proposed platform will not only be capable of providing global roaming between multiple cellular systems, but also will be able to successfully interwork with 3G Cellular, WLAN, and WiMAX networks, thus creating a truly seamless inter-network roaming experience for the user. This Chapter is organized as follows: The next section establishes the importance for WiMAX networks to interwork with the existing 3G cellular networks. Followed by this comes the section which introduces the network architectures of Mobile WiMAX networks. Subsequently follows a brief discussion on the relevant work done in the area of interworking with other networks. Next the section on the proposed architecture, which describes how the IMS has been used as a universal coupling mediator for internetworking, is presented. Followed by this is the section is on the analytical modeling and the OPNET based simulation framework with relevant performance results prior to the final conclusions. The extended interworking architecture and the analytical model presented in this Chapter has contributed to [43] and the OPNET simulation results have partly contributed to [44].

5.1 Introduction

WiMAX is the BWA technology primarily designed for WMANs [188]. The IEEE 802.16 specifies a fixed WMAN protocol for providing a wireless alternative for last mile broadband access [14]. In December 2005, the IEEE ratified the 802.16e-2005 amendment to the IEEE 802.16 standard [15]. This amendment adds features and attributes to the standard to support nomadic mobility.

Unfortunately, the emerging popularity of this BWA technology, commercially known as Mobile WiMAX, was soon envisioned as a potential challenge to the long-term viability of existing wireless technologies, including IEEE 802.11-based WLANs, broadband residential Internet technologies such as Digital Subscriber Line (DSL) and cable, and the 3G cellular technologies [189]. Due to Mobile WiMAX technology's capabilities for reaching relatively high data rates (up to 15 Mbps) and spanning over a larger coverage area (up to 4 km) with mobility support, concerns were raised if this would become a cheaper (or perhaps a better) alternative to the existing 3G cellular technology (i.e., voice and narrowband data) [190]. This has resulted into a much intense competition between the fixed wireless/wired network service providers and cellular network operators to stake their claim in the rapidly growing wireless broadband market.

On the other hand, from a cooperative perspective, this may be treated as a powerful complimentary technology and an opportunity towards the evolution of 4G networks by interworking 3G cellular systems with fixed or mobile wireless access networks [191]. Therefore, a case for developing a platform for interworking between WiMAX and 3G cellular systems may be argued as follows. To begin with, mobile WiMAX technology is still in its infancy with many existing technical challenges [192]. Furthermore, the main hindrance to its competitive edge (over the 3G cellular system) is the lack of a fully defined networking infrastructure in the current mobile WiMAX standard [193]. Therefore, it may be assumed that the mobile WiMAX technology is insufficiently equipped to be a head-to-head competitor against the 3G network [194]. However, if it is interworked with the already well established 3G cellular networks such as UMTS or CDMA2000, it is probable that WiMAX will form an important part of the future 4G network [84], [110]. Thus the motivation for the works presented in this Chapter.

In the previous Chapters (Chapter 3 and Chapter 4), a novel approach towards realizing a unified mobility and session management platform at the core network level has been proposed [40]. This framework provides a common platform for interworking between UMTS, CDMA2000, and WLANs. Therefore, this Chapter further extends the existing platform for interworking with a WiMAX network. The novelty of this approach for WiMAX-UMTS-CDMA2000-WLAN interworking is that it uses the IMS as an arbitrator for real-time session negotiation and management as initially proposed in [35]. Furthermore, it may now be successfully argued that the proposed interworking architecture and the mobility management framework will be capable of withstanding the demands of the future 4G network.

5.2 WiMAX: Beyond the Radio Interface

The IEEE 802.16 BWA technology family, often referred to as WiMAX or WMAN, is intended to provide an alternative (and perhaps a more efficient) solution for the last mile broadband access challenge [14]. In December 2005 the IEEE ratified the 802.16e amendment to the 802.16 standard [15]. Based on the IEEE 802.16e amendment, features and attributes for supporting nomadic mobility (amongst other features such as superior data rates, scalability, and lower costs) were available. The WiMAX forum is currently going beyond the air interface and defining the network architecture necessary for implementing an end-to-end Mobile WiMAX network [195]. As illustrated in Fig. 5.1, the WiMAX network model comprises of the following networking entities: mobile user terminal, Access Service Network (ASN), Connectivity Services Network (CSN), user terminal including application agents, and Mobile Subscriber Station (MSS) [110].

The MN and the Base Transceiver Stations (BTS) sit at the network edge and are responsible for over-the-air transmissions. Further into the network, the ASN interfaces the BTS and the CSN connecting it to the all-IP core network. Typically the ASN includes numerous BTSs with one or more ASN gateways. The ASN manages radio resources, MN access, mobility, security and QoS. It acts as a relay for the CSN for IP address allocation and AAA functions.

The ASN gateway may host the MIP FA. The CSN performs core network functions, include policy and admission control, IP address allocation, billing and settlement. It hosts the MIP HA, the DHCP and AAA servers, and PSTN and VoIP gateways. The CSN is also responsible for internetworking with non-WiMAX networks (e.g. 3G, DSL) and for roaming through links to other CSNs.

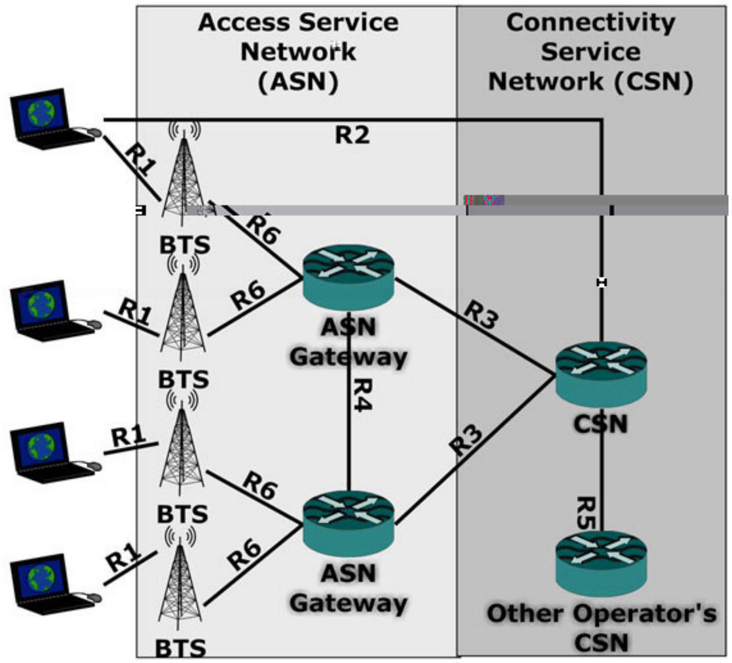


Fig. 5.1. WiMAX Network Architecture.

Specifications are being developed by the Network Working Group (NWG) within the WiMAX Forum for defining the role of the ASN and CSN and ensuring that WiMAX networks can interwork with other networks, using WiMAX or other wireless or wired access technologies.

In addition, the NWG specifications are designed to enable network operators to enjoy the benefits of vendor interoperability at the infrastructure level, to rely on a consistent client interface and, if they desire, to open their network to virtual operators, akin to existing cellular Mobile Virtual Network Operators (MVNOs) [196]. A detailed discussion on the WiMAX architecture is not included since it is beyond the scope of this thesis.

5.3 Related Work

With the above motivations, there have been some recent attempts for interworking WiMAX with the 3G cellular networks. The initial works concentrated on different flavors of the commonly known loose and tight coupling architectures [84]. Furthermore, throughout the past years there have been a series of proposals about different mobility management mechanisms as discussed subsequently.

In the case of WiMAX, MIP based approaches for mobility management have been the most widely proposed [85]. Although MIP is recognized as a simple but effective solution for mobility in mobile WiMAX networks in general, it suffers from several drawbacks [57]. Firstly, since it is a purely network layer oriented solution; it is unable to solve any Link Layer handoff problems. Secondly, triangular routing may result in extended delays. Thirdly, it may not deal with vertical handoffs due to discrepancies in the mobility management techniques of a heterogeneous networking environment (e.g., UMTS core network does not use MIP).

One way of avoiding the above drawbacks is by adopting our previously proposed unified coupling framework for heterogeneous networks [35]. This framework provides a MN the highest possible level of internetworking, with fully seamless continuity of service across heterogeneous networks. As an arbitrator for internetworking between the BWA technology (WiMAX in this case) and the 3G Cellular network (either UMTS or CDMA2000), the IMS is deployed. A clear advantage of using the IMS is its ability for real-time session negotiation and management using SIP.

Nevertheless, MIP also plays an important role for IP (terminal) mobility management in our approach. Although a similar idea of using the IMS as a coupling medium (only for session mobility management) has been recently published at a much later stage in [110], it mainly focuses on the QoS guarantee and the AAA aspects provided by the IMS. Most importantly the architecture fails to sufficiently describe a mechanism for seamless vertical handoff and mobility management. Hence, prior to introducing this novel architecture, a brief overview of the IMS and design concepts will be presented

5.4 The Interworking Architecture

The extended internetworking architecture is illustrated in Fig. 5.2. The UMTS core network is connected to the IP network through the GGSN, which also acts as its MIPv4 FA. Once the system acquisition is done by a MN connected to the UMTS network, the next step is to set up a data pipeline. The actual IP address allocation for the MN is initiated by sending the MIPv4 registration request to it's HA via the GGSN (i.e., the MIP-FA). This mechanism is based on the specifications given under [186]. The MN acts as an IMS-SIP client and sends a SIP registration message to its home system through the P-CSCF. Once authorized, a suitable S-CSCF for the MN is assigned and its subscriber profile is sent to the designated S-CSCF.

After the activation of the PDP context and the service registration, the MN is ready to establish a data session. As illustrated in Fig. 5.3, the sequence of the SIP session origination procedure can be described as follows. The mobile origination procedure is initiated by a SIP INVITE message sent from the UMTS interface of the source MN. This initial message is forwarded from the P-CSCF to the S-CSCF of the originating network, via the CSCFs of the terminating network, and finally to the destination. This SIP INVITE carries a request to follow the precondition call flow model. This is important because some clients require certain preconditions (that is, QoS levels) to be met before establishing a session.

Once this is done, the MN can start the media/data flow and the session will be in progress (via the UMTS interface). When this MN roams between WiMAX and UMTS systems, inter-system roaming takes place. Generally, the selection of the appropriate network during inter-system roaming is facilitated by an external trigger such as a network selection mechanism/algorithm (this has not been taken into consideration since it is beyond the scope of this thesis). Once the new network has been identified the MN undergoes relevant access registration procedures (in this case, the WiMAX network) and AAA procedures. Next, the vertical handoff process is initiated.

In the case of a WiMAX network, the general handoff process may either take place at layer 2 or layer 3. Layer 2 handoff, which is also known as "micro mobility," simply changes the air interface attachment point but keeps the IP attachment point unchanged. This process involves the detection of signal strength, releasing the connection with the serving BS, and establishing

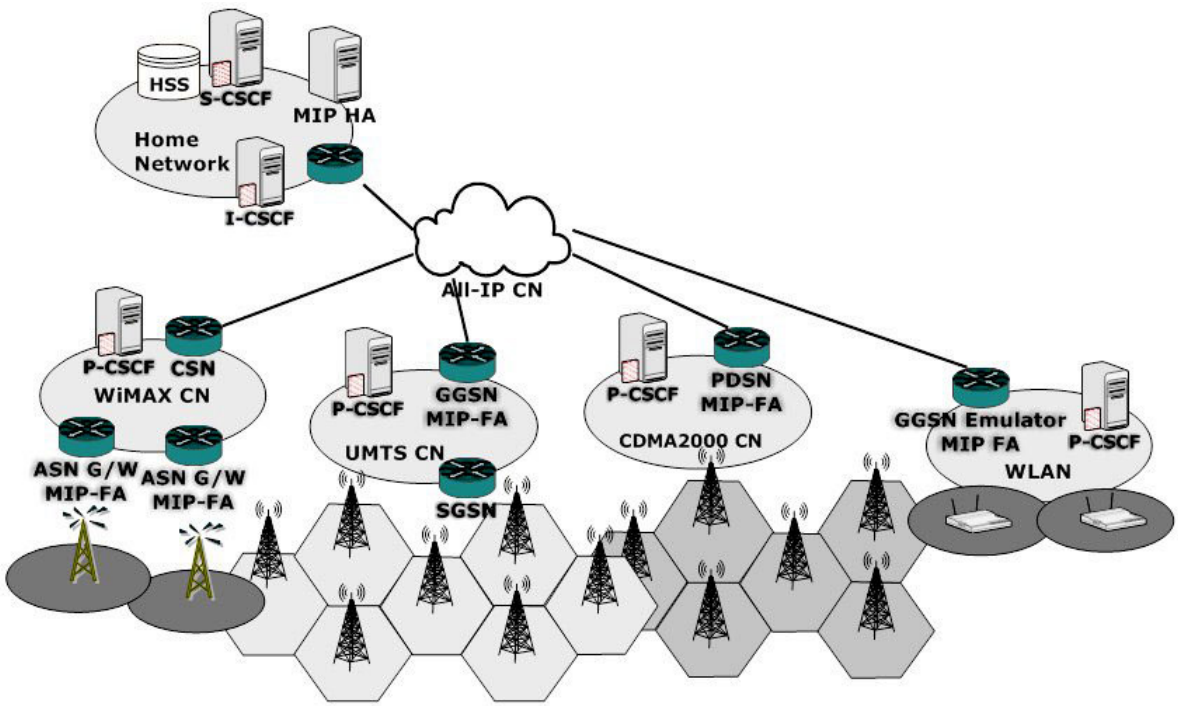


Fig. 5.2. The Extended Interworking Architecture.

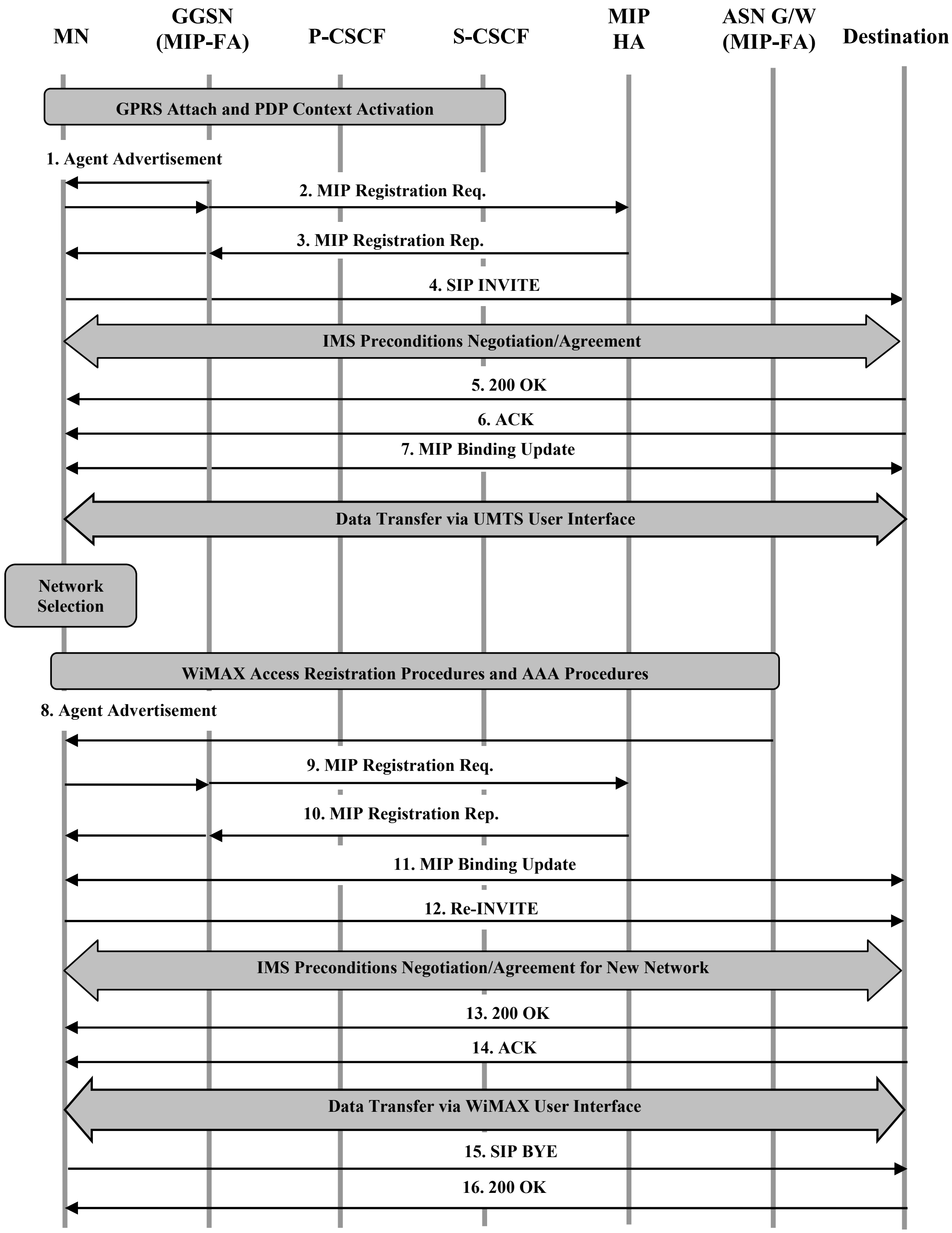


Fig. 5.3. UMTS-WLAN Session Handoff Signaling.

a connection with the target BS. Also, a Layer 2 handoff is transparent to the upper layer protocols. On the other hand, Layer 3 handoff, which is often referred as "macro mobility," changes the IP attachment point of a mobile user. During a layer 3 handoff, the MIP protocol is used to update the HA with the CoA of the MN. Therefore, during a layer 3 handoff, the MN must be registered and authenticated with the HA every time it moves from the serving BS to the target BS.

Since the proposed architecture is designed for facilitating IP and session mobility, a layer 3 handoff is performed for facilitating inter-system roaming as illustrated by Fig. 5.3. Once the MN undergoes relevant access registration and AAA procedures the WiMAX interface performs the MIP registration procedures with the ASN Gateway (MIP FA) as explained previously. This is when the ASN Gateway (MIP-FA) forwards this request (via the CSN) to the MIP-HA and the HA assigns the home IP address to the new WiMAX interface. Followed by this the exchanging of a MIP BU message between the MN and the destination for avoiding triangular routing takes place [125].

The next stage is the taking place of the IMS-SIP session handoff procedures. This requires sending a SIP ReINVITE (with same Call-ID and other identifiers corresponding to the ongoing session) to the destination SIP UAC. Followed by this is a resource/preconditions reservation for the WiMAX interface. Once this is successfully done the new session flow can be initiated. It is important to note that until such time that the new data flow is initiated via the WiMAX interface, the data flow via the UMTS interface remains active. Thus the model follows the make-before-break handoff mechanism as previously discussed in [35]. Inter-system roaming from WiMAX to UMTS can also take place in a similar manner. Furthermore, since this design is an extension to our WLAN-UMTS-CDMA2000 interworking platform, WiMAX-CDMA2000 roaming can also be accommodated within this architecture in a similar manner.

5.5 Analytical Modeling

This section provides an analytically modeled approach for evaluating the proposed extended version of the interworking architecture. The primary aim of this analytical model is to help analyze the feasibility of the platform to interwork with BWA technologies (i.e., UMTS-WiMAX interworking).

This analysis uses a similar approach, as explained in Section 3.5, for evaluating QoS metrics and measures involved in the chosen MIP and SIP combined mobility management approach. With the help of this model, QoS metrics such as reconfiguration and vertical handoff delay (from the network layer and above), transient packet loss, jitter, and signaling overhead are analyzed and compared. As previously mentioned, since the aim of this thesis is to develop a universal mobility management platform by having the IMS as a common arbitrator, delay components relating to the Physical and Link Layers of various wireless networks/links (i.e., UMTS, CDMA2000, WLAN, and WiMAX) have been overlooked. This analysis is also based on a similar assumption as in Section 3.5, that the call/session arrivals follow a Poisson process. Furthermore, it can be stated that the arguments provided in Section 3.5 for justifying VoIP session arrivals to be modeled as a Poisson process will also be similarly applicable.

5.5.1 Handoff Delay

In this section, the derived formula under Section 3.5.1 will be appropriately modified for a comparative analysis of the vertical session handoff delay for the newly interworked WiMAX interface as illustrated in Fig. 5.1 and Fig. 5.2. A modified analysis method is used for comparing the efficiency of the new MIP-SIP based vertical session handoff mechanism (delay, in this case) against the previously presented Pure-SIP based session handoff mechanism, for a data session establishment within the context of a UMTS-WiMAX interworking framework.

Similar to the approaches followed under Sections 3.5.1 and 4.5.1, the vertical handoff sub-procedural delays such as the Link Layer handoff delay, movement detection delay, and packet re-transmission delay have not been taken into consideration. Additionally, according to the proposed enhancements to the architecture for an SIP-MIP based vertical handoff (as illustrated in Fig. 5.2), there is no DHCP based address allocation delay. Thus the main contributor to the network layer based vertical handoff delay is the session re-configuration delay. Therefore, it can be expressed that the new vertical session handoff delay (at the core network level regardless of the underlying radio network and link layer delays) for the newly considered scenario mainly consists of the (previously considered) IMS based session handoff delay at the application layer and the newly introduced MIP related delay at the Network Layer. Another reason for the exclusion of the above mentioned delay components is that, these delay components may significantly vary depending on the radio network technology used, and thus not being able to provide a fair evaluation of the performance of the interworking architecture.

In order to derive an expression for a UMTS-to-WiMAX vertical handoff delay for a MIP-SIP combined method, equation (3.5) will be used. As mentioned under Section 3.1, equation (3.5) expresses the end-to-end transmission delay for a packet size S, sent from network A to network B over a number of hops via wireless and wired links. This has been directly applied to the entire MIP and IMS SIP signaling flow involved in the UMTS-to-WiMAX vertical handoff mechanism illustrated in Fig. 5.2. Therefore, the final expression in this case (i.e., a MIP and SIP combined handoff approach), can be expressed as a combination of the following end-to-end delay components as expressed in equation (5.1).

$$\begin{aligned} D_{UMTS\text{-}WIMAX} = {} & D(S_{MIPReq}, H_{UMTS\text{-}MIP\text{-}HA}) + D(S_{MIPRep}, H_{UMTS\text{-}MIP\text{-}HA}) \\ & + D(S_{MIP\text{-}BU}, H_{UMTS\text{-}CN}) + D(S_{ReINVITE}, H_{WiMAX\text{-}CN}) + D(S_{183\text{-}SP}, H_{WiMAX\text{-}CN}) \\ & + D(S_{PRACK}, H_{WiMAX\text{-}CN}) + D(S_{OK}, H_{WiMAX\text{-}CN}) + D(S_{UPDATE}, H_{WiMAX\text{-}CN}) \\ & + D(S_{OK}, H_{WiMAX\text{-}CN}) + D(S_{ACK}, H_{WiMAX\text{-}CN}) + D(S_{BYE}, H_{UMTS\text{-}CN}) \\ & + D(S_{OK}, H_{CN\text{-}UMTS}) + \Delta \end{aligned} \tag{5.1}$$

where, Δ is additional IMS (application layer) latency. Thus equation (5.1) is capable of providing the total vertical session handoff delay for a MIP and SIP combined scenario. Although this particular scenario concentrates on a vertical handoff from UMTS-to-WiMAX, previous Chapters have presented how other vertical handoff scenarios relating to different radio interfaces such as WLAN-to-UMTS, UMTS-to-CDMA2000, CDMA2000-to-UMTS could be constructed with appropriate substitutions using equation (3.5). As, mentioned under Section, 3.5.1, the important point to note here is that since it has been assumed that the channel is error free during the vertical handoff, the likelihood of the occurrence of errors has been overlooked. This is since for successful session establishment the entire message flow must take place and if any message is damaged or lost the vertical handoff process will fail.

As mentioned earlier, the above analysis assumes that the VoIP call arrivals follow a Poisson arrival process. However, recent studies on the Internet traffic have strongly indicated that the Poisson packet arrivals and exponential packet lengths studied in the classical queuing theory are basically inappropriate for modeling Internet traffic [144] [197] [198]. Furthermore, these studies point out that long-tailed distributions serve as better models for packet inter-arrival times and service lengths for Internet traffic [199] [200] [201] [202]. Therefore, for the sake of completeness, this thesis also extends the analysis for the case of a Pareto based VoIP session arrival for a MIP-SIP based mechanism under Appendix B. The results of the analysis given under Appendix B points out that, modeling VoIP session arrivals to be Poisson is an adequate assumption according to the conclusions of [147], especially when the system utilization is relatively low (i.e., less than 50%).

5.5.2 Packet Loss

The total packet loss during a UMTS-WiMAX vertical session handoff is also analyzed similarly to Section 3.5.2. Thus, the total packet loss ($Loss_{UMTS\text{-}WiMAX}$) during a MIP-SIP combined UMTS-to-WiMAX vertical session handoff can be similarly defined as the sum of all lost packets during the vertical handoff while the MN is receiving downlink data packets. It is assumed that the packet loss begins when the Layer 2 handoff is detected and all in-flight packets are lost during the vertical handoff time. Thus, it can be expressed as follows:

$$Loss_{UMTS-WiMAX} = \left[\frac{1}{2T_{ad}} + D_{UMTS-WiMAX} \right] \times \lambda_d \times N_m \tag{5.2}$$

where, T_{ad} is the time interval between P-CSCF discovery times, λ_d is the downlink packet arrival rate at the wireless interface (this depends on the used voice codec), and N_m is the average number of vertical handoffs during a single session [154].

5.5.3 Signaling Overhead/Cost

In order to keep consistency with the previous two Chapters, this Section also presents a comparative analysis for the signaling cost. Similar to the definition used under Section 3.5.3, signaling cost or overhead is defined as the accumulative traffic load on exchanging signaling messages during the MN's communication session (i.e., session establishment, session handoff, and so on). Therefore, similar to equation (3.8), the above definition can be expressed as:

$$Cost = P \times S_{message} \times H_{A-B} \tag{5.3}$$

where, P defines the probability associated with the occurrence of a particular event, $S_{message}$ is the average size of a signaling message/packet related to this event, H_{A-B} is the average number of hops/distance the signaling packet traverses between the source node A to the destination node B. Using the above definition and a mobility scenario similar to the illustration in Fig. 3.12, an expression for signaling cost is derived for a MIP-SIP based scenario.

Similar to Chapter 4, the signaling cost for the UMTS-WLAN based vertical handoff scenario illustrated in Fig. 5.3, can be expressed as the sum of the following two signaling components. The first component of signaling cost computation is associated with the entire MIP and SIP INVITE (session establishment) message sequence. As mentioned under Section 3.5.3, the computation of this signaling cost component is associated with P_1, which is the arrival probability of a session to a MN that is likely to be subjected to vertical handoff. The second component of signaling cost computation is associated with the MIP and SIP ReINVITE (vertical handoff) message sequence. Similar to what has been mentioned under Section 3.5.3, the MIP and SIP ReINVITE sequence is associated with P_2, which are the vertical handoff probability and the inter-network mobility rate. Detailed steps for the derivation for P_1 and P_2 have been omitted from this Section and the reader is referred to Section 3.5.3 for the same. Finally, the total signaling overhead incurred by vertical handoffs during a given data session can be expressed as:

$$Cost = P_1\lambda\sum_{i=1}^{n_1}\left(S_{INVITE_i} \times H_{(A-B)_i}\right)+P_2\eta\sum_{i=1}^{n_2}\left(S_{\mathrm{Re}\,INVITE_i} \times H_{(A-B)_i}\right) \tag{5.4}$$

where, λ *call/min* is the average session arrival rate, η min^{-1} is the average network mobility rate associated with a roaming MN, n_1 and n_2 represent the number of messages involved in each handoff/message exchange sequence as illustrated in Fig. 5.3. Further, if η is the average network mobility rate of a MN and λ is the average session arrival rate, λ/η may be defined as the *CMR* [164]. Thus (5.4) can be re-arranged as:

$$Cost = \left[P_1\eta\sum_{i=1}^{n_1}\left(S_{INVITE_i} \times H_{(A-B)_i}\right)\right]\frac{\lambda}{\eta} + P_2\eta\sum_{i=1}^{n_2}\left(S_{\mathrm{Re}\,Invite_i} \times H_{(A-B)_i}\right) \tag{5.5}$$

$$Cost = \left[P_1\eta\sum_{i=1}^{n_1}\left(S_{INVITE_i} \times H_{(A-B)_i}\right)\right]CMR + P_2\eta\sum_{i=1}^{n_2}\left(S_{\mathrm{Re}\,Invite_i} \times H_{(A-B)_i}\right) \tag{5.6}$$

5.6 Performance Analysis

The following numerical results are generated using MIPv4 and the 3GPP-SIP messages. In order to maintain consistency, standard SIP message sizes and other related parameters have been used, similar to the previous Chapters. This is because this interworking architecture is conceptually similar to the MIP-SIP signaling framework introduced in Chapter 4.

Furthermore, since this architecture gives the freedom to interface with different radio network interfaces without any changes to the core network, the relative distances between the elements of the core network do not change. Hence, in order to maintain consistency, the analysis uses similar relative distances for hops as per the illustration in Fig. 4.3. Although Fig. 5.4 may have similar hop counts to what is used for the analysis in Chapter 4, under the considered scenario the MIP-FA is implemented on the WiMAX gateway, as illustrated in Fig. 5.4.

Based on the values of MIP and IMS-SIP message sizes and the hop counts of Fig. 5.4, the following analytical results are derived for equations (5.1), (5.2), (5.4) and (5.6) for the scenario of a UMTS-WiMAX based vertical handoff. Furthermore, a second scenario is simulated using a fully IMS dependant Pure-SIP based mobility management mechanism as a benchmark for comparison.

This mechanism is based on SIP REFER (RFC 3515) method with relatively high signaling overheads and IMS related latencies. Since the SIP REFER method supports IP/terminal mobility as well as session mobility in providing seamless service continuation during a vertical handoff, the use of MIP for supporting IP/terminal mobility is not required. Since Chapter 3 discusses details of this Pure-SIP method, details have been omitted from this Section.

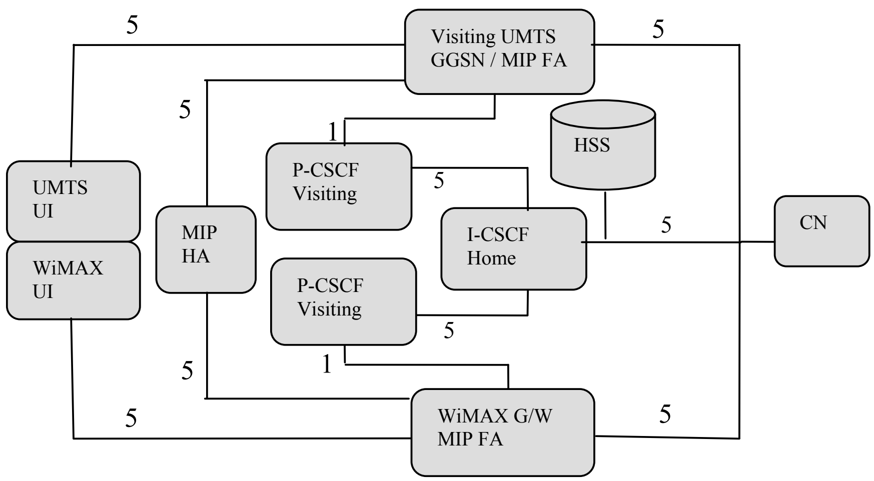


Fig. 5.4. Relative Distances in Hops.

Figure 5.5 illustrates the behavior of vertical handoff delay against increasing session handoffs for UMTS-to-WiMAX and WiMAX-to-UMTS interfaces with respect to MIP-SIP and Pure-SIP based handoff mechanism. Interestingly enough, the graphs on Fig. 5.5 show close resemblance to the graphs in Fig. 4.4 and Fig. 4.5. Therefore, WiMAX-to-UMTS shows a relatively higher vertical handoff delay in contrast to a UMTS-to-WiMAX scenario. This behavior is in line with the behavior experienced in the previous Chapter. Hence it can be confirmed that, when a session is transferred to a network with a relatively lower link bandwidth, a relatively higher vertical handoff delay may be expected.

The next important point to note for the WiMAX-to-UMTS case is that as the number of handoffs increases from 5 to 6, an extremely sharp exponential increase in delay is noted. What happens here is that due to multiple handoffs being processed by the link (with a relatively low bandwidth), the utilization of the system $\rho=\lambda/\mu \rightarrow 1$. If the handoffs are further increased to a point where $\lambda>\mu$ is reached, the arrival rate increases beyond the maximum capacity of the link, which eventually fails the link. However, this observation is not there in the case of UMTS-to-WiMAX. This is since the session is handed-off to a network with relatively high bandwidth and the system is far from reaching a state of saturation.

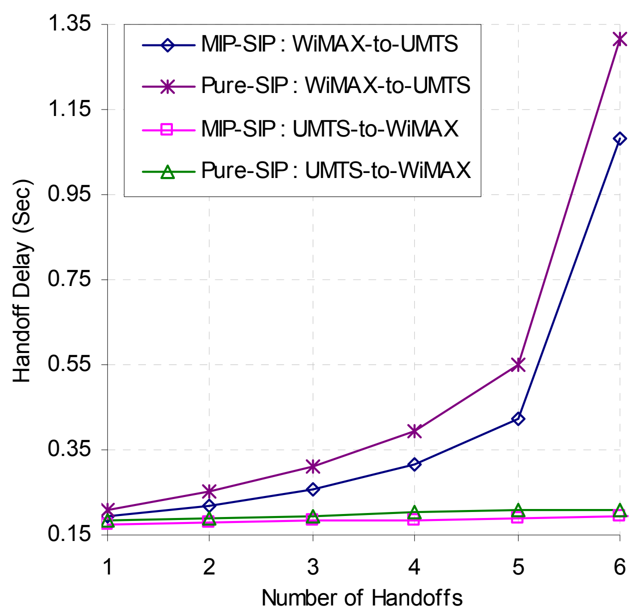


Fig. 5.5. Vertical Handoff Delay vs. Number of Session Handoffs.

Furthermore, according to the presented analytical model, for a MIP-SIP combined mechanism, the average session establishment delay over the WiMAX interface is 174 ms and the same for a Pure-SIP based mechanism is 152 ms. Next, the average vertical handoff delays based on the MIP-SIP and Pure-SIP mechanisms from UMTS-to-WiMAX are 174 ms and 183 ms, respectively. Similar to the results presented in Chapter 3 and Chapter 4, the combined MIP-SIP based handoff mechanism shows an overall lower vertical handoff delay in comparison to a Pure-SIP based approach. This is due to the Pure-SIP based approaches with relatively high number of IMS related application layer message flows and latencies.

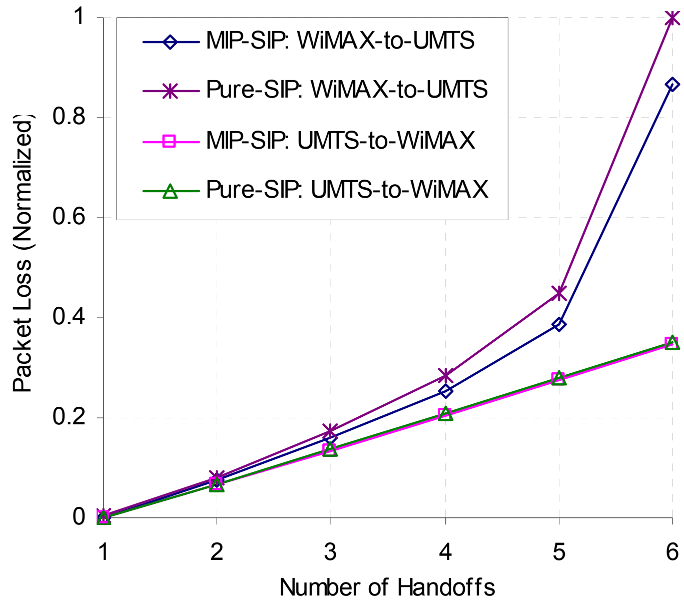


Fig. 5.6. Transient Packet Loss vs. Number of Handoffs.

Figure 5.6 illustrates the normalized transient packet loss during vertical handoffs from WiMAX-to-UMTS and vice-versa for MIP-SIP and Pure-SIP based mechanisms as the number of handoffs increases (in the case of a break-before-make handoff scenario). According to the previous explanations given, it is clear that the packet loss during a vertical handoff is directly proportional to the vertical handoff delay. Hence, the WiMAX-to-UMTS graphs indicate near exponential packet loss in Fig. 5.6. Similarly, the packet loss is relatively low for a UMTS-to-WiMAX handoff, which is in line with the corresponding handoff delay shown in Fig. 5.5. Furthermore, as expected, Pure-SIP approach shows a relatively high packet loss for all scenarios.

Figure 5.7 illustrates the comparison of the behavior of jitter, which is the variation of delay, during an UMTS-to-WiMAX and UMTS-to-WLAN vertical handoff scenarios. This graph compares the jitter result obtained in the previous Chapter for an UMTS-to-WiMAX case against similarly obtained results for an UMTS-to-WLAN case investigated in this Chapter. Despite the UMTS-to-WLAN case shows relatively higher jitter values, the overall jitter rates are still within acceptable limits for a VoIP session. Then again, beyond a certain number of handoffs, both the jitter curves tend to indicate a very close and exponentially increasing trend. Thus by comparing Fig. 5.7 against Fig. 3.18 it could be argued that the maximum number of possible successful handoffs are more dependant on the data rate of the voice codec used rather than the link bandwidths.

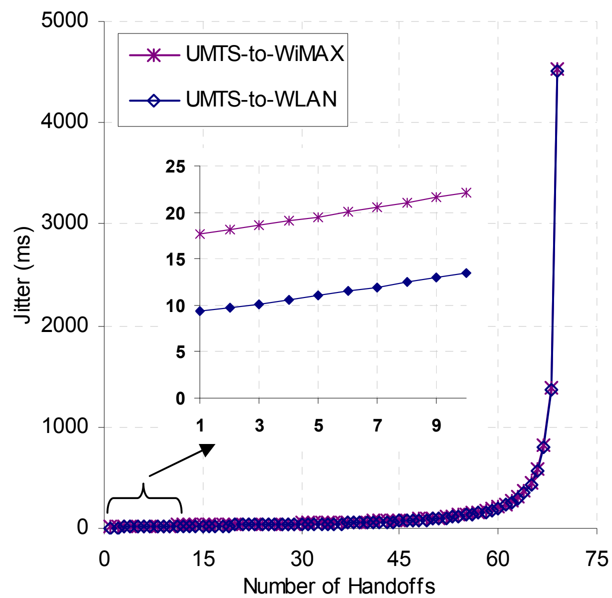


Fig. 5.7. Jitter vs. Number of Handoffs.

It must be noted that since a MIP-SIP based vertical handoff mechanism (similar to what was introduced in Chapter 4) has been used for the considered UMTS-to-WiMAX case, signaling cost analysis may indicate similar trends with the previous Chapter. Nevertheless, in order to maintain consistency with the previous Chapters, a brief analysis of signaling cost (with a different perspective) is presented under this Section.

Figure 5.8 compares the normalized signaling cost against CMR when network mobility rate $\eta\ min^{-1}$ and service rate (μC_i) are constant (say, case 1) and the same when session arrival rate $\lambda\ calls/min^{-1}$ and service rate (μC_i) are constant (say, case 2). The graphs relating to case 1 show a rather slow increase in the normalized signaling cost as the CMR linearly increases. As the CMR linearly increases, by keeping the network mobility rate $\eta\ min^{-1}$ and service rate (μC_i)

constant, the session arrival rate λ *calls/min* is expected to linearly increase. However, it exponentially reduces P_1. This eventually results in the slowing down of the signaling cost and hence such a behavior is indicated. The graphs corresponding to case 2 show how the signaling cost exponentially reduces with increasing CMR while the session arrival rate λ *calls/min* and service rate (μC_i) are constant. The explanation is that, a result of linearly increasing the CMR, the network mobility rate η min^{-1} decreases exponentially. This eventually results in an exponential reduction of the signaling cost.

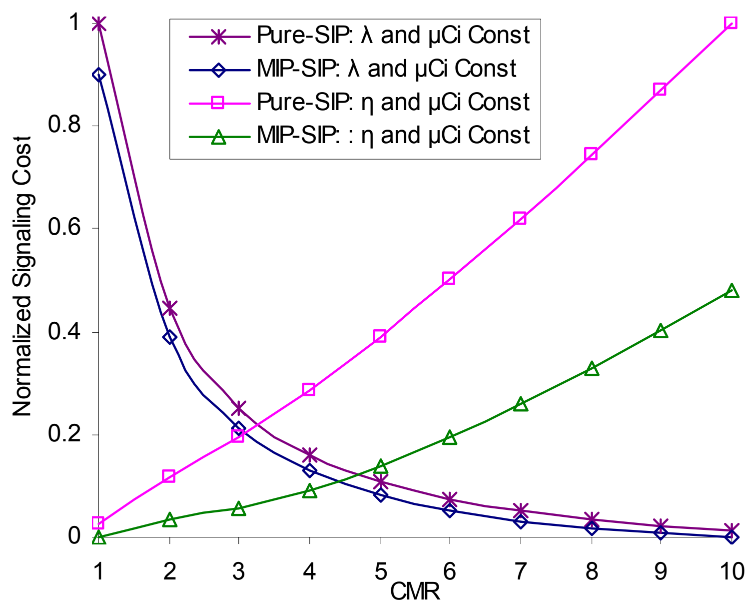


Fig. 5.8. Signaling Cost vs. Call-to-Mobility Rate (CMR).

Figure 5.9 illustrates how signaling cost behaves against network mobility rate η min^{-1} when the service rate (μC_i) and the session arrival rate λ *calls/min* are constant. As for this scenario, when network mobility rate η min^{-1} is increased, P_1 and P_2 linearly increases, which eventually increases the signaling cost linearly. Lastly, Fig. 5.10 illustrates how signaling cost behaves against increasing session arrival rate λ *calls/min* when the network mobility rate η min^{-1} and the service rate (μC_i) are constant. As the session arrival rate λ *calls/min* increases the system eventually reaches its saturation point. This is indicated by the stabilization effect of the signaling cost curves. It is also important to note that in all the above considered cases, the signaling method with lesser interactions with the IMS model (i.e., the MIP-SIP mechanism) always indicated the lowest signaling cost.

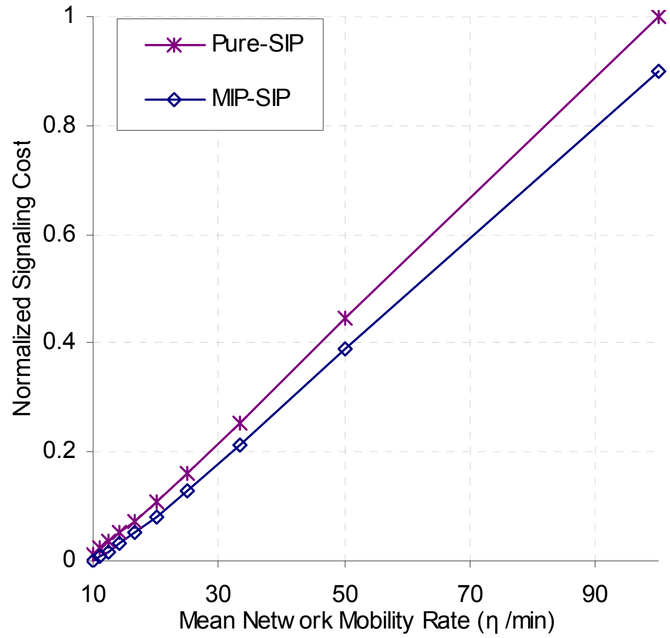


Fig. 5.9. Signaling Cost vs. η when μC_i and λ are constant.

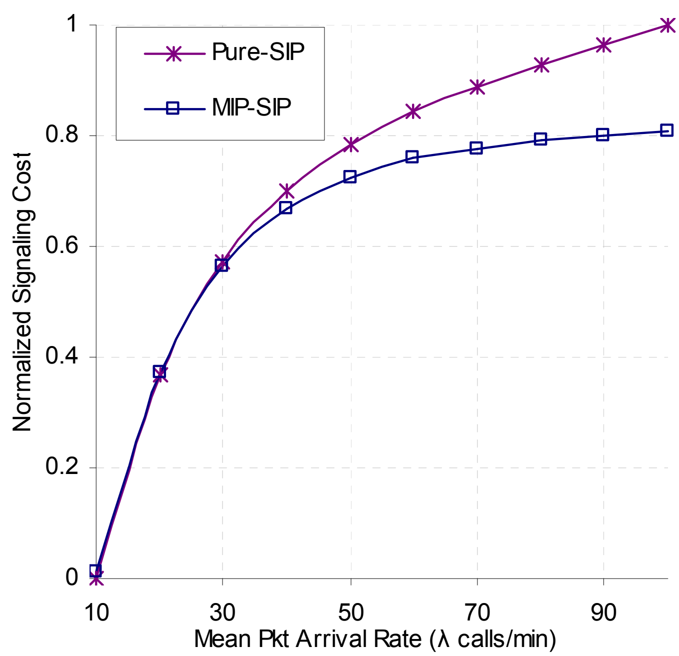


Fig. 5.10. Signaling Cost vs. λ when μC_i and η are constant.

5.7 Network Modeling and Results

5.7.1 Simulation Platform

In order to investigate the ability for interworking in the presented architecture, a similar simulation scenario as in [40] is used. However, since the OPNET Modeler 11.5 based simulation platform used in [40] does not support WiMAX capability, these simulations are performed on an upgraded OPNET Modeler 14.0 platform. Since OPNET's standard SIP components do not address the specifications of the 3GPP's IMS, substantial modifications are required. Thus a fully functional SIP-IMS model for OPNET is constructed and integrated to OPNET's existing UMTS Special Module. The newly developed SIP-IMS model is an enhanced version of the basic IMS-SIP signaling model, which is currently available under the contributed models library of the OPNET University Program [138]. Fig. 5.11 illustrates the constructed simulation scenario.

As in the previous Chapters, modifications are made for SIP UASs to function as different CSCFs, UAC processes to communicate with modified UASs, IMS-SIP based messaging and flow between the CSCFs, roaming facility between multiple domains, and facility for introducing process delay controls (i.e. for messages sent between CSCFs and the HSS queries). As a result, a UMTS network that is fully capable of using IMS based SIP signaling for session management is developed. Followed by this, below the IMS architecture, a MIP v4 framework is constructed.

The next task is to create an all-IP based heterogeneous network with MIP and SIP signaling as illustrated in Fig. 5.11. Since IMS and MIP protocols are implemented in the core network, their behavior can be considered to be independent to the underlying networks (i.e., to WiMAX, UMTS, or CDMA2000). Taking the above facts and limitations of OPNET simulator into consideration, a simple all-IP heterogeneous test bed is created by interworking a UMTS network with a WiMAX network (similar to Chapter 4). Next, measurements are collected for investigating a joint MIP-SIP vertical hand-off (say, between UMTS and WiMAX in this case). The reader is referred to [35] and [40] for specific details of this simulation platform.

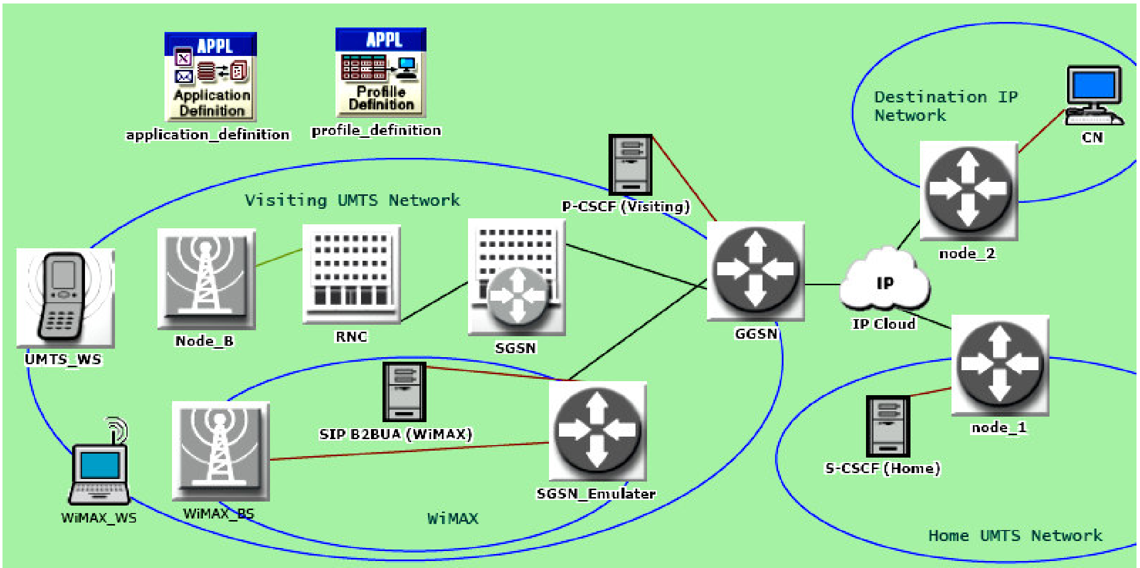


Fig. 5.11. OPNET 14.0 Simulation Model.

This platform is used for simulating and studying session establishment and vertical handoff delay, transient packet loss and jitter for a session that is handed off from WiMAX to UMTS. Five different types of voice codecs are used for voice traffic generation. These are namely; G.711 (data rate of 64 kbps), G.726 (data rate of 32 kbps), GSM (data rate of 13 kbps, G.729 (data rate of 8 kbps), and G.723.1 (data rate of 5.3 kbps). Besides, the generated voice packets are considered to have a fixed IP header of 40 bytes, which includes a 12-byte RTP header, an 8-byte UDP header, and a 20-byte IP header. Beyond this, depending on the transmission medium, an additional overhead of 6-byte fixed-length generic MAC header and a 4-byte CRC is used for WiMAX and an overhead of 6 bytes adds up if the UTRAN is used. Furthermore, no header compression option is considered at UTRAN PDCP layer, no silence suppression used, and no playout buffer used to compensate jitter in these simulations.

5.7.2 Simulation Results

Firstly the average session establishment and vertical handoff delays for the above scenarios are investigated. The average MIP-SIP based VoIP session establishment delay over the UMTS interface is 240 ms and the same over the WiMAX interface is 210 ms. The average vertical handoff delays for IMS-SIP controlled VoIP sessions are as follows: A vertical handoff from WiMAX-to-UMTS incurs 220 ms and UMTS-to-WiMAX incurs 195 ms. The large 3GPP SIP message sizes and heavy application layer based IMS latencies (e.g., HSS look-up, SIP ReINVITE message exchange, and routing all SIP signaling via the Home Network) are the major contributors towards the vertical handoff delay. Furthermore, the above results show rather close resemblance to the analytical results presented in Section 5.6. However, the simulation results obtained for the average session establishment delay and the average handoff delay are approximately 15-25% higher of the analytical results. Such a degree of variation in the simulated result is unfortunately inevitable since the system initialization delays may affect the final outcome. Nevertheless, it is a highly encouraging observation to obtain.

Secondly, the transient packet loss during a vertical handoff for different codecs is investigated. This transient packet loss is observed by simulating a break-before-make type handoff as described in detail in [35]. Fig. 5.12 and Fig. 5.13 illustrate the transient packet losses for UMTS-to-WiMAX and WiMAX-to-UMTS vertical handoffs. As indicated, G.711 and G.726 codecs experience relatively higher levels of transient packet loss. The relatively higher frame sizes used by these codecs are seen as the main contributor towards such high transient packet losses. Furthermore, if closely observed, Fig. 5.12 and Fig. 5.13 have a close resemblance to the trend shown in Fig. 3.10, which is the transient packet loss for a UMTS-to-WLAN and WLAN-to-UMTS vertical handoff scenario. Therefore, Fig. 5.14 provides a transient packet loss comparison for a UMTS-to-WiMAX handoff and a UMTS-to-WLAN handoff. For all codecs used, UMTS-to-WLAN vertical handoff experiences relatively higher packet loss.

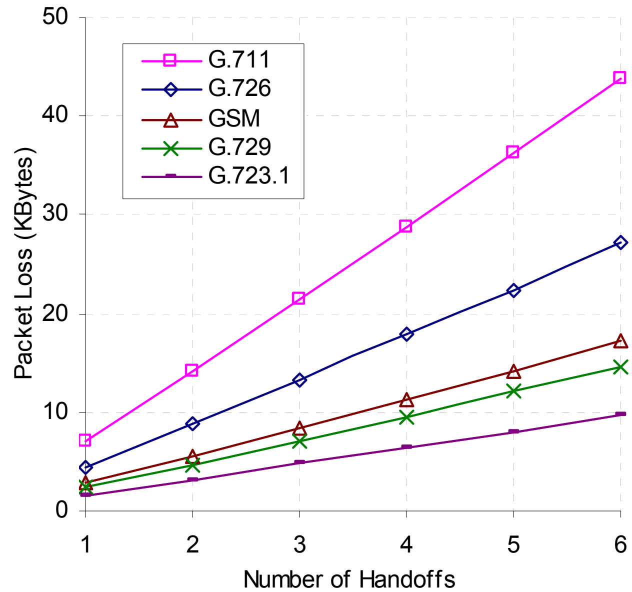


Fig. 5.12. Transient Packet Loss for a UMTS-to-WiMAX Handoff.

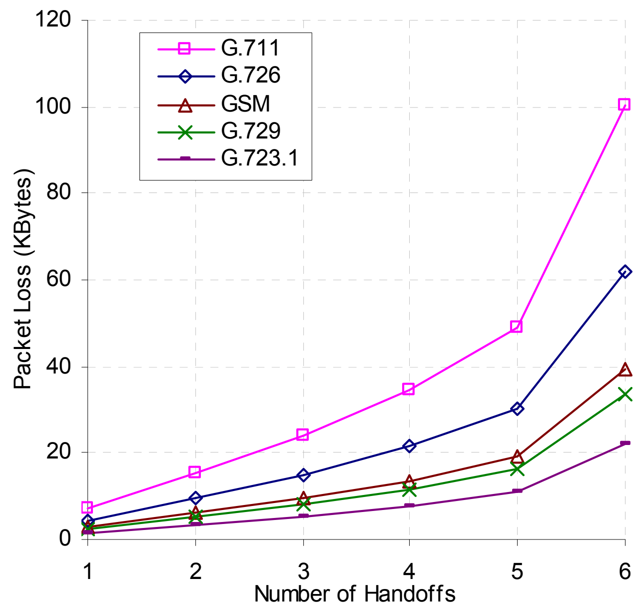


Fig. 5.13. Transient Packet Loss for a WiMAX-to-UMTS Handoff.

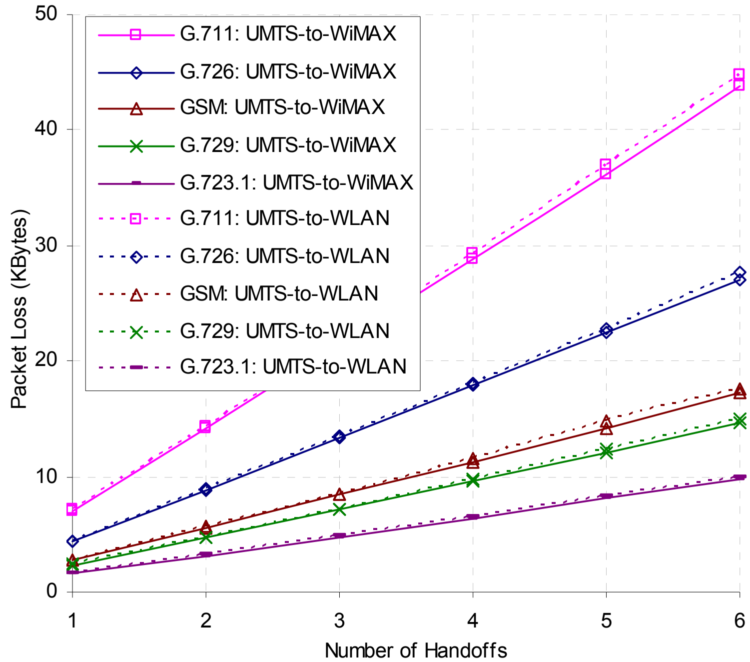


Fig. 5.14. Transient Packet Loss Comparison for a UMTS-to-WiMAX Handoff and a UMTS-to-WLAN Handoff.

Jitter, the variation of inter-arrival delay, is another factor which affects delay, especially during a vertical handoff (when the link capacity changes, etc.). The effect on jitter across different codec types is illustrated in Fig. 5.15. Fig. 5.15 also shows close resemblance to the Jitter results given in Fig. 3.11 (b). The main factors affecting the jitter during a vertical handoff for the considered VoIP data flow are the voice packet payload length and the downlink data flow rate. Since the data flow rate depends on the serving wireless interface (i.e., WiMAX or UMTS), the codecs that uses the lowest voice packet payload lengths may encounter the lowest jitter levels. Therefore, G.723.1, G.729, and GSM codecs with respective packet payload sizes of 20 bytes, 10 bytes, and 32.5 bytes result in relatively lower jitter.

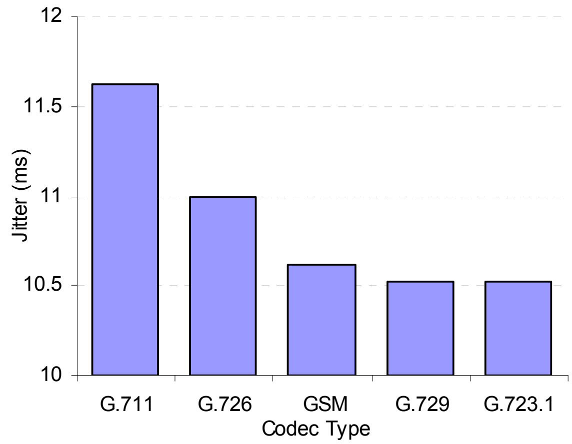


Fig. 5.15. Variation of Delay (Jitter) for a WiMAX-to-UMTS Handoff.

Furthermore, G.723.1 and G.729 also have the lowest jitter values. On the other hand, G.711 and G.726 result in relatively high transient packet loss and packet delay variation during vertical handoffs. It is further evident from these simulation results that the voice capacity over a heterogeneous network is a function of system parameters, transmission rate, voice packet payload length (depending on the codec used), and the sampling period.

5.8 Summary and Conclusions

This Chapter further developed the interworking framework to interwork between 3G cellular, WLAN, and WiMAX networks. Similar to the previous Chapter, terminal/IP mobility was managed at the Network Layer by MIP and session mobility was handled by the SIP at the Application Layer. Since the default mobility management protocol for mobile WiMAX networks is MIP, major modifications were not required for the initial mobility management framework. Therefore, it can now be concluded that the improved model is capable of interworking with a variety of heterogeneous mobile networks and thus proposes a suitable platform for a NGMN or a 4G network.

Chapter 6

Summary, Conclusions, and Future Directions

6.1 Thesis Summary and Conclusions

The objective of this thesis was to propose a novel mobility-aware architecture for interworking heterogeneous mobile data networks. More precisely, the proposed framework conveniently enables any 3G cellular technology (such as UMTS or CDMA2000 systems), WLANs, and WMANs (i.e., WiMAX systems) to interwork under a common mobility management platform. In particular, within the centralized common mobility management platform, both terminal mobility, and session mobility was managed in a real-time environment for a roaming user.

The approach used for achieving the above objectives can be summarized into three main stages as follows. Initially an in-depth review of the current literature published in the area was carried out. This was essential for identifying and formulating the problem, defining the motivations, trends, and issues.

The first stage of the research was to design and develop the initial architecture for interworking the UMTS and the WLAN system (with limited mobility). This framework used the IMS as a universal coupling mediator for real-time session negotiation and management. Since the initial goal was merely to interwork between UMTS and WLAN systems, and the design was somewhat close to a loose coupling model, the 3GPP's IMS was the obvious candidate to act as the coupling media. Furthermore, since it has already been used for signaling within the IMS, SIP was chosen as the protocol for mobility and session management. Thus the mobility management was initially Pure-SIP based and fully managed at the Application Layer.

The second stage of the research further extended the existing framework to be capable of interworking between different 3G cellular technologies (i.e., UMTS and CDMA2000) and the WLAN. However, this required major architectural changes to the initial design. As a result, a coupling framework similar to the 3GPP2's IMS was chosen. It also required re-designing of the entire mobility management framework. Instead of having a Pure-SIP based mobility management framework, a MIP and SIP combined mobility management framework was designed. Within this new framework, terminal/IP mobility was managed at the Network Layer by MIP and session mobility was handled by SIP at the Application Layer.

The third and the last stage of this research further extended the model to interwork between 3G cellular, WLAN, and BWA (WiMAX) networks. Similar to the second stage, terminal/IP mobility was managed at the network layer by MIP and session mobility was handled by SIP at the application layer. Since the default mobility management protocol for mobile WiMAX networks is MIP, major modifications were not required for the initial mobility management framework. Therefore, it can now be said that the improved model is capable of interworking with a variety of heterogeneous mobile networks and thus proposes a suitable platform for a NGMN or a 4G network.

An OPNET based simulation platform was used for modeling the proposed interworking architectures. Vertical handoff for a given VoIP session from UMTS to WLAN (and vice versa) using the proposed Pure-SIP and MIP-SIP based mobility management methods were simulated and performance measurements were taken. Further, the same simulations were carried out for UMTS to WiMAX handoffs under similar conditions (only for the MIP-SIP based vertical handoff method). All simulated scenarios proved to be capable of performing successful VoIP session handoffs between dissimilar networks whilst maintaining acceptable QoS levels. The summarized results obtained from the OPNET based simulations are tabulated in Table VI.I.

Further, an analytical model was derived for evaluating the proposed scheme for analyzing the QoS metrics and measures involved in call or session mobility management under the second stage. More precisely, QoS metrics that were analyzed are vertical handoff delay, transient packet loss, jitter, and signaling overhead/cost. The primary assumption made for this analysis was that, data call/session arrivals follow a Poisson process. The substance of the derived results can be summarized as follows: The QoS analysis indicated that a MIP-SIP based mobility management framework performs better than its predecessor, the Pure-SIP based mobility management method. Also, the analysis results indicated that the QoS performance for the investigated parameters were within the acceptable levels for real-time

TABLE VI.I

SUMMARY OF OPNET BASED SIMULATION RESULTS.

	Pure-SIP	MIP-SIP
Session establishment delay: UMTS	197 ms	240 ms
Session establishment delay: WLAN	182 ms	200 ms
Session establishment delay: WiMAX	n/a	210 ms
Vertical handoff delay: WLAN-to-UMTS	245 ms	220 ms
Vertical handoff delay: UMTS-to-WLAN	215 ms	196 ms
Vertical handoff delay: UMTS-to-WiMAX	n/a	195 ms
Vertical handoff delay: WiMAX-to-UMTS	n/a	220 ms
Transient packet loss: WLAN-to-UMTS	2.92 KBytes	2.85 KBytes
Transient packet loss: UMTS-to-WLAN	2.83 KBytes	2.79 KBytes
Transient packet loss: UMTS-to-WiMAX	2.81 KBytes	2.78 KBytes
Transient packet loss: WiMAX-to-UMTS	2.93 KBytes	2.85 KBytes
Jitter: UMTS-to-WiMAX	n/a	10.6 ms
Jitter: UMTS-to-WLAN	10.24 ms	n/a

TABLE VI.II

SUMMARY OF ANALYTICALLY MODELED RESULTS.

	Pure-SIP	MIP-SIP
Session establishment delay: UMTS	170 ms	192 ms
Session establishment delay: WLAN	155 ms	176 ms
Session establishment delay: WiMAX	152 ms	174 ms
Vertical handoff delay: WLAN-to-UMTS	210 ms	192 ms
Vertical handoff delay: UMTS-to-WLAN	186 ms	176 ms
Vertical handoff delay: UMTS-to-WiMAX	183 ms	174 ms
Vertical handoff delay: WiMAX-to-UMTS	210 ms	192 ms
Jitter: UMTS-to-WiMAX	n/a	17.71 ms
Jitter: UMTS-to-WLAN	9.21 ms	9.38 ms

VoIP conversations. The summarized results obtained from the analytical modeling are tabulated in Table VI.II.

The reason for both the analysis and OPNET simulations to concentrate on VoIP was because voice traffic is the primary concern on any telecommunications provider. However, similar analysis may be expanded for other applications based on FTP for file transferring, HTTP for web traffic, and also other multimedia application traffic for video and audio streaming. Hence the designed analytical model and the OPNET based simulation platform could successfully be used by researches and network providers as a tool or test-bed for studying the vertical handoff performance for the above mentioned applications regardless of the interworked end networks.

6.2 Future Directions

Future works in regards to improving this architecture can be seen as two folds. Firstly, to design and develop an authentication and authorization framework to the IEEE 802.21 MIH protocol for ensuring data integrity, replay protection, confidentiality, and data origin authentication. Further to reduce the latency during the authentication and key establishment prior to a vertical handover between heterogeneous access networks, which support the IEEE 802.21 protocol.

The main motivations for this work is due to the fact that in the current heterogeneous networking environments, security mechanisms including authentication and authorization are individually specified and handled by each access technology. Therefore, it is highly important for a centralized authentication and authorization framework to be defined for an inter-network roaming user. To the best of our knowledge, architectures and protocols for such a security framework has not been addressed in the ongoing development of the IEEE 802.21 MIH protocol.

The significance of this work is to improve the experience for a roaming user in a heterogeneous network by providing a unified security platform through authentication and authorization mechanisms. Additionally, having a unified security mechanism will also contribute towards minimizing the vertical handoff delay between heterogeneous access networks. This is since security-related message exchanges during vertical handover can significantly increase the handoff delay. As a result, in most cases, seamless service continuity cannot be guarantied. Additionally, this will be able to secure the existing IEEE 802.21 MIH protocol and enable authorization of MIH services. With such security in place, MIH protocol entities and MIH services will be less vulnerable to potential security attacks and threats.

Secondly the issue of simultaneous network access will be explored. With the emergence of Mobile WiMAX networks, which is sometimes considered as a 3G network, there are possibilities for a 3G cellular service and a Mobile WiMAX service to co-exist. In such situations, a roaming user may desire simultaneous access to both networks. Therefore, the proposed architecture will be further improved to facilitate this multiple network accessing feature.

Appendix A

Delay Analysis for an M/M/1 Queuing Model

A key feature of communication networks is the sharing of resources such as transmission bandwidth, storage, and processing capacity. Since the need for such resources is unscheduled and ad-hoc in nature, situations may arise when such resources are unavailable at the time of need. This is a common situation, which leads to delay or loss of service in communication networks. This appendix explains some fundamental models used for analyzing delay and performance.

A.1 Delay Analysis and Little's Law

In Queuing Theory, Little's Law relates the average time spent in the system $E[T]$ to the arrival rate λ and the average number of customers in the system $E[N]$ by the following formula:

$$E[N] = \lambda E[T] \tag{A1}$$

However, the above expression does not specify what constitutes a "system," which helps to apply the Little's Law for various scenarios. Therefore, the "system" could be an individual transmission line, a multiplexer, a switch, or even a complete network. In the following scenario, the Little's Law is successfully applied for finding the average delay experienced by a packet traversing in a packet switching network [157].

Figure A.1 shows an example of a packet switching network with inter connected packet switches. It is assumed that when a packet arrives at a packet switch, the packet is routed instantaneously and placed in a multiplexer to await transmission on an outgoing line. Thus each packet switch can be viewed as consisting of a set of multiplexers. Now the Little's Formula can be applied to the entire network. Let N_{net} be the total number of packets in the

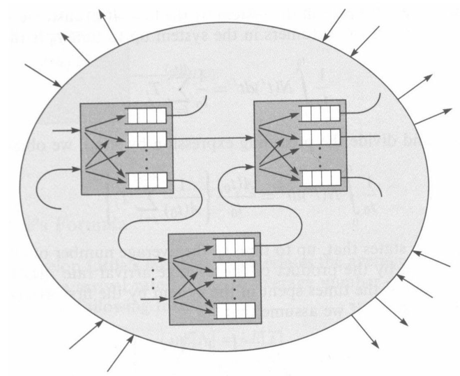


Fig. A.1. Packet-Switching Network Delay.

network, let T_{net} be the time spent by the packet in a network, and let λ_{net} be the total packet arrival rate to the network, Little's formula then states that

$$E[N_{net}] = \lambda_{net} E[T_{net}] \tag{A2}$$

The above formula implies that the average delay experienced by packets in traversing the network (i.e., for the entire system) is:

$$E[T_{net}] = E[N_{net}]/\lambda_{net} \tag{A3}$$

Now, equation (A2) can be refined by applying the Little's Law to each individual multiplexer. For the m^{th} multiplexer the Little's formula gives:

$$E[N_m] = \lambda_m E[T_m] \tag{A4}$$

where λ_m is the packet arrival rate at the multiplexer and $E[T_m]$ is the average time spent by a packet in the multiplexer. The total number of packets in the network N_{net} is equal to the sum of the packets in all the multiplexers:

$$E[N_{net}] = \sum_m E[N_m] = \sum_m \lambda_m E[T_m] \tag{A5}$$

By combining (A3), (A4), and (A5), and expression for the total delay (i.e., end-to-end daley) experienced by a packet in traversing the entire network can be obtained as:

$$E[T_{net}] = E[N_{net}]/\lambda_{net} = \frac{1}{\lambda_{net}} \sum_{m} \lambda_m E[T_m] \tag{A6}$$

Hence the average network delay depends on the overall arrival rates in the network, the arrival rate to individual multiplexers, and the delay in each multiplexer. Equation (A6) is very important since it incorporate the effects of routing (i.e., packet arrivals) and the capacity of the transmission line on the network. Therefore, it can be directly used for evaluating the delay performance of packet-switching networks.

Appendix B

Delay Analysis for Self-Similar Traffic

This section presents an analytical model for analyzing vertical session handoffs in a heterogeneous mobile networking environment by considering a long-tailed distribution. The analysis studies a G/M/1 queuing model where G is a Pareto distribution (rather than a Poisson arrival distribution) to model data sessions that are subjected to vertical handoff in a packet switched network. The vertical handoff mechanism and heterogeneous networking platform (UMTS-WiMAX) used for the analysis is based on a MIP-SIP combined mechanism as introduced under Chapter 5. The reminder of this section is organized as follows. The next section briefly introduces the importance of the concept of self-similarity in regards to Internet traffic analysis. Followed by is the section on Pareto/M/1 queuing analysis. Subsequently the vertical handoff analysis and numerical results follow prior to the concluding remarks. The analytical model and results presented under this section has contributed to [203] and [209].

B.1 Introduction

Internet traffic has been described as having one or more of the following related characteristics: self-similar (or fractal) traffic traces, long-range dependence, burstiness on multiple scales, and long or heavy-tailed packet inter-arrival times or service requirements [204].

Self-similarity implies that the traffic looks the same over any time scale. Furthermore, as first shown in [144], Poisson traffic does not have the same characteristic. Long-range dependence is defined with respect to the autocorrelation function of a stationary discrete-time stochastic process, $R(k)$. It measures the level of correlation of the process with itself and measured k periods away. The process is said to be long-range dependent if $\Sigma k\ R(k) = \infty$, thus implying that there is at best a slow and non-exponential decline in the autocorrelation function with increasing lags k.

Furthermore, it may be argued that a self-similar process is also long-range dependent. The Hurst parameter is often used to describe the degree of self-similarity in long-range processes [204]. The concept of burstiness means that packets arrive in several short inter-arrival times followed by a much longer time. Examples of long-tailed distributions are the Pareto, the log-normal, the folded Cauchy, and the DFR form of the Weibull. In this analysis, a method for studying Pareto queues is presented.

B.2 Pareto/M/1 Queuing Model

The standard form for the two-parameter Pareto distribution function defined over the nonnegative real numbers can be written as:

$$F(x) = 1 - 1/(\alpha + x)^{\beta} \; \forall(\alpha, \beta \rangle 0) \tag{B.1}$$

As a critical motivation for the subsequent procedure, such a distribution function can be directly derived as a gamma *(α,β)* mixture of ordinary exponential densities. With no loss in generality, the one-parameter version of the Pareto can be given as [205].

$$F(x) = 1 - 1/(1 + x)^{\beta} \tag{B.2}$$

The corresponding density function is:

$$f(x) = \beta/(1 + x)^{\beta+1} \tag{B.3}$$

and it is shown that the Pareto is indeed a long-tailed distribution, where β measures the initial rate of decline of the density function curve [205].

In the following scenario, a Pareto arrival distribution into the queuing system is considered. From the standard analysis of a G/M1 queue, the steady-state probability for the number of customers Q in system just before an arrival is given for all nonnegative n as [206]:

$$\Pr\{Q = n\} = q_n = (1 - r)r^n \tag{B.4}$$

For Pareto/M/1, the usual approach for obtaining the stationary delay time distributions and system size probabilities requires solving a root finding problem involving the Laplace-Stieltjes Transform (LST), $A^*(s)$, of the inter-arrival time distribution function [206], [156]. Thus r is the root of the fundamental branching process equation obtained by solving for z is:

$$z = A^*[\mu(1 - z)] \tag{B.5}$$

where $1/\mu$ is the expected service time [206]. The system utilization, ρ, which is λ/μ, where λ is the customer arrival rate, and for the problem to have a non-trivial solution, one must have $\rho < 1$. The unique root of the fundamental equation of the branching process, say r, then becomes the parameter of a geometric distribution for steady-state system sizes at the embedded arrival points [205]. These geometric probabilities are then combined with convolutions of the exponential service distribution to derive the stationary line-delay distribution.

Unfortunately, for the case of Pareto arrivals, a closed form for $A^*(s)$ does not exist. This section uses a method proposed by Harris and Marchal for finding Coxian distribution fits for arbitrary distribution functions using Laplace transform approximations [207]. It turns out that their technique, which is called as the Transform Matching Method (TMM)) works especially well for distributions defined over the full real line but without all moments. Thus use TMM for A^*(s) and then use Newton's method to solve for the root r. Once the root is found, the queue and system waiting-time distribution functions can easily be derived for $t \geq 0$ as [206]:

$$W_q(t) = 1 - re^{-\mu(1-r)t} \tag{B.6}$$

$$W(t) = 1 - e^{-\mu(1-r)t} \tag{B.7}$$

A close observation of the above queue and system waiting-time distribution functions indicates that they have the same functional form as the M/M/1 queue except with r replacing ρ. Thus the expected queue waiting time, W_q, and system waiting time, W, can be expressed as follows [206] [205]:

$$W_q = \frac{r}{\mu(1-r)} \tag{B.8}$$

$$W = \frac{1}{\mu(1-r)} \tag{B.9}$$

B.3 Vertical Handoff Analysis

An analytical model is derived for the scheme proposed under Chapter 5 for evaluating vertical session handoff performance for Internet data traffic with Pareto based arrivals.

B.3.1 Handoff Delay

A standard vertical handoff delay during mid-session mobility consists of the following sub-procedures (or delays); D_1 = link layer handoff delay, D_2 = movement detection delay, D_3 = address allocation delay, D_4 = session re-configuration delay, and D_5 = packet re-transmission delay [154]. The vertical handoff delay at the network layer (and above) are calculated independent of the link layer delay D_1 and mainly consist of D_3 and D_4. According to our proposed architecture for IMS based vertical handoff, there is no DHCP related address allocation; hence it can be argued that is D_4 the main contributor for network layer based vertical handoff delay, D. The session re-configuration delay, D_4 mainly consists of the previously mentioned IMS based session negotiation and handoff and HSS related message exchange delays.

In order to derive an expression for D, we must first derive an expression for analyzing the end-to-end transmission delay. Hence, let us assume that the end-to-end transmission delay for a packet size S sent from network A to network B over a number of hops via a wireless and wired links to be expressed as follows:

$$D(S, H_{a-b}) = D_{wl} + D_w + L_{wl} + L_w \tag{B.10}$$

where, D_{wl} is the total delay at the wireless interface (say, BS), D_w is the total delay at the wired link, L_{wl} is the latency of the wireless link, and L_w is the latency of the wired link. In order to derive D_{wl} and D_w a Pareto*/M/1* queuing model has been applied to the packet flow of the data session at the wireless BS and other networking elements of the IMS on the path of signaling and data routing of Fig. 5.2.

It is important to note that to apply the results of Pareto*/M/1* analysis, it is assumed that the service times that a packet experiences at different nodes are independent of each other. However, this assumption is untrue, since the service time is proportional to the packet length, and a packet has the same length as it traverses the network. Nevertheless, it has been found that this independence assumption can be used in large networks [156]. Using the results form the Pareto/M/1 model, expressions for D_{wl} and D_w can be expressed as follows:

$$D_{wl} = \frac{1}{\mu_{wl}(1 - r_{wl})} \tag{B.11}$$

where, μ_{wl} is the service rate and r_{wl} is the root of the fundamental branching process equation obtained by solving for z at the wireless interface. For clarity and convenience sake, the units for μ_{wl} are changed from packets/sec to bits/sec. If the probability density function of for packet size, x, in bits be $\mu e^{-\mu x}$ with a mean packet length of $1/\mu$ bits/packet, and the capacity of communication channel i be C_i bits/sec. The product μC_i is then the service rate in packets/sec. Therefore, for channel i, we have

$$D_{wl} = \frac{1}{\mu C_i (1 - r_{wl})} \tag{B.12}$$

where, D_{wl} includes both queuing and transmission delays. Also note that the mean packet size does not depend on the channel as the capacity and the input rates do. However, when D_w is considered, it can be expressed as a collection of delays of multiple Pareto*/M/1* queues. It is also assumed that if the output of several Pareto*/M/1* servers feed into the input queue of another server, the resulting input process is also a Pareto process, with mean equal to the sum of the means of the feeding process. This assumption has been derived from a similar assumption made for a G/M/1 queue in [156]. Thus the total wired network delay experienced by a packet can be expressed as:

$$D_w = \frac{1}{\lambda_w}\sum_j \lambda_j \left(\frac{1}{\mu C_j (1 - r_w)}\right) \tag{B.13}$$

where, λ_w is the total packet arrival rate to the network, λ_j is the packet arrival rate at j^{th} node, and μC_j is the service rate in packets/sec at the j^{th} node. Thus by combining equations (B.10), (B.12) and (B.13) we get:

$$D(S, H_{a-b}) = \frac{1}{\mu C_i (1 - r_{wl})} + \left\{\frac{1}{\lambda_w}\sum_j \lambda_j \left(\frac{1}{\mu C_j (1 - r_w)}\right)\right\} + L_{wl} + L_w \tag{B.14}$$

Now, an expression for the vertical handoff delay D can be expressed by applying (B.14) to the entire IMS signaling flow involved in the vertical handoff mechanism as illustrated in Fig. 5.3. Thus the final expression for D is a combination of the following end-to-end delay components as indicated in equation (B.15).

$$\begin{aligned} D_{IMS} = {} & D(S_{MIPReq}, H_{UMTS\text{-}MIP\text{-}HA}) + D(S_{MIPRep}, H_{UMTS\text{-}MIP\text{-}HA}) \\ & + D(S_{MIP\text{-}BU}, H_{UMTS\text{-}CN}) + D(S_{ReINVITE}, H_{WiMAX\text{-}CN}) \\ & + D(S_{183\text{-}SP}, H_{WiMAX\text{-}CN}) + D(S_{PRACK}, H_{WiMAX\text{-}CN}) \\ & + D(S_{OK}, H_{WiMAX\text{-}CN}) + D(S_{UPDATE}, H_{WiMAX\text{-}CN}) \\ & + D(S_{OK}, H_{WiMAX\text{-}CN}) + D(S_{ACK}, H_{WiMAX\text{-}CN}) \end{aligned}$$

$$+D(S_{BYE},H_{UMTS\text{-}CN}) + D(S_{OK},H_{CN\text{-}UMTS}) + \Delta \qquad \text{(B.15)}$$

where, Δ is the additional IMS (application layer) related latency due to HSS lookup process. The important point to note here is that the derivation of equation (B.15) has not taken into account the errors that may cause various messages to be damaged or lost. This is since for successful session establishment, the entire message flow must take place and if any message is damaged or lost the vertical handoff process will fail. Hence it has been assumed that the channel is error free during the process of the vertical handoff taking place. It is also worth reminding that make-before-break handoff is applied in the proposed handoff scenarios, which helps compensate for large handoff delays. For the purpose of a complete analysis of vertical handoff delay, the standard straight forward case of break-before-make handoff scenario is used.

B.3.2 Packet Loss

The total packet loss (*Pkt_loss*) during a vertical session handoff can be defined as the sum of all lost packets during the vertical handoff while the MN is receiving the downlink data packets. It is assumed that the packet loss begins when the Layer 2 handoff is detected and all in-flight packets are lost during the vertical handoff time. Thus, it can be expressed as:

$$Pkt\_loss = \left[\frac{1}{2T_{ad}} + D\right] \times \lambda_{wl} \times N_m \qquad \text{(B.16)}$$

where, T_{ad} is the time interval between P-CSCF discovery times, λ_{wl} is the downlink packet arrival rate at the wireless interface, and N_m is the average number of vertical handoffs during a single session [154]. N_m plays a major role in the calculation of packet loss since the packet loss due to vertical handoff is directly proportionate to the number of handoffs it is subjected within a given session.

B.4 Numerical Results

This section presents numerical results relating to the behavior of vertical handoff delay and transient packet loss against system utilization for different shape parameter values (i.e., β = 1.5, 2.5, 3.5). In order to better understand the behavior of the Pareto/M/1 queue, its performance has been compared against the known closed form values for an M/M/1 queue. The results used for the performance comparison for an M/M/1 queue is obtained from the results given under Chapter 5 on vertical session handoff analysis [43]. Table IV.I provides the typical MIPv4 and SIP message sizes and Fig. 5.4 provides the relative distances in hops used in the numerical evaluation.

Figure B.1 illustrates the graphs for WiMAX-to-UMTS vertical handoff delay against the system utilization for a Pareto/M/1 queuing analysis and a M/M/1 classical queuing analysis. M/M/1 queuing based analysis and Pareto/M/1 queuing based analysis (for β = 2.5 and 3.5) show approximately close behavioral patterns for relatively lower system utilizations (i.e., under 50%). However, beyond this point, the vertical handoff delay increases according to the nature of packet arrival patterns (i.e., Poisson or Pareto). For example, since Poisson arrivals are relatively smoother and not as bursty as Pareto arrivals, the results clearly illustrates an exponentially increasing delay. On the other hand, as the system utilization grows beyond 80%, the Pareto/M/1 queuing model tends to demonstrate its characteristic heavy tailed behavior.

On the other hand, Fig. B.2 illustrates relatively lower handoff delays for the graphs corresponding to UMTS-to-WiMAX vertical handoff delay against the system utilization for a Pareto/M/1 queuing analysis and an M/M/1 classical queuing analysis. This indicates that when a session is transferred to a network with relatively lower link bandwidth, a relatively higher vertical handoff delay may be expected. Also note that for the Pareto/M/1 queue, the long tailed arrivals actually help clear congestion. This is because every once in a while, there are unevenly distributed long inter-arrival times. However, for the case of β = 1.5, the mean inter-arrival times become finite, which eventually leads to congestion in the system. However as β gets larger, the such extreme behavior is not experienced [208].

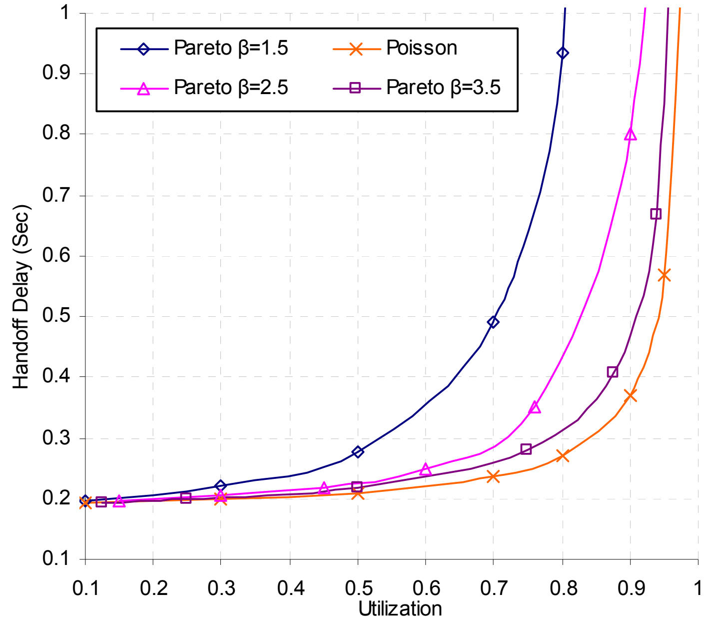


Fig. B.1. WiMAX-to-UMTS Handoff Delay vs. System Utilization.

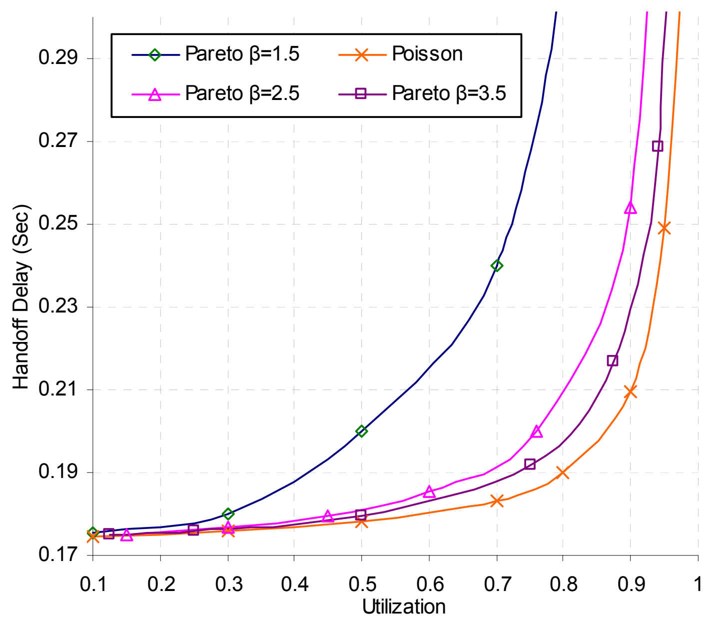


Fig. B.2. UMTS-to-WiMAX Handoff Delay vs. System Utilization.

Figure B.3 illustrates the normalized transient packet loss during vertical handoffs as the system utilization increase (in the case of a break-before-make handoff scenario). The voice codec considered for the downlink packet transmission in this case is a GSM codec. According to equation (B.16), the packet loss during a vertical handoff is directly proportional to the vertical handoff delay. Therefore, relatively high vertical handoff delays indicated by the WiMAX-to-UMTS graphs in Fig. B.1 directly relate to the two high packet loss curves in Fig. B.3. Similarly, the packet loss is relatively low in Fig. B.3 for a UMTS-to-WiMAX handoff, which is in line with the two relatively low handoff delay graphs shown in Fig. B.2. Further, the exponential and heavy-tailed behaviors can also be observed in Fig. B.3 for Poisson and Pareto based models respectively.

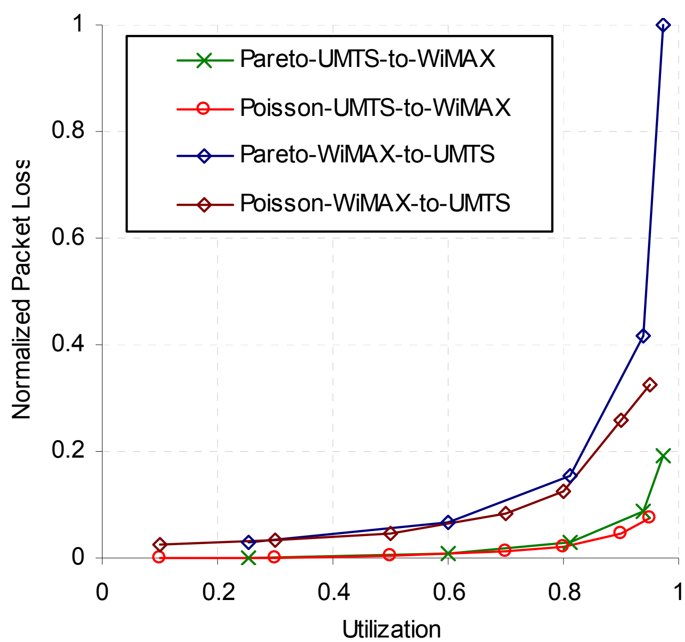


Fig.B.3. Transient Packet Loss vs. System Utilization.

References

[1] IEEE, "IEEE Std 802.11-1997 Information Technology – Telecommunications and Information Exchange between Systems-Local and Metropolitan Area Networks – specific Requirements – Part II: Wireless LAN Medium Access Control (MAC) and Physical Layer (PHY) Specifications," *IEEE Std 802.11-1997*, 1997.

[2] IEEE, "Supplement to IEEE Standard for Information Technology – Telecommunications and Information Exchange Between Systems – Local and Metropolitan Area Networks-Specific Requirements – Part II: Wireless LAN Medium Access Control (MAC) and Physical Layer (PHY) Specifications: Higher-speed Physical Layer extension in the 2.4 GHz band," *IEEE Std. 802.11b-1999*, 2000.

[3] IEEE, "Supplement to IEEE Standard for Information Technology telecommunications and information exchange between systems – local and metropolitan area networks – specific requirements. Part II: wireless LAN Medium Access Control (MAC) and Physical Layer (PHY) specifications: high – speed physical layer in the 5 GHz band," *IEEE Std. 802.11a-1999*, 1999.

[4] IEEE, "IEEE standard for Information Technology – Telecommunications and Information exchange between systems – local and metropolitan area networks – specific requirements Part II: wireless LAN medium access control (MAC) and physical layer (PHY) specifications," *IEEE Std. 802.11g-2003*, 2003.

[5] IEEE, "Wireless LAN Medium Access Control (MAC) and Physical Layer (PHY) specifications: Amendment 4: Enhancements for Higher Throughput.," *P802.11n/D3.00*, 2008.

[6] T. Paul and T. Ogunfunmi, "Wireless LAN Comes of Age: Understanding the IEEE 802.11n Amendment," *IEEE Circuits and Systems Magazine*, vol. 8, pp. 28-54, 2008.

[7] IEEE, "Part 11: Wireless Medium Access Control (MAC) and Physical Layer (PHY) specifications: Amendment 5: Spectrum and transmit power management extensions in the 5 GHz band in Europe," *IEEE Std 802.11h-2003*, 2003.

[8] D. Qiao and S. Choi, "New 802.11h mechanisms can reduce power consumption," *IT Professional*, vol. 8, no. 2, pp. 43-48, March-April 2006.

[9] C. Eklund, R. B. Marks, K. L. Stanwood, and S. Wang, "IEEE Standard 802.16: A Technical Overview of the Wireless MAN Air Interface for Broadband Wireless Access," *IEEE Communications*, vol. 40, no. 6, pp. 98-107, June 2002.

[10] IEEE, "IEEE Standard for Local and metropolitan area networks — Part 16: Air Interface for Fixed Broadband Wireless Access Systems," *IEEE Standard 802.16-2001*, 2001.

[11] IEEE, "IEEE Standard for Local and metropolitan area networks — Part 16: Air Interface for Fixed Broadband Wireless Access Systems — Amendment 2: Medium Access Control Modifications and Additional Physical Layer Specifications for 2-11 GHz," *IEEE Standard 802.16a-2003*, 2003.

[12] IEEE, "IEEE Standard for Local and metropolitan area networks — Part 16: Air Interface for Fixed Broadband Wireless Access Systems — Amendment 1: Detailed System Profiles for 10–66 GHz," *IEEE Standard 802.16c-2002*, 2002.

[13] IEEE, "Draft Amendment to IEEE Standard for Local and Metropolitan Area Networks - Part 16: Air Interface for Fixed Broadband Wireless Access Systems- Amendment 3: Detailed System Profiles for 2-11 GHz," *IEEE Project P802.16d*, 2003.

[14] IEEE, "Air Interface for Fixed Broadband Wireless Access Systems," *IEEE STD 802.16 - 2004*, October 2004.

[15] IEEE, "Air Interface for Fixed and Mobile Broadband Wireless Access Systems," *IEEE P802.16e/D12*, February 2005.

[16] 3GPP, "Universal Mobile Telecommunications System (UMTS); Technical Specifications and Technical Reports for a UTRAN-based 3GPP system (3GPP TS 21.101 version 3.17.0 Release 1999)," *ETSI TS 121 101 V3.17.0*, 2008.

[17] 3GPP, "3rd Generation Partnership Project; Technical Specification Group Services and System Aspects; General Packet Radio Service (GPRS); Service description, Stage 1 (3G TS 22.060 version 3.1.0)," *3G TS 22.060 V3.1.0*, 1999.

[18] 3GPP2, "HRPD/1XRTT and 3GPP E-UTRAN (LTE) Interworking and Inter-Technology Handoff - Stage 1 Requirements," *3GPP2 S.R0129-0*, 2008.

[19] 3GPP2, "cdma2000 High Rate Packet Data Air Interface Specification," *3GPP2 C.S0024-B*, 2006.

[20] 3GPP2, "Introduction to cdma2000 Standards for Spread Spectrum Systems," *3GPP2 C.S0001-D*, 2005.

[21] ITU-T, "Detailed specifications of the radio interfaces of International Mobile Telecommunications-2000 (IMT-2000)," *Recommendation M.1457-7*, 2008.

[22] 3GPP, "Universal Mobile Telecommunications System (UMTS); UTRA High Speed Downlink Packet Access (HSDPA); Overall description; Stage 2," *ETSI TS 125 308 V5.7.0*, 2004.

[23] A. Jamalipour, *The Wireless Mobile Internet: architectures, protocols, and services,* England: John Wiley & Sons, 2003.

[24] J.-C. Chen and T. Zhang, *IP-Based Next-Generation Wireless Networks*, John Wiley & Sons, 2004.

[25] A. K. Salkintzis, C. Fors, and R. Pazhyannur, "WLAN-GPRS integration for next-generation mobile data networks," *IEEE Wireless Communications,* vol. 9, no. 5, pp. 112-124, June 2002.

[26] K. D. Wong, M. Barton, B. Kim, V. K. Varma, S. Ramesh, G. Hayward, and J. A. Friedhoffer, "UMTS signaling over 802.11 wireless LAN," *in Proceedings of at the IEEE Vehicular Technology Conference-Fall*, Orlando, Florida, USA, October 2003.

[27] V. K. Varma, S. Ramesh, K. D. Wong, M. Barton, G. Hayward, and J. A. Friedhoffer, "Mobility management in integrated UMTS/WLAN networks," in *Proceedings of the IEEE International Conference on Communications*, Anchorage, Alaska, May 2003.

[28] A. K. Salkintzis, "Interworking techniques and architectures for WLAN/3G integration toward 4G mobile data networks," *IEEE Wireless Communications*, vol. 11, no. 3 pp. 50-61, 2004.

[29] A. K. Salkintzis, "WLAN/3G interworking architectures for next generation hybrid data networks," in *Proceedings of the IEEE International Conference on Communications*, Paris, France, June 2004.

[30] K. Pahlavan, P. Krishnamurthy, A. Hatami, M. Ylianttila, J. P. Makela, R. Pichna, and J. Vallstron, "Handoff in hybrid mobile data networks," *IEEE Personal Communications*, vol. 7, no. 2, pp. 34 - 47, 2000.

[31] 3GPP, "IP Multimedia Subsystem (IMS)," 3GPP TS 23.228 Version 6.10.0 Release 6, 2005.

[32] Yi-Bing Lin, Ai-Chun Pang, *Wireless and Mobile All-IP Networks*, John Wiley & Sons, 2005.

[33] K. S. Munasinghe, A. Jamalipour, and B. Vucetic, "Interworking between WLAN and 3G Cellular Networks: An IMS Based Architecture," in *Proceedings of the IEEE International Conference on Wireless Broadband and Ultra Wideband Communications*, Sydney, Australia, March 2006.

[34] K. S. Munasinghe and A. Jamalipour, "Interworking of WLAN-UMTS Networks: An IMS-based Platform for Session Mobility," *IEEE Communications*, vol. 46, no. 9, pp. 184-191, 2008.

[35] K. S. Munasinghe and A. Jamalipour, "A 3GPP-IMS based Approach for Converging Next Generation Mobile Data Networks," in *Proceedings of the IEEE International Conference on Communications*, Glasgow, Scotland, UK, June 2007.

[36] K. S. Munasinghe and A. Jamalipour, "Mobility Management QoS Measures Evaluation for Next Generation Mobile Networks," in *Proceedings of the Tyrrhenian Workshop on Digital Communications*, Naples, Italy, September 2007.

[37] K. S. Munasinghe and A. Jamalipour, "Mobility Management QoS Measures Evaluation for Next Generation Mobile Networks," in *Wireless Communications: CNIT Thyrrenian Symposium Series: Signals and Communications Technology*, P. Silvano, Ed. USA: Springer, pp. 265-280, 2008.

[38] K. S. Munasinghe and A. Jamalipour, "An Analytical Evaluation of Mobility Management in Integrated WLAN-UMTS Networks," *Elsevier Computers and Electrical Engineering*, 2009 (in press).

[39] K. S. Munasinghe and A. Jamalipour, "Analysis of Signaling Cost for a Roaming User in a Heterogeneous Mobile Data Network," in *Proceedings of IEEE Global Communications Conference*, New Orleans, Louisiana, USA, November 2008.

[40] K. S. Munasinghe and A. Jamalipour, "A Unified Mobility and Session Management Platform for Next Generation Mobile Networks," in *Proceedings of IEEE Global Communications Conference*, Washington D.C., USA, November 2007.

[41] K. S. Munasinghe and A. Jamalipour, "Analytical Modeling of IMS based Interworking in Heterogeneous Mobile Data Networks," in *Proceedings of International Conference*

on Signal Processing and Communication Systems, Gold Coast, Australia, December 2007.

[42] K. S. Munasinghe, M. R. Kibria, and A. Jamalipour, "Designing VoIP Session Management over Interworked WLAN-3G Cellular Networks," *IEEE Wireless Communications*, vol. 15, no. 4, pp. 86-94, 2008.

[43] K. S. Munasinghe and A. Jamalipour, "Interworked WiMAX-3G Cellular Networks: An Architecture for Mobility Management and Performance Evaluation," *IEEE Transactions on Wireless Communications*, 2008 (in press).

[44] K. S. Munasinghe and A. Jamalipour, "An Architecture for Mobility Management in Interworked 3G Cellular and WiMAX Networks," in *Proceedings of the IEEE Wireless Telecommunications Symposium*, Pomona, California, USA, April 2008.

[45] S. Bhavsar, "WLAN-Cellular Integration for Mobile Data Networks," in *Proceedings of Communications Networks and Services Research Conference*, New Brunswick, Canada, May 2003.

[46] Y. Xiao, Y. Pan, K. K. Leung, and X. Du, "Architecture, Mobility Management, and Quality of Service for Integrated 3G and WLAN Networks," *Wiley Interscience: Journal of Wireless Communications and Mobile Computing*, vol. 5, no. 7, pp. 805-823, 2005.

[47] W. Song, W. Zhuang, and A. Saleh, "Interworking of 3G cellular networks and wireless LANs," *International Journal of Wireless and Mobile Computing*, vol. 2, no.4, pp. 237-247, 2007.

[48] J.-y. Song, S.-w. Lee, and D.-h. Cho, "Hybrid coupling scheme for UMTS and wireless LAN interworking," in *Proceedings of at the IEEE Vehicular Technology Conference-Fall*, Orlando, Florida, Oct. 2003.

[49] L. Chunming and Z. Chi, "An improved interworking architecture for UMTS-WLAN tight coupling," in *Proceedings of the IEEE Wireless Communications and Networking Conference*, Atlanta, Georgia, USSA, March 2005.

[50] M. Lee, G. Kim, S. Park, S. Jun, J. Nah, and O. Song, "Efficient 3G/WLAN Interworking Techniques for Seamless Roaming Services with Location-Aware Authentication," in *Lecture Notes in Computer Science Networking*, R. Boutaba, Ed. Berlin / Heidelberg: Springer, pp. 370-381, May 2005.

[51] A. K. Salkintzis, *Mobile Internet: Enabling Technologies and Services*: CRC Press, 2004.

[52] J. M. a. K. A.-B. Xin Gang Wang, "An adaptive QoS framework for integrated cellular and WLAN networks," in *Proceedings of the PG Networking Conference*, Liverpool, UK, June 2003.

[53] D. Findlay, H. Flygare, R. Hancock, T. Haslestad, E. Hepworth, D. Higgins, and S. McCann, "3G interworking with wireless LANs," in *Proceedings of the International Conference on 3G Mobile Communication Technologie*s, London, UK, May 2002.

[54] ETSI, "Requirements and Architectures for Interworking between HIPERLAN/2 and 3rd Generation Cellular systems," *ETSI TR 101 957 Version 1.1.1*, 2001.

[55] J. C. Chen and H. W. Lin, "A gateway approach to mobility integration of GPRS and wireless LANs," *IEEE Wireless Communications*, vol. 12, no. 2, pp. 86 - 95, 2005.

[56] Y. Min-hua, L. Yu, and Z. Hui-min, "The Mobile IP handoff between hybrid networks," in *Proceedings of the IEEE International Symposium on Personal, Indoor and Mobile Radio Communications*, Beijing, China, September 2003.

[57] C. Chang, "A Mobile-IP based Mobility System for Wireless Metropolitan Area Networks," in *Proceedings of International Conference Workshops on Parallel Processing*, Oslo, Norway, June 2005.

[58] T. Sanguankotchakorn and P. Jaiton, "Effect of Triangular Routing in Mixed IPv4/IPv6 Networks," in *Proceedings of International Conference on Networking*, Cancun, Mexico, April 2008.

[59] H. H. Choi and D. H. Cho, "Mobility management based on mobile IP in mixed IPv4/IPv6 networks," in Proceedings of the *IEEE Vehicular Technology Conference-Fall*, Orlando, Florida, USA, October 2003.

[60] A. C. Pang, J. C. Chen, Y. K. Chen, and P. Agrawal, "Mobility and session management: UMTS vs. cdma2000," *IEEE Wireless Communications*, vol. 11, no. 4, pp. 30-43, August 2004.

[61] S.-L. Tsao and C.-C. Lin, "VGSN: a gateway approach to interconnect UMTS/WLAN networks," in *Proceedings of the IEEE International Symposium on Personal, Indoor and Mobile Radio Communications*, Lisbon, Portugal, September 2002.

[62] P. M. Feder, N. Y. Lee, and S. Martin-Leon, "A seamless mobile VPN data solution for UMTS and WLAN users," in *Proceedings of the International Conference on 3G Mobile Communication Technologies*, London, UK, June 2003.

[63] 3GPP, "3GPP system to Wireles Local Area Network (WLAN) interworking," *3GPP TS 23.234 version 6.4.0 Release 6*, 2005.

[64] 3GPP, "3GPP system to Wireless Local Area Network (WLAN) Interworking; Functional and architectural definition," *3GPP TR 23.934 version 1.0.0 Release 6*, 2002.

[65] 3GPP, "Feasibility study on 3GPP system to Wireles Local Area Network (WLAN) interworking," *3GPP TR 22.934 version 6.2.0 Release 6*, 2003.

[66] F. A. Phiri and M. B. Murthy, "WLAN-GPRS Tight Coupling Based Interworking Architecture with Vertical Handoff Support," *Wireless Personal Communications*, vol. 40, no. 2, pp. 137-144, 2007.

[67] D. I. Axiotis, T. Al-Gizawi, K. Peppas, E. N. Protonotarios, F. I. Lazarakis, C. Papadias, and P. I. Philippopoulos, "Services in interworking 3G and WLAN environments," *IEEE Wireless Communications*, vol. 11, no. 5, pp. 14-20, 2004.

[68] H. Hu and Y. a. Liu, "Key Technologies of WLAN Accessing 3GPP PS Based Services," in *Proceedings the IEEE Circuits and Systems Symposium on Emerging Technologies: Frontiers of Mobile and Wireless Communication*, Shanghai, China, May 2004.

[69] 3GPP2, "3GPP2 – WLAN Interworking - Stage 1 Requirements," *3GPP2 S.R0087-0*, July 2004.

[70] Y.-G. Hong, J.-S. Park, and H.-J. Kim, "IPv6 based CDMA/WLAN heterogeneous network interworking," in *Proceedings of International Conference on Advanced Communication Technology*, Korea, February 2005.

[71] Z. Huang, Yang, P., and Deng, H., "QoS based handoff for CDMA2000-WLAN interworking," in *Proceedings of ACM International Workshop on Quality of Service and Security for Wireless and Mobile Networks*, Spain, October 2006.

[72] R. S. Gerardo Gómez, *End-to-end Quality of Service Over Cellular Networks: Data Services Performance and Optimization in 2G/3G*: John Wiley and Sons, 2005.

[73] H. C. W. Kim, "A seamless handover between cdma2000 and WLAN for 3G-WLAN interworking service continuity," in *Proceedings of Advanced Industrial Conference on Telecommunications*, Lisbon, Portugal, July 2005.

[74] I. Roh, M.-H. Rim, S. Shin, and H.-S. Jang, "Radio resource requirements in the IMT-2000 network," in *Proceedings of International Conference on Management of Engineering and Technology*, Portland, OR, USA, July 2001.

[75] J. Nah, J. Yoo, J. Yu, Y. Lee, and H. H. Lee, "Mobile signaling structures of IMT-2000 exchange," in *Proceedings of the IEEE International Conference on Personal Wireless Communication*, Jaipur, India, February 1999.

[76] S. Kim, H. Cho, H. Hahm, S. Lee, and M. S. Lee, "Interoperability between UMTS and cdma2000 networks," *IEEE Wireless Communications*, vol. 10, no. 1, pp. 22-28, 2003.

[77] C. P. Makaya, Samuel, "An Interworking Architecture for Heterogeneous IP Wireless Networks," in *Proceedings of International Conference on Wireless and Mobile Communications*, French Caribbean, March 2007.

[78] C. Makaya and S. Pierre, "Reliable Integrated Architecture for Heterogeneous Mobile and Wireless Networks," *JOURNAL OF NETWORKS*, vol. 2, no. 6, pp. 24-32, 2007.

[79] D. Johnson, C. Perkins, and J. Arkko, "Mobility Support in IPv6," *IETF RFC 3775*, June 2004.

[80] H. Soliman, C. Castelluccia, K. El-Malki, and L. Bellier, "Hierarchical Mobile IPv6 Mobility Management (HMIPv6)," *IETF RFC 4140*, August 2005.

[81] TIA/EIA, "Interworking/Interoperability between DCS1900 and IS-41 Based MAPs for 1800 MHz Personal Communications Systems — Phase I," *TIA/EIA/IS-129*, July 1996.

[82] 3GPP2, "Network Interworking between GSM MAP and ANSI-41 MAP Rev. B," 3GPP2 *N.S0028 V1.0.0*, April 2002.

[83] D. S. Woo, S. S. Park, D. H. Lee, J. H. Yu, and J. T. Ihm, "Design and Implementation of the Interworking Architecture between a 3G Cellular Network and New Wireless Broadband Technology," in *Proceedings of IASTEAD Conference on Communications Systems and Networks*, Palma De Mallorca, Spain, August 2006.

[84] S. Khan, S. Khan, S. A. Mahmud, and H. Al-Raweshidy, "Supplementary Interworking Architecture for Hybrid Data Networks (UMTS-WiMAX)," in *Proceedings of the*

International Conference on Computing in the Global Information Technology, Bucharest, Romania, August 2006.

[85] M. Suknaic, M. Grgic, and B. Xovko-Cihlar, "Interconnection between WLAN and 3GPP Networks," in *Proceedings of the International Symposium ELMR*, Zadar, Croatia, June 2006.

[86] T. Ali-Yahiya, K. Sethom, and G. Pujolle, "A Case Study: IEEE 802.21 Framework Design for Service Continuity across WLAN and WMAN," in *Proceedings of the International Conference on Wireless and Optical Communications Networks*, Singapore, July 2007.

[87] IEEE, "Midia Independant Handover Services," IEEE P802.21/D01.00, Mar. 2006.

[88] L. Eastwood, S. Migaldi, Q. Xie, and V. Gupta, "Mobility using IEEE 802.21 in a Heterogeneous IEEE 802.16/802.11-based, IMT-Advanced (4G) Network," *IEEE Wireless Communications*, vol. 15, no. 2, pp. 26-34, 2008.

[89] IEEE, "Draft IEEE Standard for Local and Metropolitan Area Networks:Media Independent Handover Services," *IEEE LAN/MAN Draft IEEE P802.21/D07.00*, July 2007.

[90] A. d. l. Oliva, T. Melia, A. Vidal, C. J. Bernardos, I. Soto, and A. Banchs, "IEEE 802.21 enabled mobile terminals for optimized WLAN/3G handovers: a case study," *ACM SIGMOBILE Mobile Computing and Communications Review*, vol. 11, no. 3, pp. 29-40, 2007.

[91] A. d. l. Oliva, T. Melia, A. Banchs, I. Soto, and A. Vidal, "IEEE 802.21 (Media Independent Handover Services) Overview," *IEEE Wireless Communications*, vol. 15, no. 4, pp. 96-103, 2008.

[92] J. Case, M. Fedor, M. Schoffstall, and J. Davin, "A Simple Network Management Protocol (SNMP)," *RFC 1157*, May 1990.

[93] M. Li, K. Sandrasegaran, and T. Tung, "A Multi-Interface Proposal for IEEE 802.21 Media Independent Handover," in *Proceedings of International Conference on the Management of Mobile Business*, Toronto, Canada, July 2007.

[94] W. Lee, M. Kang, and M. Lim, "Implementation of 802.21 for Seamless Handover Across Heterogeneous Networks," in *Lecture Notes in Computer Science*. Berlin: Springer, 2007.

[95] S. McCann, M. Gast, and N. Canpolat, "IEEE P802.11 - TASK GROUP U - MEETING UPDATE: Status of Project IEEE 802.11u," IEEE, July 2008.

[96] G. Lampropoulos, A. K. Salkintzis, and N. Passas, "Media-independent handover for seamless service provision in heterogeneous networks," *IEEE Communications*, vol. 48, no. 1, pp. 64-71, 2008.

[97] T. Melia, L. Boscolo, A. Vidal, and A. de la Oliva, "IEEE 802.21 Reliable Event Service Support for Network Controlled Handover Scenarios," in *Proceedings of IEEE Global Communications Conference*, Washington D.C., USA, December 2007.

[98] Y. Wang, H. Lo, Y. Li, and W. Lee, "Seamless Handover with Buffer Prediction for Wireless Networks Based on IEEE 802.21," in *Proceedings of International Conference on Information Networking*, Pusan, Korea, January 2008.

[99] F. Iqbal, M. Y. Javed, and G. A. Shah, "IEEE 802.21 based multihoming approach to vertical handover in heterogeneous networks," in *Proceedings of IEEE Region 8 International Conference on Computational Technologies in Electrical and Electronics Engineering, Novosibirsk*, Russia, July 2008.

[100] Q. B. Mussabbir, W. Yao, Z. Niu, and X. Fu, "Optimized FMIPv6 Using IEEE 802.21 MIH Services in Vehicular Networks," *IEEE Transactions on Vehicular Technology*, vol. 56, no. 6, pp. 3379-3407, 2007.

[101] J. R. J. Postel, "File Transfer Protocol (FTP)," *RFC 959*, October 1985.

[102] R. Fielding, J. Gettys, J. Mogul, H. Frystyk, L. Masinter, P. Leach, and T. Berners-Lee, "Hypertext Transfer Protocol -- HTTP/1.1," *RFC 2616*, June 1999.

[103] M. Stahl, "DOMAIN ADMINISTRATORS GUIDE," RFC 1032, November 1987.

[104] H. Kwon, K. Rho, A. Park, and J. Ryou, "Mobility Management for UMTS-WLAN Seamless Handover; within the Framework of Subscriber Authentication," in the *Proceedings of IASTED International Conference on Communications, Network and Information Security*, Phoenix, Arizona, November 2005.

[105] F. G. Marquez, M. G. Rodriguez, T. R. Valladares, T. de Miguel, and L. A. Galindo, "Interworking of IP multimedia core networks between 3GPP and WLAN," *IEEE Wireless Communications*, vol. 12, no. 3 pp. 58-65, 2005.

[106] T. Renier, K. L. Larsen, G. Castro, and H. Schwefel, "Mis-Session Macro-Mobility in IMS-Based Networks," *IEEE Vehicular Technology Magazine*, vol. 2, no. 1, pp. 20-27, 2007.

[107] C. Gourraud, "Using IMS as a Service Framework," *IEEE Vehicular Technology Magazine*, vol. 2, no. 1, pp. 4-11, March 2007.

[108] J. O'Connell, "Service Delivery within an IMS Environment," *IEEE Vehicular Technology Magazine*, vol. 2, no. 1, pp. 12-19, March 2007.

[109] W. Wei, N. Banerjee, K. Basu, and S. K. Das, "SIP-based vertical handoff between WWANs and WLANs," *IEEE Wireless Communications*, vol. 12, no. 3, pp. 66-72, 2005.

[110] F. Xu, L. Zhang, and Z. Zhou, "Interworking of WiMAX and 3GPP Networks based on IMS," *IEEE Communications*, vol. 45, no. 3, pp. 144-150, 2007.

[111] 3GPP2, "All-IP Core Network Multimedia Domain," *3GPP2 X.S0013-002-A v1.0*, November 2005.

[112] TIA, "TIA WLAN Standards," *GSC11_GRSC4_45*, 2006.

[113] D. Vingarzan and P. Weik, "IMS Signaling over Current Wireless Networks: Experiments Using the Open IMS Core," *IEEE Vehicular Technology Magazine*, vol. 2, no. 1, March 2007.

[114] G. Camarillo and M. A. García-Martín, *The 3G IP Multimedia Subsystem (IMS): Merging the Internet and the Cellular Worlds*: Wiley, 2004.

[115] V. Koukoulidis and M. Shah, "The IP multimedia domain in wireless networks: concepts, architecture, protocols and applications," in *Proceedings of IEEE International Symposium on Multimedia Software Engineering*, Taipei, Taiwan, December 2004.

[116] 3GPP, "UMTS Release 5," *3GPP RTS/TSGS-0021101v540*, 2003.

[117] 3GPP, "UMTS Release 6," *3GPP RTS/TSGS-0021101v600*, 2005.

[118] J. Rosenberg, H. Schulzrinne, G. Camarillo, A. Johnston, J. Peterson, R. Sparks, M. Handley, and E. Schooler, "SIP: Session Initiation Protocol," *RFC 3261*, 2002.

[119] 3GPP, "IP Multimedia (IM) Subsystem Cx and Dx Interface;Signaling Flows and Message Content," *3GPP TS 29.228 Version 5.12*, 2005.

[120] H. Schulzrinne, S. Casner, R. Frederick, and V. Jacobson, "RTP: A Transport Protocol for Real-Time Applications," *RFC 3550*, 2003.

[121] J. L. P. Calhoun, E. Guttman, G. Zorn, J. Arkko, "Diameter Base Protocol," *RFC 3588*, 2003.

[122] M. Poikselkèa, *The IMS: IP Multimedia Concepts and Services in the Mobile Domain*: John Wiley, 2004.

[123] M. Poikselkä, *The IMS: IP multimedia concepts and services*: Wiley, 2006.

[124] A. Mahendran and J. Nasielski, "3GPP/3GPP2 IMS Differences," *3GPP2, S00-20020513-030*, 2002.

[125] C. Perkins, "IP Mobility Support for IPv4," *IETF RFC 3344*, 2002.

[126] D. Johnson, C. Perkins, and J. Arkko, "Mobility Support in IPv6," *IETF RFC 3775*, 2004.

[127] 3GPP2, "Wireless IP Network Standard," *3GPP2 1 PP2 P.S0001-B*, 2002.

[128] D. Choi, C. Fischer, A. Y. Lee, E. L. Lou, C. Z. Liu, and H. C. Yu, "Transition to IPv6 and support for IPv4/IPv6 interoperability in IMS," *Bell Labs Technical Journal*, vol. 10, no. 4, pp. 261-270, 2006.

[129] A. Roos, M. Hartman, and S. Dutnall, "Critical issues for roaming in 3G," *IEEE Wireless Communications*, vol. 10, pp. 29-35, 2003.

[130] 3GPP, "Technical Specification Group Core Network; NAS Functions related to Mobile Station (MS) in idle mode," *3G TS 23.122 V3.1.1 (2000-2)*, Feb. 2002.

[131] 3GPP, "Technical Specification Group Services and System Aspects; General Packet Radio Service (GPRS); Service description; Stage 2 (Release 6)," *3GPP TS 23.060 V6.11.0 (2005-12)*, December 2005.

[132] M. Handley, V. Jacobson, and C. Perkins, "SDP: Session Description Protocol," *RFC 4566*, July 2006.

[133] J. Rosenberg and H. Schulzrinne, "Reliability of Provisional Responses in the Session Initiation Protocol (SIP)," *RFC 3262*, June 2002.

[134] J. Rosenberg, "The Session Initiation Protocol (SIP) UPDATE Method," *RFC 3311*, September 2002.

[135] H. Schulzrinne, Wedlund, E., "Application-layer mobility using SIP," in *Proceedings of IEEE Service Portability and Virtual Customer Environments*, San Francisco, CA, December 2000.

[136] J. Rosenberg, Peterson, J., Hchulzrinne, H., and Camarillo, E., "Best Current Practices for Third Party Call Control (3pcc) in the Session Initiation Protocol (SIP)," *RFC 3725*, 2004.

[137] R. Sparks, "The Session Initiation Protocol (SIP) Refer Method," *RFC 3515*, 2003.

[138] A. H. Enrique Vazquez, Jose Ignacio Fernandez, "SIP-IMS Model for OPNET Modeler," *OPNET University Program Contributed Models*, 2005.

[139] K. S. Munasinghe, M. R. Kibria, and A. Jamalipour, "Designing VoIP Session Management over Interworked WLAN-3G Networks," *IEEE Wireless Communications*, vol. 15, no. 4, pp. 86-94, 2008.

[140] Nortel Networks, "Voice over packet An assessment of voice performance on packet networks," [Online] Viewed September 2008, Available: http://www.nortelnetworks.com, 2001.

[141] G. Mansfield, T. K. Roy, and N. Shiratori, "Self-similar and fractal nature of Internet traffic data," in *Proceedings of International Conference on Information Networking*, Oita, Japan, February 2001.

[142] T. Kushida, "The traffic measurement and the empirical studies for the Internet," in *Proceedings of the IEEE Global Communications Conference*, Sydney, Australia, November 1998.

[143] W. E. Leland, M. S. Taqqu, W. Willinger, and D. V. Wilson, "On the self similar nature of Ethernet traffic," in *Proceedings of ACM Sigcomm*, San Francisco, CA, USA, 1993.

[144] V. Paxon and S. Floyd, "Wide area traffic: The failure of Poisson Modeling," *IEEE/ACM Transactions on Networking*, vol. 3, no. 3, pp. 226-244, 1995.

[145] W. Leland, M. Taqqu, W. Willinger, and D. Wilson, "On the self-similar nature of ethernet traffic," *IEEE/ACM Transactions on Networking*, vol. 2, no. 1, pp. 1-15, 1994.

[146] K. Salah, "On the deployment of VoIP in Ethernet networks: methodology and case study," *Elsevier Computer Communications*, vol. 29, no. 8, pp. 1039-1054, 2005.

[147] M. Karam and F. Tobagi, "Analysis of delay and delay jitter of voice traffic in the Internet," *Computer Networks*, vol. 40, no. 6, pp. 711-726, 2002.

[148] S. Sharafeddine, N. Kongtong, and Z. Dawy, "Capacity Allocation for Voice over IP Networks Using Maximum Waiting Time Models," in *Lecture Notes in Computer Science: Telecommunications and Networking,* Berlin: Springer, pp. 660-670, July 2004.

[149] C. Wu, K. Chen, Y. Chang, and L. Lei, "Detecting VoIP Traffic based on Human Conversation Patterns," in *Proceedings of Principles, Systems and Applications of IP Telecommunications*, Heidelberg, Germany, July 2008.

[150] L. Cai, Y. Xiao, X. Shen, and J. Mark, "VoIP over WLAN: Voice capacity, admission control, QoS, and MAC," *International Journal of Communication Systems*, vol. 19, pp. 491-508, 2006.

[151] T. M. Chen, "Network Traffic Modeling," in *The handbook of computer networks*, H. Bidgoli, Ed.: Wiley, 2007.

[152] J. Cao, W. S. Cleveland, D. Lin, and D. X. Sun, "Internet Traffic Tends Toward Poisson and Independent as the Load Increases," *Bell Labs Tech. Report*, 2001.

[153] J. Roberts, "Internet Traffic, QoS and Pricing," *Proceedings of the IEEE*, vol. 92, no. 9, pp. 1389-1399, 2004.

[154] S. C. Lo, G. Lee, W. T. Chen, and J. C. Liu, "Architecture for mobility and QoS support in all-IP wireless networks," *IEEE Journal of Selected Areas of Communications*, vol. 22, no. 4, pp. 691-705, 2004.

[155] J. Korhonen, "Performance Implications of the Multi Layer Mobility in a Wireless Operator Networks," Berkeley Summer School. Helsinki University, Computer Science, 2004.

[156] L. Kleinrock, *Queuing Systems Volume 1: Theory*: John Wiley & Sons, 1975.

[157] A. Leon-Garcia and I. Widjija, *Communication Networks - Fundamental Concepts and Key Architectures*, 2 ed. New York: MacGraw Hill, 2004.

[158] S.-K. Kweon and K. G. Shin, "Real-Time Transport of MPEG Video with a Statistically Guaranteed Loss Ratio in ATM Networks," *IEEE Transactions on Parallel and Distributed Systems*, vol. 12, no. 4, pp. 387-403, April 2001.

[159] D. Tipper and M. K. Sundareshan, "Numerical Methods for Modeling Computer Networks Under Nonstationary Conditions," *IEEE Journal on Selected Areas in Communications*, vol. 8, no. 9, pp. 1682-1695, December 1990.

[160] H.-H. Choi, O. Song, and D.-H. Cho, "Seamless Handoff Scheme Based on Pre-registration and Pre-authentication for UMTS–WLAN Interworking," *Wireless Personal Communications*, vol. 41, no. 3, pp. 345-367, 2006.

[161] C. Makaya and S. Pierre, "An Architecture for Seamless Mobility Support in IP-Based Next-Generation Wireless Networks," *IEEE Transactions on Vehicular Technology*, vol. 57, no. 2 pp. 1209-1225, 2008.

[162] Y. J. Lee and I. F. Akyildiz, "A New Scheme for Reducing Link and Signaling Costs in Mobile IP," *IEEE Transactions on Computers*, vol. 52, no. 3, pp. 706-712, 2003.

[163] U. Narayan and X. Jiang, "Signaling Cost Analysis of Handoffs in Mixed IPv4/IPv6 Mobile Environment," in *Proceedings of the IEEE Global Communications Conference*, Washington, D.C., USA, November 2007.

[164] Q. Wang and M. A. Abu-Rgheff, "Interacting mobile IP and SIP for efficient mobility support in all IP wireless networks," in *Proceedings of the IEE International Conference on 3G Mobile Communication Technologies*, London, UK, October 2004.

[165] J. Li, H. Kameda, and K. Li, "Optimal dynamic mobility management for PCS networks," *IEEE/ACM Transactions on Networking*, vol. 8, no. 3, June 2001.

[166] D. Pesch, M. I. Pous, and G. Foster, "Performance evaluation of SIP-based multimedia services in UMTS," *Computer Networks*, vol. 49, no. 3, pp. 385-403, 2005.

[167] G. Agglou and R. Tafazolli, "Determining the optimal configuration for the relative distance micro discovery ad hoc routing protocol," *IEEE Transactions on Vehicular Technology*, vol. 51, no. 2, pp. 354-370, 2002.

[168] R. TENG, N. IMAI, H. MORIKAWA, and T. AOYAMA, "Relative Distance Routing with Virtual Clustering for High Density Network.," *Institute of Electronics, Information and Communication Engineer's Technical Report*, vol. 102, pp. 83-88, 2002.

[169] J. W. Roberts, "Traffic Theory and the Internet," *IEEE Communications*, vol. 39, no. 1, pp.94-99, 2001.

[170] L. Kleinrock, *Queuing Theory – Vol. 2*: Wiley, 1976.

[171] Verio, "Verio Backbone SLA Terms and Conditions," [Online] Viewed September 2008, Available: http://www.verio.com/global-ip-guarantee/, 2008.

[172] Internap, "Internap Service Level Agreement," [Online] Viewed September 2008, Available: http://www.internap.com/product/technology/performanceip/files/DS_IS_sla_0208.pdf, Sept. 2009.

[173] R. G. Cole and J. H. Rosenbluth, "Voice over IP Performance Monitoring," *ACM SIGCOMM Computer Communication Review*, vol. 31, no. 2, pp. 9-24, 2001.

[174] X. Hong, M. Gerla, G. Pei, and C. Chiang, "A group mobility model for ad hoc wireless networks," in *Proceedings of ACM international workshop on Modeling, analysis and simulation of wireless and mobile systems*, Seattle, Washington, USA, 1999.

[175] J.-L. Huang and M.-S. Chen, "On the effect of group mobility to data replication in ad hoc networks," *IEEE Transactions on Mobile Computing*, vol. 5, no. 5 pp. 492-507, 2006.

[176] M. Gerla, K. Xu, and X. Hong, "Exploiting Mobility in Large Scale Ad Hoc Wireless Networks," in *Proceedings of the IEEE Workshop on Computer Communications*, Dana Point, CA, USA, October 2003.

[177] X. Hong, M. Gerla, G. Pei, and C.-C. Chiang, "A Group Mobility Model for Ad Hoc Wireless Networks," in *Proceedings of ACM International Workshop on Modeling, Analysis, and Simulation of Wireless and Mobile Systems*, Seattle, Washington, USA, August 1999.

[178] T. Camp, J. Boleng, and V. Davies, "A Survey of Mobility Models for Ad Hoc Network Research," *Wireless Communication & Mobile Computing*, vol. 2, no. 5, pp. 483-502, 2002.

[179] C. Schindelhauer, T. Lukovszki, S. Rührup, and K. Volbert, "Worst case mobility in ad hoc networks," in *Proceedings of ACM Symposium on Parallel Algorithms and Architectures*, San Diego, California, USA, September 2003.

[180] L. Badia and N. Bui, "A Group Mobility Model Based on Nodes' Attraction for Next Generation Wireless Networks," in *Proceedings of the International conference on Mobile technology, applications & systems*, Bangkok, Thailand, October 2006.

[181] T. B. Zahariadis, K. G. Vaxevanakis, C. P. Tsantilas, N. A. Zervos, and N. A. Nikolaou, "Global roaming in next-generation networks," *IEEE Communications*, vol. 40, no. 2, February 2002.

[182] S. M. Faccin, P. Lalwaney, and B. Patil, "IP multimedia services: analysis of mobile IP and SIP interactions in 3G networks," *IEEE Communications*, vol. 42, no. 1, pp. 113-120, 2002.

[183] A. B. Johnston, *SIP: Understanding the Session Initiation Protocol*: Artech House, 2003.

[184] J.-W. Jung, R. Mudumbai, D. Montgomery, and H.-K. Kahng, "Performance evaluation of two layered mobility management using mobile IP and session initiation protocol," in *Proceedings of IEEE Global Communications Conference*, San Francisco, California, December 2003.

[185] G. Bajko, B. Bertenyi, and K. Kiss, "Multimedia sessions between 3G wireless and Internet users," in *Proceedings of IEEE International Conference on Communications*, Helsinki, Finland, June 2001.

[186] 3GPP, "Combined GSM and Mobile IP Mobility Handling in UMTS IP CN," *3GPP TR 23.923 v 3.0.0*, 2000.

[187] S. Zeadally, F. Siddiqui, N. DeepakMavatoor, and P. Randhavva, "SIP and mobile IP integration to support seamless mobility," in *Proceedings of the IEEE International Symposium on Personal, Indoor and Mobile Radio Communications*, Barcelona, Spain, September 2004.

[188] L. Nuaymi, *WiMAX: technology for broadband wireless access*: John Wiley, 2007.

[189] L. Ma, "The Competition and Cooperation of WiMAX, WLAN, and 3G," in *Proceedings of the IEE International Conference on Mobile Technology, Applications and Systems*, Guangzhou, China, November 2005.

[190] J. G. Andrews, A. Ghosh, and R. Muhamed, *Fundamentals of WiMAX: Understanding Broadband Wireless Networking*: Prentice Hall, 2007.

[191] K. R. Santhi and G. S. Kumaran, "Migration to 4G: Mobile IP based Solutions," in *Proceedings of International Conference on Internet and Web Applications and Services*, Guadeloupe, French Caribbean, February 2006.

[192] J. M. Clint Smith, *3G Wireless With WiMAX and Wi-Fi: 802.16 and 802.11*: McGraw-Hill Professional, 2005.

[193] D. Sweeney, *WiMAX Operator's Manual: Building 802.16 Wireless Networks*: Apress, 2005.

[194] J. Korhonen, "WiMAX versus 3G: threat or opportunity," *IEE Communications Engineer*, vol. 3, no. 6, pp. 38-41, 2006.

[195] D. Gray, "Mobile WiMAX - Part 1: A Technical Overview and Performance Evaluation," WiMAX Forum, August 2006.

[196] M. Paolini, "Building Eng-to-End WiMAX Networks," WiMAX Forum, April 2007.

[197] A. T. Andersen, A. Jensen, and B. F. Nielsen, "Modeling And Performance Study Of Packet-Traffic With Self–Similar Characteristics Over Several Time-Scales With Markovian Arrival Process (MAP)," in *Proceedings of the Nordic Teletraffic Seminar NTS*, 1995.

[198] S. Basu, A. Mukherjee, and S. Klivansky, "Time Series Models for Internet Traffic," in *Proceedings of IEEE INFOCOM*, San Francisco, California, March 1996.

[199] A. Feldmann, A. C. Gilbert, and W. Willinger, "Data Networks As Cascades:Investigating The Multifractal Nature Of The Internet," *ACM SIGCOMM Computer Communication Review*, vol. 28, no. 4, pp. 42-55, 1998.

[200] A. Feldmann, A. C. Gilbert, W. Willinger, and T. Kurtz, "The Changing Nature of Network Traffic: Scaling Phenomena," *Computer Communication Review*, vol. 28, no. 2, pp. 5-29, 1998.

[201] H. Fowler and W. Leland, "Local Area Network Traffic Characteristics, With Implications For Broadband Network Congestion Management," *IEEE Journal in Selected Areas in Communications*, vol. 9, no. 3, pp. 1139-1149, 1991.

[202] V. S. Frost and B. Melamed, "Traffic Modeling For Telecommunications Networks," *IEEE Communications*, vol. 32, no. 3, pp. 70-81, 1994.

[203] K. S. Munasinghe and A. Jamalipour, "Analysis of Vertical Session Handoff for Self-Similar Traffic in a Heterogeneous Mobile Data Network," in *Proceedings of International Conference on Signal Processing and Communication Systems*, Gold Coast, Australia, December 2008.

[204] T. Fowler, "Fractals, Long Range Dependence And Packet Traffic," in *Proceedings of Advanced Simulation Technologies Conference*, San Diego, CA, 1999.

[205] M. J. Fischer, D. M. B. Masi, D. Gross, and J. F. Shortle, "One-Parameter Pareto, Two-Parameter Pareto, Three-Parameter Pareto: Is there a modeling difference?" *The Telecommunications Review*, 2005.

[206] G. Gross and C. M. Harris, *Fundamentals of Queuing Theory*, 3 ed. New York: John Wiley, 1998.

[207] C. M. Harris and W. G. Marchal, "Distribution Estimation Using Laplace Transforms," *INFORMS Journal of Computing*, vol. 10, pp. 448-458, 1998.

[208] M. J. Fischer and C. M. Harris, "A Method for Analyzing Congestion in Pareto and Related Queues," *The Telecommunications Review*, vol. 2, pp. 15-22, 1999.

[209] K. S. Munasinghe and A. Jamalipour, Evaluation of Session Handoffs in a Heterogeneous Mobile Network for Pareto based Call Arrivals, Submitted for publication in *Proceedings of the IEEE Wireless and Communications Conference (WCNC 2009)*, Budapest, Hungary, April 2009.