Gestion de la mobilité et de la QoS des réseaux de la nouvelle génération

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A new approch for Mobility and QoS Management for Vertical Handovers in Next Generation Networks.

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Résumé

Les réseaux de la nouvelle génération "Next Generation Networks" sont considérées comme un réseau tout-IP contenant des différents technologies d'accès pour accommoder des services variés, et différents types de trafic permettant à l'utilisateur mobile de changer de point d'accès dans le même réseau d'accès ou entre différents réseaux sans dégradation de la qualité de service offerte. Le défi principal est d'offrir à l'utilisateur "roaming" entre différents réseaux d'accès une mobilité transparente et un "handover" rapide. D'ou le terme "Seamless Mobility". Ce type de mobilité appelé handover vertical ou "Vertical handover" est le défi principal à faire face dans les réseaux de la nouvelle génération où des technologies d'accès différentes seront intégrées. Le handover vertical doit être réalisés "seamlessly" et sans dégradation de la qualité de service QoS.

Pour offrir QoS dans le réseau tout-IP, la mobilité des utilisateurs et la gestion de ressources doivent être pré-controlées afin d'assurer à l'utilisateur un handover transparent tout en conservant la qualité de service établie dans le réseau initial. Pour garantir la qualité de service QoS et la mobilité d'un mobile se déplaçant d'un réseau à un autre de technologie d'accès différente, une gestion spéciale de mobilité et de QoS est nécessaire pour permettre au mobile de consérver la QoS déjà établie et de maintenir une connexion dans le nouveau réseau avec les ressources nécessaires avant d'initier le handover, avec minimum de délai, minimum de perte et minimum de dégradation de QoS.

Dans ce contexte, nous introduisons dans le premier chapitre l'évolution des systèmes de communications commencant par les réseaux cellulaires, réseaux UMTS, passant par 802.11 WLAN et clôturant par les réseaux en cours de développement WiMax. Tous ces différentes technologies d'accès offrent à l'utilisateur des services différents à des bandes passantes différentes, sans la possibilité de lui offrir des niveaux de QoS pre- défini en relation avec sa mobilité offerte.

Les réseaux de la nouvelle génération "Next Generation Networks" sont basés sur l'intégration des différentes technologies d'accès de manière que les caractéristiques de chaque réseau d'accès sont offertes aux utilisateurs se déplaçant dans tout le système de réseaux intégrés. Les deux technologies d'accès considérées comme complémentaires dans l'intégration des réseaux de futures générations sont WLAN et UMTS. D'une part, UMTS offre une large couverture géographique mais à de bas débits et d'autre part WLAN offre de hauts débits mais dans une couverture géographique limitée. D'où un bref overview sur les spécifications et caractéristiques de ces deus technologies d'accès est présenté au chapitre2.

Dans le chapitre 3, on présente les différentes architectures d'intégration existantes intégrant les technologies à convergence naturelle comme UMTS et WLAN et les raisons principales à forcement intégrer ces deux technologies. Les défis principaux et les coût de l'intégration sont étudiés en considérant différents protocoles de mobilité et différents scénario d'intégration.

Dans le chapitre 4, on propose d'utiliser "**Anticipated Vertical Handover** Architecture" basée sur le protocole de mobilité Fast MIPv6 dans le contexte du handover de UMTS à WLAN et vice versa. Cela permet un "seamless handover" basé sur une pre-acquisition du l'adresse CoA "Care-of-address" à travers les messages de signalisation de FMIPv6 établies avant l'initiation du handover. Avec l'application de "**Anticipated Vertical Handover**" (AVHO) dans le contexte du handover vertical, des études de simulations sont faites pour comparer AVHO au MIPv6, le protocole générique de mobilité. Dans cette proposition, les signalisations du handover sont anticipées avant le déroulement du handover basé sur un " forced handover" scénario ou le mobile est notifié pour initier le procédure du handover avant qu'il se déplace au nouveau réseau d'accès afin de minimiser le délai du handover "handover latency" et la perte des paquets. Mais la gestion de la qualité de service reste un autre challenge à surmonter.

Pour ce but, notre deuxième contribution est la proposition d'une nouvelle architecture pour intégrer WLAN et UMTS offrant QoS à l'utilisateur se déplaçant entre différents réseaux d'accès. Dans notre architecture proposée appelée "**Mobility and QoS Management Architecture**" MQMA, un nouveau module appelé "Inter-Domain Management " IDM est proposé pour réaliser la gestion de QoS et pour guider la demande du handover au réseau d'accès convenable. IDM est présent dans chaque réseau d'accès pour offrir "seamless handover" en sélectionnant dans la phase "Who's next" de l'IDM le meilleur réseau capable d'offrir et de satisfaire les paramètres de QoS en demande. Cela est réalisé par la communication avec l'IDM sélectionné en traitant le handover basé sur le protocole de gestion de Mobilité et QoS (**Mobility and QoS Management Protocol MQMP).** Des études de performances sont faites pour étudier l'efficacité de l'architecture proposée. Notre architecture montre qu'elle est générale et scalable comme elle peut être appliquée non seulement dans le cadre du handover vertical mais aussi du celui de handover horizontal.

Dans le contexte du modèle MQMA, le protocole de gestion de mobilité et QoS (**Mobility and QoS Management Protocol**) est proposé pour contrôler les échanges de messages entre les différentes entités incluses dans le modèle MQMA. MQMP est responsable du fonctionnement de l'IDM et du procédure du handover entre les différents réseaux d'accès.

L'IDM proposé est étendue dans trois sous-entités appelées IDM vertical (**VIDM**), IDM horizontal (**HIDM**), et stand-by IDM (**SIDM**). Chaque entité est responsable de guider le handover au réseau convenable afin de sélectionner le meilleur réseau. D'où le handover peut être réalisé verticalement par VIDM ou guidé horizontalement au même réseau d'accès ou au même technologie d'accès par HIDM. Dans le troisième cas, le handover est guidé au réseau WiMax par le SIDM, du fait que WiMax est considéré comme le réseau de destination en réserve (stand-by destination network) capable de satisfaire les différentes obligations du handover. Dans cette méthode, le problème de laisser tomber (dropping) les demandes du handover est résolu comme le handover n'est plus jeté mais guidé au réseau d'accès convenable, comme décrit au chapitre 5.

En conséquence, notre troisième contribution est la proposition de résoudre les problèmes de mobilité horizontale sans dégradation de la qualité de service QoS pour inter-UMTS et inter-WLAN handovers. Le but est d'avoir un handover horizontal transparent et "seamless". Les signalisations utilisées sont aussi basées sur MQMP appliqué dans le cadre de handover UMTS-UMTS et WLAN-WLAN, comme décrit au chapitre 6.

Finalement, dans le chapitre 7, l'intégration de la récente technologie WiMax dans notre modèle proposé est présentée. Dans ce scénario, le handover de UMTS à WiMax d'une part, et de WLAN à WiMax d'autre part est considéré pour le but de résoudre tous les problèmes de handover vertical et horizontal. Une telle intégration globale offre une mobilité transparente et un support de QoS pour les réseaux de la nouvelle génération accommodant plusieurs technologies d'accès.

Au chapitre 8, on conclut que l'architecture proposée comme une solution globale peut satisfaire les demandes principales du handover guidé au réseau sélectionné. En comparant à d'autre solutions de gestion de mobilité et du handover, le modèle MQMA montre son efficacité en fonction du "seamless" handover offert sans dégradation de QoS, et avec prévention du terminaison forcée du handover.

Table of contents

Chapter 3 Integration Architectures

Chapter 4 UMTS/WLAN Handover

Chapter5 Mobility and QoS management architecture MQMA

Chapter 6 Mobility and QoS Management for Horizontal Handover

Chapter 7 WiMax Integration with MQMA model

Chapter 8 Conclusion and Perspectives

Abstract

The vision of next generation networks is an all-IP network supporting heterogeneous wireless access technologies to accommodate a variety of services and traffic types and to allow the mobile user to roam within the service area or across the different networks without degrading the QoS provided. The main challenge is to provide the user roaming between different access technology with a transparent and seamless mobility. This type of mobility, known as vertical handover, is the main challenge to defeat in next generation networks. Vertical handovers should be performed seamlessly and with no QoS degradation. In order to provide QoS in the converged all-IP network, user mobility and resource reservation should be pre-managed in a way that the mobile node will not encounter a handover latency and QoS degradation when moving from one access network to another. To manage the mobility of the mobile nodes moving from one access technology to another, a special mobility and QoS management policy are needed in order for the session to be resumed on the new data path with minimum delay, packet loss, and QoS degradation.

In this context, we introduced in the first chapter, the evolution of communication systems starting with cellular networks, UMTS networks, passing through the WLAN networks and ending up with the emerging WiMax technology. All these different access technologies have provided users with different throughput and services, but without being able to provide a pre-defined QoS levels, coping with the user mobility. The next generation mobile data networks is build on integrating different access technologies so that the characteristics of each access network are offered to the user roaming in the integrated system. The two complementary access technologies which are to be integrated for this NGN vision are the UMTS and the WLAN since UMTS provides low data rates with high coverage area and WLAN provides higher data rates but in restricted geographical areas. In this frame, a brief introduction on UMTS and WLAN specifications and characteristics is presented in chapter 2.

In chapter 3, we describe the existing integration architectures combining the naturally converged networks UMTS and WLAN and the reasons behind integrating these different access technologies. The main challenges faced and the integration cost are studied, in addition to the different mobility protocols employed in the different integration scenarios.

In chapter 4, we proposed to apply **an Anticipated Vertical Handover Architecture** based on anticipated FMIPv6 mobility protocol in the context of WLAN to UMTS handovers, and vice versa. This will allow a seamless handover based on pre-CoA acquisition due to the

FMIPv6 signaling messages done prior to Handover initiation. With the Anticipated Vertical Handover (AVHO) applied in the vertical handover context, simulation results were conducted and compared to the MIPv6 mobility protocol. In this proposition, the handover signaling between UMTS and WLAN were anticipated prior to handover based on a forced handover scenario where the mobile node is notified to start the handover process before moving to the new access network in order to minimize the handover latency and packet loss. But, QoS management is another challenging issue that is still to be addressed.

To this end, our second contribution is the proposition of a new architecture to integrate WLAN and UMTS, and to provide QoS to the end user moving between different access technologies. In the proposed architecture named **Mobility and QoS Management Architecture MQMA**, a new network module called the inter-domain management **IDM** was proposed in order to perform QoS mapping and "guide" the handover request to the appropriate access network. IDM is present in each access network to provide a seamless handover and search through the IDM "Who's Next" phase for the best network capable of handling the required QoS parameters. This is performed by communicating with the selected IDM, while processing the handover based on **Mobility and QoS Management Protocol MQMP**. Performance studies were performed to study the efficiency of the proposed architecture. Our architecture shows to be general and scalable since it will apply not only to vertical handovers but also to horizontal handovers as well.

In the context of MQMA model, the **Mobility and QoS Management Protocol MQMP** was proposed to control the message exchanged between the different entities involved in the MQMA model. The MQMP will be responsible of the IDM functioning and the HO working process between different access technologies.

The proposed IDM entity was extended to three sub-entities called Vertical IDM (**VIDM**), Horizontal IDM (**HIDM**) and Stand-by IDM (**SIDM**). Each entity will guide the handover to the appropriate network ending up with the best network selection. Hence, the HO could be performed vertically by VIDM or guided horizontally to the same access technology by HIDM. In the third case, the HO is guided to the emerging WiMax network, by the SIDM, since the WiMax will be considered as the stand-by destination network capable of satisfying different HO requirements. In this method, the problem of dropping handover calls is resolved since the HO is no more rejected but guided to the suitable access network, as described in chapter 5.

As a result, our third contribution is proposed to solve the horizontal mobility issues with no QoS degradation for inter UMTS and inter WLAN handovers. In this way, horizontal handover (HHO) is also transparent and seamless. The necessary signaling transfer is based on the proposed MQMP applied for UMTS–UMTS and WLAN-WLAN handovers, as described in chapter 6.

Finally, in chapter7, the integration of the new emerging WIMAX technology to our MQMA model is also presented. In this scenario, the HO from UMTS to WiMax from one side, and from WLAN to WIMAX from the other side is also considered for the purpose of solving all the challenges of horizontal and vertical handovers. Such a global integration model will provide seamless mobility and QoS support in next generation networks combining different access technologies.

In the last chapter, we conclude that the proposed architecture with Seamless Mobility and QoS Management is a global solution that satisfies the keys requirements of the future handovers which will be guided to the selected access network. Compared to other mobility solutions, the MQMA model proved to be more efficient in terms of providing seamless handover with no QoS degradation and preventing forced HO termination.

Chapter 1

Evolution of Mobile Communications

1.1. Introduction

A variety of networks like GSM, GPRS, UMTS or WLAN, WIMAX will co-exist in the future, each offering different coverage and bandwidth to users. Continuous end-to-end connectivity must be supported and the user shall be able to specify the desired QoS while moving between different accesses technologies. A seamless integration of heterogeneous wireless access networks with QoS provision is a major challenge. Thus, next-generation wireless systems call for the integration and interoperation of mobility management techniques in heterogeneous networks. The future vision of the next generation networks is to integrate different wireless networks considered as complementary to each other. The mobile user will be able to connect to the system using the best available access network. The integration of different networks faces several challenges because of the following heterogeneities: service demands, access technologies, network architecture, mobility and handover management.
In this chapter, an overview of the mobile communication evolution is presented followed by a description of the different mobility management protocols. At the end the main challenges and requirements of an all-IP network are described.

1.2. Wireless communication evolution

The International Telecommunication Union (ITU) defines a 3G network as one that delivers, among other capabilities, improved system capacity and spectrum efficiency versus 2G systems. It supports data services at transmission rates of at least 144 Kbps in mobile (moving) environments and at least 2 Mbps in fixed (indoor) environments.
The most important 3G network proposal is the Universal Mobile Telecommunications System UMTS. The 3G cellular system (UMTS) will support real-time and non-real-time multimedia services with data rates from 144 kb/s to 2 Mb/s with wide coverage and nearly universal roaming [Hol03]. However, the costs of acquiring the necessary radio spectrum and the required network equipment upgrades are very high. These data rates are in contrast to WLAN systems such as IEEE 802.11 a/b/g, which provide affordable services and bit rates surpassing those of 3G systems, up to 11 Mb/s with 802.11b and 54 Mb/s with 802.11a/g [Wal01]. However, the coverage offered by WLANs is quite limited and lacks roaming support. The

complementary characteristics of 3G cellular systems (slow, wide coverage) and WLAN (fast, limited coverage) make it attractive to integrate these two technologies to provide ubiquitous wireless access.

1.2.1. First Generation (1G): Analog Cellular

The introduction of cellular systems in the late 1970s and early 1980s represented a quantum leap in mobile communication (especially in capacity and mobility). These 1G cellular systems still transmit only analog voice information. The most prominent 1G systems are Advanced Mobile Phone System (AMPS), Nordic Mobile Telephone (NMT), and Total Access Communication System (TACS). With the 1G introduction, the mobile market showed annual growth rates of 30 to 50 percent, rising to nearly 20 million subscribers by 1990 [Wal01].

1.2.2. Second Generation (2G): Multiple Digital Systems

The development of 2G cellular systems was driven by the need to improve transmission quality, system capacity, and coverage. Further advances in semiconductor technology and microwave devices brought digital transmission to mobile communications. Speech transmission still dominates the airways, but the demands for fax, short message, and data transmissions are growing rapidly. Supplementary services such as fraud prevention and encrypting of user data have become standard features that are comparable to those in fixed networks. 2G cellular systems include GSM, Digital AMPS (D-AMPS), code division multiple access (CDMA), and Personal Digital Communication (PDC). Different standards serve different applications with different levels of mobility, capability, and service area (paging systems, cordless telephone, wireless local loop, private mobile radio, cellular systems, and mobile satellite systems). GSM is the most successful family of cellular standards (GSM900, GSM–railway [GSM–R], GSM1800, GSM1900, and GSM400 [Wal01].

1.2.3. 2G to 3G: GSM Evolution

Phase 1 of the standardization of GSM900 was completed by the European Telecommunications Standards Institute (ETSI) in 1990 and included all necessary definitions for the GSM network operations. Several tele-services and bearer services have been defined (including data transmission up to 9.6 kbps), but only some very basic supplementary services were offered. As a result, GSM standards were enhanced in Phase 2 (1995) to incorporate a large variety of supplementary services that were comparable to digital fixed network integrated services digital network (ISDN) standards. In 1996, ETSI decided to further enhance GSM in annual Phase 2+ releases that incorporate 3G capabilities [Cha02].

GSM Phase 2+ releases have introduced important 3G features such as intelligent network (IN) services with customized application for mobile enhanced logic (CAMEL), enhanced speech compression/decompression (CODEC), enhanced full rate (EFR), and adaptive multi-rate (AMR), high–data rate services and new transmission principles with high-speed circuit-switched data (HSCSD), general packet radio service (GPRS), and enhanced data rates for GSM evolution (EDGE). UMTS is a 3G GSM successor standard that is downward-compatible with GSM, using the GSM Phase 2+ enhanced core network. Ultimately the ITU took a position on the matter, defining an IMT-2000 standard that encompassed five different radio interfaces including CDMA2000. Note that all of the IMT-2000 protocols use spread-spectrum techniques, which has implications about network installation, operation and maintenance. [Wal01]

The UMTS Forum is expecting a high grade of asymmetry between uplink and downlink for data transmission for services such as Internet access, since the exponential growth of the Internet in the last few years is the main driver for data services, with higher necessary capacity on the downlink.

Future services will range from low up to high user data rate, maximum 2 Mbps [Hol02].

1.2.4. Standardization of 3G

The International Telecommunications Union (ITU) has launched one of its most ambitious projects ever. Third generation mobile telecommunications will provide wireless access to the global telecommunication infrastructure any time and anywhere. This new framework of standards is known under the generic name of International Mobile Telecommunications-2000 (IMT-2000) and in Europe is named Universal Mobile Telecommunications System (UMTS) [Hol02].

IMT-2000/UMTS represents the culmination of ten years of study and design work, it is also considered the most exciting development in the mobile communications since the advent of digital systems back in the early1990s.

IMT-2000/UMTS identifies the following key factors as essential for the success of the next generation of mobile communications:

- Used worldwide,
- Used for all mobile applications,
- Support both packet-switched (PS) and circuit-switched (CS) data transmission,
- Offer high data rates up to 2 Mbps (depending on mobility/velocity), and
- Offer high spectrum efficiency.

Third generation will operate in all radio environments like urban and suburban areas, hilly and mountainous areas, microcell, picocell, and indoor environments to provide service to anyone, anytime, and anywhere.

This enables a much wider application range of third generation systems compared to the second generation. In addition, the ability for global roaming has to be supported in the system design.

The ITU defines a 3G network as one that delivers, among other capabilities, improved system capacity and spectrum efficiency versus 2G systems. It supports data services at transmission rates of at least 144 Kbps in mobile (moving) environments and at least 2 Mbps in fixed (indoor) environments.

The most important IMT–2000 proposals are the UMTS (W-CDMA) as the successor to GSM, CDMA2000 as the interim standard ’95 (IS–95) successor, and time division–synchronous CDMA (TD–SCDMA) (universal wireless communication–136 [UWC–136]/EDGE) as TDMA–based enhancements to D–AMPS/GSM—all of which are leading previous standards toward the ultimate goal of IMT–2000. [Hol02]

1.2.5. Wireless LANs

WLAN enables wireless networks that support data rates of 1–54 Mb/s over small areas of few thousand square meters. Global public and industry interest in Wireless LANs (WLANs) has exploded in the past year. Companies are deploying WLANs in offices and factories as replacements for wired LANs or to provide employees with access to the corporate network. Universities and hospitals are also deploying WLANs. In some countries, public WLAN service providers have launched fee-based WLAN services in public venues such as coffee shops, hotels, and airports. In the private sector, WLANs are being installed by consumers to create in-home networks and in public spaces by community groups to provide free Internet access to citizens. The list of major (and start-up) telecommunications hardware and software manufacturers, as well as service providers, supporting WLAN grows almost daily. [Cro97]

1.2.6. WiMax

WiMax stands for World wide Interoperability foe microwave access. WiMax is defined in the IEEE 802.16 standards and is being promoted by the WiMax forum.
Being a wireless radio access technology, WiMax cannot avoid being compared to WLAN. However, this technology is much more efficient in several areas. It offers a maximum useful throughput of 60 megabytes per second (Mbps) in a 20-MHz channel, as compared to 25 useful Mbps under standards 802.11 a or g. The quality of service of WiMax is also superior, thanks notably to more sophisticated control over transmission, which guarantees a good level for the services offered. It also provides greater security as the final equipment must be declared before being connected to the network.
Table 1.1 states the different WiMax characteristics together with a WLAN comparison [Cro97].

Standard	Maximum Bit Rate	Fallback Rates	Channels Provided	Frequency Band	Radio Technique
802.11	2 Mbps	1 Mbps	3	2.4 GHz	FHSS or DSSS
802.11b	11 Mbps	5.5 Mbps 2 Mbps 1 Mbps	3	2.4 GHz	DSSS
802.11a	54 Mbps	48 Mbps 36 Mbps 24 Mbps 18 Mbps 12 Mbps 9 Mbps 6 Mbps	12	5 GHz	OFDM
802.11g	54 Mbps	Same as 802.11a	3	2.4 GHz	OFDM

Table 1.1 WiMax-WLAN comparisons

1.3. Future all-IP networks

In next generation networks, the problem of providing a transparent and fast vertical handover for a mobile user roaming between different access technologies is of great importance. In fact, performing vertical handover and providing the necessary QoS and resources for an ongoing connection is the key issue in next generation networks. In order to reach a vertical Handover, many works have been conducted in order for the handover to be transparent and fast with respect to the mobile node.

An important objective in the project of the Next Generation NG systems is to support applications and services with performance and quality of service (QoS) equivalents or even with better level that the presented by the wired networks, and also to improve the connectivity with others wireless/wired systems.

With 3G systems deployed, it is necessary to consider how they will evolve to include a much wider range of users, applications, and economic development. As far as the next-generation networks are concerned, the following points are considered [Gao03]:

- Transition to an all-IP network infrastructure
- Support of heterogeneous wireless access technologies, such as UTRAN, WLANs, wireless personal area networks (WPANs), and 3G.
- Seamless handovers across both homogeneous and heterogeneous wireless technologies
- Mobility and quality of service (QoS) support
- Deployment of new protocols for services such as AAA (e.g., DIAMETER) and their interworking with existing technologies, such as the home/visited location register (HLR/VLR) and Mobile Application Part (MAP) in Global System for Mobile Communications (GSM), and Remote Authentication Dial-In User Service (RADIUS) in the Internet
- Support of different types of mobility (terminal, session and personal mobility)
- Mechanisms to support service roaming

1.4. Challenges and requirements

In multi access networking a user should be able to select the "best" access network at any given time depending on availability and suitability for applications [Ier05]. This scenario will typically be deployed in hot-spots where several access networks are available such as airports, conferences or at home. While it is desirable to maintain the QoS during access technology change, applications must be able to adapt to match resource availability in the new access network. Such adaptation processes should be not noticeable to the user.

1.4.1. Decision making

The design of handoff management techniques in NG all-IP-based wireless systems has many challenges such as the reduction of both signaling and power overheads, the QoS guarantees during the handoff process, the handover latency, resources and routes setup delay, packet loss rate, and efficient use of network resources [Sot04].

The decision making process "when to switch to what access technology" requires a cooperative model between the terminal and the network, because it also depends on the availability of a given access network, its capabilities and resource availability to maintain the

QoS. User preferences or cost implications may also influence this decision. Operator policies and user contracts with network/service providers play a significant role, too.

1.4.2. Mobility management protocols

In the all-IP network architecture, and in order to perform inter-technology handovers between different network technologies, mobility protocols are required in order to execute such handovers.
Mobility protocols exist at different layers of the Internet protocol stack. Each layer exhibits individual behavior.

-Mobile IP, a network layer protocol, is designed to support host mobility at the macro-mobility level. Its goal is to provide the ability of a host to stay connected to the internet regardless of its location**.**
The Internet Protocol version 4 (IPv4) is a fundamental network layer protocol that contains addressing information and some control information that enables data packets to be routed. The addressing information consists of a 32-bit unique identifier called the IP address. Every client on the Internet is identified by a unique IP address. The routing information contains the IP destination address. The major problems with Mobile IPv4 are triangular routing and tunneling overhead. In addition, Mobile IPv4 deployment requires the implementation of foreign agents.

To reduce the service degradation that a mobile node could suffer due to a change in its point of attachment *Fast Handovers for Mobile IPv6* has been proposed.
Using anticipated FMIPv6, the handover is prepared in advance. Assuming we receive the fast binding acknowledgement (F-BAck) via the PAR, i.e, the overlapping area is designed considering the mobile speed to make it possible, and then we perform the handover. The latency will be proportional to the difference between receiving the F-BAck and the reception of the first packet forwarded to the New Access Router (NAR).

Two different mechanisms have been described: anticipated and tunnel-based handover.
Tunnel-based handover relies on link layer triggers to potentially obtain better results than Anticipated Handover, introducing though a link layer dependence that could make the solution unfeasible for some link layer technologies.
In principle, a link layer independent solution would be a more desirable solution. Therefore, we have focused on the performance study of the *Anticipated Handover* proposal, which is solely based on network layer information. Anticipated Handover proposes a ‘make before break’ approach.

-SCTP is a transport layer mobility protocol providing mobility management [Ste00]. Mobility management in the transport layer is accomplished by the use of Stream Control Transmission Protocol (SCTP) and its currently proposed Dynamic Address Reconfiguration (DAR) extension. SCTP with its DAR extension is called Mobile SCTP (mSCTP).

mSCTP is a transport layer protocol similar to Transmission Control Protocol (TCP) that operates on top of the unreliable connection-less packet network. It provides unicast end- to-end communication between two or more applications running in separate hosts and offers connection-oriented, reliable transportation of independently sequenced message streams.

The biggest difference to TCP is multi-homing, the concept of several streams within a connection (multi-streaming) and the transportation of sequence of messages instead of sequence of bytes.

mSCTP is capable of handling several multiple IP addresses at both endpoints while keeping the end-to-end connection intact. These addresses are considered as logically different paths between the endpoints. During initiation of the connection, lists of addresses are exchanged between the endpoints. Both endpoints must be able to receive messages from any of the IP addresses related to the endpoints. One address is chosen as the primary address and is used as the destination for normal transmission. The other addresses are used for retransmissions only. The SCTP DAR extension enables the endpoints to add delete and change the primary address dynamically in an active connection without affecting the established connection.

Unlike techniques based on MIP or SIP, the SCTP-based vertical handover scheme does not require the addition of components such as home/foreign agents or SIP server to the existing networks but it is sufficient to use SCTP DAR as the transport layer protocol beside TCP and UDP. This requires that all the internet nodes must support SCTP in their TCP/IP protocol stack which is a step much more difficult than migration towards IPv6.
The architecture of the integrated system would be very simple because no third party other than the endpoints participates in vertical handover (e.g. GGSN, SGSN or AR).

A viable application-layer mobility protocol is the IETF-developed signaling protocol Session Initiation Protocol (SIP).

SIP is a signaling protocol mainly used to establish, modify, and terminate multimedia sessions consisting of multiple media streams, unicast as well as multicast.
Table 1.2 lists the most distinct differences between the mobility protocols [Scho1].

Protocol	Mobile IPv4	Mobile IPv6	mSCTP	SIP
Layer	Network	Network	Transport	Application
Transparency	Yes	Yes	Yes	Yes
Transport services	TCP/UDP	TCP/UDP	SCTP	UDP
Deployment	Home agent in home network Foreign agent in foreign network Mobile IPv4 supported by home agent, foreign agent and client	Home agent in home network Mobile IPv6 supported by home agent and client	mSCTP supported by client and server	SIP server in home network SIP supported by SIP server, client and server

Table 1.2 Comparison of Mobility Protocols

1.4.3. QoS provisioning

The vision of next generation networks is an all-IP network supporting heterogeneous wireless access technologies. The main challenge is to provide the user roaming between different access technology with a transparent and seamless mobility. NG mobility should be performed seamlessly and with no QoS degradation. In order to provide QOS in the converged all-IP network, user mobility and resource reservation should be pre-managed in a way that the mobile node will not encounter a handover latency and QoS degradation when moving from one access network to another.

Thus, QoS provisioning between the heterogeneous access networks is a real challenge in the integration of in NGN.

1.4.4. Mobile Device Requirement

Demand for high performance, multifunction handheld devices such as mobile phones, PDAs and laptop computers continue to grow. Next generation handheld devices will possibly have an optional access to WLAN, 2.5G/3G cellular technologies and WiMax. These subsystems must be designed in a way that the mobile user will be free to select the best network technology, the cheapest and the faster one. Besides, the ability of the mobile to switch between different access technologies should be highly considered.

1.5. Conclusion

The network of the future wireless mobile generation will be based on pure IP- network leading to a global interconnection and integration between the various telecommunications networks. This will require new integration architectures and mobility protocols to provide seamless roaming and QOS provisioning to the end user.

Since UMTS and WLAN are viewed as the future complementary access technologies, existing integration architectures are presented in the next chapter.

In chapter 3, MIPv6 and Anticipated Vertical HO based on FMIPv6 are compared in the context of UMTS to WLAN handovers and vice versa.

In chapter 4, a new architecture with Seamless Mobility and QoS Management is proposed, based on Inter-Domain Manager module IDM responsible of providing seamless handover by mapping the required resources and QoS level of the mobile node to the appropriate IDM in the new domain with respect to the transferred context and user profile. The mobile user will then be "guided" according to the selected access network. To satisfy the required QoS, a specific mobility and admission control strategy and resource mapping are done prior to handover.

In chapter 5, the horizontal handover was also solved and presented as an alternate solution for vertical handovers.

In chapter 6, the evolving WiMax technology is considered as an extension to our proposal to prove its scalability and efficiency in the next heterogeneous IP network combining different access technologies.

Chapter 2

UMTS and WLAN Overview

2.1. Introduction

Standing for "Universal Mobile Telecommunications Systems", UMTS represents an evolution in terms of services and data speeds from today's "second generation" mobile networks. As a key member of the "global family" of third generation (3G) mobile technologies identified by the ITU, UMTS is the natural evolutionary choice for operators of GSM networks, currently representing a customer base of more than 850 million end users in 195 countries and representing over 70% of today's digital wireless market. This chapter covers UMTS advantages, its network architecture and its compatibility with the GSM and GPRS systems. It also analyzes the UMTS necessary functions such as transmission requirements and its different management procedures.

Universal Mobile Telecommunications System (UMTS), also referred to as Wideband Code Division Multiple Access (WCDMA), is one of the most significant advances in the evolution of telecommunications into third-generation (3G) networks.

Using fresh radio spectrum to support increased numbers of customers in line with industry forecasts of demand for data services over the next decade and beyond, "UMTS" is synonymous with a choice of WCDMA radio access technology that has already been selected by approaching 120 licensees worldwide.

The world's leading equipment manufacturers are now presenting their first WCDMA/ UMTS handset models, with many of them featuring in-built cameras. Most models in this first wave of UMTS terminal designs are multi-band and multi-mode, allowing users to switch seamlessly between UMTS, GPRS and GSM services in different frequency bands as they travel around the world [Hol02].

2.1.1 UMTS Data Rates

UMTS has three basic data rates; these data rates vary depending on the radii of the cells and the mobility speed. They are classified as follows:
•144 kbps-Satellite and rural outdoor formed of macro cells of 15 km radius,
•384 kbps-Urban outdoor formed of micro cells of 500m radius at a traveling speed of up to 120 km/h,
•2048 kbps-Indoor formed of Pico-cells of 100m radius and low range outdoor at slow speeds up to 10 km/h.

2.1.2. The Frequency Bands of UMTS

The ITU decided on 1992 to reserve 230 MHz of the used spectrum of the IMT-2000 divided into two bands: [Hol03]

•**1920-1980** and **2110-2170** MHz Frequency Division Duplex (FDD, W-CDMA) Paired uplink and downlink, channel spacing is 5 MHz and raster is 200 kHz. An Operator needs 3 - 4 channels (2x15 MHz or 2x20 MHz) to be able to build a high-speed, high-capacity network.
•**1900-1920** and **2010-2025** MHz Time Division Duplex (TDD, TD/CDMA) Unpaired, channel spacing is 5 MHz and raster is 200 kHz. Tx and Rx are not separated in frequency.
•**1980-2010** and **2170-2200** MHz Satellite uplink and downlink.

2.2. Technical Evolution towards UMTS

As discussed in the first chapter, the evolution of mobile communications through the different decades was classified into generations where each generation was defined by a set of different standards and demands. The evolution of the transmission rates is shown in Figure 2.1 [Hol02].

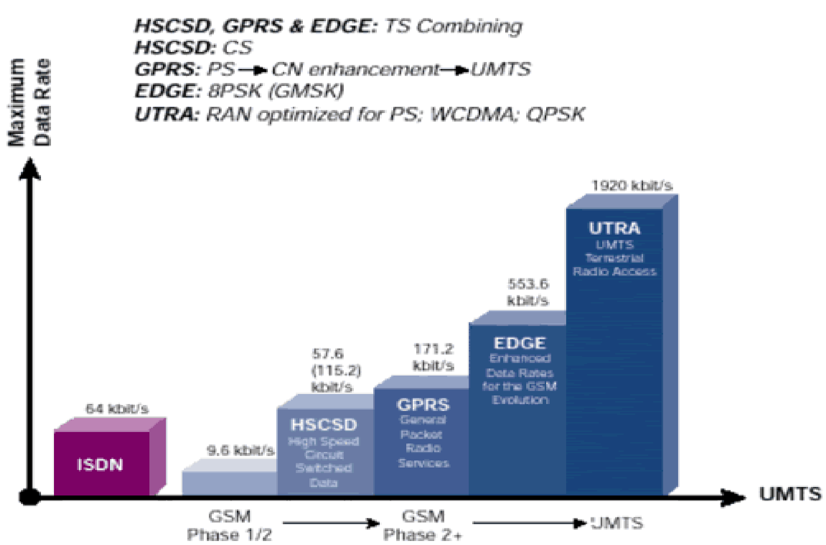

Figure 2.1 The Transmission Rates evolution

The first digital cellular system, the GSM offered a data rate of 9.6 Kbps knowing that the access scheme applied was a combination of FDMA and TDMA. Further technical

improvements were introduced to the existing GSM system to reach an average data rate of 57.6 Kbps offered by HSCSD and with further technical improvements data rates of 171.2 and 553.6 Kbps were offered by GPRS and EDGE simultaneously.
Coming to the 3G system, the UMTS, where a different access scheme was applied which is the CDMA that enabled a data rate of 2Mbps.

2.3. UMTS Network Architecture

These specifications of the first release of UMTS (UMTS Release 99) relate to the addition of the UMTS Radio Access Network (**UTRAN**), which is typically added to circuit-switched voice infrastructure and GPRS "Internet access"[3GP99].
Taking a few steps back, a GSM network would logically look like as shown in Figure 2.2.

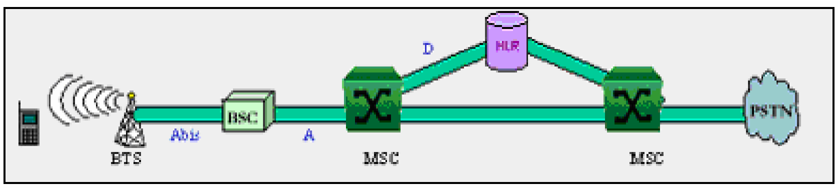

Figure 2.2 GSM Network

An addition of a GPRS overlay network would logically look like as shown in Figure 2.3 where an overlay for packet switched domain was added.

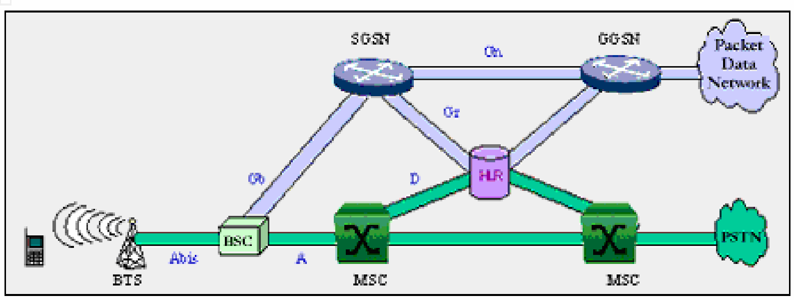

Figure 2.3 GSM/GPRS Network

This overlay network effectively increased the bandwidth of the core network to allow high-speed data transfer with an "always-on" connection. The restricting factor for end-to-end high-speed data transfers became the Radio Access Network.

The addition of UMTS R99 effectively adds a new "front-end" high-speed network to the voice and data networks, removing the last major "speed bump". After the addition of UMTS R99 a wireless network looks like as shown in figure 2.4 [3GP99].

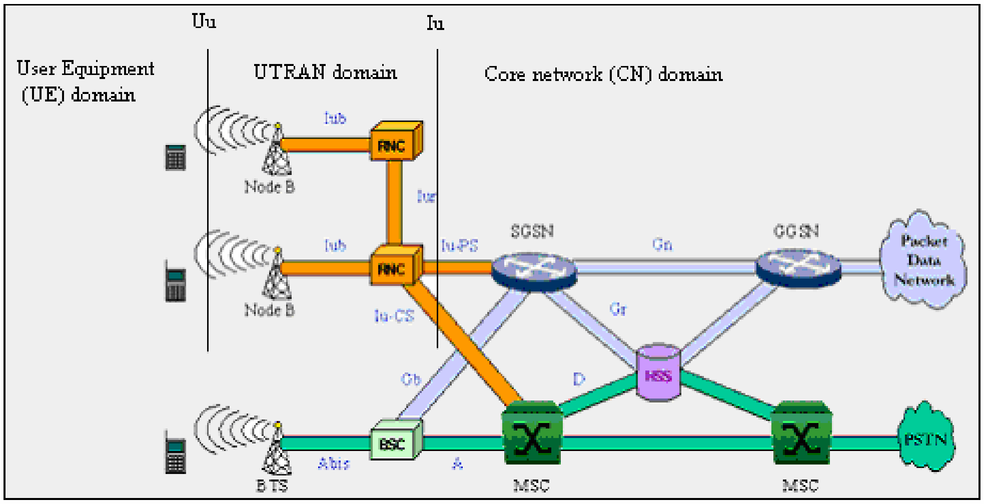

Figure 2.4 UMTS Release 99 Network

The UMTS network architecture (Release 99) consists of three domains: The User Equipment (UE) domain, the UMTS Terrestrial Radio Access Network (UTRAN) domain and the Core Network (CN) domain.

The UE domain represents the equipment used by the user to access UMTS services while the UTRAN domain and the CN domain, together known as the infrastructure domain, consist of the physical nodes which perform the various functions required to terminate the radio interface and to support the telecommunication services requirements of the user.

The main reason why the UTRAN can offer a high-speed connection compared to the GSM/GPRS is in the air interface mechanisms. In GSM/GPRS networks, modulation schemes known as Time Division Multiple Access (TDMA) and Frequency Division Multiple Access (FDMA) are used. In UMTS networks, the modulation scheme is known as Wide-band Code Division Multiple Access (WCDMA), which has two basic modes of operation: Frequency Division Duplex (FDD) and Time Division Duplex (TDD). These UMTS modulation schemes are inherently more efficient than their GSM/GPRS counterparts, which in turn enable faster connections [Hol02].

The new network elements introduced as part of UMTS R99 [3GP99] are:
•the Radio Network Controller **(RNC**),
•the Node B, and
•the UMTS UE.

The UMTS UE is based on the same principles as the GSM MS. It consists of the Mobile equipment (ME) and the UMTS subscriber identity module (SIM) card (USIM).

The UTRAN domain handles all radio-related functionality. It consists of one or more Radio Network Sub-systems (RNS) where each RNS consists of one or more Node Bs and one Radio Network Controller (RNC).

Radio Network Controller (RNC)

The RNC in UMTS networks provides functions equivalent to the Base Station Controller (BSC) functions in GSM/GPRS networks. The major difference is that RNCs have more intelligence built-in than their GSM/GPRS counterparts. For example, RNCs can autonomously manage handovers without involving MSCs and SGSNs. This was something not possible using standard BSCs in GSM/GPRS networks.
The RNC is responsible for control of the radio resources in its area providing central control for the RNS elements (RNC and Node Bs). One RNC will control multiple Node Bs.

The RNC handles protocol exchanges between Iu, Iur, and Iub interfaces and is responsible for centralized operation and maintenance (O&M) of the entire RNS with access to the OSS. The user’s circuit-switched and packet-switched data coming from Iu–CS and Iu–PS interfaces are multiplexed together for multimedia transmission via Iur, Iub, and Uu interfaces to and from the UE.
The RNC uses the Iur interface, which has no equivalent in GSM BSS, to autonomously handle 100 percent of the RRM (Radio Resource Management), eliminating that burden from the CN. Serving control functions such as admission, RRC(Radio Resource Control) connection to the UE, congestion and handover/macro diversity are managed entirely by a single serving RNC (SRNC).

Node B

The Node B, also known as a Base Station and equivalent to the Base Transceiver Station (BTS) from GSM, converts the signals of the radio interface into a data stream and forwards it to the RNC over the Iub interface. In the opposite direction, it prepares incoming data from the RNC for transport over the radio interface.
Depending on sectoring (omni/sector cells), one or more cells may be served by a Node B.
The main task of Node B is the conversion of data to and from the Uu radio interface, including forward error correction (FEC), rate adaptation, W–CDMA spreading / despreading, and quadrature phase shift keying (QPSK) modulation on the air interface. It measures quality and strength of the connection and determines the frame error rate (FER), transmitting these data to the RNC as a measurement report for handover and macro diversity combining. The Node B is also responsible for the FDD softer handover.

New Interfaces

The new interfaces added as part of R99 are (figure 2.5):

- **Iu-CS**: This is the circuit-switched connection for carrying (typically) voice traffic and signaling between the UTRAN and the core voice network. The equivalent interface in GSM/GPRS networks is the A-interface.

- **Iu-PS**: This is the packet-switched connection for carrying (typically) data traffic and signaling between the UTRAN and the core data GPRS network. The equivalent interface in GSM/GPRS networks is the Gb interface.

- **Iur**: The primary purpose of Iur is to support inter-MSC mobility. When a mobile subscriber moves between areas served by different RNCs, the mobile subscriber's data is now transferred to the new RNC via Iur. The original RNC is known as the Serving RNC and the new RNC is known as the Drift RNC. There is no equivalent interface in GSM/GPRS networks.

- **Iub**: This is the interface used by an RNC to control multiple Node B's. The equivalent interface in GSM/GPRS networks is the Abis interface. The Iub interface is in the main standardized and open, unlike the Abis interface in GSM/GPRS.

- **Uu**: This is the interface between the User Equipment and the network. That is, it is the UMTS air interface. The equivalent interface in GSM/GPRS networks is the Um interface.

2.3.1. The Core Network Domain

The core network can be divided into three domains which are the Circuit Switched, the Packet Switched and the common Circuit Switched /Packet Switched domain [Hol02].

•The Circuit Switched domain

This domain is used for the access of services in circuit mode. It contains:

-The Mobile Services Switching Center (MSC)
The MSC performs the telephony switching functions of the system. It controls calls to and from other telephone and data systems.
-The Gateway Mobile Services Switching Center (GMSC)
A Gateway is a node interconnecting two networks. The GMSC is the interface between the mobile cellular network and the PSTN. It is in charge of routing calls from the fixed network towards a GSM user. The GMSC is often implemented in the same machines as the MSC.
-Visitor Location Register (VLR)
The VLR contains information from a subscriber's HLR necessary in order to provide the subscribed services to visiting users.

•The Packet Switched domain

This domain is used for the access of services in packet mode. It contains:

-The SGSN (Serving GPRS Support Node)
It has the same functions as MSC/VLR but it is used for packet communication. It is responsible for the delivering of data packets from and to the MN’s within its geographical service area .Its tasks include packet routing and transfer, mobility management, authentication and charging functions.

-The GGSN (Gateway GPRS Support Node)
It has the same functions as the GMSC but it is used for packet communication. It is the gateway between an external Packet Data Network (PDN) and the GPRS core network. In the case of an external IP network, the GGSN is seen as an ordinary router serving all IP addresses of the MS(s).Additionally, the GGSN stores the current SGSN addresses of the user and his or

her profile in its location register. The GGSN also deals with session management and charging functions.

-The Short Message Service Gateway MSC (SMS-GMSC) and Short Message Service Interworking MSC (SMS-IWMSC)
They are connected to the SGSN via the Gd interface to enable the SGSN to support SMS.

-The Charging Gateway Functionality (CGF)
It provides a mechanism to transfer charging information from the SGSN and GGSN to the network operator's chosen Billing Systems.

• The common domain PS/CS

-Home Location Register (HLR)
The HLR is considered as a very important database that stores information of the subscribers belonging to the covering area of a MSC.

-The EIR (Equipment Identification Register)
The EIR is also used for security purposes. It is a register containing information about the valid mobile equipments.

-The AUC (Authentication Center)
The AUC register is used for security purposes. It provides the parameters needed for authentication of the mobiles and encryption functions [Hol03].

2.4. The UMTS functions

The description of the UMTS network is focused on the different functions to fulfill by the network and not on its physical components. In UMTS, five main functions can be defined: [Hol02]

- Transmission,
- Radio Resources Management (RR),
- Mobility Management (MM),
- Communication Management (CM), and
- Operation, Administration and Maintenance (OAM).

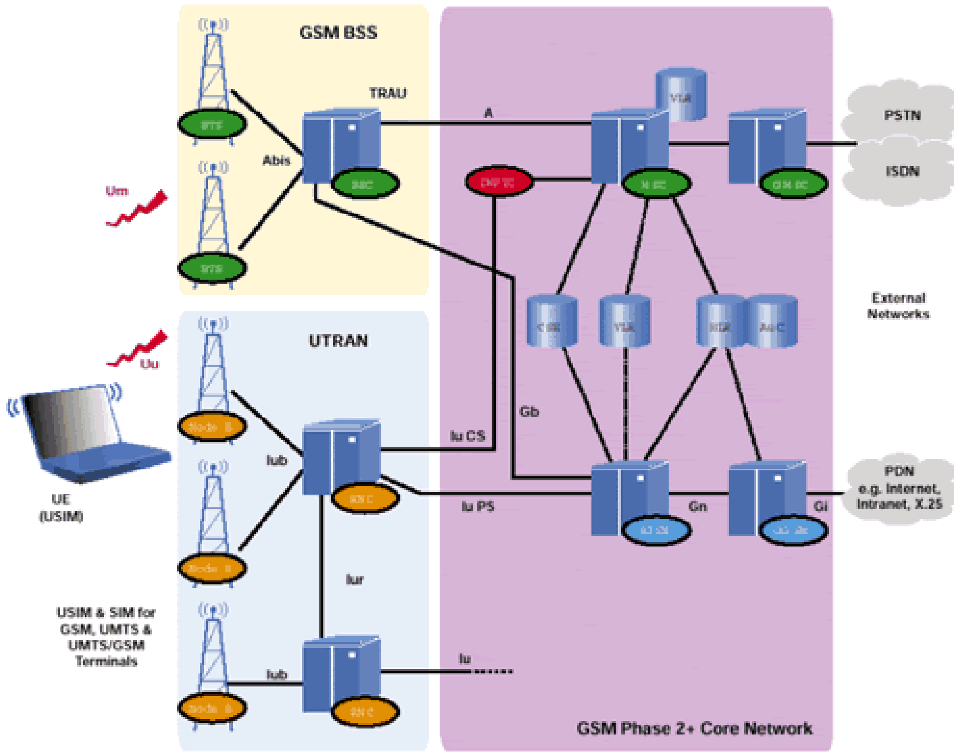

Figure 2.5 Interworking of UMTS and GPRS

2.5. UMTS Protocol Stack

The UMTS protocol stack differs from one interface to another depending on the role and the function of each interface. It is divided into User and Control planes which share the same Transport plane as shown in Figure 2.6.

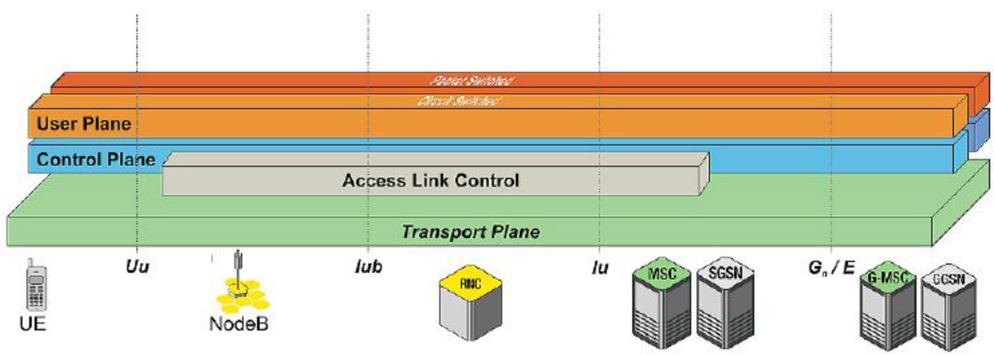

Figure 2.6 An Abstract View at the UMTS Protocol Stack

The **Physical Layer** offers information transfer services in the form of transport channels .In the physical layer all the signal processing functions, channel coding , interleaving, modulation, spreading, synchronization, etc. are performed. Its role also is the mapping of transport channels onto physical channels.

The **Data Link Control Layer** is divided into the following sub layers:

- Medium Access control (MAC) sublayer,
- Radio Link Control (RLC) sublayer,
- Packet Data Convergence Protocol (PDPC) sublayer, and
- Broadcast Multicast Control (BMC) sublayer.

The **MAC** sublayer performs the data transfer services on logical channels, which are defined with respect to the type of information which is transferred on them.

The **RLC** sublayer provides error correction by retransmission (e.g. Selective Repeat, Go Back N, or a Stop-and-Wait ARQ) in acknowledged data transfer mode, and is also responsible for segmentation and assembly of user data, controls the appropriate sequence of data blocks and ensures avoiding block duplication.

The **PDCP** sublayer provides transmission and reception of network Protocol Data Units (PDUs) in acknowledged or unacknowledged and transparent RLC modes.

The **BMC** sublayer performs broadcast and multicast transmission service in transparent or unacknowledged mode[Hol02].

Finally the lowest sublayer of the Network Layer is the **Radio Resource Control (RRC)** sublayer. The **RRC** sublayer fulfils the following functions: broadcasting of system information, radio resource handling, control of requested quality of service, and measurement reporting and control.

2.5.1. The User planes

As defined previously, it is the plane in which all user data such as encoded voice or packet data are transported. It includes the Data Stream(s) and the Data Bearer(s) for the Data Stream(s). Figure 2.7 shows the User plane for Packet Switched Services [Hol03].

The highest layer “application” is located at the top of the mobile station’s protocol stack. Common applications include those based on the Internet Protocol (IP) such as Hypertext Transmission Protocol (HTTP) over the Transmission Control Protocol (TCP). GPRS maintains a logical link, over which user data packets are transmitted. In the GSM recommendations, two common packet types (IP and X.25) are explicitly mentioned. The GPRS task is to accept the user data packet at one access point (the GGSN), and deliver it to another access point (the GPRS mobile station). The following paragraphs offer a closer look at the interfaces and protocol layers affected by a user data packet’s transport from the external PDN to the mobile [3GP99].

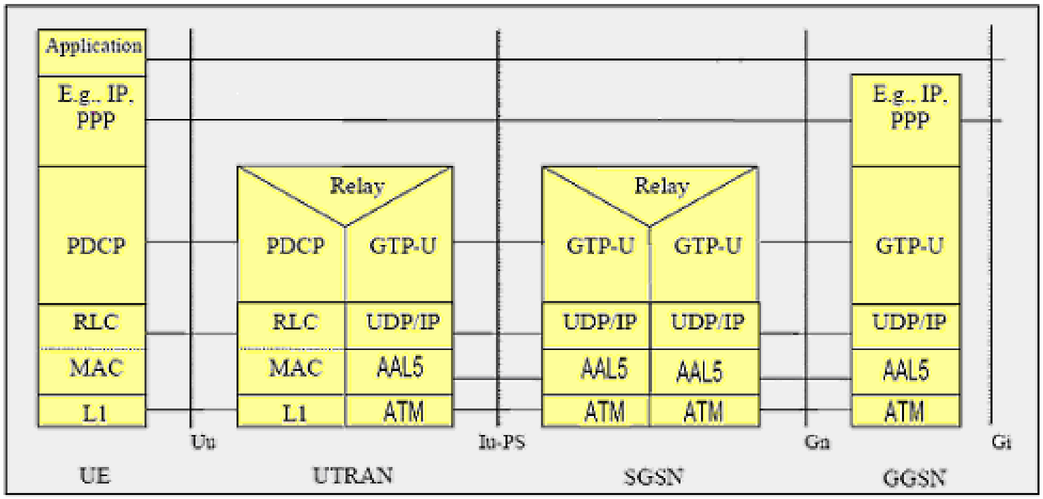

Figure 2.7 User Plane for packet switched services

2.6. Radio Resources Management (RR)

The role of the RR function is to establish, maintain and release communication links between mobile stations and the different switching centers of the heterogeneous network.

The elements that are mainly concerned with the RR function are the mobile station and the Node B or BTS. However, as the RR function is also in charge of maintaining a connection even if the user moves from one cell to another. Thus handovers are concerned with the RR functions.

The RR is also responsible for the management of the frequency spectrum and the reaction of the network to changing radio environment conditions. Some of the main RR procedures that assure its responsibilities are:

- Channel assignment, change and release,
- Handover,
- Frequency hopping,
- Power-level control,
- Discontinuous transmission and reception,
- Timing advance.

2.7. Mobility Management

The Mobility Management (MM) functions in a GPRS / UMTS network are responsible for registering a MS/UE with the network. This naturally implies that the MM also handles functions relating to user identification, user authentication and subscriber management. The MM is also responsible for maintaining the current location of the MS in the GPRS PLMN [3GP99].

The information is present in what is known as a MM context that is created in the MS and the SGSN in a PS connection whereas it is created in the MS and the MSC/VLR in a CS connection.

MM performs the following procedures:
• Attach Function,
• Detach Function,
• PDP contexts Function,
• Location Management Function,
• Handover Management.

Attach function
The attach procedure is always initiated by the UE. It starts when the user turns on the UE. When the UE finds a UE to attach into, it synchronizes with it and then attempts to attach by sending an attach request to the network (RNC). After an authentication procedure, the network responds by accepting the UE and changes its state into connected.

Detach function
When the UE no longer requires the services of the UMTS network, it can explicitly move to the network detachment state and detach from the network by sending a detach request. Network detachment can also be initiated by the network.

PDP context functions
To set up a connection with a PDN (public data network which is usually the internet), a PS connection has to be established. The UE activates the PDP context in the GGSN. A PDP context is a range of settings that defines which data networks a user may use for exchanging data. The list of permitted PDP contexts is stored in the HLR.

Location management function
Location management is the mechanism of keeping track of a user's location outside an active connection. To transfer an incoming connection to an inactive user, the network must continuously be up-to-date with the user's location.

Handover management function
Handover management is the mechanism of handing over an active connection from one cell to another.

To understand the dynamics of UMTS, it is useful to consider the different usage scenarios. The usage scenarios describe how the different parts of the UMTS network interact before, during and after communication. The usage scenarios for UMTS are found based on the UE service states [Hol02].
The UE exists in three service states: detached, connected and idle. The UE is in the detached state when it is switched off and there is no communication between the UE and the network. The UE cannot send or receive anything. In order for the user to make use of the network the UE needs to attach to the network by switching on, selecting a cell to which it can attach, and attaching to that cell. When the UE is attached to the network, it moves to the connected state and starts communication, or moves on to the idle state and becomes inactive.

The states differ depending on whether the UE is in CS or PS mode. Thus there are six important usage scenarios to consider: network attachment, CS connection, CS idle, PS connection, PS idle and network detachment. For UMTS PS service domain, these states are renamed as PMM-DETACHED, PMM-IDLE and PMM-CONNECTED, respectively.

UMTS proved to be the most important proposal of IMT-2000 as the successor of GSM due to its compatibility with the existing GSM and GPRS systems and its satisfaction of the 3G standards. It is a 3G network that delivers, among other capabilities, improved system capacity and spectrum efficiency versus 2G systems. It supports data services at transmission rates of at least 144 Kbps in mobile (moving) environments and at least 2 Mbps in fixed (indoor) environments and assures Global coverage.

Therefore, UMTS is the natural evolutionary choice for operators of GSM networks, currently representing a customer base of more than 850 million end users in 195 countries and representing over 70% of today's digital wireless market. As discussed in chapter one WLAN and UMTS proved to be complementary systems thus their integration would be a wise choice, but before discussing the integration, WLAN will be interpreted in the next section.

2.8. WLAN overview

A WLAN -also referred to as IEEE 802.11 standard- is an on-premise data communication system that reduces the need for wired connections and makes new applications possible, there by adding new flexibility to networking. Mobile WLAN users can access information and network resources as they attend meetings, collaborate with other users, or move to other campus location, but the benefits of WLANs extend beyond user mobility and productivity to enable portable LANs. WLANs have proven their effectiveness in markets and are now experiencing broader applicability in a wide range of business settings. Wireless LANs offer the following productivity, service, convenience, and cost advantages over traditional wired networks. In general WLANs have the following advantages [Cro97]:

- Mobility
- Installation Speed and Simplicity
- Installation Flexibility
- Reduced Cost-of-Ownership
- Scalability

2.8.1. Infrastructure-based WLAN

In infrastructure WLANs, multiple **access points (APs)** link the WLAN to the wired network (figure 2.8) and allow users to efficiently share network resources.
The access points not only provide communication with the wired network but also mediate wireless network traffic in the immediate neighborhood. Multiple access points can provide wireless coverage for an entire building or campus. In fact they work like base stations in cellular systems.
The 802.11b WLAN network architecture basically consists of one or more Basic Service Sets (BSSs) and a Distribution System (DS) [Wal01].

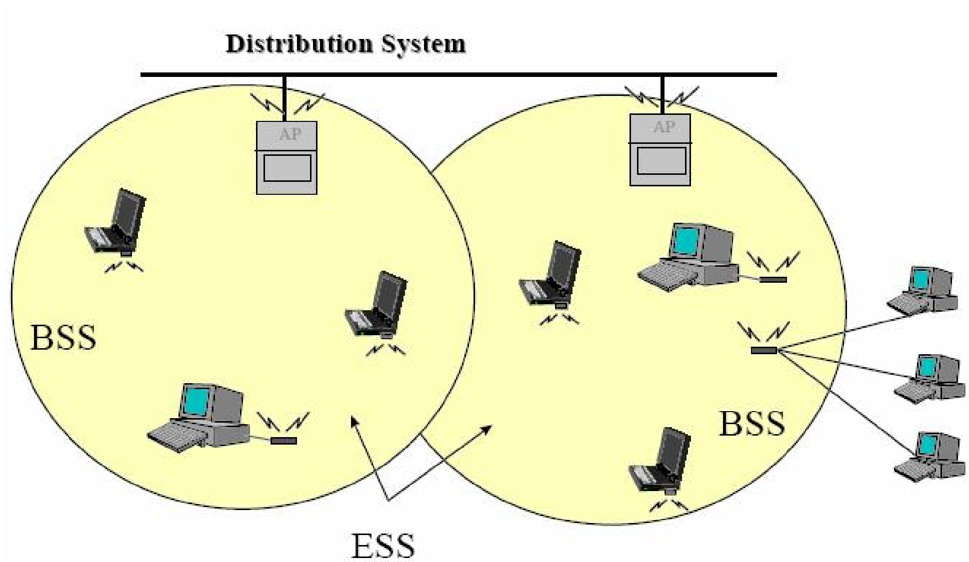

Figure 2.8 Infrastructure WLAN

The basic part of the network architecture is the BSS. It consists of a group of Stations (STAs) that are under direct control of a single coordination function. The STAs are computing devices (laptops, handheld computers etc.) with wireless network interfaces. The STAs communicates through a wireless medium. The geographical area covered by the BSS is known as the Basic Service Area (BSA). The BSA is analogous to a cell in the UMTS network. In an infrastructure network all STAs communicate by channeling all traffic through a centralized Access Point (AP). The AP controls the communication in the BSS as well as providing network connectivity between other BSSs and thus has a bridging function. The AP is analogous to the Node B in the UMTS network [Wal01].

A common distribution system (DS) integrates multiple BSSs. The DS does not specify any particular backbone technology and can be wired to a wide range of mediums. The integration of multiple BSSs using a DS is called an Extended Service Set (ESS). The ESS provides not only access for multiple wireless users but also gateway access for wireless users into a wired network such as the Internet.

2.9. The 802.11 protocol stack

A protocol view of the 802.11 protocol stack is given in Figure 2.9 [Cro97]. The physical layer corresponds to the OSI physical layer fairly well, but the data link layer is split into two or more sublayers. The MAC (medium access control) sublayer determines how the channel is allocated, that is, who gets to transmit next. Above it is the LLC (logical link control) sublayer whose job is to hide the differences between different 802.11 variants and make them indistinguishable as the network layer is concerned.

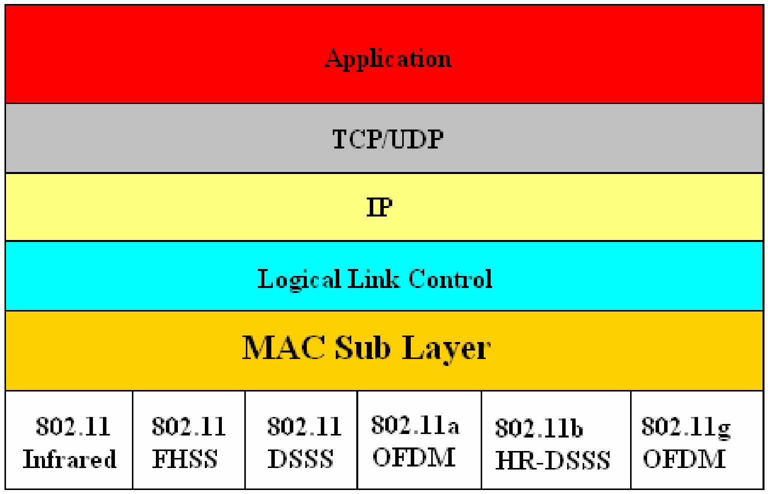

Figure 2.9 802.11 protocol stack

2.9.1. The 802.11 Physical Layer

The 1997 802.11 [Cro97] standard specifies three transmission techniques allowed in the physical layer. The infrared method uses much the same technology as television remote controls do. The other two use short range radio, using techniques called **FHSS** (Frequency Hopping Spread Spectrum) and **DSSS** (Direct Sequence Spread Spectrum). All of these techniques provides 1 or 2 Mbps and at enough low power that they do not conflict too much. Later, two new techniques were introduced to achieve higher data rates, these are **OFDM** (Orthogonal Frequency Division Multiplexing) and **HR-DSS (high rate direct sequence spread spectrum)**. They operate up to 54 and 11 Mbps respectively.

2.9.2. The 802.11 MAC sublayer protocol

802.11 supports two modes of operation. The first, called **DCF (Distributed Coordination Function)**, does not use any kind of central control. The other, called **PCF (Point Coordination Function)**, uses the base station to control all activity in its cell. All implementations must support DCF but PCF is optional.

2.10. Mobility management

The WLAN specification handles mobility management in a very simple way. There are no distinct definitions of location management and handover management but to illustrate the differences to UMTS mobility management this section looks more into how WLAN actually deals with location management and handover management.

The WLAN location management is quite different from UMTS location managemt in the sense that when a STA has first associated with an AP, the AP continually knows the location of the STA, as the STA is required to be within the reach of the AP, and no further mechanisms are required in order for the AP to be able to forward frames destined for the STA [Wal01].

2.11. Conclusion

This chapter presents a brief overview on UMTS and WLAN which are considered to be the complementary access technologies to be integrated in NGN.

The next chapter will present the different existing integration architecture proposed and the different mobility management protocols accompanying the different integration proposals.

Chapter 3

Integration architecture

3.1. Introduction

Next generation access network will be based on various radio access systems such as UMTS and WLAN. In the UMTS network, the packet switched data network is based on GPRS network which is capable of providing data transmission with medium speed over wide area, supporting highly mobile users. From the other side, WLAN can offer high data rates in small geographical areas or hot spots, and are expected to be widely deployed in the next generation networks. The handover between these two access technologies also known as vertical handover, should not only consider seamless mobility but also resource availability and seamless service continuity in the new access network.
In this chapter, the integration architectures between heterogeneous access networks, such as WLAN and UMTS are considered under different mobility management protocols such as MIPv4, MIPv6,… After reviewing the most important standardization activities for inter-working between WLANs and cellular networks, we present a comparison of the different existing architectures for WLAN/UMTS integration.

3.2. Challenges and requirements

The future wireless mobile generation will be based on pure IP- network leading to a global interconnection and integration between the various telecommunications networks. This will require new integration architectures and mobility protocols to provide seamless roaming and QOS provisioning
UMTS and WLAN are viewed as the future complementary access technologies. While third-generation (3G) cellular systems promise competitive data rates, at speeds of up to 300 kb/s initially and increasing up to 2 Mb/s, with the same always-on connectivity of wired technology, mobile network operators are turning to wireless local area network (WLAN) technology. This interest in WLAN technology raises from the rapid evolution and successful deployment of WLAN systems worldwide, as well as the very high data rates (in excess of 100 Mb/s) [Ana00].

Both WLAN and 3G are capable of providing higher-speed wireless connections that cannot be offered by earlier cellular technologies [Axi04]. However, each technology has significant market applications. WLANs can cover only small areas and allow limited mobility, but provide higher data rates. Therefore, WLANs are well suited to hotspot coverage, where there is a high density of demand for high-data-rate wireless services requiring limited mobility. On the other hand, 3G wireless systems, with their wide coverage, and high mobility, are more suited to areas with moderate or low-density demand for wireless usage requiring high mobility .In that sense; the two wireless networks are complementary.

A multimode 3G/WLAN terminal can access high-bandwidth data services where WLAN coverage is offered, while accessing wide area networks using 3G at other places.

However, this approach alone will only allow limited multi-access functionality. To make multi-access solutions effective, we need an integrated solution providing seamless mobility between access technologies, allowing continuity of existing sessions, integrated authentication, integrated billing, terminal mobility, and service mobility. Thus the following requirements are to be considered [Axi04]:

- ***Common billing:*** The simplest form of interworking should provide a common bill to the subscriber.
- ***3GPP system-based access control:*** authentication, authorization, and accounting (AAA) for subscribers in the WLAN and UMTS to be mutually recognized.
- ***Access to 3GPP GPRS-based services:*** The goal of this scenario is to allow the cellular operator to extend access to its services to subscribers in a WLAN and vice versa.
- ***Service continuity:*** this consists of maintaining service continuity across the different access technologies.
- ***Seamless services:*** it consists of providing service continuity between the access technologies in a transparent manner with respect to the end user.
- ***Authentication and privacy*:** In an integrated environment, where the subscriber has multiple access network options, there are two possible configurations relating to the integration of WLAN and 3G networks:

    1. The first is the case in which the 3G operator owns the WLAN. In operator owned WLANs, the operator has authentication in place for their users, which they can leverage in the WLAN space.

    2. The second is the case in which the Wireless Internet Service Provider (**WISP**) or enterprise is the provider of the WLAN. When the 3G operator does not own the WLAN, the billing and authentication would still be provided by the 3G operator. In an enterprise WLAN, the enterprise may choose its own authentication and billing system.

3.3. Seamless roaming and Mobility

The issue of session mobility and seamless roaming is extremely important. Ideally, no user intervention would be required to perform the switchover from WLAN to UMTS. Moreover, the user would not perceive this handover. When the user moves back into the coverage of a WLAN system, the flow is handed back to the WLAN network. The mobility function is distinct from roaming in that Mobility requires no user intervention and preserves any IP-based session during handovers between cellular data and WLAN. With roaming, the switchover between WLAN and cellular data requires explicit user intervention and in most cases would result in teardown of any existing sessions [ETS01].

There are a few different strategies to solve the problem of session mobility. At the network layer **MIP** (Mobile IP) from the **IETF** (Internet Engineering Task Force) provides a scheme for host mobility, including support for TCP and UDP connections. MIP uses agents (home agents and foreign agents) that assign care-of-addresses to mobile hosts so that these mobile hosts can be accessed from a foreign network [Per02]. Another approach to network mobility is **SIP** (Session Initiation protocol). Unlike MIP, SIP is an application layer protocol that allows for a host to move networks without interrupting a session. In this scenario, a mobile host receives a new address when joining the new network and then sends an invitation to restart the session to its corresponding host [Ros02]. These schemes work well for horizontal handoffs (i.e. between adjacent hotspots that act as foreign subnets) but they can have long handover latency, causing existing sessions to timeout. This situation is not ideal for the vertical (i.e. between heterogeneous networks) handoff we require from WLAN to 3G.

Another scheme that can address the problem of vertical handoff and session mobility is the use of SCTP (Stream Control Transmission Protocol). This protocol was designed for use with VoIP (Voice over IP) and has been specified by 3GPP (3rd Generation Partnership Project) to carry all signaling traffic on UMTS [3GP99]. One of its attractive features is known as multi-homing. This enables packets to be transmitted over multiple interfaces identified by multiple IP addresses [Ste00]. SCTP will send packets to a destination IP address designated the primary address like TCP and UDP, but has the ability to redirect packets to an alternative IP address if the primary IP address becomes unreachable – say during vertical handover. Thus we can use multi-homing when performing the handover between two heterogeneous networks, by allowing our host to have multiple IP addresses associated with one or more networks. If the QoS (Quality of Service) declines during handover, packets can be rerouted to the alternative secondary addresses that will still refer to our host [Ste001].

3.4. Existing integration architectures

Several approaches have been proposed for inter-working between WLANs and cellular networks depending on how much inter-dependence is required between the two wireless networks.

The European Telecommunications Standards Institute **(**ETSI**)** specifies in [ETS01] two generic approaches for inter-working: called loose coupling and tight coupling.
With loose coupling, the WLAN is deployed as an access network complementary to the UMTS network. In this case, the WLAN utilizes the subscriber databases in the UMTS network but features no data interfaces to the UMTS core network. Considering the simplified UMTS reference diagram displayed in Figure 3.1, we may argue that the loose coupling between the UMTS and the WLAN is carried out at the Gi reference point. This means that with loose coupling the WLAN bypasses the UMTS network and provides direct data access to the external packet data networks (PDN**s**).

On the other hand, with tight coupling the WLAN is connected to the UMTS core network in the same manner as any other radio access network (RAN), such as UMTS RAN and UMTS terrestrial RAN (UTRAN). In this case, the WLAN data traffic goes through the UMTS core network before reaching the external PDNs. As shown in figure 3.1, with tight coupling the WLAN is connected to either Gb or Iu-ps reference points.

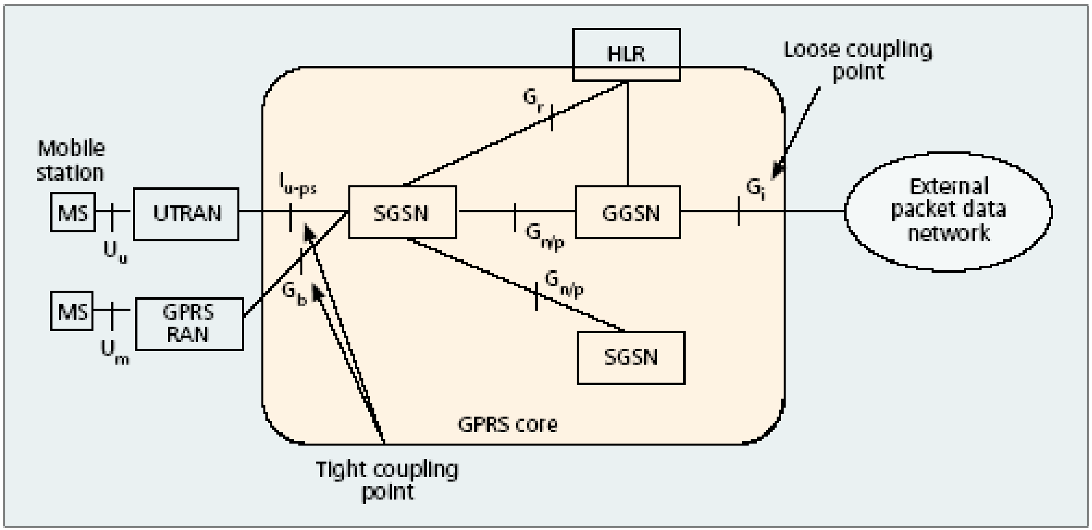

Figure 3.1 Loose and tight coupling.

Many architectures were proposed to ensure the interoperability between WLAN and GPRS, relying on MIPv4 as the mobility protocol for vertical handoff aiming to provide seamless roaming between different access technologies [Tsa02]. MIPv4 has three main functions consisting of mobile agent discovery, registration with the Home Agent, and delivery of datagrams using tunneling to the mobile node through the Foreign Agent FA [Per02]. In this case, the foreign agent FA in GPRS network resides in the GGSN, FA in the WLAN reside in an access router and the HA is located the access router of another WLAN network where the operator's IP network resides.

3.4.1. Loose coupling

The loose coupling architecture approach to the integration of WLAN and UMTS systems involves minimal interaction between the two networks. The only major requirement in this architecture is that both systems have access to a shared database of subscriber details that will be used for AAA functions (Authentication, Authorization, Accounting functions).

Figure 3.1 shows the loosely coupled architecture. Notice that UMTS originating traffic passes through the node-B, RNC, SGSN, GGSN and terminates in the internet. When the User moves to a WLAN, the traffic passes directly to the Core IP network (internet) through the access routers. Thus in the loose coupling approach, the WLAN is effectively provided as an access network complementary to the cellular data network. This means that only the signaling traffic travels through the core UMTS network and not the data traffic.

If you consider the scenario where a cellular operator provides access to WLAN hotspots as part of its service to the customer: the customer has entered a hotspot area with his
3G / WLAN dual compatible device; when he connects to the WLAN, the latter will access the customer's details as stored on the operator's database of subscriber details. This allows the user to be authenticated and billed properly [ETS01]. The IETF (Internet Engineering Task Force) is responsible for the security and mobility issues (e.g. Mobile IP) in this scenario as the user is effectively connecting directly to Internet via the WLAN and not the UMTS network. The WLAN data traffic does not pass through the GPRS core network but it goes directly to the operator's IP network (and/or Internet). It is coupled with the GPRS network in the operator's IP network. The cellular access gateway (CAG) acts as an authenticator for WLAN users. It interacts with the HLR similar to the way SGSN interacts with the HLR. CAG provides the AAA server functionality in the cellular operator's IP core. CAG interacts with the home location register (HLR) to obtain the authentication. When the MS moves from GPRS to WLAN, it performs a MIP registration via the FA that resides in the WLAN. The FA completes the registration with the HA, providing a care-of address to the HA to be used as a forwarding address for packets destined to the MS. The FA then associates the care-of address with that of the MS and acts as a proxy on behalf of the MS for the life of the registration. When the MS moves from WLAN to GPRS, the same process will be done with FA being in the GGSN node. An advantage of this scheme is that the user will have just one common bill from the cellular operator even though he accessed two separate services. Also the two heterogeneous network services (UMTS, WLAN) can be engineered as separate entities (as they were originally designed) and thus less time is spent developing complicated inter-working schemes between the two.

A drawback of this scheme is that two new entities are needed: The billing mediator and the CAG cellular access gateway.
The billing mediator must be developed to handle billing when a cellular subscriber uses the WLAN. This is more complicated especially when the WLAN is not owned by the cellular operator. For each WLAN that the cellular operator wishes to provide access to, an agreement must be made between the operator and the WLAN owners to facilitate correct billing. The CAG is necessary to reuse SIM based authentication against the HLR, a way of authentication that could improve functionality and ease of use.

This architecture has appeal for operators (especially in the short term) because of its relative simplicity and ease of implementation, both financially and technically, making it the best choice for the 3G operators [ETS01].

3.4.2. Tight Coupling

In this scenario, the 802.11 network is an integral part of the 3G network, thus representing an alternative or complementary Radio Access Network (RAN) to the pre-existing cellular one. Essentially the 802.11 network will appear to the 3G core network as another 3G access network. In this way, the customer is allowed to access both UMTS and external services via their WLAN interface.

When the user is covered by both technologies, the customer will be able to use the most preferable radio technology as demanded for his/her requirements. Using this solution implies the adoption of user terminals implementing common 3G features. It will also be required to be equipped with different radio technologies (UMTS and 802.11) to access the wireless medium.

For UMTS, the **IWU** inter-working unit network element will appear as an SGSN [Bud03]. This gateway will hide the details on the 802.11 network to the 3G core and will also implement all the 3G protocols required in a 3G radio access network. This can be seen clearly from figure 3.2.

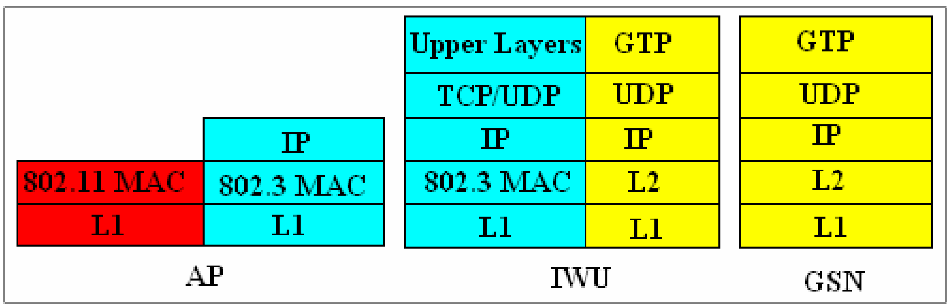

Figure 3.2 Tight coupling- IWU

In general, the tight coupled network architectures provide an approach to solving the problem of integrating 802.11 WLAN and 3G with the following characteristics [ETS01]:

- Seamless service continuation across WLAN and 3G. Users should be able to maintain their data sessions as they move between the WLAN and 3G network
- Reuse of the 3G AAA
- Reuse of the 3G infrastructure (e.g. Subscriber databases, billing systems etc.)
- Increased security since 3G authentication can be applied onto of the WLAN authentication
- Common provisioning and customer care
- Access to core 3G services such as SMS and MMS [Sal02] .

This architecture, depending on the pre-existing architecture of the 3G cellular system, leads to the necessity that the WLAN to the 3G core networks are to be owned by the same operator. This is a major drawback, which puts a strong limitation on the actual commercial availability of this kind of integration. Besides, while loose coupling approach uses MIPv4 to provide session mobility across GPRS and WLAN, Tight coupling approach uses GPRS mobility management for session mobility.

3.4.3. Other integration scenarios

The mobile SCTP (mSCTP) is a transport layer protocol similar to Transmission Control Protocol (TCP) that operates on top of the unreliable connection-less packet network. It provides unicast end- to-end communication between two or more applications running in

separate hosts and offers connection-oriented, reliable transportation of independently sequenced message streams [Rie02].
The mSCTP scenario starts with a client located in its home network at location A where it has established communication with a server in a foreign network through the Internet. Figure 3.3 illustrates the mSCTP connection initialization [Rie02].

During ongoing communication between the client and server, the client initially in the cellular network detects a weak WLAN signal. As the client moves into the coverage area of the WLAN network, it receives an IP address WLAN_IP from the local space either by contacting DHCP or by IPv6 address auto-configuration. The client is now able to establish a link with the server with its second IP address and thus become multi-homed, i.e. reachable by the way of two different networks.

•The client therefore tells the server via the first link using ASCONF (Address Configuration) messages that it is reachable by a second IP address (1). In other words, it adds the newly assigned IP address to the association identifying the connection to the server.

•The server responds to this by returning an acknowledgment (2).

•As the client leaves the coverage area of the UMTS network, the client tells the server to set the newly assigned IP address to the primary IP address (3), which the server responds to with an acknowledgment (4).

•The new primary IP address WLAN_IP now becomes the destination address for further communication and the communication can continue unaffected over the new link.

•When the user moves from WLAN to UMTS, the client tells the server to set the UMTS_IP as the primary IP address (5), FS responds with an Ack (6) and the traffic between MC and FS is routed through UMTS.

•If the MC looses the signal from the WLAN cell, it starts the delete IP address process. The MC sends an ASCONF message with parameters set to “delete IP address” and WLAN_IP to request that the FS release the address WLAN_IP from its host routing table (7). After the MC receives an ACK from the FS (8), it deletes WLAN_IP from its address list, and WLAN_IP is released from the association [Rie02].

SIP is the IETF-developed signaling protocol Session Initiation Protocol (SIP) at the application layer [Ros02].

The adopted scenario is that the home SIP server is located in the WLAN network.
The SIP protocol starts with a client located in its home network (WLAN) at location A where it has established communication with a server in a foreign network through the Internet. Figure 3.5 illustrates the SIP connection initialization [Ros02].

•As the client attaches to the home WLAN network, its user agent sends a location update to the registrar in the home SIP server (1).

•The registrar processes the update message and forwards it to the location service, which stores the information. The home SIP server in return sends an acknowledgement (2).

•The client now wants to communicate with a server located in a foreign network, so its user agent sends an invite request to its home SIP server. The home SIP server recognizes that the request is not meant for it and forwards it to the SIP server belonging to the server's domain (3).

•The redirect server in the server's SIP server receives the request and consults the location service to find the location of the server. Most often the location service is able to find the address in the registration table. In some cases the location service can however only return an address of another redirect server. The location service returns the address to the redirect server, which returns it to the client's user agent (through its home SIP server) (4).

•The client's user agent confirms the response with an acknowledgement (5).
•The client's user agent now has the latest address of the server and is able to send its invite request to the server's user agent (6).

•Alternatively, instead of redirecting the request (4-6), a proxy server could forward the request to the server (7).

•The server's user agent acknowledges the request (8) regardless of it has gone through a redirect or a proxy server, and the two nodes can begin communicating (9).

The client now changes its position from location A in the home WLAN network to location B in a foreign UMTS network during the communication. To maintain the connection, SIP implements a two-fold location update.

•As the client moves from location A in the home WLAN network to location B in a foreign UMTS network during communication with the server (messages (1) and (2)), it must update its location.

•Its user agent therefore sends a location update to the home SIP server so that new invite requests can be redirected correctly (3).

•The registrar processes the update message and forwards it to the location service, which stores the information. The home SIP server responds with an acknowledgment (4).

•Then the client sends a new invite request to the server's user agent using the same call identifiers as in the original connection setup (5). The request contains the new address, which tells the server's user agent where it wants to receive future SIP messages.

•The server's user agent acknowledges the request (6) and the communication continues unaffected (7).

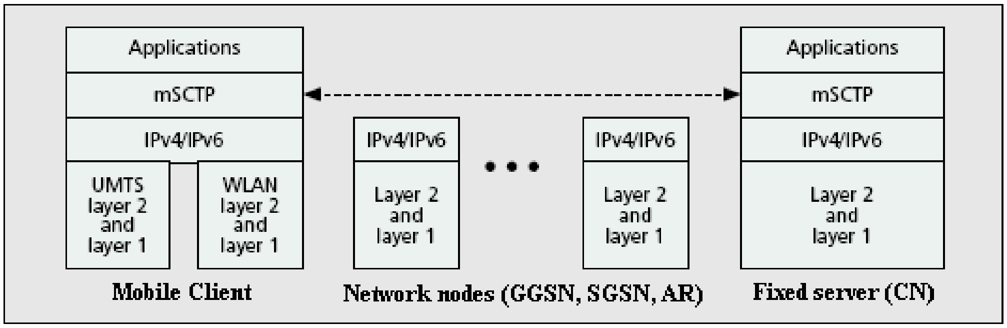

Figure 3.3 mSCTP interaction with the existing nodes

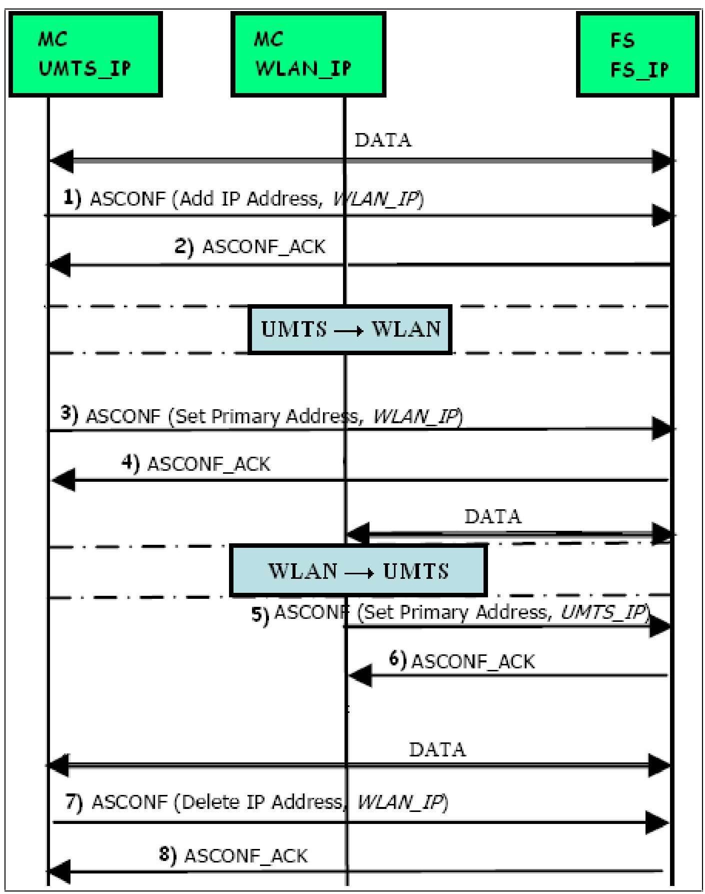

Figure 3.4 Signaling exchange for mSCTP

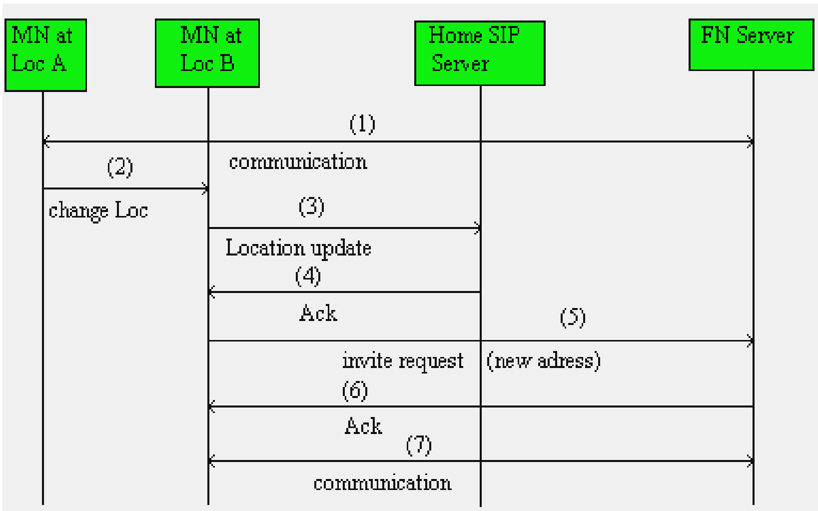

Figure 3.5 Signaling for SIP connection initialization

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3.5. Comparative study

Mobile IP has some shortcomings when it comes to delay-sensitive multimedia applications. The triangular routing adds handover delays and the tunneling overhead adds extra bytes to the packet header. It is more suited for long-lived TCP connections like telnet, ftp, etc. In comparison, SIP is much more suited for real-time communication over UDP. It is, however, less suited for TCP-based application. SIP is therefore often used either as a partially replacement for Mobile IP or as a complement, where SIP handles UDP connections and Mobile IP handles TCP connections.

With tight coupling the WLAN connects to the SGSN either via an already known interface or a new interface, specified for best possible performance with WLANs.
Tight coupling provides firm coupling between WLAN and 3G, and its main advantage is enhanced mobility across the two domains [Axi04]. Also, tight coupling offers reuse of 3G authentication, authorization, and accounting, (AAA) and protects the operator's investment by reusing the 3G core network resources, subscriber databases, billing systems, and so on. What is more, it can support 3G core services such as SMS and lawful interception for WLAN subscribers. Tight coupling is still mainly for WLANs owned by cellular operators and cannot easily handle third party WLANs. There is also some cost concerns related to tight coupling.
There are several advantages to the loosely coupled integration method. It allows the independent deployment and traffic engineering of 802.11 and 3G networks. 3G carriers can benefit from other providers' 802.11 deployments without extensive capital investments. At the same time, they can continue to deploy 3G networks using well-established engineering techniques and tools. Also, while roaming agreements with many partners can result in widespread coverage, including key hot-spot areas, subscribers benefit from having just one service provider for all network access. They no longer need to create separate accounts with providers in different regions, or covering different access technologies.

Finally, unlike the tightly coupled approach, loose coupling allows a wireless ISP to provide its own public 802.11 hot spot, interoperate through roaming agreements with public 802.11 and 3G service providers, or manage a privately installed enterprise Wireless LAN. The loosely coupled approach offers several architectural advantages over the tightly coupled approach, with hardly any disadvantage.

In general, the best choice of architectures should be decided by a couple of factors. When the wireless network architecture is composed of a number of WLAN operators and cellular operators then loosely coupled architecture would be better. If the cellular operator owns the WLAN then tightly coupled architecture would be better [Axi02].

3.6. Conclusion

In this chapter, we presented the existing integration architectures combining the naturally converged networks UMTS and WLAN. In addition, the different mobility management protocols accompanying these integration architectures were discussed.

A comparative study was given for the different solutions. But due to the fact that MIP is considered as the generic mobility management protocol, and that the integration architecture should be based on the current infrastructure with minimum change. We focused on the MIPv4 and MIPv6 and their extensions for mobility management in NGN proposed in the IETF framework to support micro mobility such as Fast Handover for Mobile IPv6. Thus, in the next chapter, FMIPv6 is described in the context of vertical handover between WLAN and GPRS, as a first step. The scenario of integration and the handoff process are described. The proposed anticipated vertical handover architecture is based on anticipated FMIPv6 mobility protocol in the context of WLAN to UMTS handovers, and vice versa. This will allow a seamless handover based on prior CoA acquisition and the FMIPv6 signaling messages done prior to HO.

Chapter 4

UMTS/WLAN Handover

4.1. Introduction

In this chapter, the mobility between the two network technologies WLAN and UMTS is considered in order to realize seamless mobility management in the loose coupled integration architecture. The requirements for handling handovers which include Terminal and Network requirements, as well as handover management are presented. The Anticipated Vertical Handover AVHO was proposed and tested in the context of mobility between WLAN and UMTS, the simulation results were compared to the generic mobile IP protocol MIPv6.

4.2. Handover Requirements

In order to perform handovers, there are a few basic requirements that must be in place. The requirements can largely be divided into two: terminal and network requirements.

4.2.1. Terminal Requirements

The mobile terminal, must be a dual-mode terminal set up for mobile radio communication and wireless networking in terms of incorporated USIM and 802.11b wireless LAN card, and signed up with one or more network operators. It should be able to operate on both networks and support handover between the two networks.

4.2.2. Network Requirements

The inter-working involves two possible types of ownership/management configurations: the cellular operator configuration and the wireless Internet service provider (WISP) configuration.

The first configuration is the case in which the cellular (UMTS) operator owns and manages the WLAN. The cellular operator can enhance its data service capabilities with high speed data connectivity in strategic locations such as airports and hotels by augmenting its cellular data system with WLAN. The cellular operator has authentication and billing mechanisms in place for its users, which can be reused in the WLAN [Pah00].

The second configuration is the case in which a WISP or enterprise owns and manages the WLAN. Irrespective of network ownership/management configuration roaming between the networks is main issue in next generation network.

4.3. Handover procedure

There are many parameters to take into account when considering vertical handovers. In the handover procedure, some measurements need to be performed according to the physical parameters of the link. Then a handover decision is taken based on the measurement report. Finally the handover is executed.

4.3.1. Handover measurements parameters

The first step of the handover procedure is the measurements of several parameters that are required to assess the current status of the existing connection between the client and the serving cell and of the quality of other available cells.
The measurements can in theory be performed by one of the two entities: the client or the network. The measurements are usually carried out continuously [Zha03].
The measurements include both some static user preferences and profiles and some dynamic measurements. The static and dynamic parameters are divided into four classifications: application, user preference and user context, terminal, and network.

4.3.2. Handover Decision

Based on the measurement report from the measuring entity, the handover decision entity decides whether a handover is required or not. The handover decision can be made by one of the two entities: the client (mobile terminal-controlled mode) or the network (network-controlled mode) [Wan99].

4.3.2.1. Mobile terminal-controlled mode

The client measures the signal strength and quality from the serving base station and candidate base stations. If it finds a better candidate than the current on e.g. in terms of signal strength, it initiates a handover.
The network can broadcast various parameters to influence this process, but it is the client that makes the handover decision. The mobile terminal-controlled mode is therefore a completely decentralized mode without any centralized handover controller logical entity.

4.3.2.2. Network-controlled mode

The network measures the signal strength and quality from the client and explicitly orders the client to connect to a specific cell if required. The client makes no measurements. This method results in intense network signaling and limited radio resources, which leads to latency and eventually long handover times. The centralized structure means limitations on scalability and flexibility since the network entities only can handle up to a certain amount of traffic.
Finally, in terms of inter-technology handovers, the network needs some kind of information from the client to base its decision on, since it is the only entity that has knowledge of multiple networks. This is where the network-controlled mode fails. This is, however, compensated for in the mobile-assisted network-controlled mode [Wan99].
A variant of the network-controlled mode is the mobile-assisted network-controlled mode.

4.3.2.3. Mobile-assisted network-controlled mode

The network measures the signal strength and quality analogous to in the network-controlled mode. The network measurements are additionally supplemented with measurements from the client in order to make a more competent decision and compensate for the short-comings of the network-controlled mode in terms of inter-technology handovers. [Wan99].

In UMTS networks, the handover decision is based on the mobile-assisted network controlled mode, where the network entity, the RNC, decides on the handover based on measurement results from the client.

Opposite in WLAN networks, the handover decision is based on mobile terminal-controlled mode, where the client decides on the handover [Pah00].

The decision process may be initiated by one of the following handover triggers: signal quality, terminal mobility, cell load, and applications. Therefore a handover takes place in the following cases:

- If the signal quality of the serving cell lies below a given threshold, and the signal quality of an alternative cell is higher than the current one, then a handover may be triggered.

- If the client moves faster or slower than what is optimal for the serving cell, a handover may be triggered. This could be in the case where the client moves high speed into a WLAN cell that innately does not handle fast speed very well.

- The client can move out of the coverage of the radio system and loose radio connection. A handover to another radio system will then be required to maintain the radio connection.

- If the overall load of the serving cell is high and the available bandwidth thus becomes too small, the network can trigger a handover to a less loaded cell to balance the load.

- If the serving cell cannot support a starting application e.g. in terms of bandwidth or if the application running on the client downgrades the offered QoS e.g. in terms ofguaranteed bit rate, then the network can trigger a handover to another network that can support the application.

- The user can also implicitly trigger a handover by changing application from e.g. a low-bandwidth application to a high-bandwidth application [Bar01].

4.4. Mobile IP Support in the UMTS/WLAN integrated system

A generic mobility handling protocol that allows roaming between different types of access networks would allow the user to conveniently move between fixed and mobile networks, between public and private as well as between with different access technologies. As previously described, many integration architectures were based on MIPv4 and MIPv6 as the mobility management protocol [Bud03, Axi04]. These integrations schemes suffer from packet loss and significant HO latency. In what follows an anticipated vertical handover based on a pre-managed handover is presented. Once the MS detects the availability of a new access network, the AVHO which is based on FMIPv6 handover procedure will enable the MS to be still connected to the previous access network while preparing the handover to the new access network. AVHO is presented and compared to that of MIPv6. We considered the loose coupling architecture where unlike the tight coupling architecture based on GPRS Mobility Management, the mobility management between the two network technologies is supported by the MIP. The proposed anticipated vertical Handover (AVHO) is based on FMIPv6, which is an extension to MIP to support micro-mobility. The latter is based on a prepared handover between AP and Node B in WLAN and UMTS respectively, applied in the context of loose coupling architecture.

4.5. Anticipated Vertical Handover (AVHO)

We consider the handover between WLAN and UMTS or vice versa to be mobile-initiated handover (Figure 4.1). In this case, the mobile will initiate the RtSolPr message to notify the OAR that it is about to perform handover.

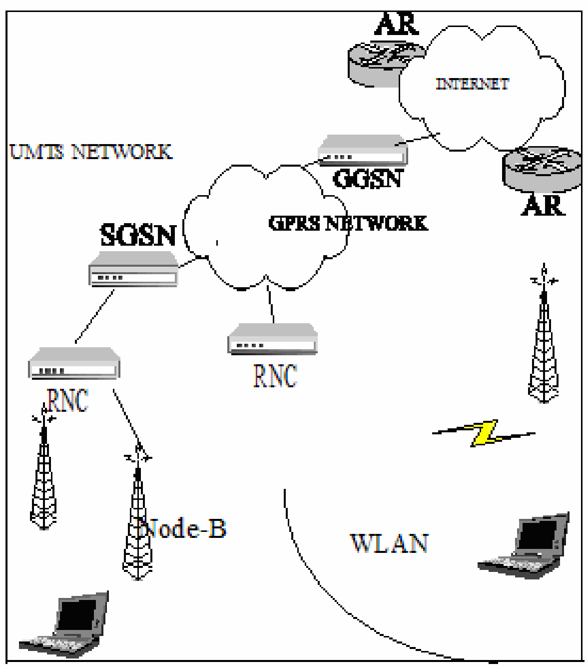


Figure 4.1 Topology of the simulated scenario

The main difference between MIPv6 and AVHO is that the latter is based on the FMIPv6 signaling messages, and the handover is prepared in advance or a "make before break" approach. With MIPv6 the mobile stops receiving packets on the old link, initiates a registration request after it detects the new base station, and then resumes the packets

reception. With AVHO the mobile initiates the registration request while still receiving packets on the old link and after receiving the COA it starts receiving the packets on the new link.
In AVHO, after the Handover initiation stage, the Mobile Node sends Fast Binding Update to the old AR by using its CoA just before it carries out Handover. The Mobile Node then receives Fast Binding Acknowledgement of the old AR by indicating the updating is achieved. Indeed, the old AR sends the F-BAck message to the Mobile Node by building a temporary tunnel. The old AR can also send the F-Back message to the Mobile Node on its old connection (to ensure the message reception by the Mobile Node), [Koo04a].

4.6. AVHO Scenario

4.6.1. From WLAN to UMTS

Old access router exchanges messages to initiate the handover with the new access router located in UMTS network. This last one will reply with the Ack together with the care of address that will be used by the mobile node in the new access network. In this case, PDP context activation and FMIP registration will occur in the GPRS network. Because of the low data rate in GPRS compared to that of WLAN, the time it takes the HI and Hack to reach the new access router and the old one respectively, will be slower compared to the WLAN rate, but this will not really affect the handover performance since the mobile node will be still communicating with the Old Access Router. Once the handover initiation phase is over, the mobile node will be connected to the new access router in the GPRS network and starts to receive data packets after the NFA message is sent to the new access router.

4.6.2. From UMTS to WLAN

The handover scenario when a mobile node is performing handover from UMTS to WLAN with AVHO signaling will occur as follows.
Old access router in the GPRS network exchanges messages to initiate the handover with the new access router located in WLAN network. This last one will reply with the Ack together with the care of address that will be used by the mobile node in the new access network. In this case, PDP context deactivation will occur and the FMIP registration will occur in the WLAN network. Because of the higher data rate available in WLAN networks compared to that of GPRS, the HI and Hack signaling messages will be faster. Once the handover initiation phase is over, the mobile node will be connected to the new access router and starts to receive data packets after the NFA message is sent to the new access router in the new access network.

4.7. Simulation Study

To achieve the goal of the simulation, we need a umts node-B, a WLAN access point and a mobile node able to roam between the two wireless networks.

- With MIPv6 the mobile stops receiving packets on the old link, initiates a registration request after it detects the new foreign base station, and then resumes the packets reception.
- With AVHO the mobile initiates the registration request while still receiving packets on

the old link and after receiving the COA it starts receiving the packets on the old link.
The mobile node is enabled with interface switching capability to move between the different access networks.
The handover is based on the physical parameters detected at the interface such as the signal power.

The UMTS node-B is characterized by its wide coverage ranging from few kms down to 0.5 km while the AP has a small coverage area reaching few hundreds of meters.
Also the power transmitted by nodeB is higher than that of AP, for example node-B transmits 3W/channel, while the power transmitted by AP is 0.036 W
To determine the reception threshold of the receiver, we used the following settings:
-The UMTS node transmission power is 3.0 W
-Range is 440m
-Two Ray Ground model.

Table 4.1 shows how the nodes are configured to trigger the handover from WLAN to UMTS or vice versa based on the network coverage and the received signal power.

Node Type	Pt in Watt	Rx in Watt	Range in m
NodeB	3.0	4.6×10^{-13}	440
AP	0.036	4.6×10^{-13}	245
Mobile	0.036	5.0×10^{-11}	400

Table 4.1 Nodes configuration

4.7.1. Simulation parameters

To achieve the goal of the simulation which is the vertical handover between UMTS and WLAN, we have defined the following settings:

- A correspondent node **(cn_)**, which may be a typical internet web server.
- A UMTS base station **(nodeB_)**.
- A WLAN access point **(AP)**.
- A router **(router_)** acting like an internet node that connects the correspondent node to the AP and nodeB_.
- A Mobile node **(MN)**, that roams between the two wireless networks with 2 interfaces.

-UMTS Node B:

- The umts nodeB_is characterized by its wide coverage, in other words the transmission power is higher than that of a WLAN access point. A typical nodeB_ transmits power in the range of 3W per channel.
- The reception threshold is set to a minimal value, so that nodeB_ can hear the mobile node from anywhere within the topology.
- The data rates supported by the UMTS link is variable in real life, it ranges between 2Mbps for stationary nodes, 384 Kbps for moving mobiles and 144Kbps for moving vehicles. The typical data rate is 384Kbps, so we have chosen it for the UMTS interface.

-WLAN AP:

The settings of an access point AP are completely different from those of nodeB_, for example: the WLAN access point is characterized by the high available data rate with a peak of 11Mbps (802.11b), but unfortunately its coverage is limited.

- The WLAN AP has a limited coverage, in other words the transmission power is lower than that of a UMTS nodeB. A typical AP transmits power in the range of 0.03W.
- The reception threshold is set to a minimal value, so that AP can hear the mobile node from anywhere within the topology.
- The peak data rates supported by 802.11b is about 11 Mbps at the MAC sub layer, but in fact this rate can never be achieved at the application layer because of the contentions, packets overhead, and error detecting codes. For that reason only about 4.0 Mbps are available for the users at the application layer and since this bandwidth is shared between different users, we have chosen a data rate of 2Mbps for the WLAN link and this choice proved to be a logical one.

- Mobile node MN:

The mobile node must have the ability to roam between the two heterogeneous networks and to communicate with the two wireless interfaces. The Dual interface of the mobile node pays our attention to the following remarks:

- The WLAN link provides a higher data rate, consequently it is considered as the primary link while the UMTS link is standby because it can support a wide coverage area.
- The mobile node will use the WLAN link as long as available until the WLAN signal fades totally, at that time, the mobile switches to the UMTS link.
- Similarly - for the inverse scenario - the mobile node will switch directly to the WLAN interface as soon as it detects a WLAN hotspot.

The transmission power of the mobile node is set to 0.036 W and its sensitivity is set to **$5\ x10^{-11}$ W,** this will result in a range of 950m for nodeB and 200m for access point as mentioned in section.

The correspondent node cn_ is a usual internet node for example a download server, while the router is the node that links the cn_ to nodeB_ and AP.

4.7.2. Handover Simulation

In all what follows we will describe the process of handover from WLAN hotspot to a UMTS cell, this must be evident because the AP is supposed to be the home network and the mobile must move out of the WLAN coverage, while the reverse procedure (meaning handover from UMTS to WLAN) follows the same steps but initially the nodeB_ is supposed to be the home network. The links between the wired nodes are full duplex with a data rate of 10 Mbps and a propagation delay of 1.8ms.The queue type is Drop tail.

The scenario is as follows:

- The TCP agent is created at the cn_ and it is attached to the cn_.
- A TCP sink is created at the mobile_ and it is attached to the mobile_.
- The TCP Agent is connected to the TCP sink.
- An FTP application is opened over the TCP Agent
- The FTP application is started at instant 5.0
- At instant 5.00 the mobile_ starts an FTP download from the cn_ .
- At instant 6.00 the mobile_ starts moving away from the AP towards the nodeB_ .
- At instant 12.00 the simulation stops.

Figures 4.2 and 4.3 show the packets sequence number versus time while performing handover. From the graph, the handover delay from UMTS to WLAN is determined by calculating the time of the last packet received on the previous network and the time the first packet is received on the new link in the new network. Thus the delay are calculated and equal to 0.502129 sec and 0.284463 sec for figures 4.2 and 4.3 respectively.

In figures 4.4 and 4.5, the throughput was measured based on the following scenario:
At instant 5.0 the mobile starts the FTP session with the correspondent node, at that time he is under the coverage of a UMTS nodeB offering him a data rate of approximately 310Kbps.
During the handover process the mobile suffers from a drop in the bandwidth down to 200Kbps, then he switches to the new link and enjoys a data rate of approximately 1.1 Mbps .
The WLAN link is able to offer data rates up to 2Mbps in our configuration, but the rate arriving at the AP from cn_ is no more than 1.1 Mbps, in other words the traffic flow is about 1.1 Mbps and the available resources are 2Mbps. It is clear that the throughput performance is better than the previous case of MIPv6, as there is no significant drop in the throughput performance.

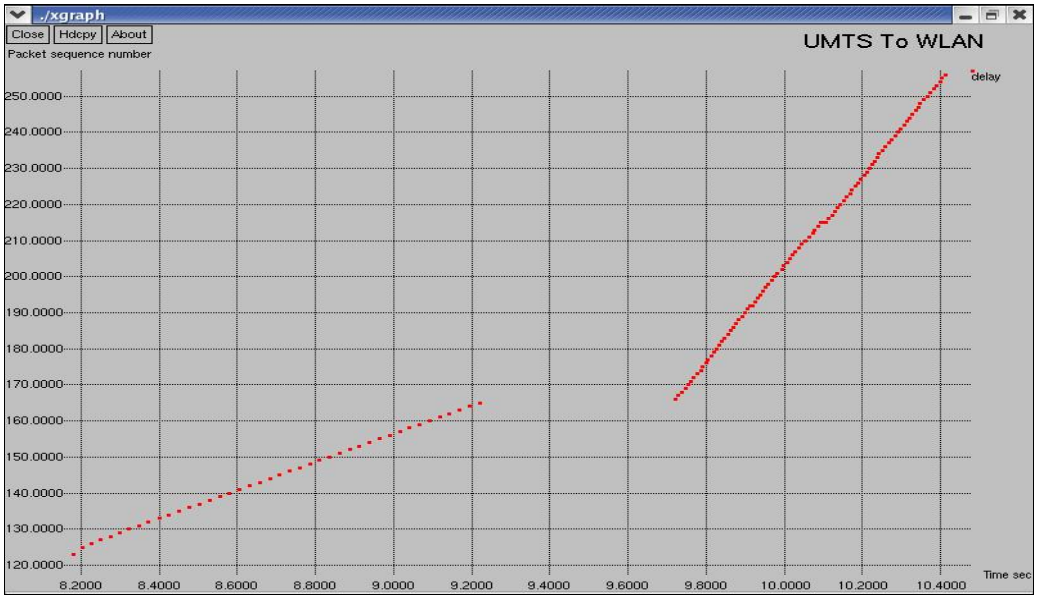


Figure 4.2: Packet sequence number Vs time using MIPv6

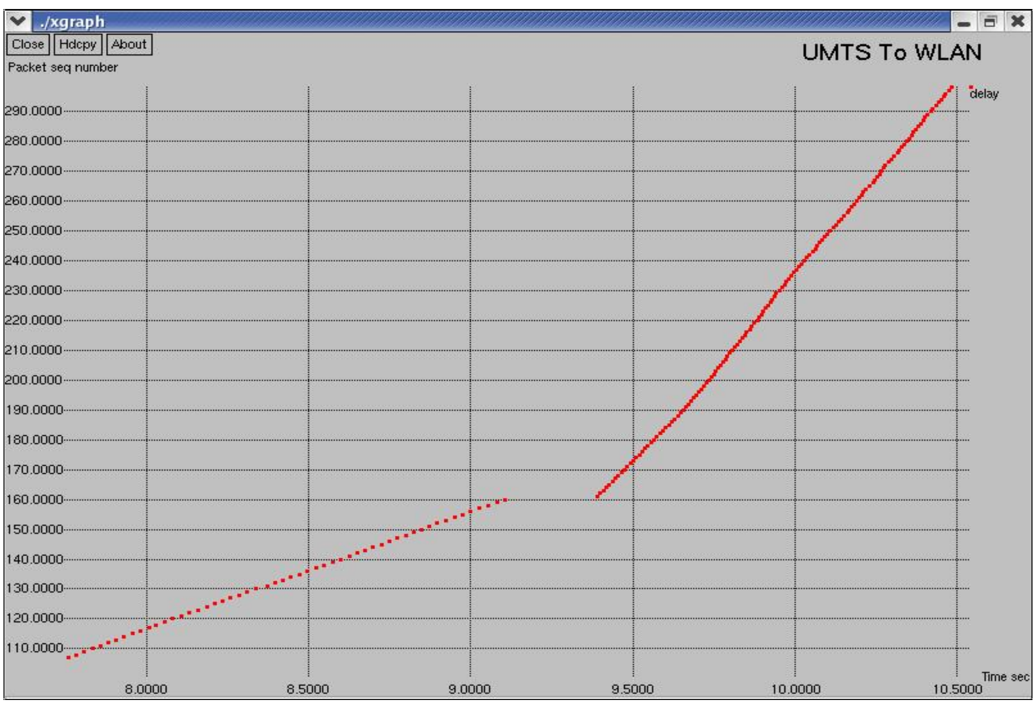


Figure 4.3 HO Delay from UMTS to WLAN using AVHO-FMIPv6

Figure 4.4 Throughput performance from UMTS to WLAN using mipv6

Figure 4.5 Throughput performance, from UMTS and WLAN using AVHO

In figures 4.6 and 4.7 shows what will happen to the data rate when passing from WLAN to the UMTS network, and the delay the user will encounter to perform the vertical handover. The Handover Delays are equal to 0.421996 **s** with MIP, and 0.247848s with AVHO.

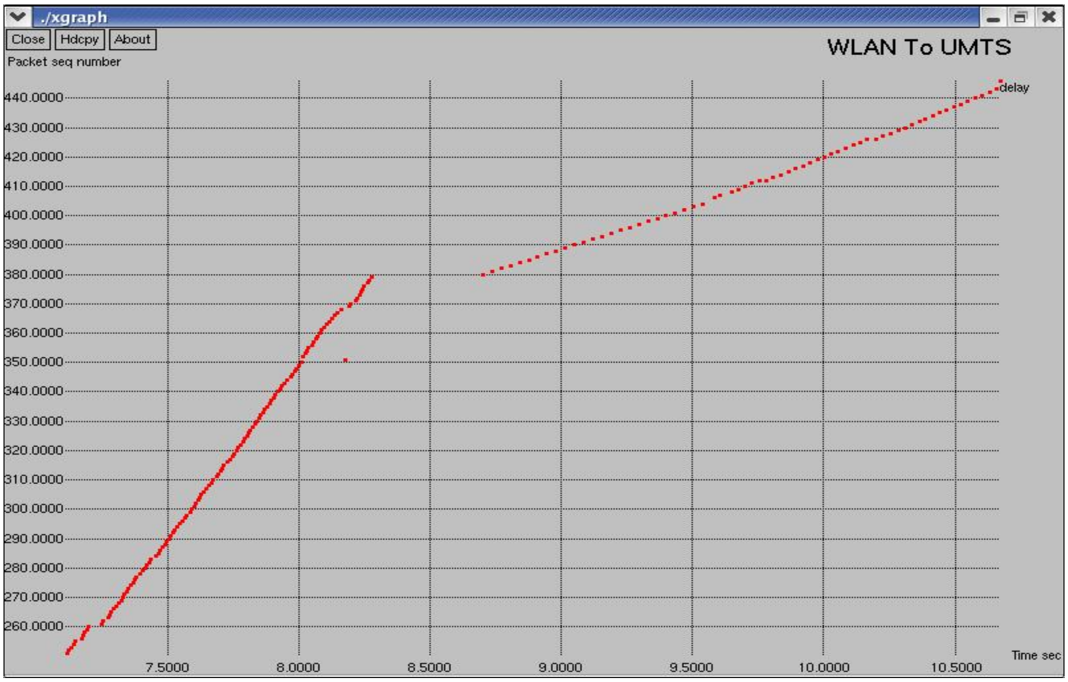


Figure 4.6 Packet sequence number Vs time with MIPv6

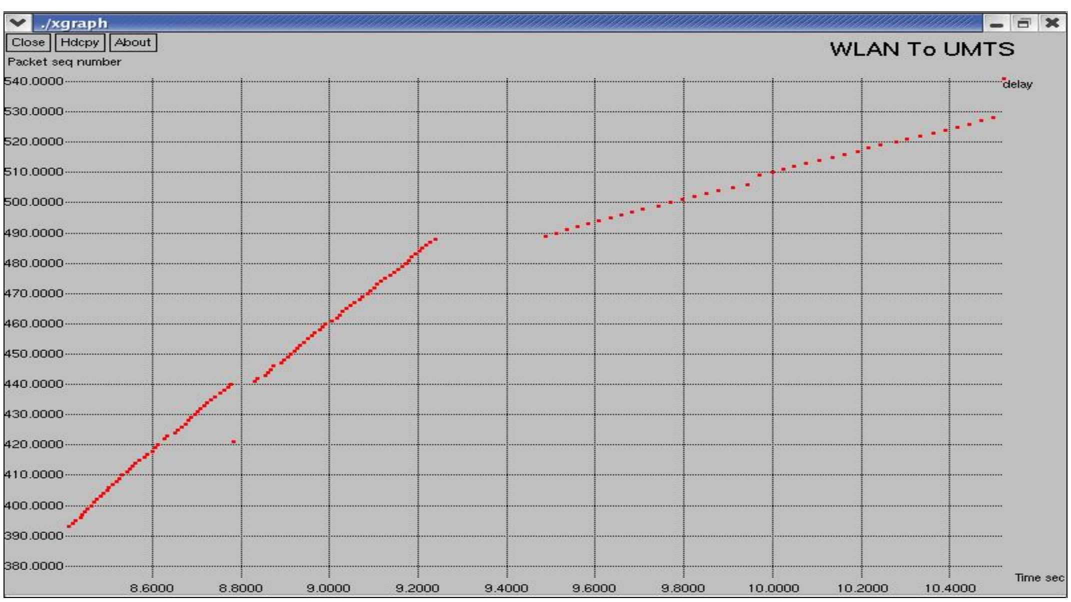


Figure 4.7 handover delay using AVHO

4.7.3. Throughput performance

As we know, the UMTS link has a poor bandwidth when compared to the WLAN link, thus when the user passes from the WLAN AP, he will loose the high data rates offered by this network. Instantaneous bit rates are shown in figure 4.8.

At instant 5.0 the mobile starts the FTP session with the correspondent node, at that time he is under the coverage of a WLAN AP offering him a high data rate of approximately 1.0 Mbps. The UMTS link is unable to support the high data rates offered by WLAN, so the main concern during the handover process is to maintain the session and prevent any disconnection. In figure 4.9, Instantaneous bit rate are also shown in the case of AVHO.

During the handover process the mobile looses the high throughput due to the difference between the two technologies, but this decrease never drops below 300kbps. This gives the best performance between the cases discussed so far.

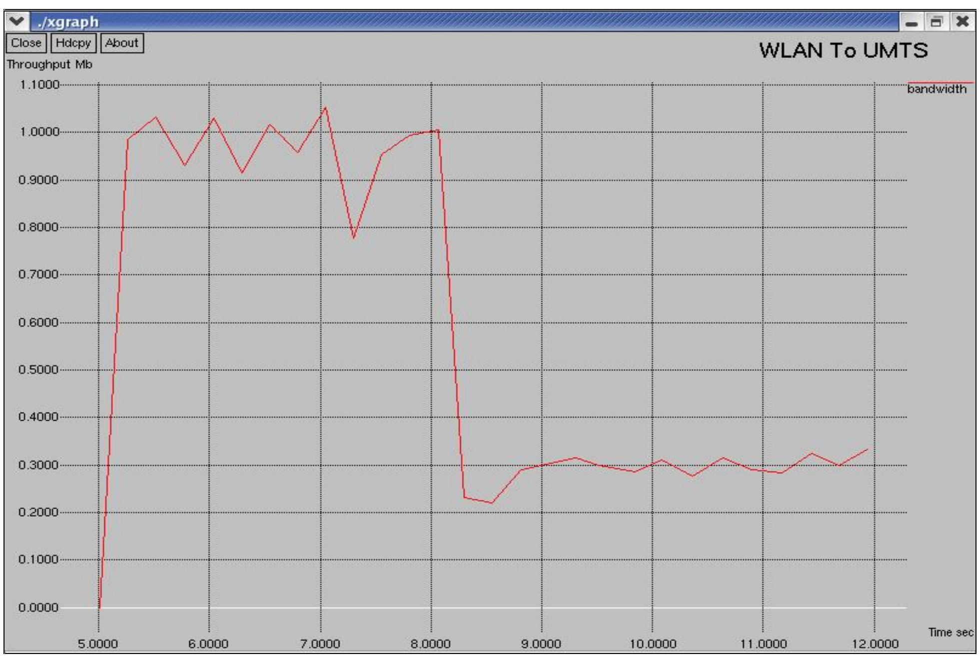


Figure 4.8 Throughput performance (WLAN/UMTS, using mipv6)

4.8. Conclusion

From the above simulation, the results show that AVHO reduces the handover delay significantly: In the case of UMTS to WLAN handover the delay is reduced by 47.5%, and in the case of WLAN to UMTS handover the delay is reduced by 41.3%.

In this study, we didn't consider the handover with QOS provisioning. QoS degradation occurred, and the user was unable to keep the application running with the same performance level. In other word, the handover was not transparent to the user. In the next chapter we will be focusing on providing QoS guarantees to the integrated system. We propose a new architecture based on AVHO with QoS called Mobility and QoS Management Architecture MQMA to integrate WLAN and UMTS, and to provide QoS to the end user moving between different access technologies.

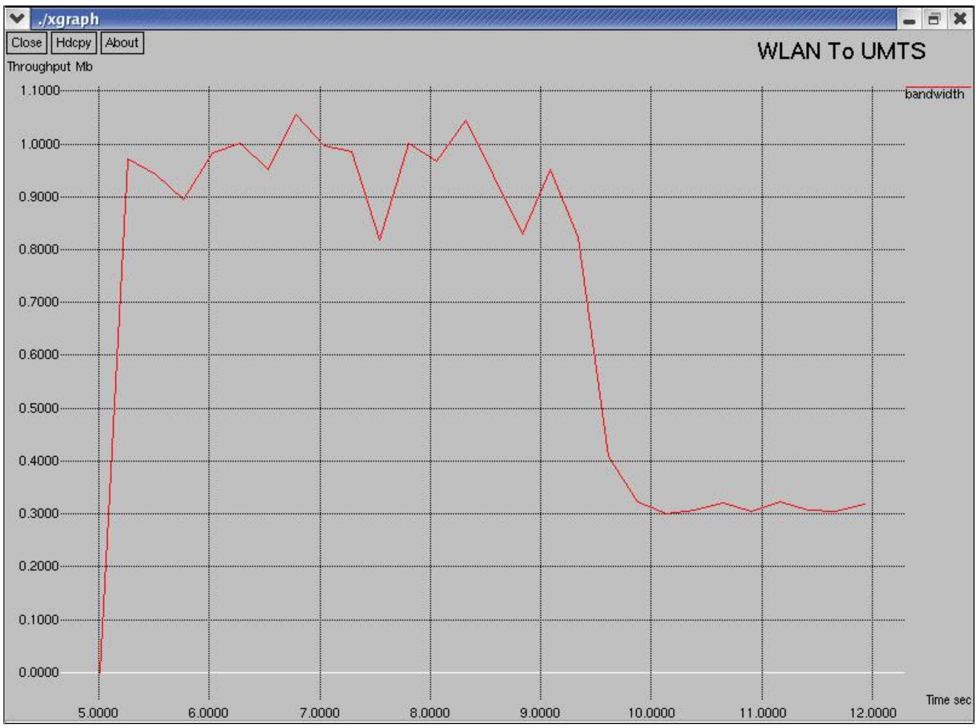


Figure 4.9 Throughput performance (WLAN/UMTS, using AVHO)

Chapter 5

Mobility and QoS Management Architecture MQMA

5.1. Introduction

In the previous chapter, AVHO was proposed in order to decrease the handover latency in the next generation networks. The handover latency is significantly reduced due to the fact that AVHO is like FMIPv6 which allows a Mobile Node to be connected to a new point of attachment and start to receive packets directly forwarded to the New Access Router (NAR) once the Mobile Node (MN) acknowledges its neighborhood by sending Fast Neighborhood Acknowledgment FNACK. Otherwise, the MN is still receiving packets from the Old Access Router (OAR) [Tsi01]. In our proposal, we will consider a mobility and QoS management protocol based AVHO, relying on the FMIPv6 approach, extended to provide seamless handover with the required QoS level.

The Mobility and QoS Management Architecture MQMA proposed in this chapter will provide not only fast handover but also a seamless handover with minimum QoS degradation. The proposition is based on Inter-Domain Manager module IDM responsible of providing seamless handover by mapping the required QOS parameters of the mobile node to the appropriate IDM in the new domain. The mobile user will then be "guided" according to the selected access network with respect to the transferred context and user profile.

5.2. MQMA overview

We propose a new integration architecture composed of a core network, based on Diffserv, connected to different access networks controlled by an Inter-Domain Manager module IDM. IDM is present in each access network operating under the proposed MQMP (Mobility and QoS Management Protocol), to provide a seamless handover with the required QoS level.

In the proposed model in figure 5.1, the core network is based on Diffserv model that maps the incoming QOS requirement to its service classes. In the concept of service classes, each class supports specific applications by marking the packet entering the Diffserv domain in the IP header field called DSCP [Yoo00]. The signalization messages between the IDM's should be treated within the Diffserv core network as high priority traffic in order for the handover latency to be minimized. Therefore, the IDM traffic should be mapped to the EF class providing low loss rate, low latency, assured bandwidth and low jitter [Myk03].

5.3. IDM description

The Inter-Domain Manager IDM, which is an internet gateway between the Diffserv core network and the corresponding access network, is responsible of storing the mobile profile and the current context of a session, and making the correct mapping with the Next IDM (NIDM) in order to provide the necessary resources to satisfy the requested QoS. Each IDM keeps a data base of possible IDM in the neighbored access network with the corresponding context and QoS parameters. The IDM in each domain will also search for the available access routers as part of its traffic monitoring responsibilities (figure 5.2). Each IDM communicates with other IDM's in the visited network in order to provide the new data path with the required QOS. Each IDM will perform the following tasks:

- Stores the QoS attributes of a current session based on the context feature sent in the context data message.
- Stores the profile of the neighbored IDM and their corresponding traffic parameters (data rate, traffic load, resource availability…).
- Selects the next access network based on the previously collected data.
- Treats and accepts (accordingly) the handover request based on the "Who's next" phase.

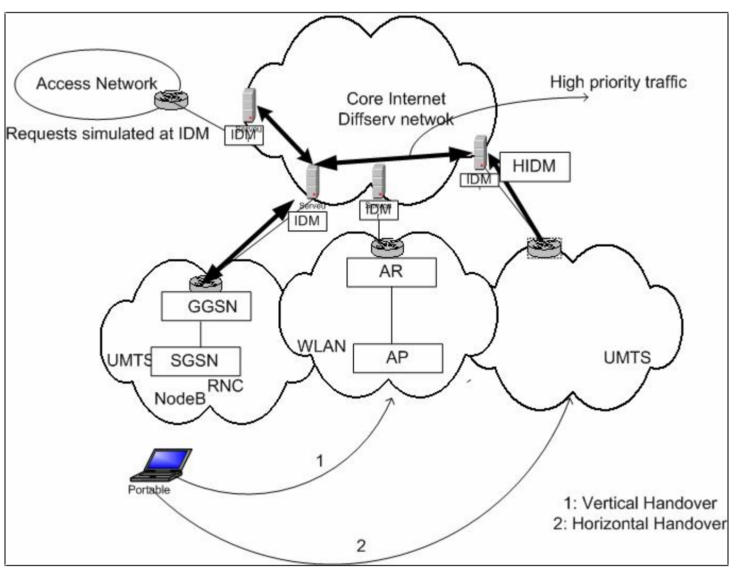


Figure 5.1 Integration Architecture with IDM

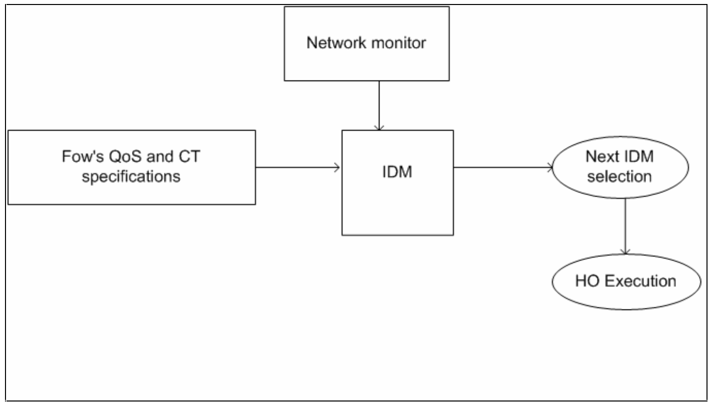


Figure 5.2 IDM operation

Figure 5.2 shows how the IDM selection process is based on the current request of QoS and context transfer specifications before the handover is executed. The monitored parameters are the MN's capabilities, user profile, service cost, QoS requirements, network coverage, data rates…Based on both inputs, the handover is "guided" to the new IDM and the handover execution could be done.

5.3.1. IDM functionalities

The IDM is the basic component of the MQMA model. HIDM operates as a handover management unit, a seamless roaming unit and a seamless vertical to horizontal handover.

- **HO management unit**: the IDM will manage the HO by searching for the next network capable of offering the end user the required QoS parameters. The best network selection is based on the network conditions, context transfer…If the profile is highly privileged, the HO could be guided through the SIDM to WiMax network in order to accept the HO with the requested QoS.

- **Seamless roaming unit**: all the signaling should be done before the MN moves, or before being out of coverage. This is done by first initiating the HO request when the MN is still connected to the network. This pre-active behavior will guarantee that the mobile node will do all the signaling issues before being disconnected from its current network (before loosing coverage), and without degrading the QoS parameters in the original network.

- **Seamless vertical to horizontal handover**: To reduce HO failure probability which occur if the IDM queue is full (since it is modeled as M/M/C/C/, the HO call could be rejected). In this case, the mobile node will not wait until all the calls in the queue are served. This will cause a significant delay since the HO request is not directly treated. The solution is to forward the HO call to the HIDM responsible for horizontal HO, which will perform the same procedure in terms of reserving resources and scanning for the appropriate IDM through the "Who's Next" phase. In this case, the HO is performed horizontally. The VIDM could also forward the call to an SIDM which will guide the HO to WiMax network (as described in chapter 6).

5.4. Mobility and QoS Management Protocol (MQMP)

The Mobility and QoS Management Protocol MQMP consists of an anticipated fast handover operation and context transfer exchanged between the home IDM and the selected IDM in the new domain capable of providing seamless handover. Each domain in the access network is managed by an IDM which operates under the MRMP protocol. Each IDM is connected to a number of access routers. The handover execution and context transfer operations are performed through a series of message exchanged between IDM's and access routers as shown in figure 5.3. The context transfer messages carry all the feature context of a session, with a context identifier and authorization key [Lou05].

To maintain knowledge of resource utilization in the entire domain, the IDM is monitoring traffic matrix, it will then use this knowledge to complete its profile necessary for admission control and resource provisioning when executing handover [Niy05]. IDM's will store the Context Data and the profile of a MN. Once the MN has the requested Context Transfer mapped, it then performs the handover according to figure 5.3. The handover treatment is based on the mobile user profile and the feature context carried by the context transfer messages through the IDM. The IDM will scan for a new IDM on the new access network in order to satisfy the required parameters corresponding to the current context.

5.4.1. MQMP operation

The main protocol operation consists of providing an anticipated handover based on FMIPv6 and context transfer in order to avoid QoS degradation in the new network by mapping the required context and profile to the New IDM (NIDM). This could be done through some of the FMIPv6 signaling messages and the Context Transfer Protocol messages. The general protocol operation is as shown in figure 5.3. Figure 5.3 shows that before the mobile node executes the handover, it will start with the normal FMIPv6 signaling procedure by sending Router Solicitation for Proxy Advertisement (RtSolPr), sent from MN to the previous access router PAR [Koo04a]. This message should also be sent to the current IDM which is responsible of managing mobility and resources in the current domain. This message will indicate to the old Access Router and the IDM that the Mobile Node would like to perform a handover. The IDM will interpret this message as a handover request so that it will start to search for another IDM. The MN sends Context Transfer Activation Request (CTAR), defined in [Lou05], to the PIDM, too. The latter will reply with a Context Transfer Request (CTREQ) to the PAR in order to get the current context data of the mobile node requesting the handover, since the IDM of each domain stores a context entry per each mobile node. So PAR will reply to the PIDM with the Context Transfer Data (CTD) relative to the user profile and the set the necessary parameters of the services currently in use by the mobile node. PIDM will use the obtained context transfer data to scan for available IDM according to resource availability and traffic load on the connected access routers (figure 5.3). Once the required context is mapped to the corresponding NIDM, it will send to the PAR a Context Transfer Data Reply (CTDR), and it will configure the new access router to accept handover by sending a CTD message and a Handover Initiation message (HI) to both the NAR and the NIDM. The selected NAR will send back a Handover Ack (HAck) with the New Care-of-address to the NIDM. This care of address is transferred through the mobile node through the RrRtAdv message used in FMIPv6 as a response to the RtSolPr. When RrRtAdv is sent to the MN, it will respond by sending a Fast Binding Update FBU to the PIDM. PIDM will send a Fast Binding Acknowledgment FBAck to both NAR and MN. The MN will finally send a Fast Neighbor Advertisement (FNA) to the NAR to notify him of its presence in the new access network [Koo04a].

The context transfer data are sent before the handover is performed to the visited network, in addition to the mobile user profile and the AAA context of the mobile node. Once the required context parameters are mapped into the NIDM in the new network, the mobile user can perform handover without QoS degradation due to the appropriate selection of NIDM and accordingly the new access router in the new access network.

Contexts are identified by FPT (Feature Profile Type), which is a 16-bit unsigned integer [Loo05]. The context type numbers are assigned by IANA, and handled according to the message specifications [Loo05]. FPT are transferred by data blocks used for transferring the actual feature context. Context Transfer Data (CTD) Message is sent from PIDM to PAR and includes feature data (CTP data). This message handles the desired list of feature contexts.

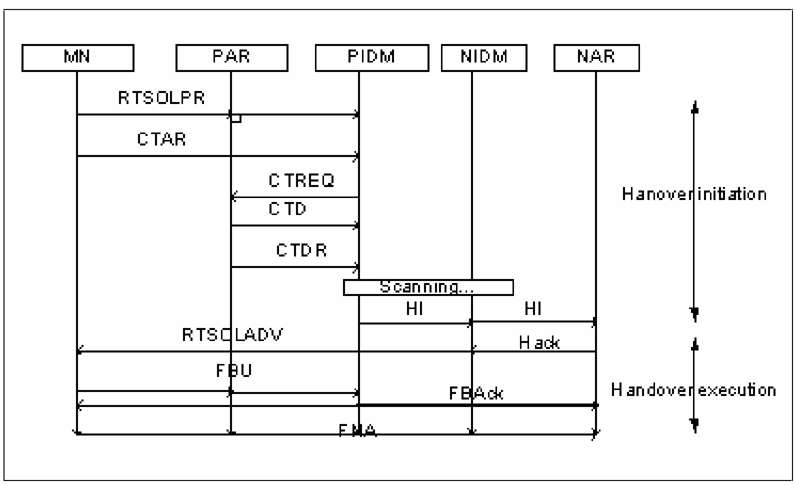


Figure 5.3 Mobility and QoS Management Protocol

In the proposed protocol, the PAR and PIDM are both involved in the FMIPv6 message exchange as shown in figure 5.3. The FMIPv6 protocol messages are carried by ICMPv6 and mobility header messages [Pos81]. All the ICMPv6 messages have a common Type field to specify the mobility protocol used the subtype to specify the message type exchanged [Koo04b]. The following messages are ICMPv6 messages: PrRtAdv, RtSolPr, HI, HACK. The other messages as FBU, FBACK, and FNA are carried by the IPv6 header type called mobility header as specified in [Koo04b].

In FMIPv6 operation, a Proxy Router Advertisement (PrRtAdv) message should be received by the MN as a response to RtSolPr [Tsi01]. But in our proposed scenario, it is sent back to the MN after the context transfer messages occur. Therefore to avoid retransmissions after the timer expires, the retransmission timer should be increased to at least five times the typical round trip time (RTT) between PIDM and NIDM (figure 5.3).

The context transfer messages used in the proposed scenario are similar to that described in [Kem02]. The transport of these messages is based on ICMPv6 protocol where the type and code are set to the specific type of CTP messages and the context data transfer is added in the data option field [Loo05]. For example, the type field in the ICMP message is the experimental Mobility ICMP type and the Sub-type field is the experimental mobility ICMP subtype for CTP [Loo05].

5.4.1.1. Handover initiation

In the case of vertical handover, before the MN is out of coverage in the current network (WLAN), it will start to initiate the handover operation in a way that switching to the new network (UMTS) is done after the handover operation is finished. In this scenario, no packet loss and minimum QoS degradation will occur. In this case the handover is called "Forced Handover" since it is triggered by physical events such as the network coverage [Xav03] as in the case of vertical handover from WLAN to UMTS. Handover could also be initiated based on the user policies and preferences such as the discovery of faster and cheaper network similar to the case of handovers from cellular networks to WLAN. This type of handover is called the "User handover". In both cases, the handover is initiated by sending a message from the MN to the IDM as show in figure 5.4.

5.4.1.2. Handover execution

During handover execution, all the necessary settings such as CoA acquisition and necessary QoS mapping and updates are done in order to successfully perform the handover. The main steps are described as follows:

- Handover request is to be treated by PAR and PIDM…It is served either fully where the vertical handover is performed, or partially where horizontal handover could be performed.

- Context transfer signaling, all the messages are to request Ack of reception, otherwise they are retransmitted. This phase should be refreshed in the current access network to be used for handover.

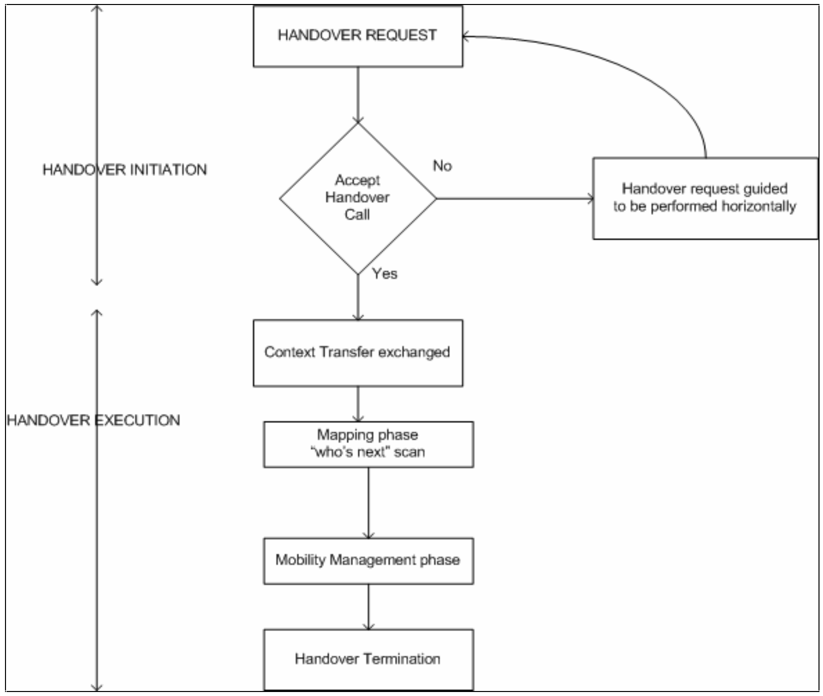


Figure 5.4 Handover request handling

- Scanning process (the "Who's next?" phase), if failed, and no IDM is available, the HO is either guided to another WLAN or UMTS, performing Horizontal Handover (HHO).
- Acquiring CoA during the FMIPv6 phase: if failed, the resources in the NAN are freed after a time interval equal to FMIPv6 time operation. The HO call is then not terminated, and the scanning process is reinitiated again until the HO call is terminated.
- Neighboring updates to start communicating with the new AR.

5.5. Handover working process

The IDM will accept the handover call to be treated if the user profile is allowed to perform the handover or according to a specific admission control strategy [Niy05]. Once accepted, the request is treated by starting the context transfer exchange and mapping the requested QoS parameters into the next selected network. The request is classified into two classes. If the application is tolerant to degradation, the NIDM selected will satisfy partially the HO call and if more resources are freed the call will be fully accepted. On the other hand, if the application is of high sensitivity and not tolerant to degradation, the HO call will be guided to a NIDM which will fully accept the HO call and the requested context will be fully satisfied. This is done in the "WHO'S NEXT" phase. Next, the mobility management will take place and the CoA is configured and updates to the PIDM and PAR are done. The handover termination phase is where the mobile node will start to communicate with the NAR on the New Access Network NAN.

Local calls and handover calls are handled by the IDM, therefore the IDM capacity and its admission control strategy should be carefully examined to decrease the blocking probability and dropping of handover call. Thus the IDM modeling is of high importance in the analysis of handover performance.

The "Who's next " phase is the scanning phase where the previous IDM (PIDM) will select the next IDM in the destination network capable of providing the required QoS and performance level based on the context transferred and QoS parameters. The scanning process is based on the IDM profile and available resources. The IDM profile is classified into class 1, accepting non tolerant to degradation applications, and class2 for tolerant to delay and degradation, as shown in table 5.1. After the scanning process the MN is guided to either IDM of class 1 or IDM of class 2.

User profile	**IDM class**	**HO call**
Tolerant to degradation	**Class 2**	**Partially then fully accepted (if possible)**
Non-tolerant to degradation	**Class 1**	**Fully accepted**
Not allowed to access NAN	**Rejected**	**Rejected**

Table 5.1 Mapping IDM classes and User's profile

5.6. MQMA specifications

Compared to the existing integration architectures such as tight and loose coupling from one side [ETS01], and Moby dick project [Jah04] from the other side, our solution has the following advantages: while the mobility management is controlled by MIPv6 for loose coupling and Moby dick, the mobility Management in MQMA is controlled by the Mobility and QoS Management Protocol which is based on the IDM interception to the FMIPv6 messages and Context transfer messages. The HO management is through the IDM "Who's next" phase, so that no separate entity will handle this mission such as the SGSN. The QoS is provided through the IDM, so that no additional broker is needed. This is in contrast to Mobydick [Jah04] QoS management where the QoS negotiations are performed after the handover. The HO may not be rejected (unless the user is nor authorized); it will be performed according to the required context and QoS (either HHO or VHO). The system architecture is also based on the IDM entity which should be available at each access network as the gateway to the main IP cloud.

A brief comparison is presented in table 5.2, and an analytical study is presented in what follows to study the performance of the proposed solution.

	Tight architecture	**Loose architecture**	**Moby dick project**	**MQMA**
Mobility Management	GPRS Mobility Management	MIP	MIP6	MQMP (FMIP6+CT+IDM)
HO Management	SGSN	CAG	HA, AAAC, QoS broker	IDM (Who's next) phase
QOS Provision	Not ensured	Not ensured	Possible through QoS broker negotiation	Guaranteed through IDM mapping phase
HO call rejection	Possible	Possible	Possible	Not possible
System Architecture	cellular operator-dependent	New entities: CAG, billing mediator…	New entities: QoS broker, AAAC, NMS…distributed in each domain.	IDM entity (between the core network and the access network)

Table 5.2 Comparison of MQMA with existing solutions

5.6.1. MQMA Description

VIDM, HIDM and SIDM are present at each GGSN in the UMTS network and at each gateway in the WLAN network and they are accordingly able to guide the HO call to the best network based on its context, the available resources, and respective IDM load. The IDM's are connected to the IP cloud as illustrated in the figure 5.5.

- Different network operators communicate through the IDM's connected in the core IP network.

- Each wireless network is connected to an IDM through which it is connected to the internet. The inter-system communication is supported by the IDM communication, each IDM is either at the GGSN node (UMTS network) or the gateway node (WLAN network).

- VIDM holds the data table of the adjacent IDM's whose access technology is different. This table is scanned when performing VHO.

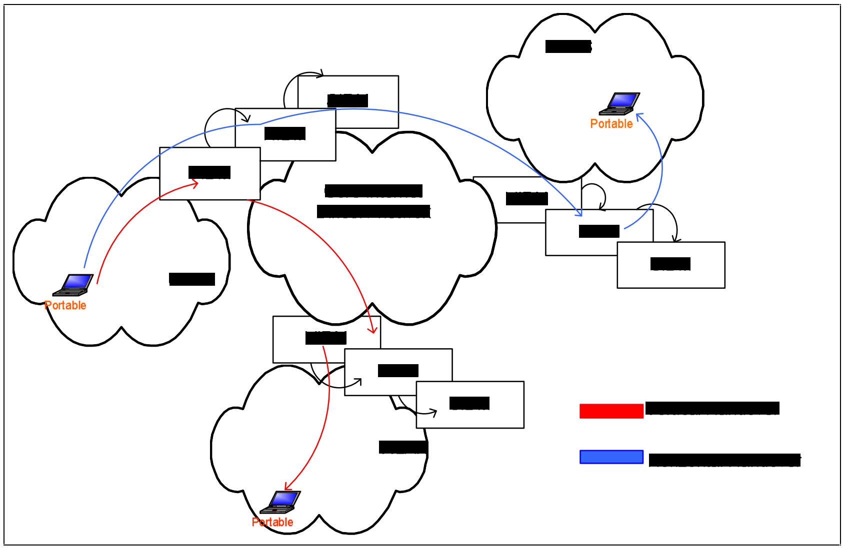


Figure 5.5 MQMA with VIDM, HIDM and SIDM integration

- The HIDM holds a data table of all the IDM's managing the same kind of access networks, the IDM of a WLAN network holds in its table all the IDM's in the adjacent WLAN's. In the same way the IDM of the UMTS network holds in its table all the IDM's in the adjacent UMTS's. For a new operator to operate under the IDM functionalities it should be connected to the corresponding IDM (WLAN or UMTS). This IDM holds all the network parameters, traffic load, and network capacity of the managed network. The IDM will communicate with the neighboring IDM's through the IP network. This is more scalable since it is not practical to create SLA with all the operators. SLA is done through the IDM's.

- Seamless mobility and QoS support are provided through the IDM inter-communication feature.

5.6.2. MQMA characteristics

The proposed integration architecture satisfies the following characteristics:

- Based on the IDM mapping in the "scanning phase", best network selection is supported which will provide the requested QoS according to the context transfer specifications.

- Mechanisms in the admission control to ensure high admission for HO call (privileging vertical HO calls over local calls).

- Mechanisms to ensure seamless mobility of the mobile node roaming between different access networks.

- IP is the infrastructure used which is scalable and cost effective.

- IDM acts as a gateway connecting each access network to the internet.

- VIDM's are to realize the HO between different access networks; otherwise it's done through the HIDM.

- When the profile exchanged between the two IDM's is done in the core network which is a diffserv network as shown in the above figure. The traffic of signaling exchanged between the two IDM's is considered to occur in the Diffserv network, where it should be treated as high priority traffic treated in the EF class. This class provides low loss rate, low latency, assured bandwidth and low jitter. So if the signaling exchange is mapped into this class, and recorded in the IDM, no additional payload is added during handover. Figure 5.6 illustrates the inter IDM exchange.

- Number of roaming users is high, the VIDM queue can not handle all the HO call, and the HO blocking probability is increased.

- The VIDM strategy is to forward the HO call to HIDM to perform horizontal HO.

- The SIDM is also used if the HIDM queue is full, and horizontal handover could not be performed.

- Hierarchical functionality: If the VIDM in the "Who's next" phase didn't find an appropriate IDM, the VIDM will guide the HO call to another access network by through the HIDM. In the same way, if the HO is not accepted (the "Who's next" didn't find an appropriate IDM), SIDM will guide the HO to WiMax as shown in chapter6.

5.7. Scenario and model description

The IDM admission control strategy is based on priority basis. Two kinds of priority admission are adopted depending on whether the HO is initiated in the WLAN or in the UMTS networks. Since WLAN has a higher bandwidth and can accommodate more calls, a one-threshold hysteresis scheme is adopted from WLAN to UMTS. In this hysterisis model, the handover calls are privilege over the local calls, so that VHO dropping probability is lowered.
On the other hand, the HO from UMTS to WLAN is based on a 2-threshold hysterisis, since it has a lower bandwidth and accordingly lower IDM capacity. The HO calls should be privileged over the local calls.

5.7.1. From WLAN to UMTS

Since WLAN has more bandwidth and can accommodate more calls, the following Markovien model is used. In this model the HO calls are privileged over the local calls. In fact the local calls are accepted until the threshold T, and if the number of calls is greater then T, then only the HO calls are accepted. The modeled system is as shown in figure 5.7.

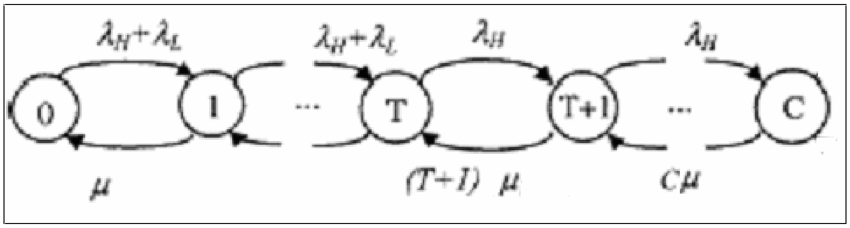


Figure 5.6 IDM model (WLAN to UMTS)

We assume the following:

- Each IDM is modeled as M/M/C/C queuing system.
- Local calls are admitted if they are less then C channels occupied.
- Handover calls are dropped by VIDM (but guided to HIDM) if all the C channels are occupied.
- C-T interval channels are the guard band channels used only for handover calls.
- The total capacity the IDM is C
- The time of stay in IDM is exponentially distributed of rate $1/\mu$.
- The local traffic is a poisson process of rate λ_l
- The handoff request is a poisson process of rate λ_h
- $\lambda_h=0.2\lambda_l$ or the local calls are 5 times the HO request arrivals.
- The threshold T=40% C
- Assuming $\mu h = \mu l = \mu$, the equal time of service is due to the fact that the IDM processing time is unique in bith cases.

The stationary probabilities for the different states are as follows according to [Fad02], where S_1=T and N=C:

- For i = 1,2, … T:

$$\pi_i = \frac{1}{i!}\left(\frac{\lambda_h+\lambda_l}{\mu}\right)^i \pi_0$$

- For i=S_1+1 to C:

$$\pi_i = \frac{1}{i!}\left(\frac{\lambda_h+\lambda_l}{\mu}\right)^{s1}\left(\frac{\lambda_h}{\mu}\right)^{i-s1} \pi_0$$

- At state 0:
  P(0)= π_0 is given by

$$P(0)=\left(\sum_{i=0}^{T}\frac{((\lambda n+\lambda h)/\mu)^i}{i!}+\sum_{i=T+1}^{C}\frac{[(\lambda n+\lambda h)/\mu]^T(\lambda h/\mu)^{i-T}}{i!}\right)^{-1}$$

- The blocking probability is:

$$P_B=\sum_{i=T}^{C}P(i)$$, where P(i)= π_i

- The HO dropping probability is

$P_{D=}$ P(C) = π_N

Two cases were considered: the high priority calls represent 25% of the total charge in case of WLAN, where due to its high bandwidth it can accommodate more calls and accordingly more HO calls C=60. For the UMTS, the high priority traffic represent 10%, since it has lower bandwidth, and accordingly HO calls are lower.
The blocking probability of the high priority and the low priority are:

$$Bl_{HPR} = \pi_N$$

$$Bl_{BPr} = \sum_{i=s_1}^{N} \pi_i$$

These probability increases with the increase of the system load. For the lower priority, the blocking probability increases with the decrease of the threshold S1=T.
Figures 5.7 and 5.8 show the blocking and dropping probabilities PB and PD in function of the IDM load. It is clear that with the increase of the IDM load, the BP is increased and it will approach 1when the IDM load overcome the threshold T, chosen to be equal to 40% in the analysis. This choice of 40% is due to the fact that the HO calls initiated from WLAN and having enjoyed the high offered throughput should be served with priority, at the same time the local calls should not be severely blocked. Figures 5.8 shows that the handover dropping probability will increase with IDM load and at C, HO calls are dropped from VIDM. Effectively they are not, since they will be guided to HIDM as will be described in the next chapter.

Figures 5.9 and 5.10 show the blocking and dropping probabilities PB and PD in function of the threshold T. The choice of the threshold will highly affects the handover dropping probability as well as the blocking probability of the local calls as discussed earlier.
The variation of these two probabilities with respect to the threshold is given in the figures below. The maximum allowed threshold is when T=C. As T increases, the blocking probability decreases, since the calls (local requests) are accepted until T is reached; whereas, the dropping probability is increased since the probability of dropping the handover calls is increased with no priority scheme.

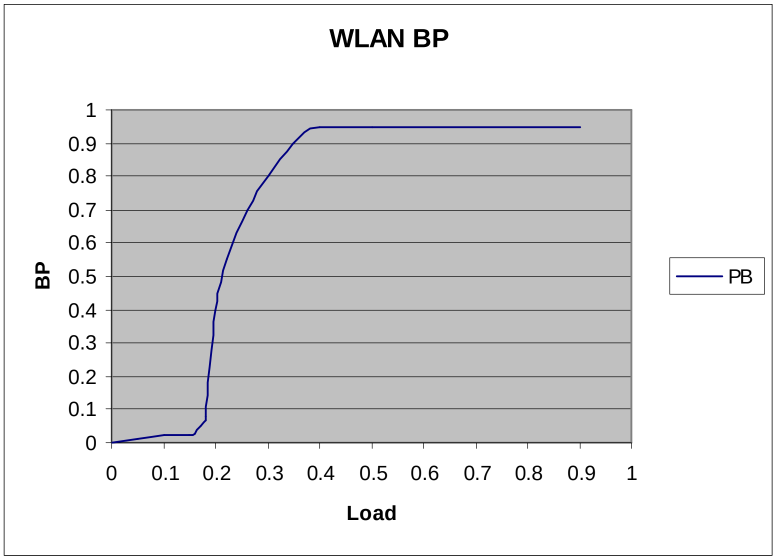


Figure 5.7 Blocking probability in function of IDM load (WLAN)

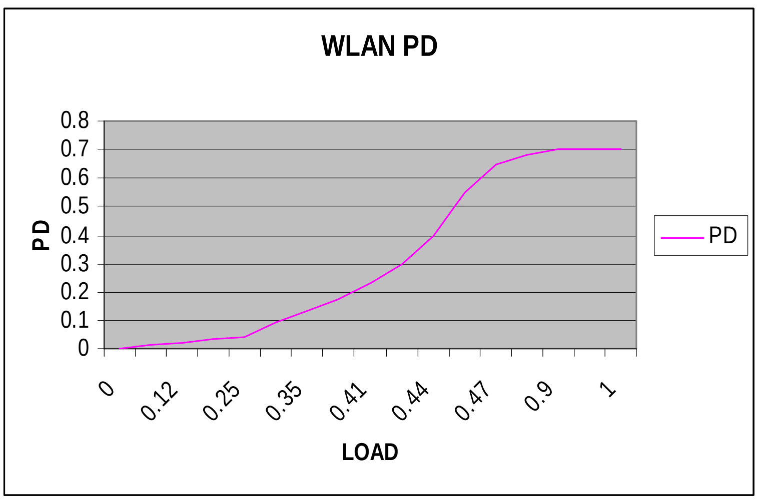


Figure 5.8 Dropping Probability of handover calls

Figures 5.9 and 5.10 show the blocking and dropping probabilities PB and PD in function of the threshold T. The choice of the threshold will highly affects the handover dropping probability as well as the blocking probability of the incoming local calls.
The variation of these two probabilities with respect to the threshold is given in the figures below. The maximum allowed threshold is when T=C. as T increases, the blocking probability decreases, since the calls(local requests) are accepted until T is reached; whereas, the dropping probability is increased since the probability of dropping the handover calls is increased with no priority scheme. IDM "who's next phase" could follow the dynamic threshold change policy.

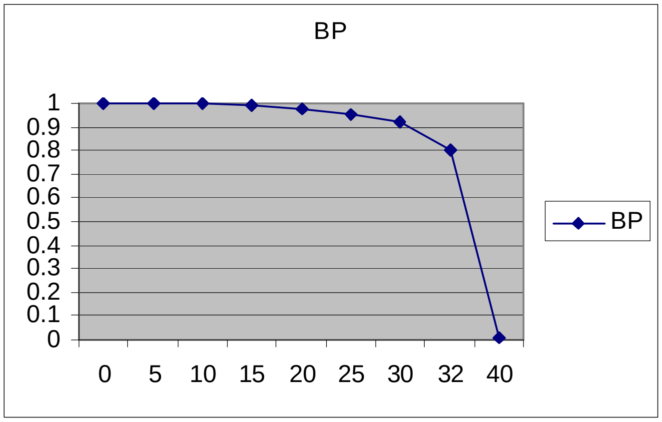


Figure 5.9 BP versus threshold T

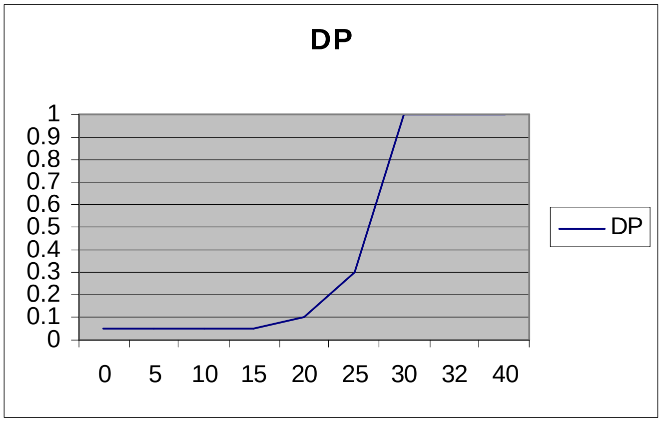


Figure 5.10 DP versus threshold T

5.7.2. UMTS to WLAN

In this scenario, the same policy is applied in order to privilege the HO calls over the local traffic. But the main difference is that in the UMTS network the bandwidth is lower than that of the WLAN so that the capacity C of the IDM is lower then that considered the WLAN. Therefore a 2-thresold hysterisis is introduced to improve the performance and decrease the blocking probability of the local traffic and the dropping probabilities of the HO calls as shown in the following associated Markov Chain [Bay00].

Our system is a server-based queuing system with priority control based on 2 classes of requests treated following a hysteresis model. We consider a finite capacity queuing system based on 2 thresholds. More precisely, we have a forward threshold and a reverse threshold S_2 and S_1.
S_1 and S_2 are the backward and forward threshold respectively between 0 and C.

As long as the buffer occupancy is less than the forward threshold S_1, the two requests are served. If the number of requests goes beyond S_1, only HO requests are accepted. If all C channels are busy, then the VHO call is blocked (but the HO could still be performed horizontally). If the number of calls is less than S_1, the system will not directly allow the local requests to be served until the backward threshold S_2 is reached.

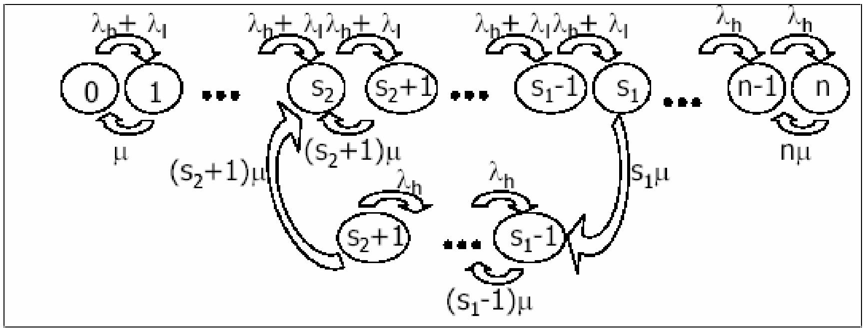


Figure 5.11 IDM model (UMTS to WLAN)

Based on [Fad02], the state probability of state i is Π_i for level 1 and Π_i' for level 2 (the states between S_2 and S_1), and the equilibrium equations are:

- State 0:

$$\lambda_0 \pi_0 = \mu_1 \pi_1 \Rightarrow \pi_1 = \frac{\lambda_h + \lambda_l}{\mu} \pi_0$$

- State 1

$$(\lambda_1 + \mu_1)\pi_1 = \lambda_0 \pi_0 + \mu_2 \pi_2 \Rightarrow \lambda_1 \pi_1 = \mu_2 \pi_2 \Rightarrow \pi_2 = \frac{\lambda_h + \lambda_l}{2\mu} \pi_1 = \frac{1}{2}\left(\frac{\lambda_h + \lambda_l}{\mu}\right)^2 \pi_0$$

- For $1<K<S_2$:

$$\pi_k = \frac{1}{k!}\left(\frac{\lambda_h + \lambda_l}{\mu}\right)^k \pi_0 = A(\lambda_h, \lambda_l, \mu, k)\pi_0 = A_k \pi_0$$

- At state N

$$\mu_N \pi_N = \lambda_{N-1} \pi_{N-1} \Rightarrow \pi_{N-1} = \frac{N\mu}{\lambda_h} \pi_N .$$

- $S_1+1 \leq k<N$:

$$\pi_k = \frac{N!}{k!}\left(\frac{\mu}{\lambda_h}\right)^{N-k} \pi_N = B(\lambda_h, \mu, N, k)\pi_N = B_k \pi_N$$

- At state S_1-1:

$$\pi_{s_1-2} = \left(\frac{\mu(s_1-1)}{\lambda_h + \lambda_l} + 1\right)\pi_{s_1-1}$$

- $s_2+1 \le k < s_1-1$:

$$\pi_k = \sum_{i=0}^{s_1-1-k} \frac{\mu^i}{(\lambda_h+\lambda_l)^i}\frac{(k+i)!}{k!}\pi_{s_1-1} = C(\lambda_h,\lambda_l,\mu,s_1,k)\pi_{s_1-1} = C_k\pi_{s_1-1}$$

$$\pi'_{s_2+2} = \frac{\lambda_h+\mu(s_2+1)}{\mu(s_2+2)}\pi'_{s_2+1}$$

$$\pi'_{s_2+k} = \left[\frac{(\lambda_h+\mu(s_2+1))\lambda_h^{k-2}}{\mu^{k-1}\prod_{i=2}^{k}(s_2+i)} + \sum_{i=0}^{k-3}\frac{\lambda_h^i(s_2+1)}{\mu^i\prod_{s=0}^{i}(s_2+k-s)}\right]\pi'_{s_2+1}$$

- And finally,

$$\pi_n = \frac{(\lambda_h+\lambda_l+s_2\mu)A_{s_2} - (\lambda_h+\lambda_l)A_{s_2-1}}{\mu(s_2+1)(C_{s_2+1}F+E)}\pi_0 = G\pi_0$$

From the above [Fad02], $\pi 0$ is derived in function of all the above transition probabilities whose sum should be equal to 1.

The blocking probability is reduced for the local calls, since S_1 is increased, and the dropping probability of the HO calls is also reduced since due to second threshold, only the HO are accepted and the system is not directly overloaded and the interval S_2-S_1 is higher.

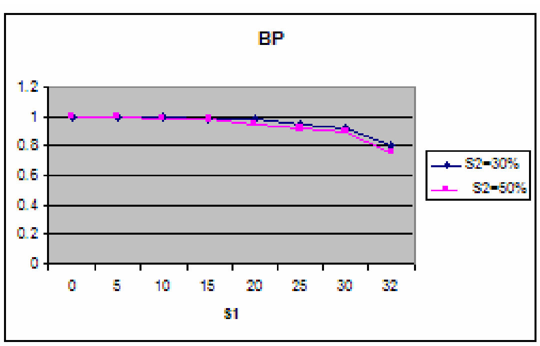


Figure 5.12 S_1 and S_2 effect on BP

In figure 5.12, S_2 is chosen to be 30% and 50% of the total capacity C, and the blocking probability was measured with increasing S_1.

As S_1 is increased, and for higher S_2, the blocking probability will decrease. Whereas, if S_2 is lowered to 30%, the BP will increase. Figure 5.13 shows the effect of varying S_1 for different values of S_2. As S_1 increases, with S_2=50, the dropping probability is increased, since C- S_2 is lower. And similarly if S_2 is set to 30, the DP is lower.

Figure 5.14 shows the variation of the dropping and blocking probabilities versus the IDM load. The results are as expected from the previous section.

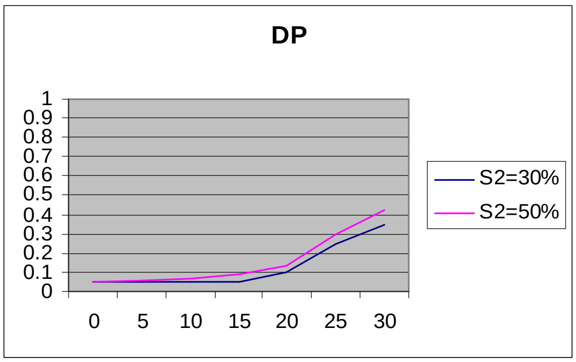


Figure 5.13 Thresholds S_1 and S_2 effect on DP

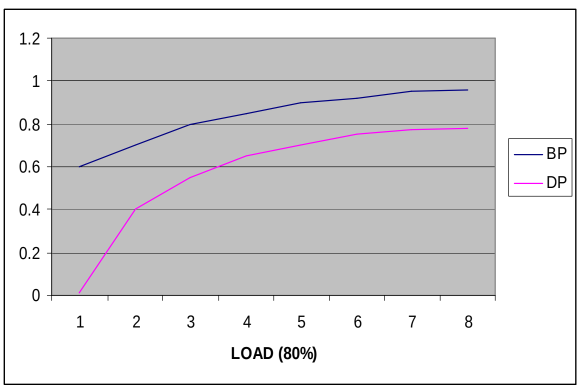


Figure 5.14 PD and PB vs. IDM load.

5.8. Simulation Results

In order to evaluate the performance of our MQMA model, we simulate the proposed model studied in ns-2.27 [VIN]. in order to study the performance results of our proposed model and compare the results. The simulated topology is as shown in figure 5.15. The simulation parameters are specified as follows in table 5.3.

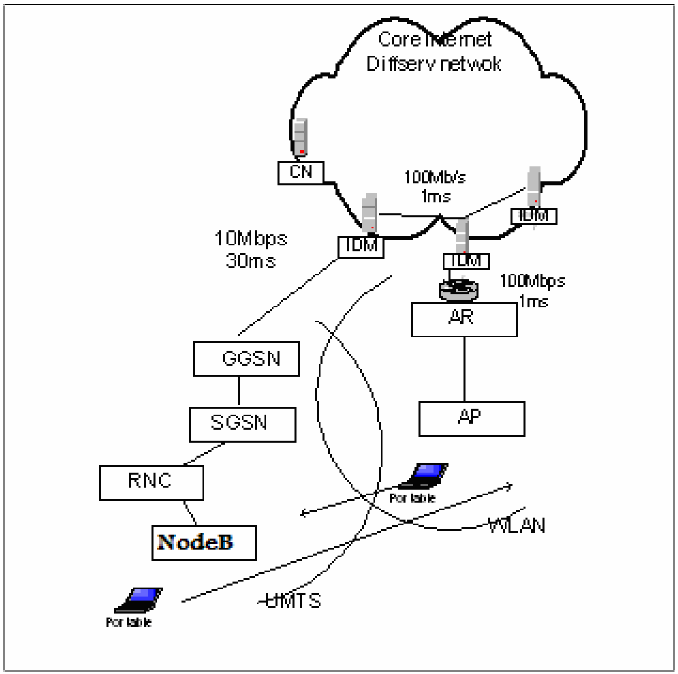


Figure 5.15 Simulated topology

Parameters/Access network	WLAN	UMTS
Bandwidth(Mbps)	10	2
Coverage	300	1000
Mobility	Medium	High
Nodes involved	AP-AR-IDM	NodeB-RNC-IDM-

Table 5.3 Simulated parameters

The EURANE, Enhanced UMTS Radio Access Network Extension, was used to simulate the UMTS network and the nodes involved [Eur].
The nodes used to represent AP are set as defined in 802.11b standard [Cro97], with Two Ray Ground propagation model. The Mobile Node (MN) has two interfaces and performs switching while performing handover either from UMTS to WLAN or WLAN to UMTS. The IDM processing time is 200µs. The IDM simulated node is based on M/M/C/C model, with capacity C. Since C is a function of the total number of users and the network bandwidth, we take C_{UMTS}=60%*C_{WLAN}, so that the capacity of IDM in WLAN is higher than that of the IDM in UMTS. To simulate the load at a specific IDM, many requests are started from the neighbored IDM nodes to represent the local request and the HO requests. If the number of request is less then the IDM capacity, the HO request is guided vertically. This is illustrated in the vertical handover from WLAN to UMTS or vice versa. Otherwise, the handover is guided horizontally. The types of traffic generated are: the HO represented by real time traffic represented by a CBR source of 140kbps, packet size is 150 octets. The local traffic is represented as CBR source with 130kbps.

-Traffic scenario:

-the local traffic is initiated to increase the IDM load. When the mobile node will loose network coverage to start the HO process at t=80s, this background traffic is introduced on order to charge the network.

-Simulation time is 300s

-At t=80s, the handover started. The mobile node will move from one access point to another in different domain based on AVHO described in chapter 3, and the MQMP protocol described earlier.

-The following assumptions were made:

From WLAN to UMTS, no hysterisis is considered. From UMTS to WLAN, hysterisis is considered and the topology is as shown above.

In this case, we simulate an additional node where only the handover requests are treated and it is not affected by the local traffic treated at the IDM. This is just to prioritize the handover and isolate them from any local traffic and realize the hyterisis model.

-In the M/M/C/C/ model, the handover call is of high priority compared to the local calls.

-The traffic exchanged locally has lower priority. C-T is the guard band channels used only for handover calls.

-A new call request is admitted if they are less then C channels occupied.

-A handover request is rejected if all the C channels are occupied.

-The local traffic is a poisson process of rate λ_l

-The handoff request is a poisson process of rate λ_h

-The time of stay in IDM is exponentially distributed of rate $1/\mu$.

The system is simulated in ns2.27 [VIN], and the above parameters were used to calculate the RTT in UMTS and WLAN, and HO latency so that the total delay is known and the performance parameters are calculated.

Based on the operation process shown in figure 5.16, and based on the specific parameters of the UMTS and the WLAN networks shown in table 5.3, we calculated the delay involved in the handoff from WLAN to UMTS and vice versa.

End-to-end delay was calculated in function of RTT time (simulated in both UMTS and WLAN). Since the signaling exchange will occur in the WLAN or UMTS network, the exchange should be considered for these different networks separately. Control packets exchanged between IDM are of 1500 octets (maximum MTU). The link between IDM is 100Mb/s in the IP network.

The analysis was decomposed into four phases. Phase1 is the initiation phase composed of FMIP initiation messages and the Context transfer messages exchanged. Phase2 is the selection phase based on the "WHO'S NEXT" operation of the involved IDM. This phase is related to the processing time of the IDM node assumed to be 200microsec. Phase 3 is the reservation phase according to the selected access network (WLAN or UMTS) and the user profile, and finally phase4 where the FMIPv6 operation is resumed by the exchange of the 9 last messages for updates and HO termination.

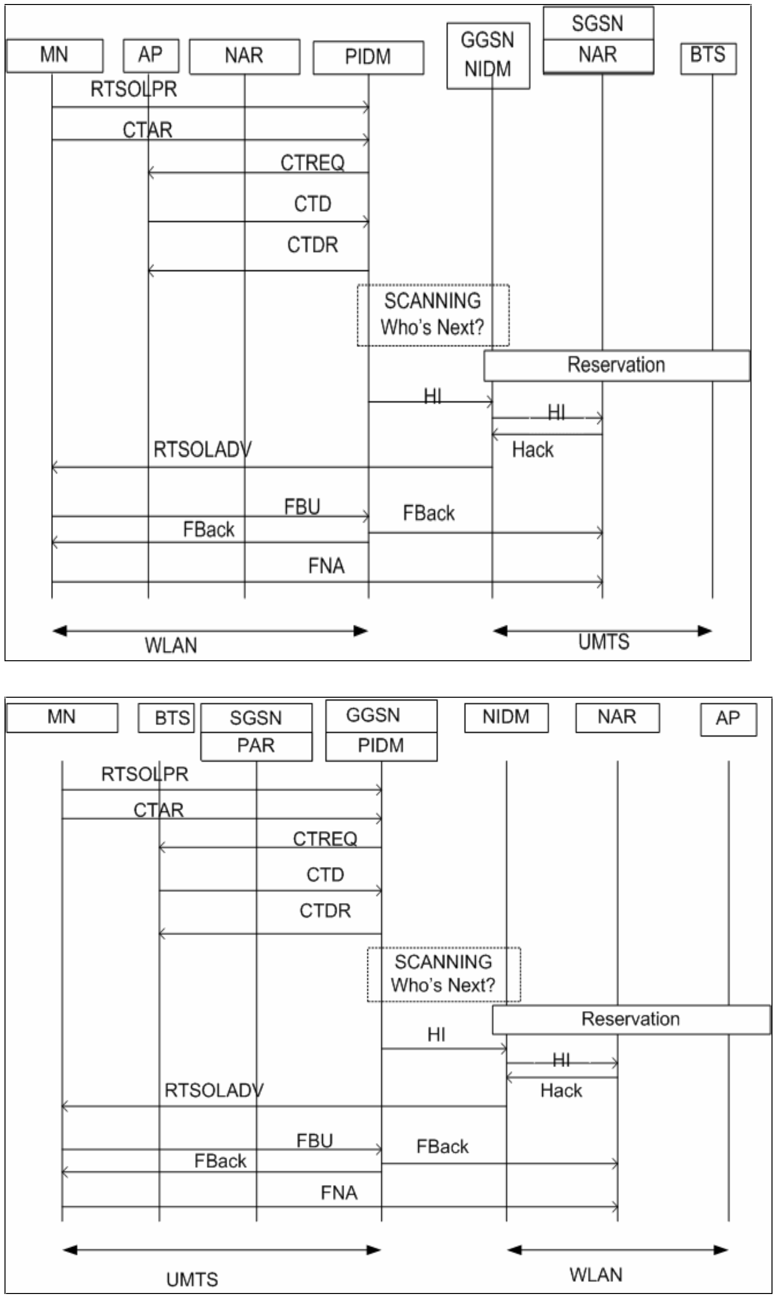


Figure 5.16 MQMP operation

Average values of these execution and initiation phases were obtained for several HO scenarios where the significant difference is according to whether the handover occurs in the UMTS or WLAN. Table 5.4 shows the total delay for the HO to be terminated. The delay values are higher but still very close to those obtained with MIPv6 signaling, or FMIP without CT [Raj03]. The overall system performance is improved due to the QoS offered according to the above described architecture.

T_1 is the timer limitation for resource reservation set at the IDM node: timer is fixed to 2*the time required for the mobile node to perform handover after the new network is selected. If the handover is not performed, the timer expires and the resources are freed. During the handover exchange, if the resources are reserved but the HO didn't continue due to transmission error or others, the resources are freed after the timer expires, the IDM will reinitiate the "Who's next" phase and allocate another IDM resources. If the HO failed and no available resources were found, horizontal HO will take place.

Measuring the average time take by each phase, the total handover delay could be determined.

Simulation shows that the time between the last packet received on the PIDM and the first packet received on the NIDM is determined by phase4 which is the phase where the MN disconnect from the PAR and reconnect to the NAR which is approximately 240ms in WLAN and 800ms in UMTS.

The packet ID received by the MN shows also no packet loss. Packets are also forwarded from PIDM to NIDM in the core network, once the mobile leaves the PAN. Then these packets are forwarded to the MN (time to forward a message is 20µs).

	WLAN	UMTS
Phase1	100ms	500ms
Phase2	200µs	200µs
Phase3	150ms	500ms
Phase4	240ms	800ms
Total HHO delay (ms)	WLAN to UMTS 840.2	UMTS to WLAN 1450.2

Table 5.4 Simulated results

Figure 5.17 show the variation of blocking and dropping probabilities in function of the IDM load from UMTS to WLAN handovers.

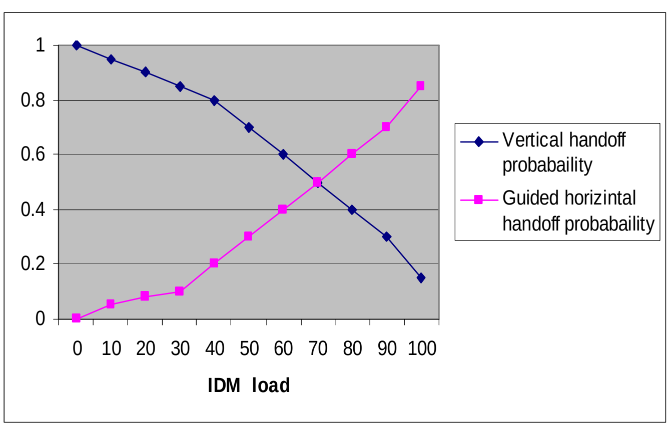


Figure 5.17 blocking and dropping probabilities

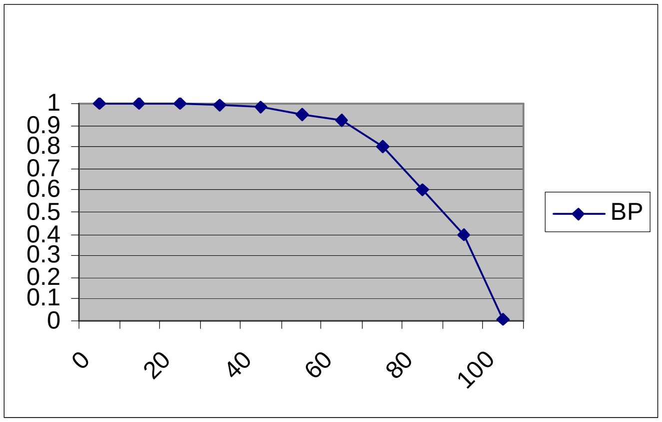


Figure 5.18 PB in function of the threshold T

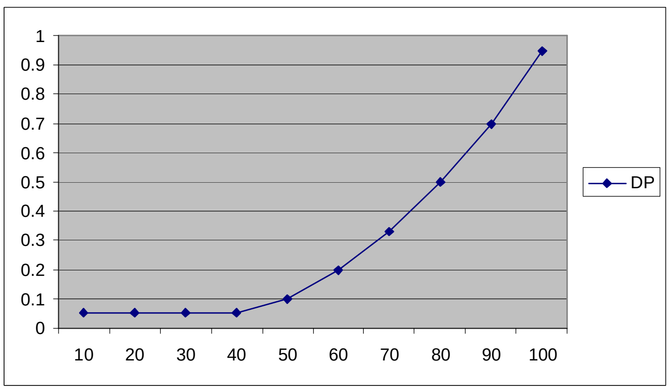


Figure 5.19 PD in function of the threshold T

The threshold effect is consistent with the analytical result at 75% of the load as expected.

Figures 5.18 and 5.19 show the blocking and dropping probabilities PB and PD in function of the threshold T. The choice of the threshold will highly affects the handover dropping probability as well as the blocking probability of the incoming local calls.

The variation of these two probabilities with respect to the threshold is given in the figures below. The maximum allowed threshold is when T=C. as T increases, the blocking probability decreases, since the calls(local requests) are accepted until T is reached; whereas, the dropping probability is increased since the probability of dropping the handover calls is increased with no priority scheme.

In figure 5.20, S_2 is chosen to be 30% and 50% of the total capacity C, and the blocking probability was measured with increasing S_1 from WLAN to UMTS scenario.

As S_1 is increased, and for higher S_2, the blocking probability will decrease; whereas, if S_2 is lowered to 30%, the BP will increase.

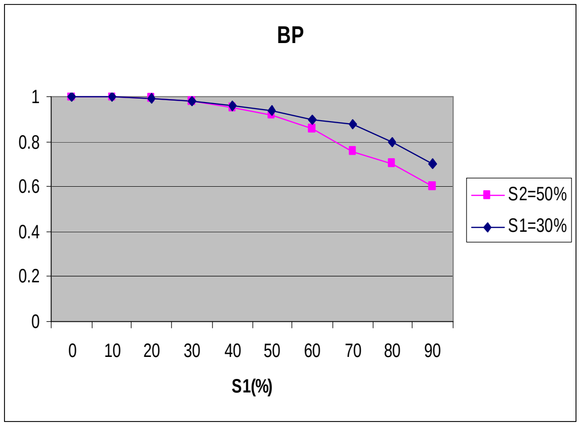


Figure 5.20 BP Vs S_1

5.9. Conclusion

In this chapter, we present a vertical handover architecture for next generation all-IP networks. The proposed MQMA manages the mobility and QoS context between heterogeneous access networks. To provide the required level of QoS context when moving between different access technologies, an IDM gateway was introduced between the Diffserv network and the heterogeneous access networks to serve the handover request. The IDM is responsible of supporting mobility and resource management by guiding the mobile node to the appropriate network based on the IDM "Who's next" phase.

Based on our MQMA model, we can deduce the following conclusions:

- The NGN integration based on our proposal MQMA is able to integrate any heterogeneous wireless system by just registering with the IDM and enabling inter IDM communication to perform vertical handovers seamlessly.

- The next network selection processed by the IDM is transparent to the different the access technologies, and handled by the IDM.

- The HO call could be accepted horizontally (for inter-handovers scenarios) and guided to the same access technology, thus providing the QoS context for Horizontal Handovers HHO.

In the next chapter, the horizontal handover is considered in order to prove the efficiency of our proposal in next generation networks integration. Horizontal mobility issues with no QoS degradation for inter UMTS and inter WLAN handovers are studied. In this way, MQMA will be extended to provide horizontal handover (HHO) for the same purpose to perform transparent and seamless handovers. In addition, the HHO will also serve to perform vertical handovers in the case where no resources are available for the VHO to take place. This is the MQMA application for vertical to horizontal handovers.

Chapter 6

Mobility and QoS Management for Vertical-to-Horizontal Handovers

6.1. Introduction

In the previous chapter, we proposed the Mobility and QoS Management Protocol MQMP and showed its performance in vertical handovers. In this chapter, we will present the inter handover mobility and show how the horizontal handover could be resolved using the previously described protocol and architecture. As we have seen in the previous chapter, the vertical handover could be guided to horizontal handover or to the same access technology if no sufficient resources are available to perform the vertical handover. In this case of horizontal handover, the QoS is highly provisioned and the user profile is respected since the HIDM will interfere in the handover process to make seamless and transparent handover.

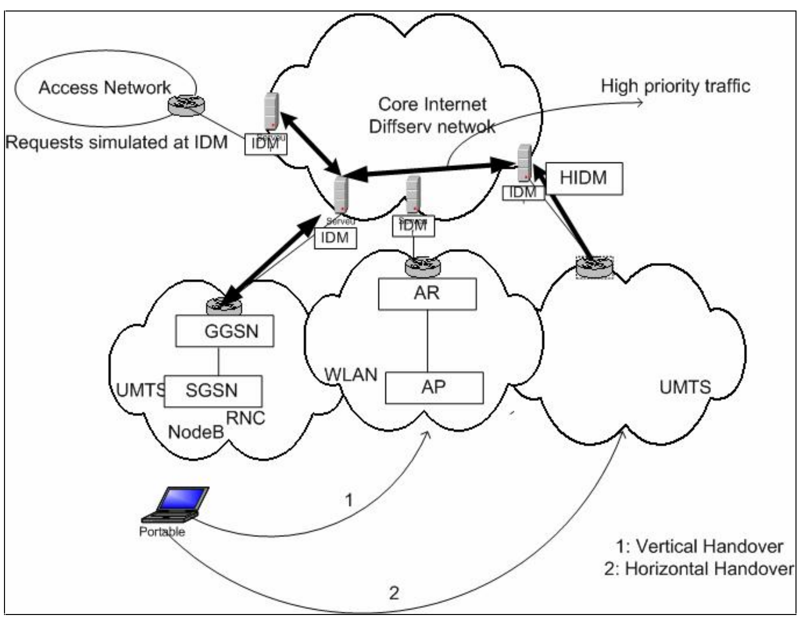


Figure 6.1 MQMA for horizontal and vertical handovers

6.2 HIDM functionalities

The HIDM functionalities are similar to VIDM, with the exception that they will perform HHO between same access technologies. In this case the HO is performed through the HIDM in the original network which will communicate with the HIDM in the next network, as shown in figure 6.1. The HIDM operates as a handover management unit, a seamless roaming unit and a seamless horizontal to vertical handover.

- **HO management unit**: the IDM will manage the HO by searching for the next network capable of offering the end user the required QoS parameters. the best network selection is based on the network conditions, context transfer…If the profile is highly privileged, the HO could be guided through the SIDM to WiMax network in order to accept the HO with the requested QoS.

- **Seamless roaming unit**: all the signaling should be done before the MN moves, or before being out of coverage. This is done by first initiating the HO request when the MN is still connected to the network. This pre-active behavior, based on the make-before break approach, will guarantee that the mobile node will do all the signaling issues before being disconnected from its current network (before loosing coverage), and without degrading the QoS parameters in the original network.

- **Seamless horizontal to vertical handover**: To reduce HO failure probability which occur if the IDM queue is full (since it is modeled as M/M/C/C/, the HO call could be rejected). In this case, the mobile node will not wait until all the calls in the queue are served. This will cause a significant delay since the HO request is not directly treated. The solution is to guide the HHO call to the SIDM responsible of WiMax HO. SIDM will perform the same procedure in terms of reserving resources and scanning for the appropriate IDM through the "Who's Next" phase. In this case, the HO is performed vertically since the SIDM guide the HO to WiMax network (as described in chapter 6).

6.3. The inter-IDM Model

The proposed architecture is shown in figure 6.1 where three types of IDM are presented: VIDM, HIDM and SIDM. We described in the previous chapter the role of the VIDM while performing handovers. VIDM role is to accomplish successfully vertical handovers between heterogeneous access networks. HIDM is concerned with the inter handovers or the mobility belonging to the same access technologies. SIDM will be described in the next chapter.

The second IDM level or HIDM will receive HO call from the users performing HHO and from VIDM forwarding the blocked HO to HIDM to successfully guide the mobile requesting handover.
For micro mobility, the signaling exchange between IDM's belonging to the same access network, and the handover performance measures depend on WLAN and UMTS networks for inter-WLAN and inter-UMTS handovers .

6.3.1. HIDM modeling

At each HIDM, the traffic is of 2 types: The previous HO requests (from HIDM) considered as the traffic of higher priority. The local HO requests and the local traffic considered as the traffic of low priority.
In the core network, the traffic between 2 HIDM's is transferred through the core network as high priority traffic. Figure 6.1 shows the interconnection between homogeneous and heterogeneous access network together with HIDM and VIDM interception.

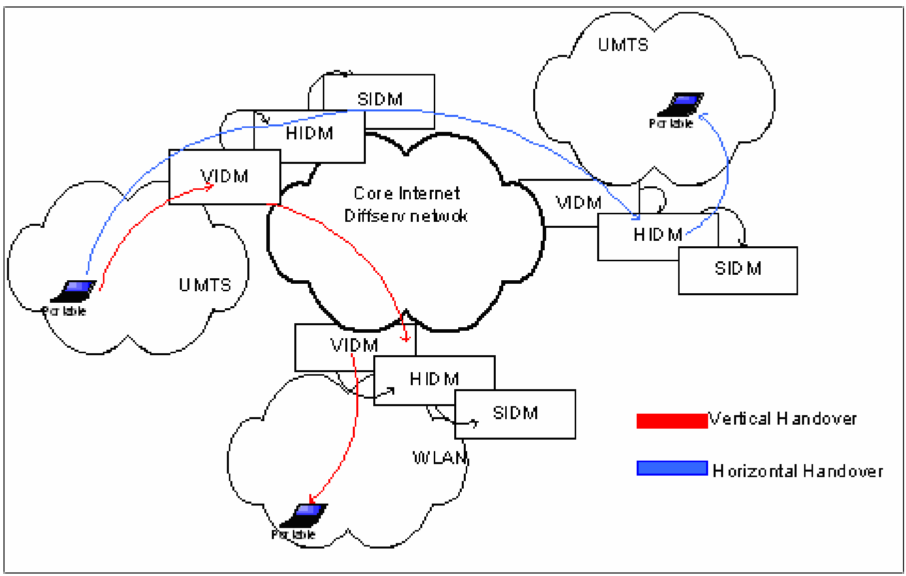


Figure 6.2 Horizontal handover

6.4. Inter WLAN model

As shown in figure 6.2, the horizontal handover is performed following the MQMA operations:

- MN sends RtsolPR to PAR to indicate HO initiation and to start the operation of acquiring a new address and performing seamless HO.
- MN sends CTAR to PAR and PIDM prior to HO. this message contains MN IP address and PIDM IP address, and the entire context to be transferred (by default). It also contains token to be used by NIDM for verification…
- CTReq from PIDM to PAR: to update the context already available in the entry of the PIDM
- PAR starts to send CTD to PIDM with MN IP address
- PIDM will start scanning (based on CTD) for a NIDM whose resources will satisfy the required context of MN asking for HO, once the authentication token verified.

- HI and all the FMIPv6 signaling proceed to perform handover to the new access network.

CTDR reply to notify that the CTD was successful (if CTD has the A flag ON)
CTAA Reply to notify that CTAR received, from NIDM to MN. It contains list of FTP's that were not successfully transferred.

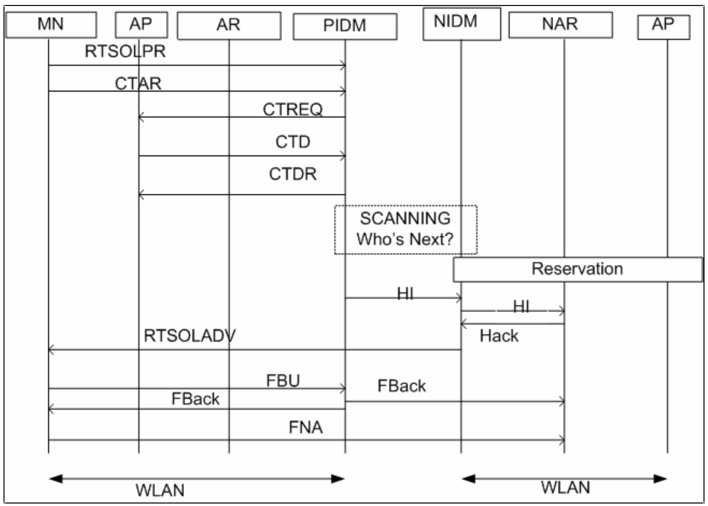


Figure 6.3 Inter WLAN model

The mobility and QoS Management Protocol MQMP is applied for the inter WLAN model as shown in figure 6.3.The same analysis is applied on the handover for inter WLAN and inter UMTS.
The IDM modeling is as considered in the previous chapter, but with equal IDM capacity in both initial and destination networks on the case of horizontal handover.
Since WLAN has more bandwidth and can accommodate more calls, the following Markovien model is used. In this model the guided HO calls are privileged over the local HO calls. In fact the local calls are accepted until the threshold T is reached, but the guided HO call will be accepted even if the T channels are occupied, thus only the guided HO calls are accepted between C and T.

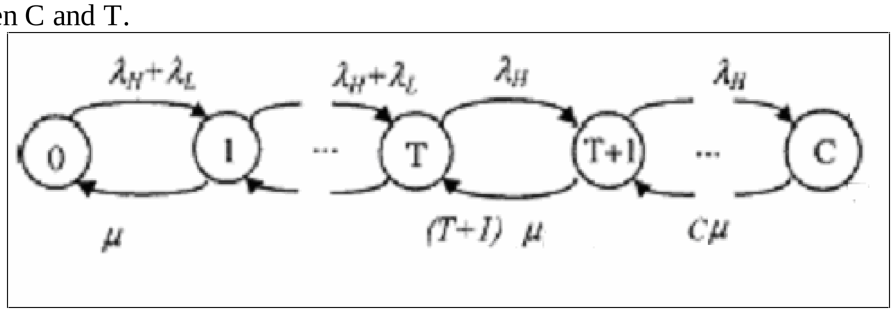


Figure 6.4 IDM model for WLAN handovers

Assuming the service time is equal, $\mu_h = \mu_l = \mu$, the stationary probabilities of the different states are [Fad02]:

- For i = 1,2, … T

$$\pi_i = \frac{1}{i!}\left(\frac{\lambda_h + \lambda_l}{\mu}\right)^i \pi_0$$

Similarly to the previous analysis in chapter 4, and based on [Fad02],

$$\pi_0 = \left[1 + \sum_{i=1}^{s_1} \frac{1}{i!}\left(\frac{\lambda_h + \lambda_l}{\mu}\right)^i + \sum_{i=s_1+1}^{n} \frac{1}{i!}\left(\frac{\lambda_h + \lambda_l}{\mu}\right)^{s_1}\left(\frac{\lambda_h}{\mu}\right)^{i-s_1}\right]^{-1}$$

Where S_1 is the first threshold corresponding to T and the dropping and blocking probabilities are

$$P_B = \sum_{i=T}^{C} P(i)$$

$$P_D = P(C)$$

As expected, the inter WLAN model has the same BP and DP compared to VHO since the IDM in interest is that of the original network. This latter could be either UMTS or WLAN according to whether the HO is performed vertically or horizontally. The modeling of the IDM is the same, hence the dropping probability and the blocking probability that when horizontal handovers are:

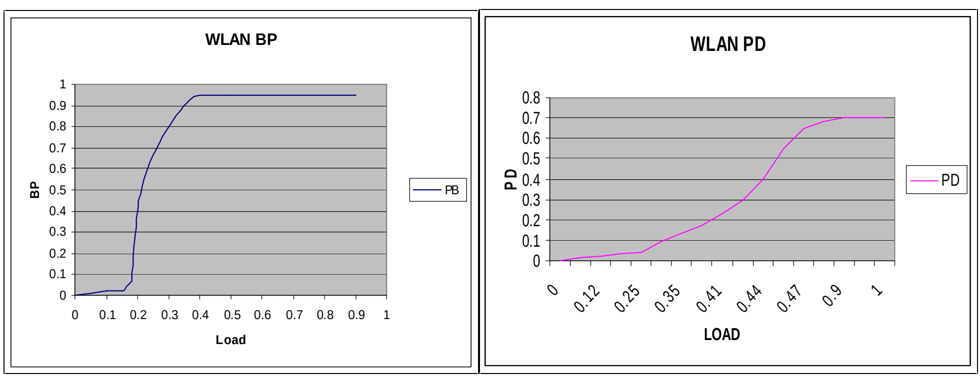


Figure 6.5 DP and BP Vs IDM load

6.5. Inter UMTS model

In this case, the guided HO calls are privileged over the local HO traffic. In the UMTS network, the capacity is lower then that of WLAN due to the lower bandwidth. Therefore a 2-thresold hysterisis is introduced to improve the performance and decrease the blocking probability of both the local traffic and the HO calls.

We used the following associated Markov Chain to model the inter UMTS handover.

Our system is a server-based queuing system with priority control based on 2 classes of requests treated following a hysteresis model. We consider a finite capacity queuing system based on 2 thresholds. More precisely, we have a forward threshold and a reverse threshold S_2 and S_1.

S_1 and S_2 are the backward and forward threshold respectively between 0 and C.

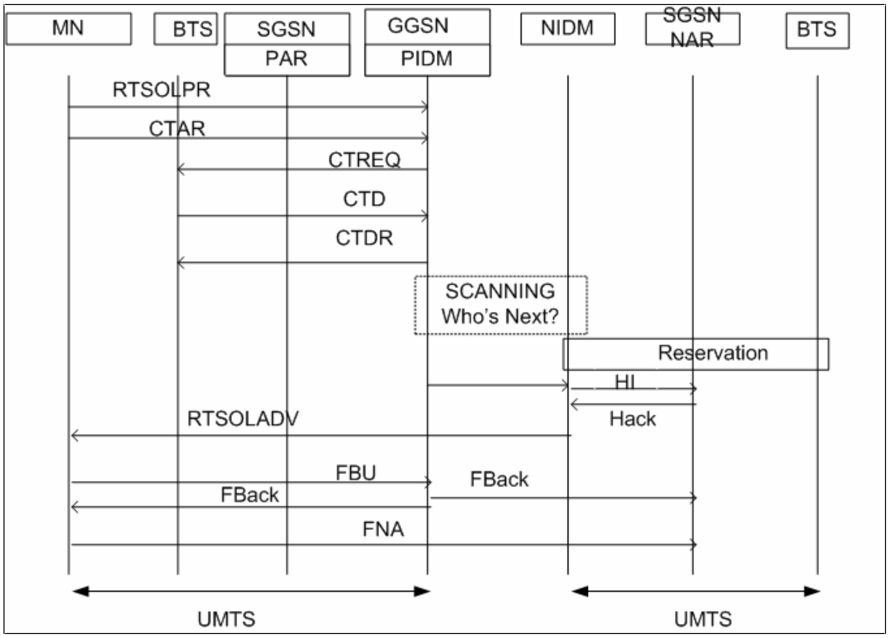


Figure 6.6 UMTS to UMTS handover

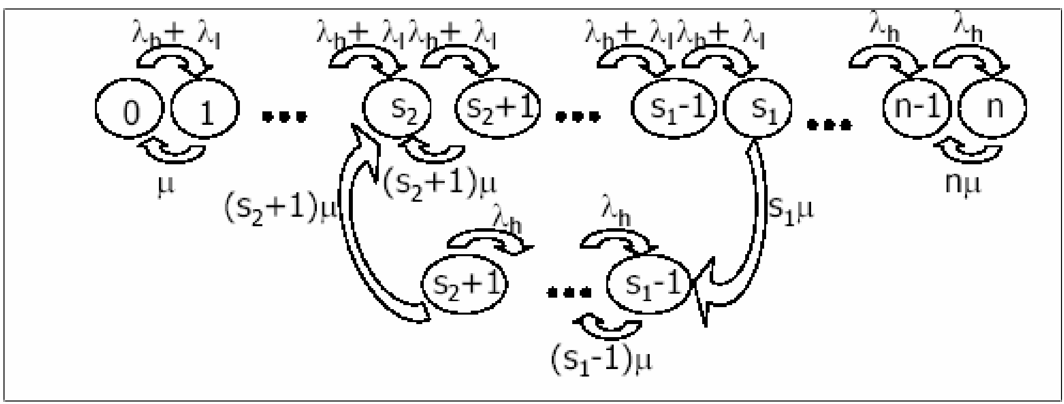


Figure 6.7 IDM model for UMTS handover

In the applied model, as long as the buffer occupancy is less than the forward threshold S_1, the two requests are served. If the number of requests goes beyond S_1, only handover requests are accepted. If all C channels are then the handover call is blocked. The hysteresis issue relies on the concept of privileging class 1 requests over those of class 2 requests. In that way, if the number of calls is less than S_1, the system will not directly allow the local request to be served until the backward thresholdS_2 is reached. In the case of inter UMTS model, the BP and DP are as shown in figure 6.8.

6.6. Simulation results

Inter WLAN and UMTS could be simulated using the previous analysis made in the previous chapter. End-to-end delay was calculated in function of RTT time (simulated in both UMTS and WLAN). Since the signaling exchange will occur in the WLAN or UMTS network, the exchange should be considered for these different networks separately. Control packets exchanged between IDM are of 1500 octets (maximum MTU). The link between IDM is 100Mb/s in the IP network.

The analysis was decomposed into four phases. Phase1 is the initiation phase composed of FMIP initiation messages and the Context transfer messages exchanged. Phase2 is the selection phase based on the "WHO'S NEXT" operation of the involved IDM. This phase is related to the processing time of the IDM node assumed to be 200microsec. Phase 3 is the reservation phase according to the destination access network (WLAN or UMTS) and the user profile, and finally in phase4, the FMIPv6 operation is resumed by the exchange of the 9 last messages for updates and HO termination.

Average values of these execution and initiation phases were obtained for several HO scenarios where the significant difference is according to whether the handover occurs in the UMTS or WLAN. Table 4.3 shows the total delay for the HO to be terminated. The delay values are higher but still very close to those obtained with MIPv6 signaling, or FMIP without CT [Raj03]. The overall system performance is improved due to the QoS offered according to the above described architecture.

T_1 is the timer limitation for resource reservation set at the IDM node: timer is fixed to 2*the time required for the mobile node to perform handover after the new network is selected. If the handover is not performed, the timer expires and the resources are freed. During the handover exchange, if the resources are reserved but the HO didn’t continue due to transmission error or others, the resources are freed after the timer expires. The IDM will reinitiate the "Who's next" phase and allocate another IDM resources. If the HO failed and no available resources were found, horizontal HO will take place.

	WLAN	UMTS
Bandwidth (Mbps)	10	2
Coverage (m)	300	1000
Mobility	medium	High
Nodes involved	AP-AR-IDM	NodeB-RNC-IDM-

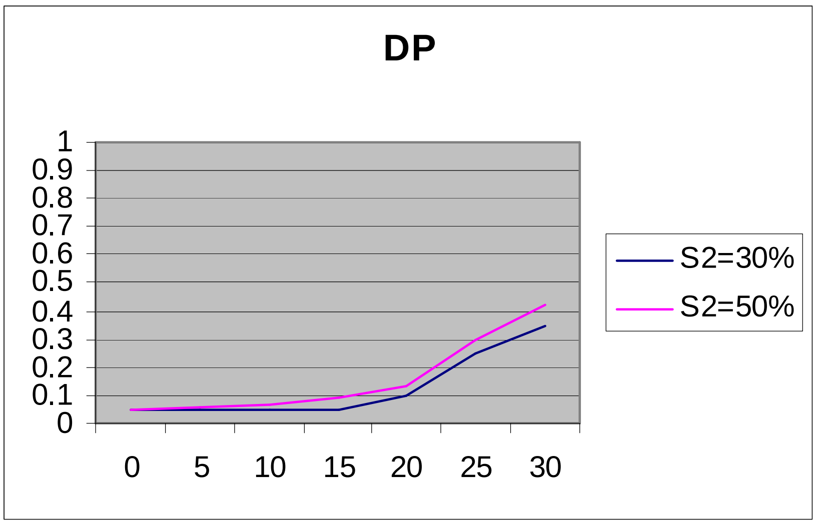
Table 6.1 Simulated parameters

Measuring the average time taken by each phase, the total handover delay could be determined. Simulation shows that the time between the last packet received on the PIDM and the first packet received on the NIDM is determined by phase4 which is the phase where the MN disconnect from the PAR and reconnect to the NAR which is approximately 240ms in WLAN and 800ms in UMTS.

Packets are also forwarded from PIDM to NIDM in the core network, once the mobile leaves the PAN. Then these packets are forwarded to the MN (time to forward a message is 20µs).

	WLAN	UMTS
Phase1	100ms	500ms
Phase2	200µs	200µs
Phase3	150ms	500ms
Phase4	240ms	800ms
Total HHO delay (ms)	490.2	1800.2

Table 6.2 Simulated results



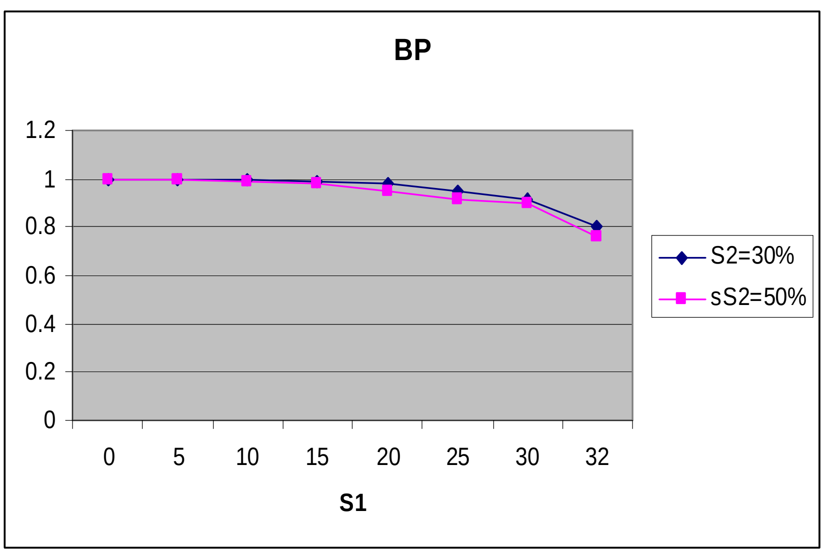


Figure 6.8 PB and PD vs. Thresholds S_1 and S_2

6.7 Conclusion

The horizontal handover considered in this chapter is the complementary part of our proposed solution for vertical handovers. In our proposition, we considered that a HHO could occur when the VHO could not be performed successfully or seamlessly. In this term, when no available resources are possible to perform the VHO requesting specific context parameters, the HHO will take place. The proposed solution of QoS context provisioning and seamless handover is also applicable for horizontal handover. As a result, the inter-access technology handover or horizontal handover (HHO) could also be performed seamlessly between same access technologies.

In the case of no available resources are possible to perform HHO, the emerging WiMax network is considered for the next generation networks integration so that the handover is performed vertically based on our proposed MQMA. As a result, WiMax is considered for the vertical handover as a stand-by destination network from UMTS or WLAN as illustrated in the next chapter.

Chapter 7

WiMax Integration with MQMA model

7.1. Introduction

In next generation networks tending to integrate different access technologies, for the purpose of offering the user the best possible service, the evolving WiMax 802.16 will play an important role in this integration. In fact, where WiFi provides high bandwidth but short coverage, and current cellular systems provide high coverage but low bandwidth, WiMax will provide both.

Worldwide Interoperability for Microwave Access, or WiMax, intended mainly for the exchange of data at home or in the office, has the potential to provide a significant improvement in cost and performance compared to existing wireless broadband access systems. IEEE 802.16e will enable a new set of high-speed nomadic and mobile data services over a wide metropolitan area with lower- cost solutions, higher performance and reliability. Coverage will be based on large cells interconnected to provide the user with seamless high data rate (several Mbit/s). The ability to maintain connection while moving across cell borders is a prerequisite for mobility and will be included as a requirement in 802.16e system profile [Fin04].

802.16e adds hand-off capability, thereby supporting portability and mobility. Operating in the 2-11GHz range, it is designed for point-to-multipoint applications and does not require LOS [Int04]. Since our purpose is the all IP wireless network mobility, we will only consider the 802.16e where the mobility is supported. 802.16e supports different types of handoff ranging from hard to soft and it is up to the operator to choose among them.

Hard handoff use a break before make approach, where the user is connected to one base station at a time. This is less complex than soft handovers but has high latency.
Soft handovers are comparable to those used in cellular networks and allow the user device to retain connection to a base station until it is associated with a new one "make before break" approach, thus reducing latency. The WiMax forum does not expect to include roaming requirements in 802.16e system profile, as roaming is a higher level capability that goes beyond the scope of the certification program, which focuses on the physical and the mac layers. The service providers working groups and the network working group within WiMax forum are

working towards identifying the functional requirements for roaming and establishing a roaming platform. The WiMax forum expects that initial products will support only simple mobility which uses hard handoff and does nor support real time applications.

WIMAX-802.11e supports three types of mobility. The first one is the portability as laptops enhanced with PCMCIA or mini cards at walking speed and performing hard handoff. The second type is the simple mobility with PCMCIA laptops at low speed, and performing hard handoff. And finally the full mobility with PCMCIA laptops moving at high speed with soft handoff.

Being a wireless radio access technology, WiMax cannot avoid being compared to WiFi. However, this technology is much more efficient in several areas. It offers a maximum useful throughput of 60 megabytes per second (Mbps) in a 20-MHz channel, as compared to 25 useful Mbps under standards 802.11 a or g. The quality of service of WiMax is also superior, it also provides greater security as the final equipment must be declared before being connected to the network. These differences in performance must not incite us to jump to the conclusion that WiMax will replace WiFi. If the standard is adopted, these two technologies from the same family should cohabit and complete each other. In addition, optimal coverage, very high throughput, reduced costs, a wider range of services, etc. WiMax has all the advantages to impose itself and to compete with UMTS, or third generation mobile telephony, which proposes similar applications. WiMax and UMTS are also the complementary technologies. Thus, this chapter will present the integration of 802.16 standards with WiFi and UMTS. This is in the context of proving that our proposal is a scalable and could be applied in considering the vertical handover between WiMax, WLAN, and UMTS access technology. The WiMax will be the destination of the guided handover in order to provide seamless handover and the required QoS instead of dropping the handover call as will be described in this chapter.

7.2. WIMAX 802.16e specifications

Standard 802.16e will allow for the use of WiMax in situations of mobility. Travelling speeds could exceed 100 km/h, and the advantage of maintaining sessions when changing connections, or handover is possible [Fin04]. Furthermore, this new technology will also be capable of receiving the best 802 standard signal, which is choosing between WiFi and WiMax.

The non-line-of-sight technology, IEEE 802.16e is based on orthogonal frequency division multiplexing (OFDM) and OFDM with multiple access (OFDMA) bringing improved levels of spectral efficiency, data throughput, and capacity compared to previous generations of radio technologies. IEEE 802.16e standards have flexible channel bandwidths between 1.5 and 20 MHz to facilitate transmission over longer ranges and to different types of subscriber platforms.

7.2.1. Seamless mobility

Ensuring seamless mobility is another challenge presented by WiMAX systems and must be considered from two perspectives: handover and security.

Handover: The challenge with handover in a WiMAX system lies in the fact that smooth handover requires a “make before break” sequence – that is, before a mobile device moves out of range of one base station, it begins communicating with a second base station before disconnecting from the first. Existing OFDM-based systems use the opposite “break before make” approach, in which the mobile device momentarily disconnects from the first base station before it starts communicating with the next. This “break before make” approach worked well for best-effort packet data services, such as web browsing, but for interactive multimedia traffic such as voice and video, it would result in delay, interruption, or disconnection and therefore an unacceptable user experience.

The “make before break” scheme relies on FBSS and MDHO, described earlier. Spearheaded by Nortel, both are advanced OFDM mobility technology innovations that have been contributed to the IEEE 802.16e standard, in collaboration with other companies.

7.2. 2. Physical characteristics

To ensure that users constantly receive the fastest data rate possible, 802.16e employs fast link adaptation and modulation/ coding techniques, which enable WiMax systems to dynamically adapt the coding modulation — for instance, quaternary phase shift keying (QPSK) or quadrature amplitude modulation (QAM) — to accommodate varying conditions that can often occur on the radio link between the user and the base station [Gag05]. These conditions could include such impairments as interference from buildings or trees or fading of the signal as the user moves farther away from the base station. This diagram shows the OFDMA PHY (physical) coding modulation set operating at 20 MHz bandwidth, and the corresponding peak data rates.
802.16 supports [Gago5]:

- nomadic and mobile operation with wide area coverage or fixed/hot spot applications
- OFDMA in time division duplex (TDD) and frequency division duplex (FDD) operations
- scalable OFDM/OFDMA with carrier requirements from 1.25 MHz to 20 MHz bandwidth
- flexible frequency reuse pattern
- fast link adaptation and modulation/ coding
- high-efficiency coding and error correction schemes
- multiple dimension of diversity
- MIMO (multiple-input multiple-output) technology

7.2.3. WiMax architecture

A broadband wireless access WiMAX-enabled network includes the following key elements: The base station provides connectivity over the radio link and manages radio link resources. It is responsible for physical layer functions (e.g., adaptive modulation and coding); radio resource management and scheduling; radio link retransmission (ARQ/HARQ); packet segmentation/reassembly; packing/unpacking; and traffic encryption and frame authentication.

The mobile control point (MCP) provides the control and mobility anchor point for a mobile station (MS) as it moves between base stations (BSs) in the access **network.** The MCP is responsible for device and subscriber authentication; service authorization; security key management; accounting; handover and macro diversity coordination; downlink traffic replication and distribution; and uplink traffic selection and forwarding. It is important to understand that unlike BSCs or RNCs in CDMA networks, the MCP in a WiMax network does not contain base station control functions; these functions reside in the base stations themselves.

Network Operations and Support Services (NOSS) include functions required to operate and maintain the wireless access network. These include element management; authentication, authorization, and accounting (AAA) services; and MS IP configuration services.

A datagram distribution network provides full connectivity between all MCPs and all BSs in the access network. It may be a third-party network or may be built as an overlay on a third-party network. Its primary functions are traffic aggregation and distribution; the latter is implemented as a datagram delivery service, meaning that an IP (or Ethernet) datagram injected at an ingress point is delivered error-free to the egress point associated with the destination address in the datagram header. Since part of the distribution network may be a third-party network, all traffic between an MCP and BS is carried in a secured tunnel (such as IPsec ESP).

Mobile stations can move between BSs while maintaining service continuity. Session and mobility management (SMM) is introduced into a separate network entity called the mobile control point (MCP) to redirect traffic to the BS currently serving an MS and to manage the handover between base stations [Fin04].

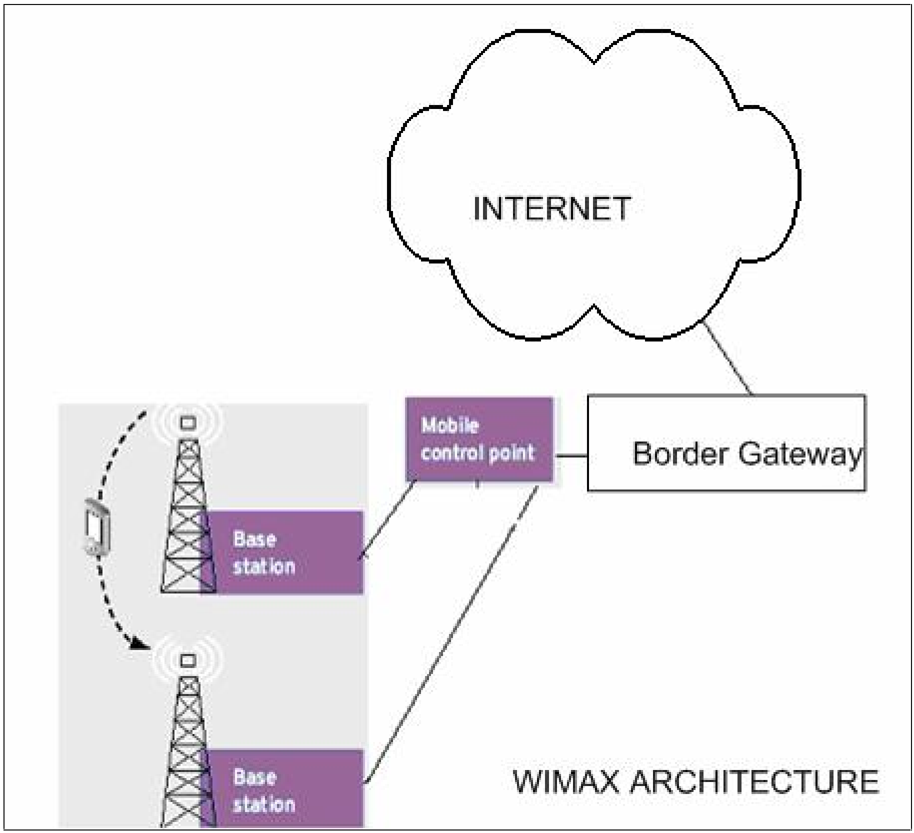


Figure 7.1 WiMax architecture

7.3. WIMAX integration in Next Generation Networks

As stated in chapter 4, WiMax is considered as the alternated destination network when performing vertical handover from WLAN to UMTS or UMTS to WLAN. In fact, the mobile user is guided to the WIMAX network through the SIDM when the vertical handover to the UMTS or WLAN networks could not be performed. The reason behind this choice is either a loaded UMTS or WLAN network, or the required QoS and user profile could not be respected in the next network.

SIDM is responsible of guiding the mobile user to the WiMax network. It is located as gateway between WiMax and the internet cloud as shown in figure 7.2.

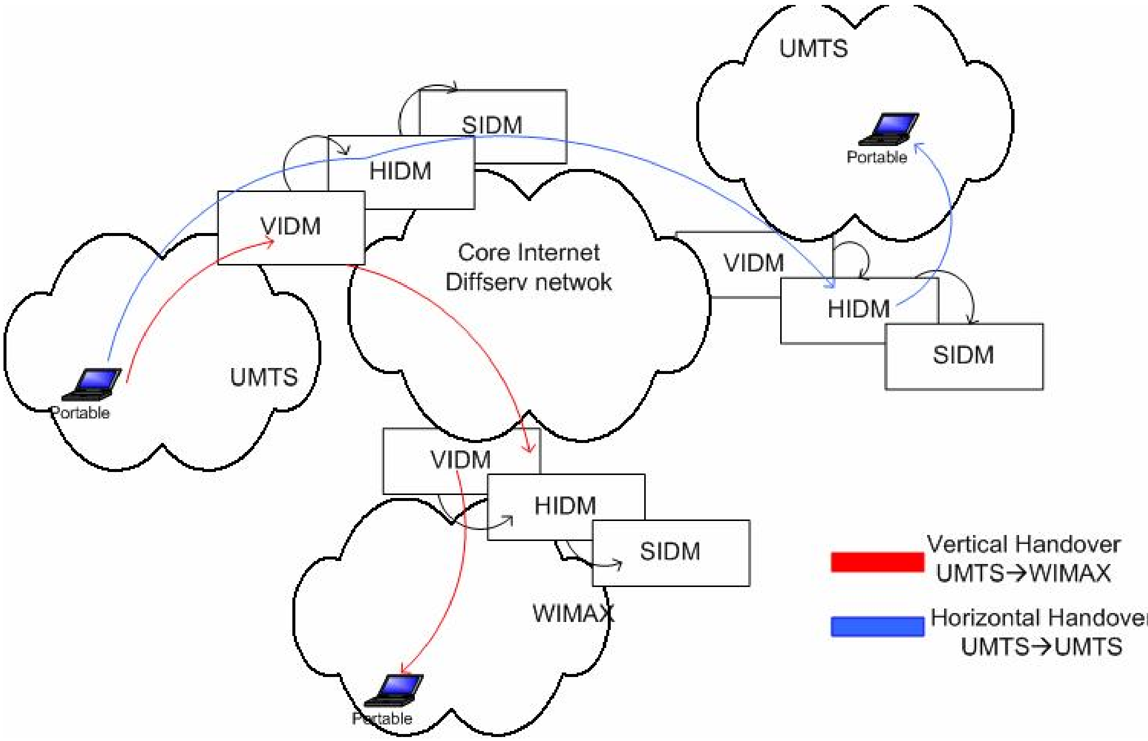


Figure 7.2 Integration Architecture

7.3.1. WiMax handovers

While WiFi provides high bandwidth but not distance, and current cellular systems provide distance but not high bandwidth, WiMax will provide both.

The handover decision from WLAN or UMTS to WiMax is mainly based on the reasons described earlier specifically its reduced cost, wider coverage and wider range of service…

When the vertical handover could not be performed due to the IDM capacity or inability to satisfy the required QoS, the handover is guided to WiMax where, due to its wider coverage and high capacity, it can accommodate higher number of vertical handovers.

In what follows the handover scenario from WLAN to WiMax and from UMTS to WIMAX is presented. Figures 7.3, 7.4, 7.5 and 7.6 show the messages exchanged based on the proposed Mobility and protocol. Our proposition shows to be scalable and easily deployed for different network technologies. Once deployed, WiMax could be easily integrated in our proposed architecture and the handover could be performed with the required QoS and according to the user profile as explained in the proposed protocol.

The main operations involved in the vertical handover process are shown in Figures 7.3, 7.4, 7.5 and 7.6. The transport of these messages is based on ICMPv6 protocol where the type and code are set to the specific type of CTP messages and the context data transfer is added in the data option field.

The scenario of the exchange is as follows:

- MN sends RTSOLPR to PAR to indicate HO initiation and to start the operation of acquiring a new address. The PAR will not reply directly (as in the normal FMIPv6 signaling flow) but it will wait for the IDM to select the appropriate NAR.
- MN sends Context Transfer Activation Request CTAR to PAR and PIDM prior to HO. this message contains MN IP address and PIDM IP address, and the entire context to be transferred (by default). It also contains token to be used by NIDM for verification.

- Context Transfer Request CTREQ is sent from PIDM to PAR to update the context already available in the entry of the PIDM

- PAR starts to send Context Transfer Data CTD to PIDM with the feature data context parameters.

- The "Who's next phase" is started. PIDM will start scanning (based on CTD) for a NIDM whose resources will satisfy the required context of MN asking for HO.

- Handover Initiation HI is sent from the PIDM to NIDM and NAR to configure the new CoA.
- All the FMIPv6 signaling flow will now proceed to perform the handover request to the selected access network.

- MN will finally send a Fast Neighbor Advertisement (FNA) to the NAR-MCP to notify him of its presence in the new access network.

- The packets destined to the old access router are no more buffered in PAR but in the PIDM which will forward these packets to the NIDM, as soon as the mobile node starts to communicate with the NAR.

The handover performed from UMTS and WLAN to WiMax is much more possible than the handover from WiMax to UMTS or WLAN, since WiMax has much more bandwidth and higher coverage than UMTS and WLAN. The messages exchanged in the described handover scenario are similar to those exchanged in the vertical handover with the difference of the nodes involved. The new entities involved in WiMax handovers are NOSS, MCP and SIDM responsible of performing seamless handovers. In our handover scenario, we assume that the node in the WiMax network are able to reserve the required resources so that the SIDM "Who's next phase" is done successfully.

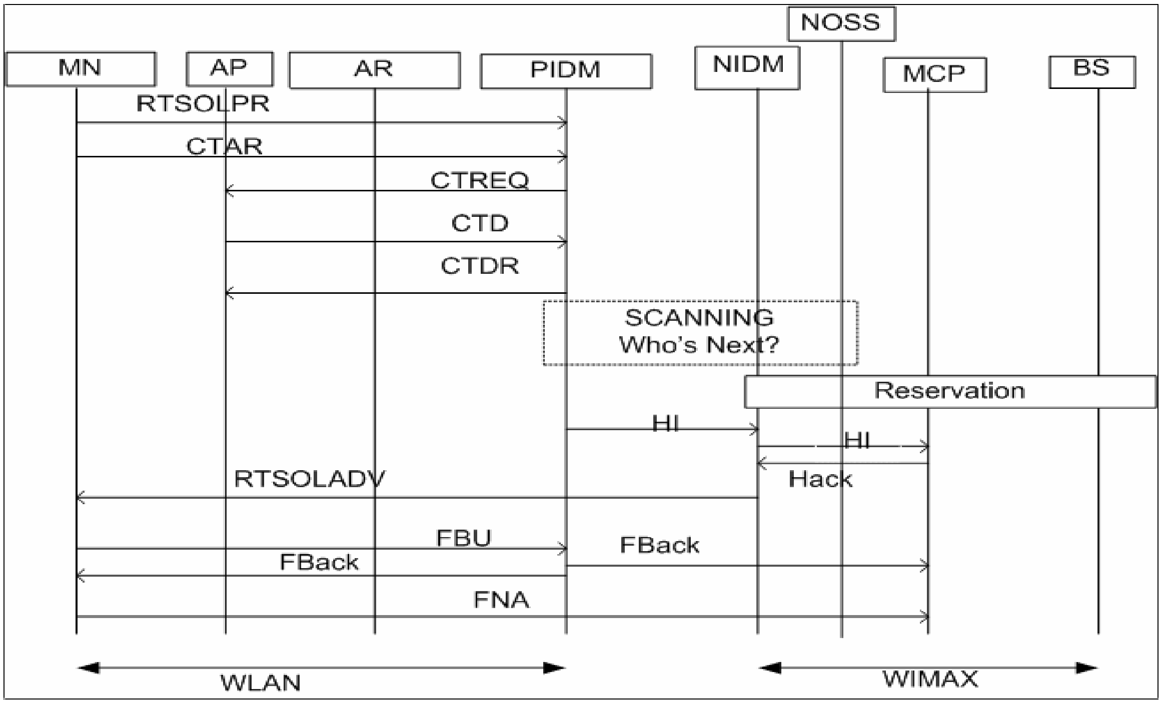


Figure 7.3 Handover from WLAN to WiMax

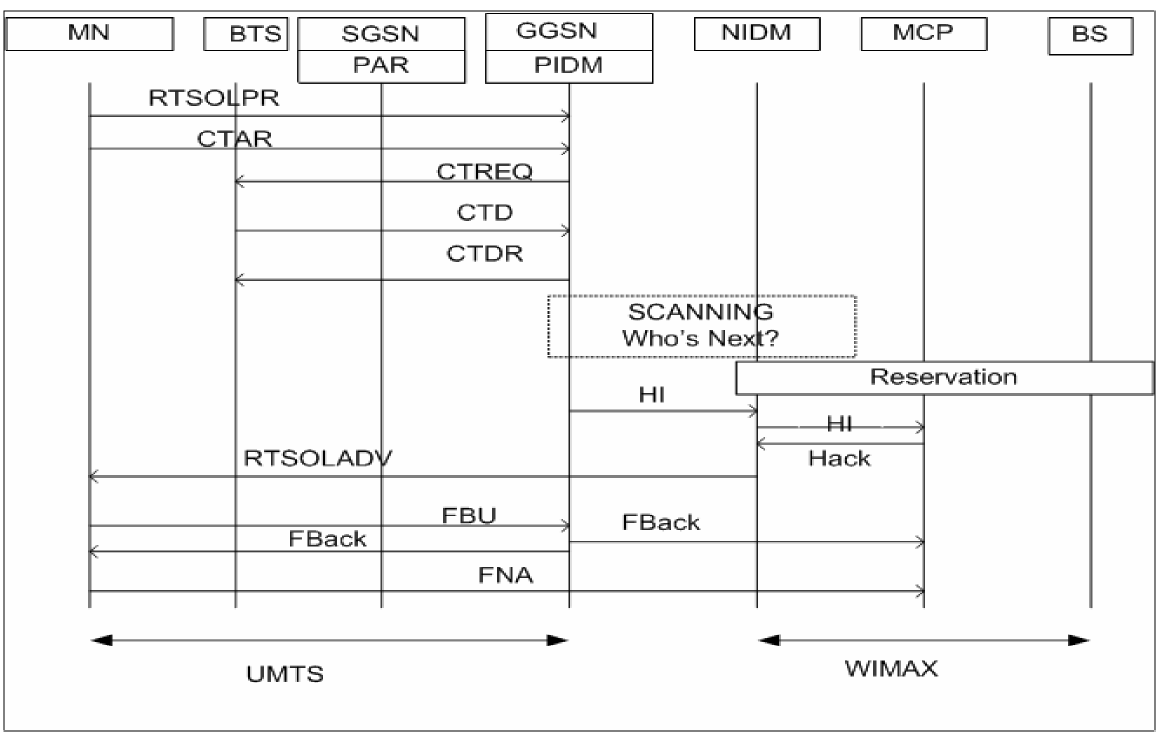


Figure 7.4 Handover from UMTS to WiMax

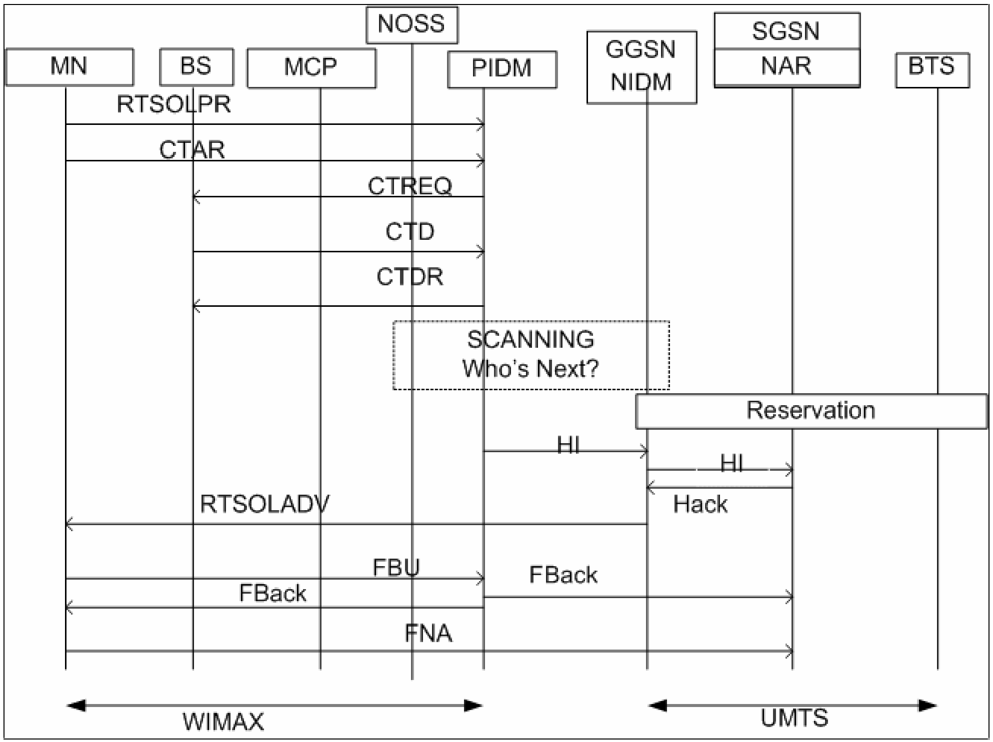


Figure 7.5 Handover form WiMax to UMTS

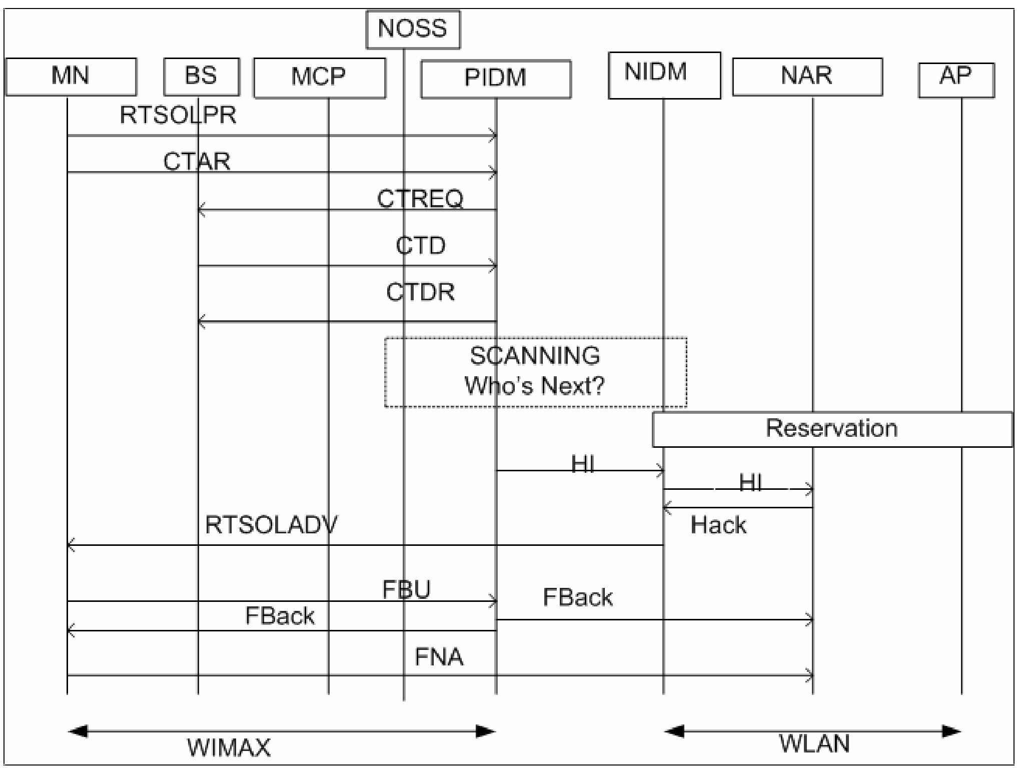


Figure 7.6 Handover from WiMax to WLAN

7.4. WiMax handovers evaluation

The proposed integration architecture with IDM support for seamless handover and QoS provisioning is based on the previously described protocol MQMP. In the above design, the mobile node can perform handover from WLAN to WIMAX or from UMTS to WiMax since the last will enable higher throughput and higher coverage compared to UMTS and WLAN.

As described earlier for vertical handovers, the SIDM is now responsible to scan all the table entries to find the next IDM capable of providing the mobile with the required parameters ensuring a seamless handover. VIDM will either conduct the handover call to the desired destination network WLAN or UMTS, or to WIMAX network if VIDM is saturated and no more calls could be accepted according to the "Who's next Policy".

The handover from UMTS→WIMAX and WLAN →WIMAX is modeled on the following basis: Since WiMax has more bandwidth and can accommodate more calls; the following Markovien (figure 7.7) model is used. In this model the HO calls are privileged over the local calls. The local calls are accepted until the threshold T is reached. When the number of calls is greater then T, then the only the HO calls (vertical calls from UMTS or WLAN) are accepted.

This priority model is similar to that described in vertical and horizontal handover for the purpose to privilege the handover calls.

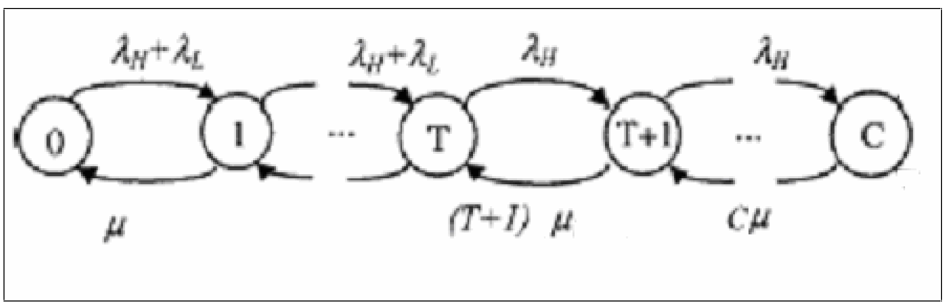


Figure 7.7 SIDM admission Model (UMTS, WLAN to WiMax)

The following assumptions were made:
The IDM is modeled as M/M/C/C queuing system.
C-T interval channels are the guard band channels used only for handover calls.
The total number of channels or the IDM capacity is $C_{WiMax}=3*C_{WLAN}$
The local traffic is a poisson process of rate λ_l and the handoff request is a poisson process of rate λ_h
The time of stay in IDM is exponentially distributed of rate $1/\mu$.
New calls are 5 times the HO request arrivals.

As in the case from UMTS to WLAN, the blocking and dropping probabilities PB and PD in function of the threshold T are as shown in figures 7.8 and 7.9. The choice of the threshold will highly affects the handover dropping probability as well as the blocking probability of the incoming local calls. The variation of these two probabilities with respect to the threshold is given in the figures below. The maximum allowed threshold is when T=C. as T increases, the blocking probability decreases, since the calls (local requests) are accepted until T is reached; whereas, the dropping probability is increased since the probability of dropping the handover calls is increased with no priority scheme.

UMTS to WIMAX has a higher BP due to the limited UMTS capacity. On the other hand, WLAN to WiMax has a lower BP with the increasing IDM load, sine more bandwidth and higher IDM capacity are available.

UMTS to WIMAX has higher DP due to the higher number of handovers to a cheaper and faster network. Whereas the handover from WLAN to WiMax has lower DP, since the users in WLAN already benefits from a low cost and high bandwidth. Figure 7.10 show the results.

On the other side, the handover from WiMax to either UMTS or WLAN is not considered since our focus was to perform a handover based on the user preferences or the network parameters. In both cases, the user in a WiMax network will enjoy the high bandwidth and the high coverage compared to WLAN and WiMax, so the handover request from WiMax to other access technologies will be rarely considered.

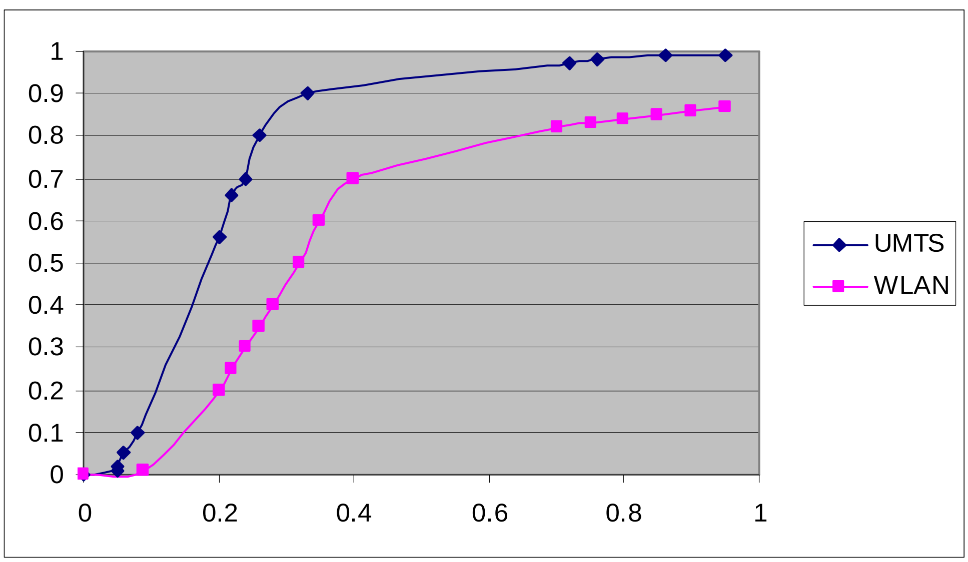


Figure 7.8 Blocking probability from UMTS/WLAN to WiMAX

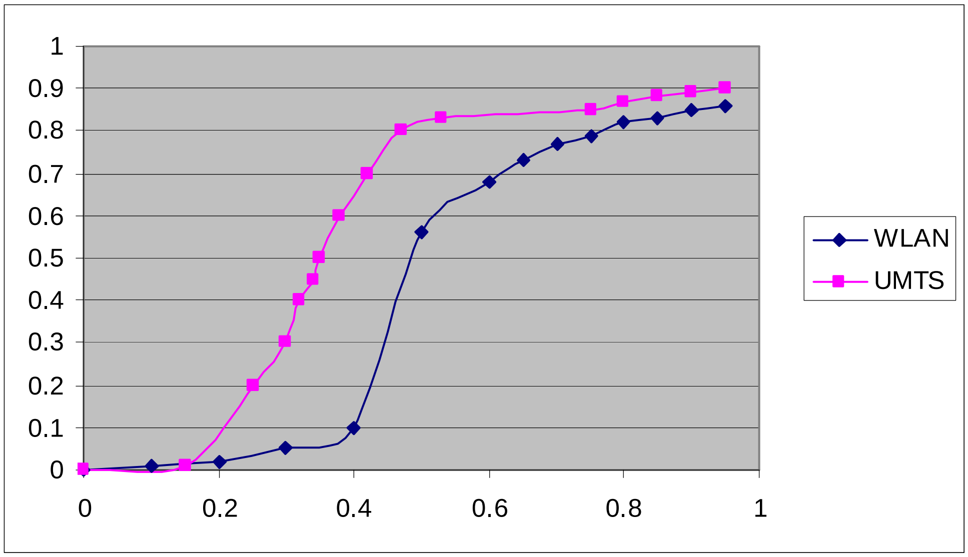


Figure 7.9 Dropping probability from UMTS/WLAN to WiMax

7.5 Simulation study

The WLAN and UMTS networks are simulated as AP described in chapter 4.
For WIMAX: the following settings were made:
802.16 MAC is OFDM, throughput is 20Mbps, propagation modelis Two Ray Ground, and the coverage is 1000m as described in table 7.1.

	WLAN	UMTS	WiMax
Bandwidth (Mbps)	10	2	100
Coverage (m)	300	1000	1000
Mobility	medium	High	medium
Nodes involved	AP-AR-IDM	NodeB-RNC-IDM-	BS-MCP-IDM

Table 7.1: Simulated networks parameters

The mobile is modeled as capable of receiving the best standard signal (802) which is choosing between wifi and WiMax from one side and the UMTS network on the other side.

Traffic Scenario:

-the local traffic is initiated to increase the IDM load. When the mobile node will loose network coverage to start the HO process at t=80s, this background traffic is introduced on order to charge the network.

-Simulation time is 300s
-At t=80s, the handover started. The mobile node will move from one access point to another in different domain based on AVHO described in chapter 3, and the MQMP protocol described earlier. The HO represented by real time traffic represented by a CBR source of 140kbps, packet size is 150 octets. The local traffic is represented as CBR source with 130kbps.

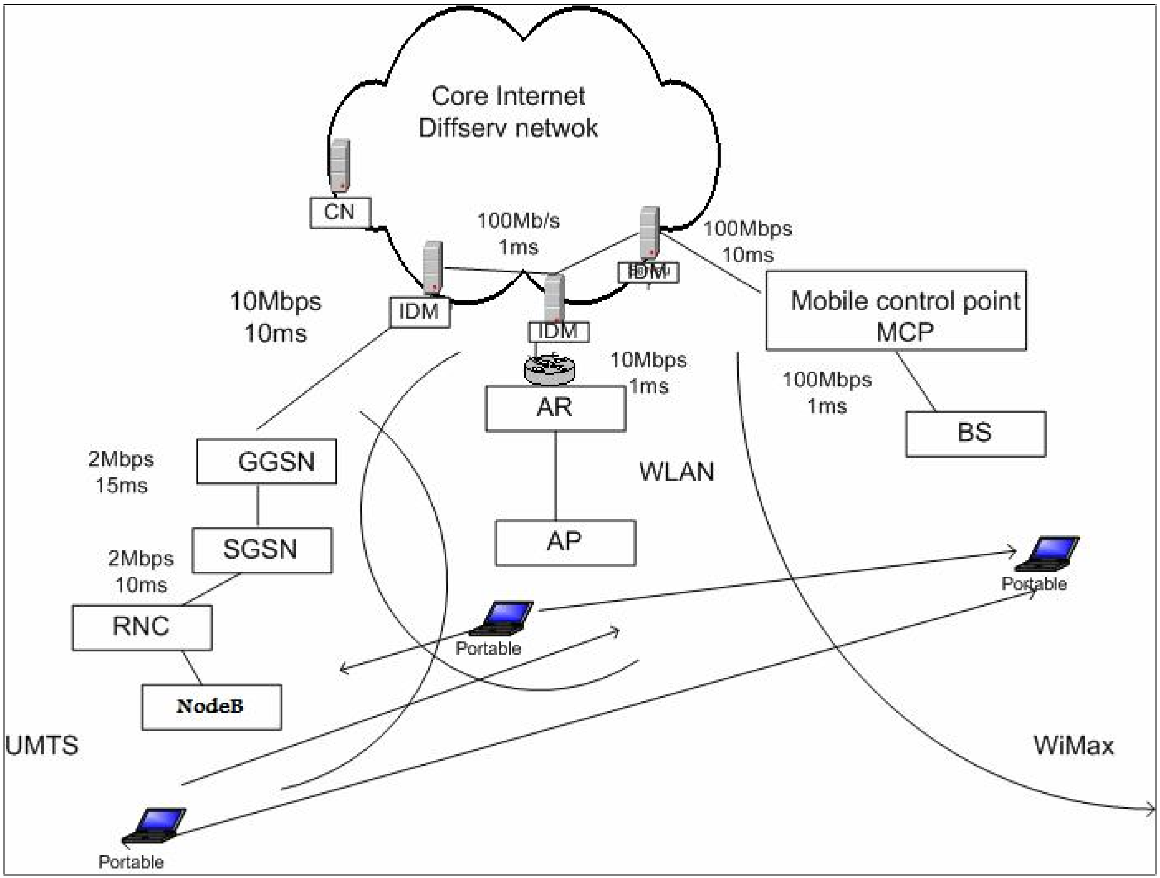


Figure 7.10: The simulated topology

The QoS mapping is based mapping the WiMax service classes with UMTS and WLAN classes as shown on table 7.2.
The WiMax service classes are: best effort, non real time with loose delay requirements, real time with variable packet size, real time with fixed packet size.

UMTS	WLAN	WIMAX
Conversational	Low latency and low jitter	Real time
Streaming	Low latency	Real time
Interactive	Low loss	Non real time

Background	Best-effort	Best-effort

Table 7.2 QoS mapping (WiMax-WLAN-UMTS)

The analysis was decomposed into four phases. Phase1 is the initiation phase composed of FMIP initiation messages and the Context transfer messages exchanged. Phase2 is the selection phase based on the "WHO'S NEXT" operation of the involved IDM. This phase is related to the processing time of the IDM node assumed to be 200microsec. Phase 3 is the reservation phase in the WiMax, and finally phase4 where the FMIPv6 operation is resumed by the exchange of the 9 last messages for updates and HO termination.

The results of the above scenario are shown in table 7.3. The WLAN and UMTS results are similar to those obtained in the previous chapter. The WiMax results shows that Phase1 could either happened in WLAN or UMTS, phase 2 is related to the SIDM processing time which is higher or equal to VIDM and HIDM. Phase 3 is the reservation process related to the WiMax network which is faster than UMTS and WLAN due the WiMax higher bandwidth. Phase 4 depends on whether the initial network is WLAN or UMTS as obtained. Table 7.4 shows the total handover delay from WLAN to WiMax and UMTS to WiMax.

	WLAN	UMTS	WiMax
Phase1	100ms	500ms	20ms
Phase2	200μs	200μs	200μs
Phase3	150ms	500ms	15ms
Phase4	240ms	800ms	24ms

Table 7.3 Simulated results

	WLAN to WiMax	UMTS to WiMax
Total HO TIME (ms)	274.2ms	1024.2ms

Table 7.4 HO results

7.6 Conclusion

In this chapter, we extend our MQMA model to include the emerging WiMax for 2 reasons. The first one is to solve the problem of VHO from WLAN to UMTS and vice versa, as described in chapter 4. In this context, WiMax mobility was presented as a stand-by destination network in order for the user requesting the HO not to be blocked. As described earlier, the user is guided to WiMax if no resources are available in UMTS or WLAN to perform the requested HO. the second reason is to show that the MQMA model is a global architecture capable of integrating different access technologies. This integration is to provide the NGN mobile user roaming between different access technologies with a seamless HO.

Chapter 8

Conclusion and perspectives

In this paper, we present a new integration architecture that manages the mobility and QoS between heterogeneous access networks (WLAN, UMTS and WiMax). In the proposed architecture, a solution to provide QOS when moving between different access technologies is presented due to an Inter-Domain Manager module IDM gateway between the Diffserv network and the heterogeneous access networks.

Unlike the previous existing architectures, after the handover is performed, changes in QoS parameters may occur if the new data path does not have the required resources necessary to serve the mobile user according to the required QoS. In this case, the related packets will receive the default forwarding treatment, and QoS degradation will occur.

To establish proper QoS treatment for the MN's packets along the new data paths, the proposed Mobility and QoS Management Architecture MQMA is based on integrating different access networks to a common Diffserv core network through Inter-Domain Management modules IDM. IDM is present in each access network to provide a seamless handover and map the QoS parameters of the visited network to the required QoS parameters in the current network by communicating with the selected IDM while processing the handover based on FMIPv6 and context transfer protocols functionalities.

The main challenge in NGN is that in order to provide QOS in the converged all-IP network, user mobility and resource reservation should be pre-managed in a way that the mobile node will not encounter a handover latency which causes a QoS degradation when moving from one access network to another or to guarantee minimum level of QoS.

An architecture with Seamless Mobility and QoS Management is proposed, based on Inter-Domain Manager module IDM responsible of providing seamless handover and guaranteed QOS level to the mobile user.

For this model special Mobility and QoS protocol was proposed to control the handover and the message exchanged between the IDM entities. It is based partly on Anticipated Vertical Handover derived from the FMIPv6 protocol in its "make-before-break" approach, and on the

Context Transfer Message protocol with the IDM interference and interpretation of the exchanged messages.

Handover is first considered vertically, and the IDM was modeled to accept the HO calls of higher priority with respect to the local calls.

VIDM, which is responsible of vertical HO, could guide the HO call to HIDM so that it could be performed seamlessly and with minimum Ho latency. In this frame, another IDM is introduced to guide the HO to the WiMAx network if it could be served with higher performance.

Simulations were conducted to calculate the total handover time in the different networks based on our IDM model. Besides, the HO dropping probability was studied with respect to the IDM load which represents the number of users requesting HO. this probability is the "Guided HO Probability" which will determine the possible HO scenario, whether vertical or horizontal.

Perspectives

After proposing a future global architecture accommodating different access technologies and providing the user with the ability to roam seamlessly across them, the key issue is to have the WiMax 802.16e specifications completed and deployed in order to study the possibility of a user enjoying the high data rate and high coverage area of WiMax networks to perform handover either to a WLAN or to a UMTS network. This latter case was not considered in our study.

It would be interesting to extend the studied analytical model of the studied performance to include the number of IDM's, the usability percentage of the IDM in different access networks so that statistics could be made, and the IDM capacity could be fixed accordingly.

We are also aware that the mobile mobility history and behavior will also help in determining the threshold of the priority model. A dynamic threshold behavior will operate perfectly in such environments where the user profile and history are updated and broadcasted to the designated IDM.

Introducing intelligent network nodes in our architecture will highly influence the inter-IDM communication in the overall system. IDM's intelligent agents will record the availability of free resources, and guide the HO without interfering with another IDM entity. This will reduce the HO delay and the overall network overload.

List of figures

List of tables

Glossary

3GPP	3rd Generation Partnership Project
AUC	Authentication Center
CAG	Cellular Access Gateway
CN	Core Network
CoA	Care-of-address
ESS	Extended Service Set
ETSI	European Telecommunications Standards Institute
EURANE	Enhanced UMTS Radio Access Network Extension
FA	Foreign Agent
F-Back	Fast Binding Acknowledgement
F-BU	Fast Binding Update
F-NA	Fast Neighbor Advertisement
GGSN	Gateway GPRS Support Node
GPRS	General Packet Radio Service
HA	Home Agent
Hack	Handover Acknowledge
HI	Handover Initiate
HIDM	Horizontal IDM
ICMP	Internet Control Message Protocol
IDM	Inter-Domain Management Module
ITU	International Telecommunication Union
MQMA	Mobility and QoS Management Architecture
MQMP	Mobility and QoS Management protocol
mSCTP	mobile SCTP
NAR	New Access Router
OFDM	Orthogonal Frequency Division Multiplexing
PAR	Previous Access Router
PrRtAdv	Proxy Router Advertisement
QoS	Quality of Service
RtSolPr	Router Solicitation for Proxy
SCTP	Stream Control Transmission Protocol
SGSN	Serving GPRS Support Node
SIDM	Stand-by IDM
SIP	Session Initiation Protocol
SLA	Service Layer Agreement

SNR	Signal-to-Noise Ratio
STA	Station
UE	User Equipment
UMTS	Universal Mobile Telecommunication System
UTRAN	UMTS Terrestrial Radio Access Network
VIDM	Vertical IDM
WiFi	Wireless Fidelity
WiMAX	Worldwide Interoperability for Microwave Access

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