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異質無線網路服務品質確保之研究

QoS Management in Heterogeneous Wireless Networks

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To my son, Alex

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陳明僑
97.3.25
於國立東華大學資訊工程研究所

摘要

過去幾年，無線網路技術的突飛猛進，不僅取代了有線網路，也形成了一個異質無線網路的環境。隨著個人電腦與手持裝置的普及，使用者追求隨處可得，隨處可用的網路資源需求與便利性也逐漸增高。隨之而起的是，各種多媒體網路服務的蓬勃發展，網路頻寬資源的需求增加，使用者需求也更趨精緻及高品質。為了能夠滿足使用者的需求，在此異質網路上提供一個服務品質保證的機制是刻不容緩的議題。

現代化的家庭網路應用，特別是針對即時性（Real-Time）的應用服務，在有限的網路資源下，需要一個品質保證的服務（Quality of Service，QoS）。在有線及無線的異質性網路中，關心的服務品質也不同，例如在無線網路環境中比較關心位元錯誤率，資料重傳以及封包遺失的問題，但這些服務品質控制參數在有線環境中的敏感度卻不高。另外，不同的應用程式所需控制的服務品質參數也有所差異，例如，網路電話較關心延遲變化率及點對點的延遲時間，線上電影則較關心封包遺失率影響影片的解析度。在本論文中，我們配合現有的 UPnP QoS 規範（Quality of Service V1.0 for UPnP），並且以 RMD（Resource Management in Diffserv）架構為基礎，設計出未來應用在 OSGi Home Gateway 上的可適性 QoS 管理機制，用以監控網路流量及網路壅塞時的頻寬管理，針對有線及無線網路異質性的服務品質參數來做進一步控制調整，進而達到家庭異質網路服務的品質確保。由實驗模擬結果顯示，RMD 網路加上可適性 QoS 管理機制後，整體延遲變化率減少 0.1391 毫秒，延遲時間減少 0.0066 秒，封包遺失率也減少 5.43%。針對不同的即時性應用程式，高解析度影片播放的封包遺失率降低了 4.53%，產出增加 1.2%；一般解析度影片播放的封包遺失率降低了 1.89%；網路電話的延遲變化率改進了 0.0407 毫秒，延遲時間減少 0.0209 秒。

自 2000 年起，第三代行動通訊系統(UMTS)開始進入商業行銷。UMTS 最大的改變，在於能提供寬頻的數據傳輸，突破以往只能提供語音服務和低速數據服務的限制。在行動通訊新舊世代交替之際，GSM/GPRS 須與 UMTS 整合使其發揮最大的效能。基於提高 Timeslot 利用率的考量，我們研究在整合系統內無線電資源

有效率地分配方法。在新的分配方法內，規劃數據傳送以 UMTS 系統為優先服務系統，語音服務則優先以 GSM 系統來承載。當 UMTS 的細胞負荷過重時，系統優先使用 UMTS 來承載語音服務，數據服務則優先由 GPRS 來載送。如此可降低 UMTS 的干擾量以避免影響使用者的傳輸品質，並可提高無線電頻譜使用率，並增加系統容量。由實驗模擬結果顯示，整合系統在每個使用者要求 64 kbps 的傳輸速度下，可有效提高 7.14%整合系統的容量，若是在每個使用者要求 128 kbps 傳輸速度的情況下，整合系統的容量可提高 13.33%容量有效調節系統資源。

異質網路的整合應用一直都是重要的研究課題，本論文也提出一個分散式多重代理人系統，提供雙模行動終端設備在第三代行動通訊系統(3G UMTS)以及無線區域網路(WLAN)間的網路資源排程機制。提出一個兩階層的控制機制，以 UMTS 作 High tier，IEEE 802.11 為 Low tier，架構一個 Two tier 的網路使用環境，結合 UMTS 之高移動性、低傳輸頻寬以及 WLAN 之低花費、高傳輸頻寬的特性，截長補短，以使得使用者能夠在任何時候依其需求使用最佳的網路作為其接取網路，進而與遠端的 IP 網路進行連接及通訊。依據模擬結果分析，延遲時間表現介於 UMTS 網路的 0.0809s 與 WLAN 網路的 0.00124s 之間；在 Jitter 的表現上，則較單一的 UMTS 和 WLAN 要大。然而，Jitter 的表現僅在使用者頻繁於兩種網路換區時方有顯著的影響。以傳輸效能上的表現，雖然雙網環境不盡全然地優於 UMTS 網路以及 WLAN 網路環境，卻能夠同時結合二者在高移動性與高傳輸頻寬等方面的優點。

關鍵詞：服務品質保證、異質無線網路、數位家庭閘道系統、資源管理、分散式多重代理人系統

Abstract

There has been a tremendous growth in wireless network technologies over the past decade. Wireless infrastructure has been deployed to complement and expand the wired infrastructure. There are many options available for users who want to access data and information wirelessly. These heterogeneous wireless technologies are similar in that they both allow users to access data on their PCs or PDAs without using network or modem cables. These technologies are also rather different in their uses and applications. Therefore, a QoS mechanism is needed to address in such heterogeneous wireless network environment to satisfy users' requirements.

An OSGi (Open Services Gateway Initiative) home gateway system manages the integration of heterogeneous home networks protocols and devices to develop ubiquitous applications. Wired and wireless heterogeneous home networks have different QoS concerns. For instance, jitter and latency are important concerns in web phones, while packet loss ratio is important in on-line video. This dissertation adopts UPnP QoS specification version 1.0 to design an adaptive QoS management mechanism based on the RMD (Resource Management in Diffserv) architecture. This dissertation monitors real-time network traffic, and adaptively controls the bandwidth, to satisfy the minimum requirement of each application in home network congestion. In the simulation results, we can find that after we build the QoS management mechanism into the RMD, total jitter, latency, and packet loss ratio has been decreased 0.1391 ms, 6.6 ms, and 5.43% respectively. To address different kinds of real-time applications, we decrease 4.53% packet loss ratio and increase 1.2% throughput for the high definition video streaming, decrease 1.89% packet loss ratio for standard definition video streaming, and for the VoIP (Voice over IP) we also decrease the jitter and latency to 0.0407 ms and 20.9 ms respectively.

2G/2.5G systems are designed to achieve 90-95% coverage. Most operators construct 3G base stations on previous 2G/2.5G stations. Consequently, integrated radio resource management is comprised of timeslot-limited and interference-limited systems. To improve capacity, this dissertation presents a novel resource management based on timeslot utilization in an integrated GPRS/UMTS service network. Although

this approach gives UMTS high priority for sending data services, it also makes voice services available over GSM. In the simulation results, each data session of 64 kbps increases 7.14% in the capacity of the integrated system. Each data session of 128 kbps increases 13.33% in the capacity of the integrated system.

WLAN services are inexpensive and have a high bandwidth, while UMTS services provide wider coverage area and higher mobility. Based on intelligent deduction, this dissertation presents a novel service scheduling scheme for WLAN/UMTS dual-mode networks. The proposed system, Distributed Multi-Agent System (DMAS), consists of a set of problem-solving agents that autonomously process their own tasks and interoperate with one another by a shared database to reach a suitable schedule for dual-mode network services. A two-level control mechanism comprising local-control and meta-control is presented to achieve a high degree of goodness in service scheduling. In the simulation result, the delay of integrated WLAN/UMTS system is between 80.9ms in UMTS and 12.4ms in WLAN. The jitter of integrated WLAN/UMTS system is greater than the jitter in pure WLAN (0.00061) and pure UMTS (0.0049). However, the effect in jitter happen only when mobile users handoff frequently between two different networks. Although the analyses of delay and jitter in integrated WLAN/UMTS network are not all better than in pure WLAN and in pure UMTS, integrated WLAN/UMTS system combines simultaneously the advantages of UMTS and WLAN.

Keywords: QoS, Heterogeneous Wireless Network, OSGi, Resource Management, Distributed Multi-Agent System

Contents

List of Figures

List of Tables

Chapter 1
Introduction

There has been a tremendous growth in wireless network technologies over the past decade. Wireless infrastructure has been deployed to be the complement and expansion of the wired infrastructure. There are many options available for users who want to access data and information wirelessly. These heterogeneous wireless technologies are similar in that they all allow users to access data on their PCs or PDAs without using network or modem cables. These technologies are also rather different in their uses and applications. Therefore, a QoS mechanism is needed to address in such heterogeneous wireless network environment to satisfy users' requirements.

As shown in Fig. 1-1, the wireless networks consist of a number of different types and coverage areas of wireless segments: (1)Wireless Personal Area Network (WPAN), e.g., Bluetooth, IEEE 802.15, provides the network access within a very small coverage area for interconnecting devices centered on an individual person's workspace. WPANs are commonly used as cable replacement among devices, such as between a headset and a cellular phone. (2)Wireless Local Area Network (WLAN), e.g., IEEE 802.11a/b/g, HiperLAN/2, provides the network access in larger coverage area. WLANs have become popular due to convenience of installation and are widely used to share data within home, office, train station and airport. (3)Wireless Metropolitan Area Network (WMAN), e.g., IEEE 802.16a/d/e, provides network access to an entire city within large coverage area. (4)Wireless Wide Area Network (WWAN), e.g., GSM, GPRS, UMTS, provides the network access in large area that is generally offered on a nationwide level.

Wireless networks enable the ability to communicate anywhere at anytime and changes the way people work and their life styles. Wireless deployment penetrates to various segments, e.g., home, office, hotspot, mall, train station and airport. Different multimedia services and real-time applications that have been traditionally used in wired networks are also increasingly being used in wireless networks. Each application requires some service level of commitment from the wireless networks in order to operate successfully **[1-2]**. These requirements are expressed in term of Quality of Service (QoS). QoS support in wireless networks is a challenging design problem since the wireless network bandwidth is a limited and variable resource. Therefore, each segment of wireless networks requires QoS mechanisms to manage network resource to each application appropriately.

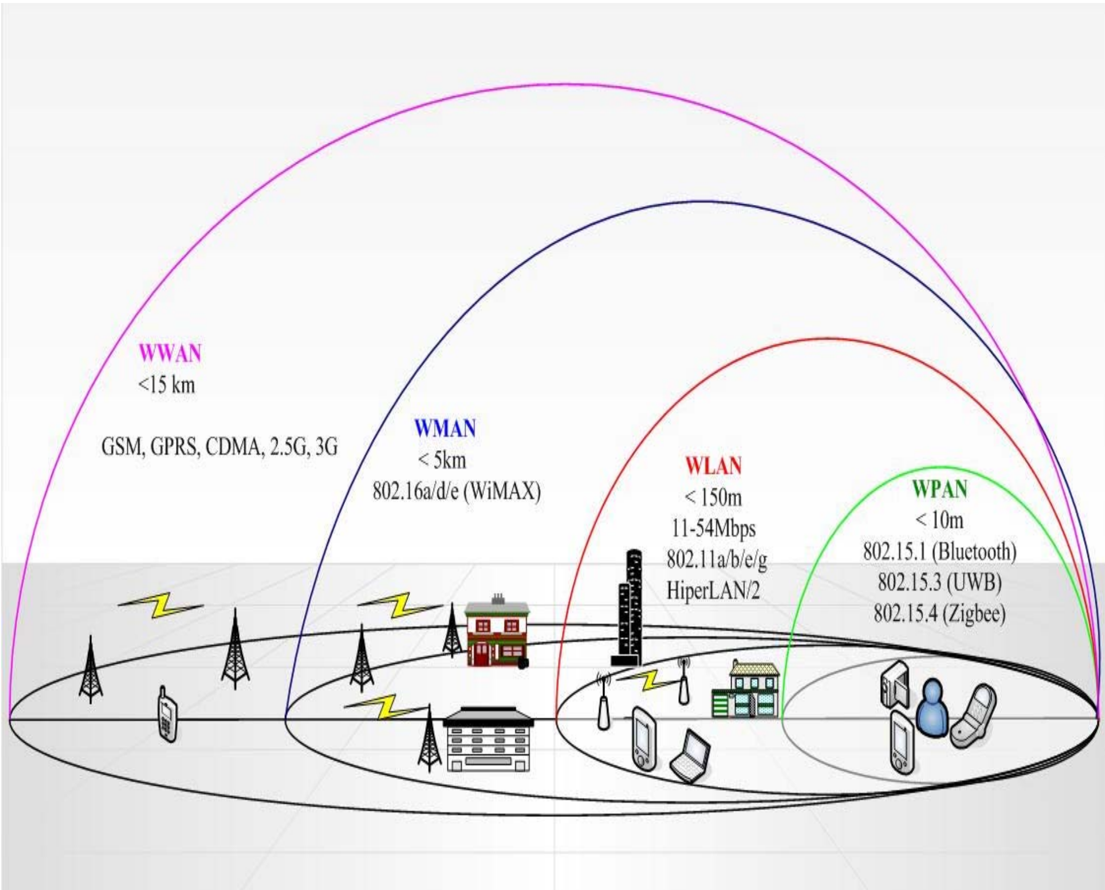


Figure 1.1: Wireless Network Technologies

1.1 Motivation

This dissertation is motivated by two major aspects. Firstly, the evolution of wireless technologies that forms a heterogeneous wireless networks environment. Secondly, the ability of necessary QoS mechanisms that supports such heterogeneous wireless networks environment.

Although wireless technology is likely to become the dominant home network technology, it has an unstable bandwidth. Hence, some mechanisms are needed to inform the applications those changes in bandwidth in real time. It is necessary a resource admission control mechanism to execute either in the network setup time or in run time. Therefore, a dynamic resource reservation mechanism is required to address the QoS in heterogeneous home network applications.

In GSM systems, each cell supports dedicated timeslots for voice connections and data connections based on circuit switching. Each cell has several channels, which are generally frequency and timeslot pairs. Each voice connection is assigned one channel. Other voice connections (or handoff connections) are admitted when the cell has free channels, and no further calls are admitted when all channels are occupied. The radio resource management (RRM) algorithm of GSM is simple and straightforward. To enhance data connection services, standard GPRS over GSM supports multi-timeslot data access, in which several data connections share particular time-slots. Namely, GSM and GPRS offer particular timeslots for voice and data connections, and RRM algorithms for GSM/GPRS are based on the time-slot system. The UMTS consumes fewer air interface resources than GSM/GPRS. In UMTS, power control techniques determine the power required to sustain a given data rate. In either of these two cases, stations with poor channel conditions require more air interface resources to sustain a given data rate than stations with good channel conditions. Qualities of air interface resource depend on radio interference and transmit power. The RRM algorithms in UMTS are limited by interference and transmit power. Therefore, a radio resource scheduling between GSM/GPRS and UMTS is required to combine timeslot-limited and interference-limited systems.

The ability of 3G UMTS networks to operate seamlessly with existing WLAN is critical for their widespread deployment. Mobile users adopt 3G networks to browse the Internet or communicate with one another. Hence, multiple standards coexist in the same service environment, otherwise a combined multi-function system is generated for future wireless communications systems. Therefore, the interworking of systems must be optimized. Various systems have their own properties. High-tier systems like UMTS have a high mobility and low transmission bandwidth. Conversely, low-tier systems such as WLAN have low service costs and low mobility. The major challenge is to present a novel discipline for service scheduling in WLAN/UMTS dual-mode networks to achieve optimal service quality.

1.2 Research Objectives

Our research goal is to develop QoS management mechanisms for heterogeneous networks and resource management mechanisms in dual-mode services networks.

1.2.1 QoS Management in Heterogeneous Home Networks

WLANs have been deployed popularly in home networks. The home network system consists of a residential gateway and a variety of networked appliances, and includes both wired and wireless networks. Users can control these digital home devices by either wired or wireless links. This dissertation has introduced a QoS mechanism to manage heterogeneous home networks. Based on an OSGi (Open Services Gateway Initiative) home gateway system, the mechanism can manage the integration of heterogeneous home networks protocols and devices to develop ubiquitous applications. The proposed adaptive QoS manager combines the wired and wireless QoS concerns into admission control. Different home applications have different key QoS requirements in congested home networks. The QoS attributes are integrated to perform adaptive admission control in wired-wireless heterogeneous home networks. Each application is guaranteed to have the minimum QoS requirement.

1.2.2 Resource Management in Dual-Mode Service Networks

Third-generation (3G) networking technology has taken many years to be deployed worldwide. The ability to operate seamlessly between existing GSM/GPRS and 3G UMTS networks is critical for its widespread adoption. To reduce construction cost, most operators constructed their new 3G base stations in the same buildings of previous generation base stations. Under the scenario of the same base station with 3G system and GSM/GPRS system, GSM/GPRS can compensate the coverage hold caused by the limitation of 3G transmitting power. An integrated GPRS and UMTS system should effectively adopt UMTS radio resources for high data rate applications, and ensure continued service when UMTS coverage is unavailable. This dissertation presents a resource management based on timeslot utilization in an integrated GPRS/UMTS service network.

WLAN services are inexpensive and have a high bandwidth, while UMTS services provide wider coverage area and high mobility. Based on intelligent deduction, this thesis presents a service scheduling scheme for WLAN/UMTS dual-mode networks. The proposed system, Distributed Multi-Agent System (DMAS), consists of a set of problem-solving agents that autonomously process their own tasks and interoperate with one another by a shared database to reach a suitable schedule for dual-mode network services. A two-level control mechanism comprising local-control and meta-control is presented to achieve a high degree of goodness in service scheduling.

1.3 Organization of the Dissertation

As shown in **figure 1-2**, the remainder of this dissertation is organized as follows. Chapter 2 introduces the fundamental technologies, standards and concepts of QoS in wireless networks. Chapter 3 introduces the proposed QoS mechanism in heterogeneous home networks. Chapter 4 presents the resource management in an integrated GPRS/UMTS service network. Chapter 5 describes the proposed distributed multi-agent scheduling scheme to enhance WLAN/UMTS dual-mode services. Chapter 6 draws conclusion and future works.

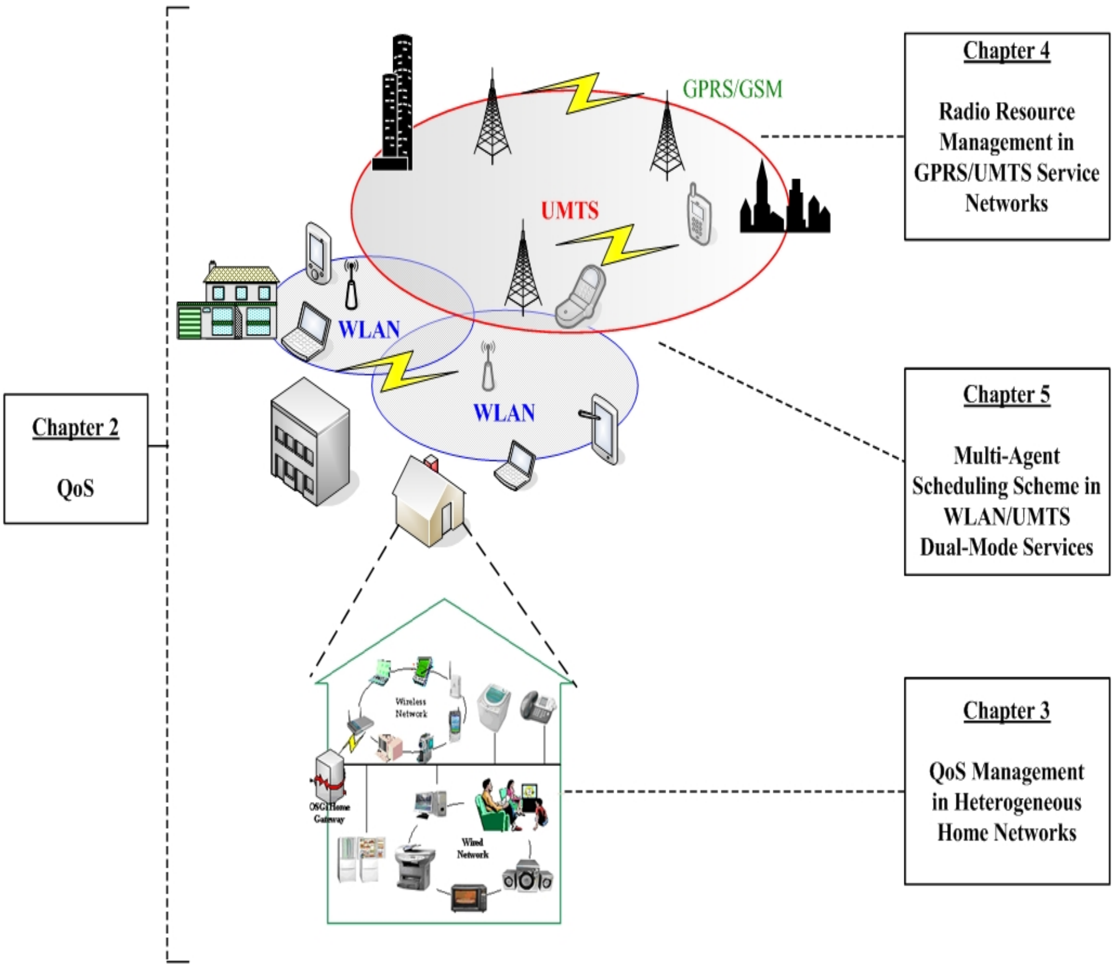


Figure 1.2: Organization of the Dissertation

Chapter 2
Background and Related Work

This chapter first presents an overview of the WLAN, GSM (Global System for Mobile communications), GPRS (General Packet Radio Service) and UMTS (Universal Mobile Telecommunications System) networks and standards as well as interworking mechanisms between WLAN and UMTS. Then this chapter also introduces QoS (Quality of Service) mechanisms required for the implementation of such wireless networks.

2.1 WLAN

In 1997, the IEEE 802.11 system was born, allowing work with the data rates of 1 and 2 Mbps. The service data rate was up to 10Mbps until 1999, and then IEEE 802.11b was generalized. The data traffic in IEEE 802.11b can be transported at 5.5Mbps and 11Mbps in 2.4GHz band. In IEEE 802.11a, the data rate is up to 54Mbps in 5GHz band. Now, the product of IEEE 802.11g is manufactured and the standards of 802.11e and 802.11i are already specified.

IEEE 802.11 is one of the important members of IEEE 802 that is a series of specifications for local area network technologies. **Figure 2.1** shows the elements of IEEE 802 and the relationships between them **[3]**.

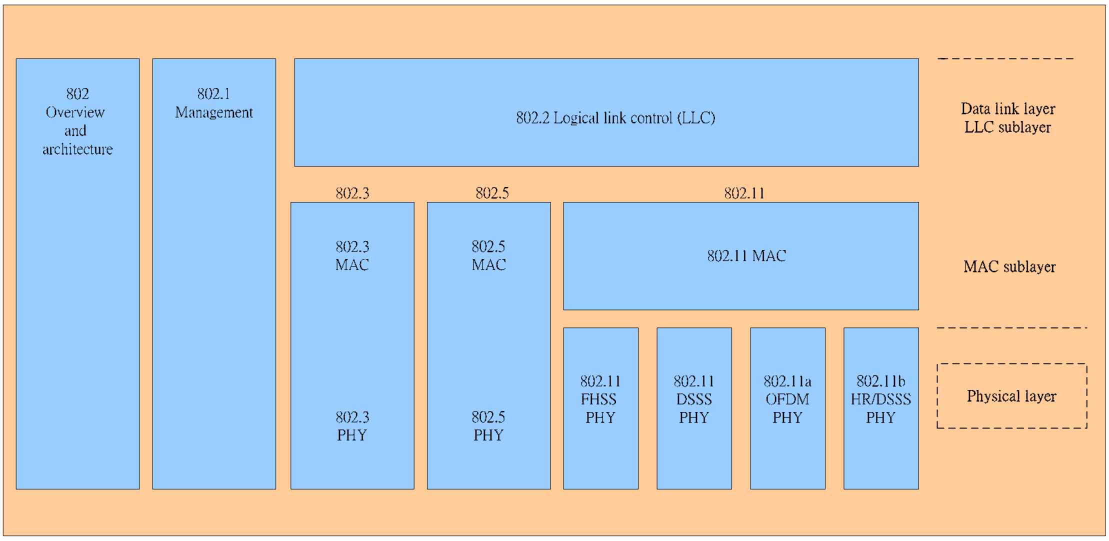


Figure 2.1: The IEEE 802 Family and Its Relation to the OSI Model

There are several major physical components in IEEE 802.11 system. They are：

Station (STA)：The object is of the communication. It is also named mobile station in general.

Access Point (AP)：It is a transporter in physical between wired line and radio wave. The mobile station transports its data packets by radio wave to one access point, and the access point is responsible for the relay to the wired network or other access points. In the reverse, it's the same role for relay when the data packets are to be transported to the mobile station.

Distribution System：When several access points are connected with wired line, they can be a large coverage area and must communicate with each other. The distribution system is a component of 802.11 in logical. It is used to forward the frames to their destination.

The IEEE 802.11 protocol supports the operation of wireless LANs in two different modes that are named infrastructure and ad-hoc networking. In infrastructure networking, access points are used for the communication between mobile stations. If one mobile station wants to communicate with the other one, it must transfers its data packets to the access point. After receiving the data packets, the access point transfers them to the destination station. In ad-hoc networking, two mobile stations can communicate with each other without going through any access points.

2.2 GSM System

GSM is the first and most successful digital cellular system (Second generation or 2G mobile system), illustrated in **Figure 2.2**. GSM commercial service started to July 1991, mobile phones were available only in 1992. By 1993, GSM subscribers can roam in European, Australia and South Africa. In January 2002, there were more than 470 GSM operators in 172 countries with 646 million subscribers. Mobile Traffic is growing faster than fixed network traffic. Today, GSM technology is in commercial operating by more than one in six of the world's population and it is estimated that at the end of Jan 2004 there were over 1 billion GSM subscribers across more than 200 countries of the world and it is predicted that in 10 years the number of mobiles will exceed the number of PSTN (Public Switched Telephone Network) lines **[4]**.

GSM standardization includes services, subsystem interfaces, and protocol architecture. It has been implemented mainly in Europe and Asia Pacific in the 900 MHz range. Variants of GSM have been deployed for different frequency ranges and applications. GSM is used in the 900 and 1800 MHz bands all over the world except North America (1900 MHz band) **[5]**. Services provided by GSM include voice, circuit switched data, and SMS (short message service). GSM uses TDMA (Time Division Multiple Access) techniques in the air interface. GSM allows up to eight subscribers to share the same 200 kHz radio carrier by allocating a unique timeslot to each subscriber **[6]**, illustrated in **Figure 2.2**. One timeslot with GSM only provides 9.6 kbit/s for the data service **[7]**. On the other hand, GSM supports the low-speed data service, like e-mail and web browsing. It is insufficient to support the multimedia service.

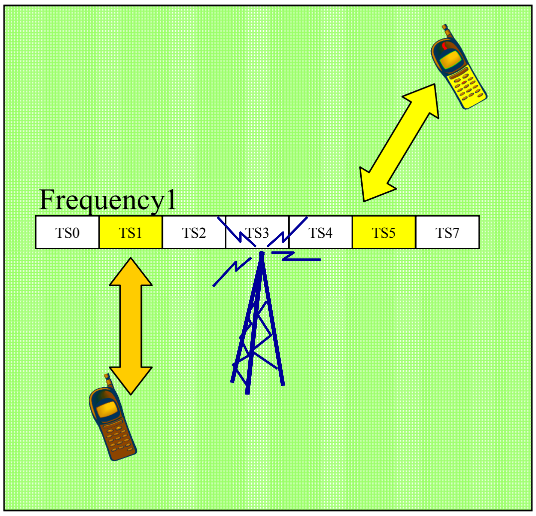


Figure 2.2: Time Division Multiple Access (TDMA)

Figure 2.3 shows the architecture of GSM network. The architecture of GSM is separated by BSS (Base Station Subsystem) and NSS (Network Switching Sub system).

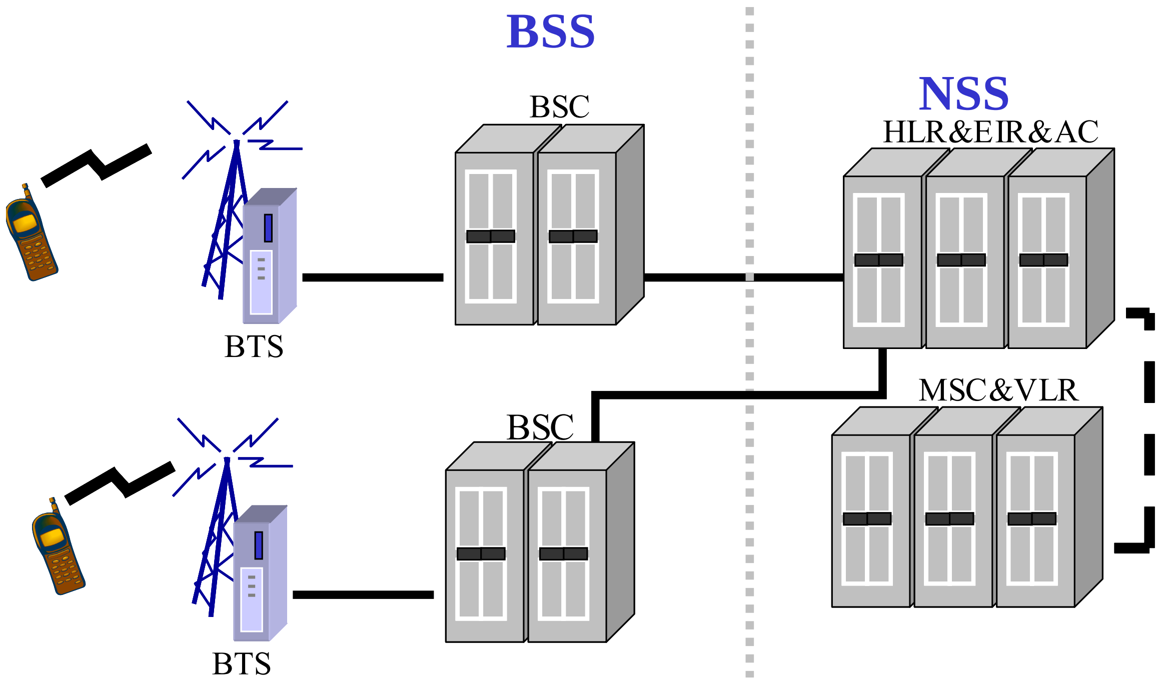


Figure 2.3: GSM Network Architecture

The GSM radio access network called BSS consists of the following elements **[8]**:
BSC *(Base Station Controller)* is responsible for radio path and RRM (Radio Resource Management). The main functions are the connection establishment among subscribers and NSS, mobility management, statistical raw data collection, air interface signaling support.
BTS *(Base Transceiver Station)* is the network radio terminal forming the air interface that subscribers use for network access and communication. It takes care of air interface signaling, air interface ciphering and speech processing.

NSS (Network Switching Subsystem), the switching part of the GSM network contains the following elements:
MSC *(Mobile Switching Center)* performs the traffic path connections and is responsible for the majority of the connection management related entities.
VLR *(Visitor Location Register)* contains subscribers and security information of the active subscribers located in the radio network part. The nature of the data in the VLR content is not constant. When the subscribers change their location(s), the VLR data changes respectively.
HLR *(Home Location Register)* is the static data storage of the subscriber information. The HLR also contains the subscriber location information, but the accuracy of this information is on the VLR level.
AC *(Authentication Center)* maintains the security information of the active subscribers.
EIR *(Equipment Identity Register)* maintains security information related to the mobile equipment, like the series number of the mobile phone.

BSS maintains the radio interfaces among NSS and subscribers. BSS decides whether subscribers should handoff to the other cells according to the measurement reports. NSS manages the mobility of subscribers and the communication connection. There is a database to store the information about subscribers in NSS. NSS can identify the legal subscribers in order to allocate the network resource **[9]**.

2.3 GPRS System

GPRS is a product of the Phase 2+ specifications developed by ETSI (European Telecommunications Standard Institute). GPRS is the way to transfer packet data over GSM air interface. GPRS requires the IP backbone to handle packet switching and connections to the Internet and other data networks. The basic packet switched data core consists of two essential elements: SGSN (Serving GPRS Support Node) and GGSN (Gateway GPRS Support Node). **Figure 2.4** shows the architecture of GPRS network **[10]**.

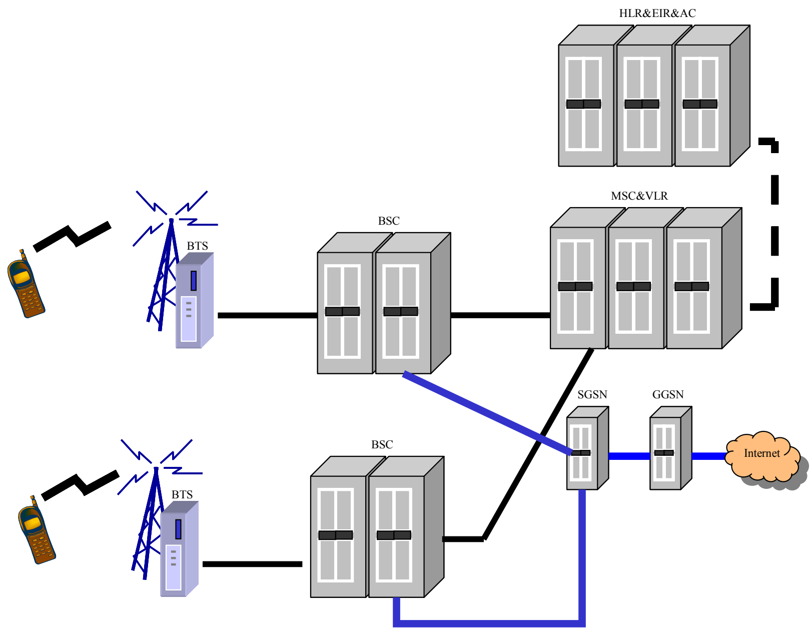


Figure 2.4: GPRS Architecture

The radio access interface of GPRS inherits the GSM network. GPRS contains the original GSM protocol and the new GPRS protocol. The data traffic and signaling is controlled by GGSN and SGSN. The subscriber database is still managed by HLR and VLR. The signaling links between the GPRS nodes and the GSM blocks will be SS7 (Signaling System #7). The signaling between GPRS nodes follows the GPRS protocol stacks as defined by the specification, illustrated in **Figure 2.5**.

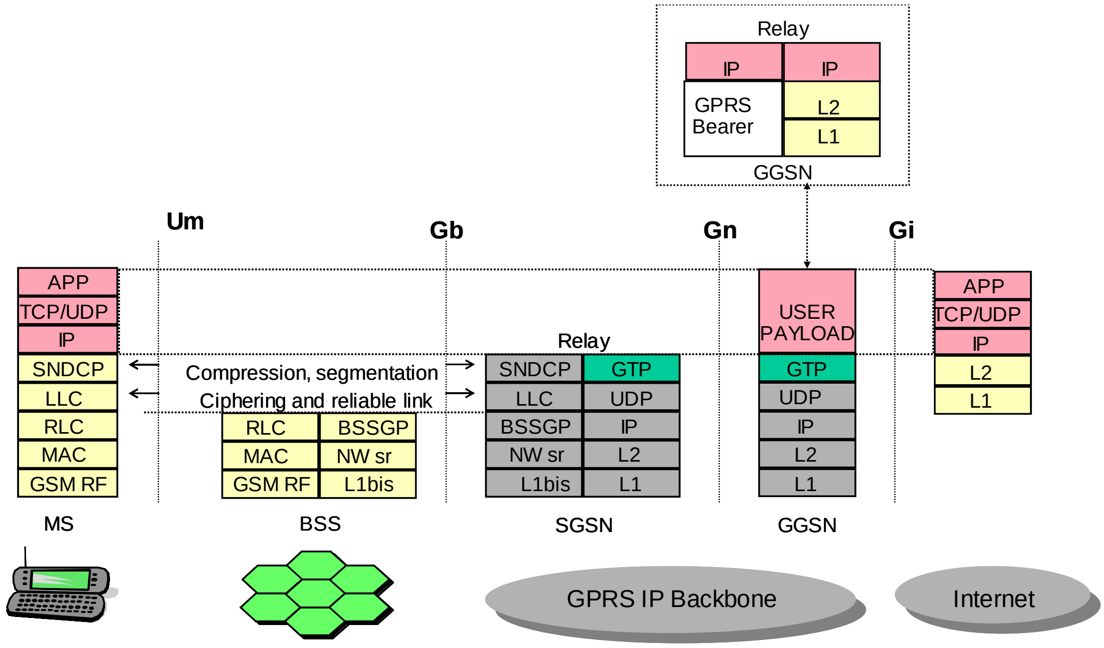


Figure 2.5: GPRS Protocol Stack

The GPRS data and signaling transmission plane consists of some standard protocol like IP, and GPRS specific protocols. We explain the protocols implemented by GPRS.

The Gn interference:

GTP *(GPRS Tunneling Protocol)*: it deals with IP datagram coming from the external network and tunnels it across the GPRS service nodes. There are multi GGSN and SGSN interfaces. Hence, for every packet the GTP provides a Tunnel Identifier (TID) which identifies the destination and transaction to which the packet/datagram belongs to. The transactions are identified using some logical identifiers as well as the IMSI (International Mobile Subscriber Identity) **[11]**.

TCP/UDP: TCP (Transmission Control Protocol) is used to transfer PDUs across the Gn interface with reliability (acknowledgement and retransmissions). UDP (User Datagram Protocol) carries PDUs across the Gn interface for all signaling information and subscriber data that does not require reliability.

IP (Internet Protocol): it is used to route subscriber data and signaling information across the Gn interface. The IP datagram size will be limited to the Physical Layer Maximum Transmission Unit (MTU) capability. An IP datagram can be as large as 65,535 bytes, but the Physical Layer MTU is less than this, therefore, the fragmentation is needed. The GSN (GGSN or SGSN) has to decide the MTU and they carry out the fragmentation. The address which is used on the IP layer of Gn interface will be source to final destination, this means even if there are intermediate GSNs for routing intention, the destination address will be the final GSN address.

The Gb interference:

SNDCP (Sub Network Dependent Convergence Protocol): it is used among SGSN and subscribers to convert Network layer PDU into a suitable format for underlying network architecture. The data units over the Gn interface to the SGSN are known as Network Layer PDUs (N-PDUs). GPRS supports several network layer protocols providing the protocol transparency for the subscribers of the service. The set of protocol entities sitting above SNDCP consists of commonly used network protocols. These all use the same SNDCP entity, which performs multiplexing of data coming from different sources before sending it to the end subscriber using the services of other layers like LLC. The SNDCP needs to index the Network Layer PDP (Packet Data Protocol) to discriminate different Network Layer Services protocols **[12]**. This index is the NSAPI (Network Service Access Point Identifier). SNDCP provides functions like:

- Multiplexing N-PDUs from one or several network layer entities onto the appropriate LLC connection
- Buffering of N-PDU for acknowledged service
- Compression and decompression of the protocol information and subscriber data
- Segmentation and reassembly of the compressed data to the maximum length of the LLC-PDU
- Negation of the control parameters (XID) among SNDCP entities

LLC (Logical Link Control): Used to provide a highly reliable ciphered logical link between SGSN and subscribers. It uses both Acknowledged and Unacknowledged mode of frame transmission depending of negotiated QoS for the end subscriber. LLC manages the frame retransmission, buffering, information length based on subscriber's NSAPI which is again decided based on negotiated QoS delay class **[13]**.

BSSGP (Base Station GPRS Protocol): It is responsible for the routing of information between the SGSN and BSS **[14]**. It deals with some elements of Quality of Service but does not carry out any form of error correction. The primary function is to provide radio-related information to be used by the Radio Link Control (RLC) and

Medium Access Control (MAC) functions on the air interface. The LLC uses the services of BSSGP for data transfer, the GMM (GPRS Mobility Management) deals with paging and radio status request. The Relay function at the BSS provides transfer of LLC frames between the RLC/MAC layer and the BSSGP layer. The BSSGP sends the information to the Network Services with some routing information to decide the destination (the BSS from SGSN end or SGSN from BSS end). The identifiers include:

- **BVCI (BSSGP Virtual Connection Identifier)**: BSSGP sends this information to the Network Services layers for efficiently routing signaling/data information to the correct peer functional entities which could point to single user data, or point to multipoint data (data send to multiple users) or signaling functional entities. Each BVCI is unique between two peer entities
- **LSP (Link Selection Parameter)**: Used in conjunction with the BVCI to aid in selecting physical link for load sharing process
- **NSEI (Network Service Entity Identifier)**: The Network Service Entity at the BSS and the SGSN provides the network management functionality required for the operation of the Gb interface. The NSEI together with BVCI uniquely identifies a BSSGP virtual connection

NS (Network Service): This layer uses Frame Relay across the Gb interface and could be a point-to-point connection between the SGSN and the BSS or a Frame Relay Network **[15]**. This layer uses a look table made up of DLCI (Data Link Connection Identifier) which is used to indicate the routing path between the SGSN and the BSS. The initial value of the DLCI derived from the BVCI, NSEI and LSP supplied by the BSSGP layer. This value will change as the frame passes through the Frame Relay Network and reach the final destination

The Um interference:

RLC (Radio Link Control): The RLC function is responsible for

- Transferring LLC-PDU between the LLC layer and the MAC function
- Segmentation of LLC-PDU into RLC data blocks and re-assembly of RLC data block (to fit into TDMA Frame blocks)
- Segmentation and Re-assembly of RLC/MAC control messages into RLC/MAC control blocks

- Backward error correction for selective transmission of RLC data blocks

RLC segmentation is a process of taking one or more LLC-PDUs and chopping them into smaller RLC blocks. The LLC-PDUs are collectively known as a Temporary Block Flow (TBF) which allocates resources to one or more Packet Data Channel (PDCH). The TBF is temporary and it is maintained only for the data transfer. Each TBF is assigned a Temporary Flow Identity (TFI) by the network and which is unique among concurrent TBFs. The same TFI value may be used concurrently for TBFs in opposite directions. The RLC data blocks consist of an RLC header, an RLC data unit and spare bits. The RLC data block along with a MAC header may be encoded using one of the four defined coding scheme. The coding scheme is critical in deciding the segmentation process.

MAC (Medium Access Control): MAC controls the access signaling across the air interface. Access signaling includes management of the shared transmission resources (assignment of the radio block to multiple subscribers on the same timeslot). MAC achieves these functionalities by placing a header in front of the RLC header in RLC/MAC data and control blocks. The MAC header contains several elements, certain elements are direction specific (downlink/uplink). The key parameters of MAC header are:

- Uplink Status Flag (USF): It is sent in all downlink RLC/MAC blocks and indicates the owner or user of the next uplink Radio block on the same timeslot
- Relative Reserved Block Period (RRBP): Signifies a single uplink block in which subscribers will transmit control information
- Payload Type (PT): Type of data (control block or data block) contained in the remainder of the RLC/MAC block
- Countdown Value: It is sent by the mobile to allow the network to calculate the number of RLC data blocks remaining in the current uplink TBF **[16]**

The GGSN is the node that is adopted by the packet data network due to evaluation of the PDP address. It contains routing information for PS-attached subscribers. The routing information is used to tunnel to the subscriber's current point of attachment. The GGSN may request location information from the HLR via the optional Gc interface. The GGSN is the first point of PDN interconnection with a GSM PLMN supporting GPRS (i.e., the Gi reference point is supported by the GGSN). GGSN functionality is common for GSM and UMTS. The main functions of GGSN are:

- Routing, encapsulation, compression and encryption
- VPN Tunneling(Intranet)/GTP Tunneling(SGSN) management
- Billing record counts and sends to the charging Gateway
- The interface between GPRS and PDN (IP/X25) management

The SGSN supports GPRS for GSM (i.e., the Gb interface is supported by the SGSN) and/or UMTS (i.e., the Iu interface is supported by the SGSN). At PS attach, the SGSN establishes a mobility management context containing information pertaining to mobility and security for the subscriber. At PDP Context Activation, the SGSN establishes a PDP context, to be used for routing purposes, with the GGSN that the subscriber will be using **[17]**.

When every packet is transmitted through GGSN, the packet needs to encase the header. In GPRS core network, the header of a packet contains IP addresses and GTP. The service node can route the packet by IP address, illustrated in **Figure 2.6**.

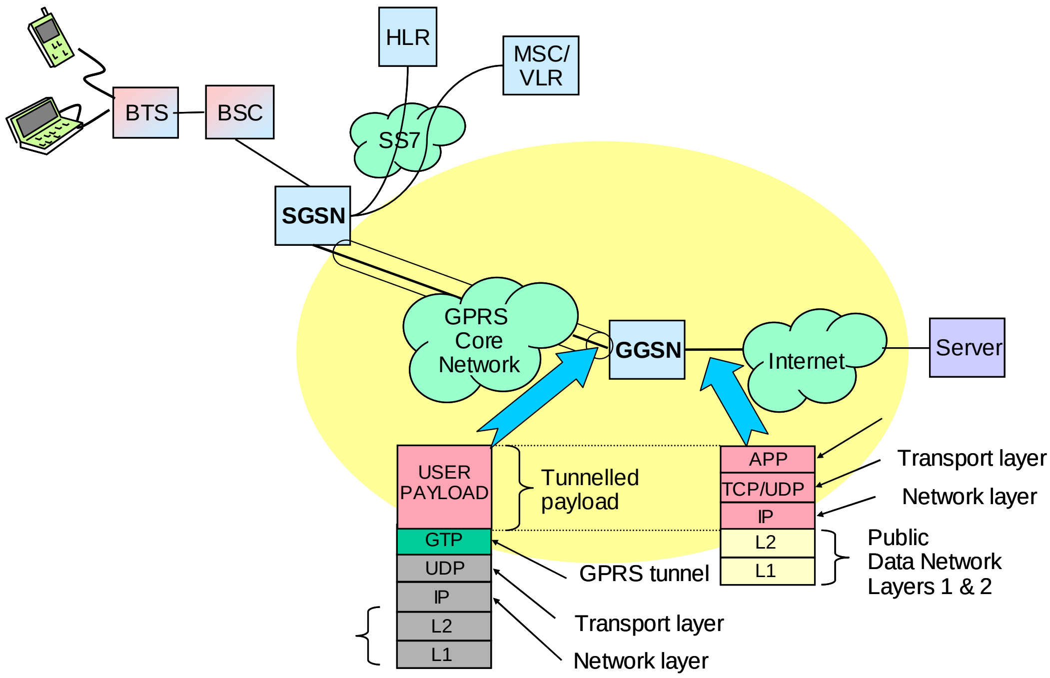


Figure 2.6: GTP Payload

The packet is translated into the Transport layer, Network layer, and Physical layer when SGSN processes the packet. SGSN transmits the packet to the correct subscriber according to the GTP (GPRS Tunnel) information. There is an IMSI (International Mobile Subscriber Identity) to identify subscribers in the GTP tunnel, illustrated in **Figure 2.7**.

	Bits							
Octets	8	7	6	5	4	3	2	1
1	Version			PT	Spare '111'			LFN
2	Message Types							
3-4	Length							
5-6	Sequence Number							
7-8	Flow Label							
9	SNDCP N-PDU LLC Frame Number							
10	X	X	X	X	X	X	X	X
11-12	reserved							
13-20	TID=IMSI+NSAPI							

Figure 2.7: GTP Header

GPRS can offer the data rate of over 150kbit/s. GPRS enables efficient use of radio resources by allowing many data subscribers to share the air interface on a statistical basis. The radio resources will only be used when data is actually being transmitted or received, even though the subscribers remain connected all the time. Call set-up will be almost instantaneous and subscribers can be charged on the basis of actual data transmitted, rather then based on connection time.

2.4 UMTS System

The 3G/UMTS specifications defined that the new air interface and system capabilities should inherit the existing 2G systems, such as GSM and GPRS. UMTS is the ability to support the multimedia data communication. Unlike in GSM, where the architecture has been reasonably stable for many years, UMTS systems are expected quickly to evolve towards All-IP technologies. The standards dictate the configuration of the open interfaces and the function of each subsystems; however, the implementation is vendor/operator specific **[18]**. UMTS Terrestrial Radio Access Network (UTRAN) has been defined with more capable elements and protocols. New elements in the coexisting GSM/UMTS networks are: Node B, RNC, and interface modules. Other network elements need only partial software or hardware upgrade. The **Figure 2.8** below shows a typical Release 99 network architecture **[19]**.

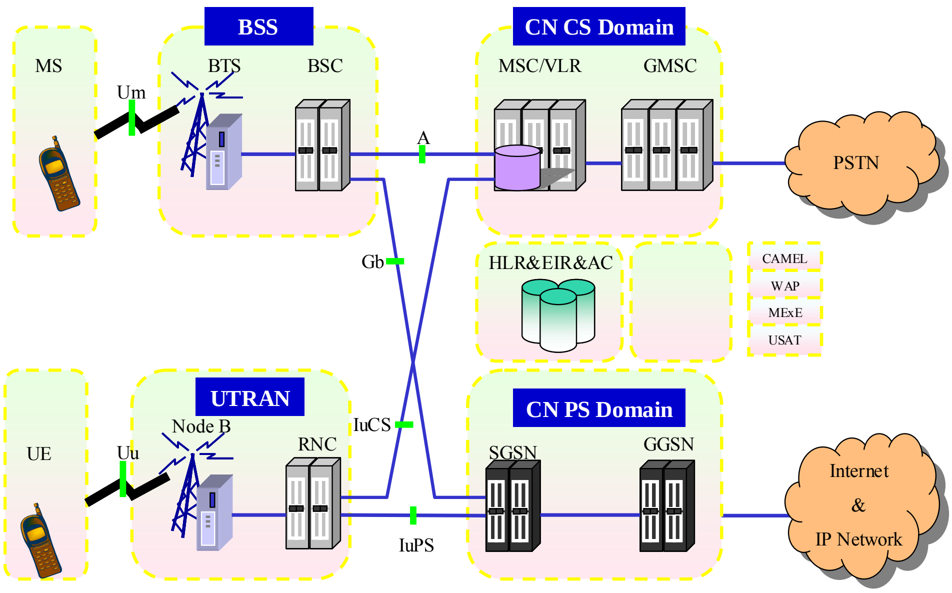
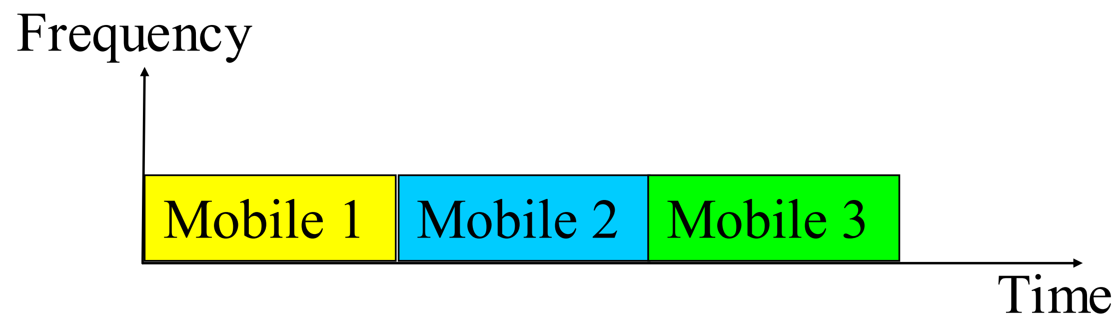
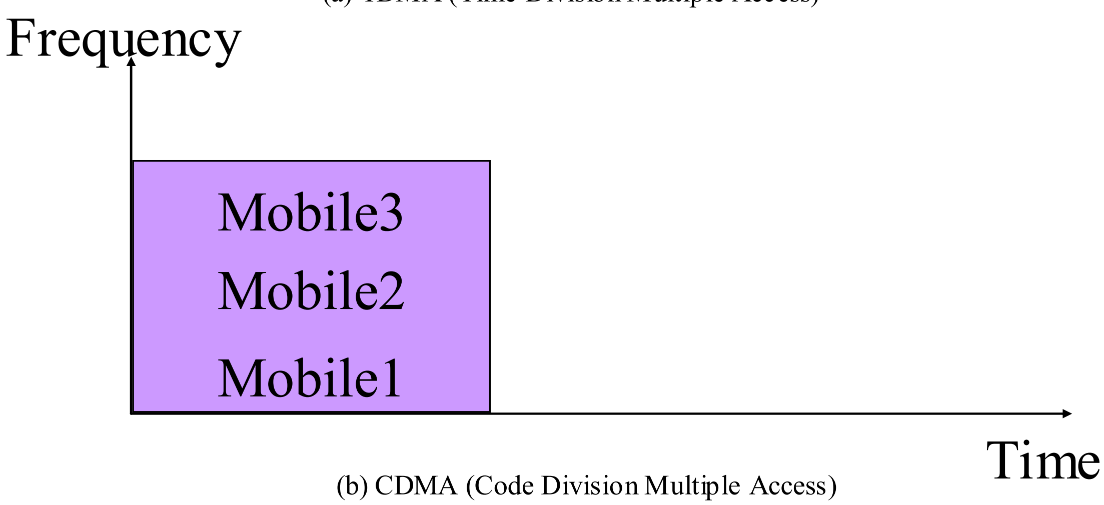


Figure 2.8: Release 99 Architecture

There are many obvious differences between the GSM/GPRS and UMTS air interface. GSM/GPRS uses TDMA for the technology of radio access and UMTS uses CDMA for the technology of radio access. For the reason of simplicity, we can say that there are two distinct categories of differences in UMTS: the multi-services to a subscriber and the air interface itself, illustrated in **Figure 2.9**.



(a) TDMA (Time Division Multiple Access)



(b) CDMA (Code Division Multiple Access)

Figure 2.9: (a) TDMA and (b) CDMA

In CDMA (Code Division Multiple Access), all subscribers share the same frequency and time, but are separated by codes. A common example to explain this phenomenon is to think them as an international gathering, where people from different nationalities are talking to their fellow countrymen. Although people are from different countries (for example, two people from Taiwan), they can understand each other, as they are capable of detecting the language being used. On the other hand, other people conversation is noise for the Taiwanese. If the surrounding noise is whispered, they can share a lot of information with each other without having to repeat anything. However, as more and more people start talking (in different languages), the noise becomes so loud that the conversation becomes more difficult to keep. Eventually, if the noise becomes very loud, it will be impossible to hear each other. Hence, there is no communication. In the same way, each group of subscribers in CDMA is given a shared code. Many codes occupy the same channel, but only users associated with a particular code can understand each other. In CDMA, the spectrum is split into channels, illustrated in **Figure 2.10**. Each channel carries several subscribers of variable capacity needs, separated by a code.

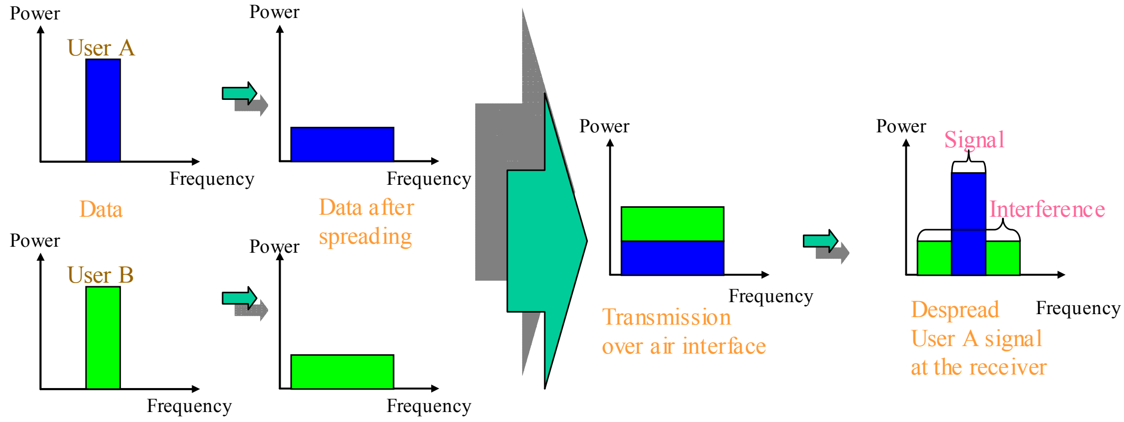


Figure 2.10: Spreading and Sharing the Same Frequency Space

The number of subscribers that shares the same frequency space is limited by the number of codes, as well as by the amount of interference in the cell (the region of coverage). Also, the subscriber may have variable bit rates. This means that some subscribers need more capacity (bit rate and power) to transfer information faster than other subscriber.

If a base station that has a block of information that it needs to send to a mobile user. This block of information could contain speech, video, packet data and signal. If we transmit the block of data, we could reduce the amount of the power needed to transmit the information by spreading it along a wide frequency band. Similar to making a cake, once you add the topping, it sits on top of the cake in a pile. By using a knife, the topping is spread out to cover the whole surface area of the cake. The bigger the surface of the cake, the better you are able to spread the topping, as illustrated in **Figure 2.11**.

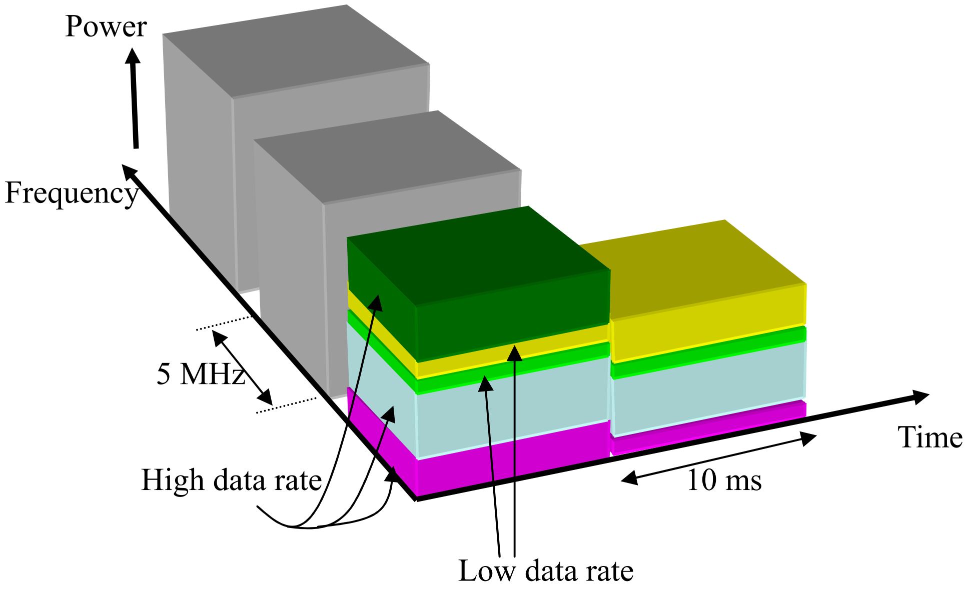


Figure 2.11: UMTS Trades-off Range against Number of Subscribers and Data Rate

In UMTS, the frequency bandwidth is fixed by specifications (4.2 - 5 MHz). However, the power and the spreading factor are variables. The spreading factor indicates to what degree we are able to spread the data over the fixed frequency band **[20]**.

The UMTS multi-service environment can support bit rates from 12.2kbit/s to 2M bit/s. The services are classed to Real Time (RT) and Non-Real Time (NRT), each of these have a different quality class and different error ratios, BLER (Bit Loss Error Ratio), and BER (Bit Error Ratio). The delay sensitivity is from 100ms to seconds. There is an asymmetric traffic in the uplink and downlink traffic. As a result, more transmission capacity is needed.

In UMTS, the size of the cell actually changes. As more capacity (that is, more voice calls or higher data rates) is applied to a cell, the actual diameter will shrink. This phenomenon is referred as cell breathing.

The single cell capacity of UMTS is based upon the traffic load and the neighbor cell interference. All neighboring cells use the same frequency, and therefore the concept of gaining through soft handovers is introduced. The usage of soft handovers increases the load in a cell, but the overall effect is positive since the interference is reduced. Also, in CDMA, fast power control mechanisms are used to ensure high capacity. Very fast and accurate transmit power control is required.

Finally, the differences between GSM/GPRS and UMTS are shown in **Table 2.1**.

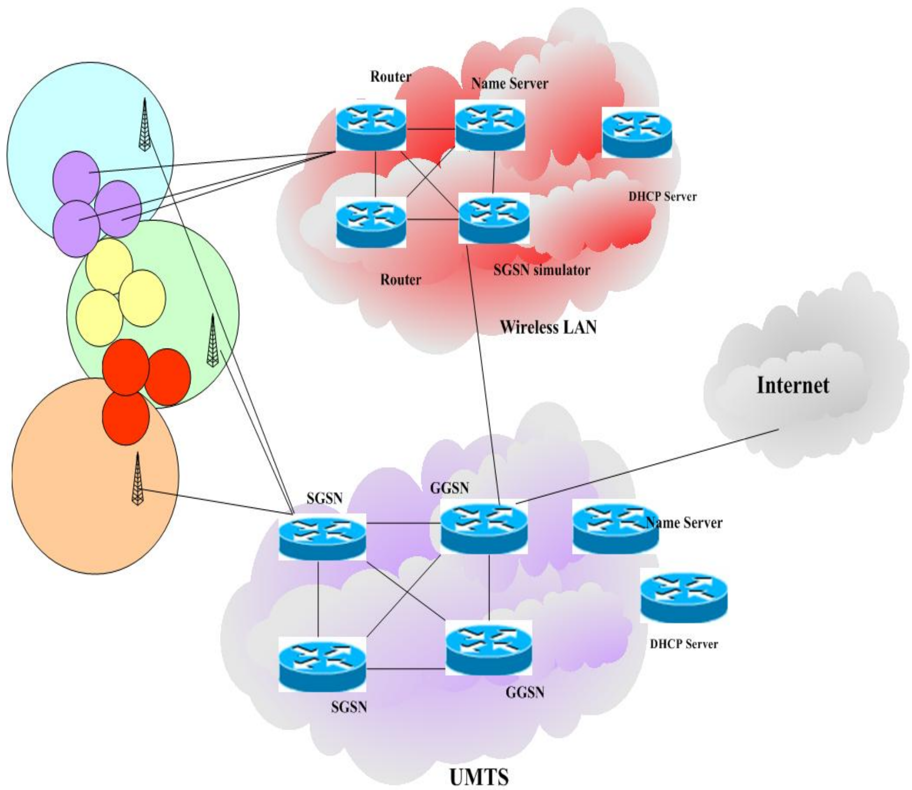
Table 2.1: The Differences between GSM/GPRS and UMTS

	UMTS	GPRS
Carrier Space	5MHz	200kHz
Frequency Reuse Factor	1	1~18
Power Control Frequency	1500Hz	2Hz or lower
Quality Control	Radio resource management algorithms	Network planning (frequency planning)
Frequency Diversity	5MHz bandwidth gives multipath diversity with Rake receiver	Frequency hopping
Packet Data	Load-based packet scheduling	Timeslots based scheduling with GPRS
Downlink Transmit Diversity	Support to improve downlink capacity	Not support by the standard, but can be applied

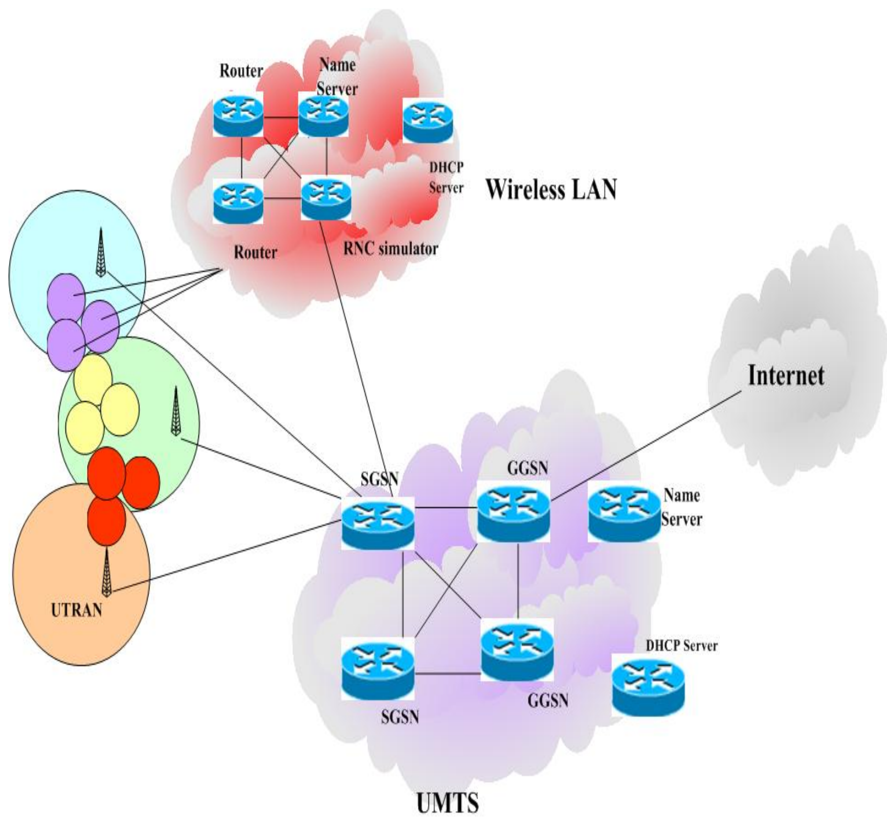
2.5 Inter-working System Architectures

In general, there are three possible kinds of integrated UMTS and WLAN architecture. They are loose coupling, tight coupling, and peer networks **[21]**. Tight coupling is the architecture where the fewest changes are made from the original UMTS network. In loose coupling, data traffic transmission is more efficient without going through the UMTS core network. In both tight coupling and loose coupling, the signal of UMTS can be carried over WLAN. In handoff, it's more efficient in tight coupling environment than in loose coupling architecture. With regards to multiple mobility management schemes, it will be used in an integrated WLAN and UMTS network. For example, they can be GPRS, 802.11 mobility management, Session Initiation Protocol (SIP), and Mobile IP. GPRS is responsible for the mobility management in the packet switched domain of UMTS. IEEE 802.11 mobility management is the mechanism that can offer mobility between different APs in WLAN. SIP is another popular scheme to handle the mobility of mobile node at many levels. Mobile IP provides global mobility for the inter-working of WLAN and UMTS.

In tight coupling architecture, the WLAN can emulate the role of one RNC or one SGSN **(Figure 2.12(a), (b))**. At first, we consider that it plays the role of SGSN and we call it SGSN simulator to differentiate from the true SGSN in reality. There will be many WLANs to be connected to the SGSN simulator and the SGSN simulator will be connected to the UMTS core network. WLAN and UMTS may separately belong to different operators in operation. In this architecture, different Routing Area Identity (RAI) values of WLAN and UMTS will coexist because they belong to different routing areas. When mobile nodes move and handoff from WLAN to UMTS or from UMTS to WLAN, the event of inter-SGSN routing area that updated with different RAI values will come up. If the mobile nodes of WLAN and UMTS are under the same GGSN node, they will get their IP address assigned from the same IP pool. By this way, the mobility between WLAN and UMTS UTRAN will not result in additional process to change different IP address. The signaling traffics and data traffics to and from the WLAN will go through the UMTS core network and the Home Subscriber Server (HSS) in UMTS will be responsible for the management of users' location.



(a)



(b)

Figure 2.12: Tight Coupling Architecture

Most part of the mobility in tight coupling scenario follows UMTS specifications. Many process and working steps are the same as or similar to related event in pure UMTS network. When a mobile node moves to one area where a WLAN network can be accessed, it will be possible that the mobile node detects that the higher bandwidth WLAN is available to use as access network instead of lower bandwidth UMTS network. At first, the mobile node associates with a WLAN AP. Secondly, an inter-SGSN routing area update procedure will be run and the SGSN simulator will be the new SGSN of the mobile node with the original SGSN as old SGSN. After the prior two steps, the mobile node will be able to connect to the UMTS core network through 802.11 WLAN. If there is any corresponding node in other network, the data packets that be delivered from the corresponding node to the mobile node can also arrive to the WLAN through SGSN simulator and be received by the mobile node.

Besides of playing the role of SGSN, the simulator in WLAN system can be one emulator of RNC and have the functions of RNC in UMTS network. It will be connected with SGSN to the core network by the Iu interface. The WLAN system can be treated as one or many RNC and several UTRANs of UMTS in practice. The mobile node must go through SGSN and GGSN in UMTS to access other outside network. As the UE in one pure UMTS system, the mobile node under WLAN gets its IP address from the IP pool in GGSN. If the WLAN and UMTS both belong to the same operator, AAA functions (Authentication, Authorization, and Accounting) can be carried out as in one pure UMTS environment.

Loose coupling is a master/slave architecture. The UMTS is the master and the WLAN acts as the slave one. There is one new node in this architecture, GSN simulator. The GSN simulator has some functions which combine GGSN with SGSN. In other words, The GSN simulator plays the role of GGSN and SGSN in operation. In the following figure, IEEE 802.11 WLAN is a visiting network to the UMTS core network **(Figure 2.13)**. It can be deployed by the UMTS operator or by the other operators. One GSN simulator can manage several WLAN. UMTS and WLAN are not the same routing area in mobility management. They are assumed to be in different IP address domains. The main difference between loose coupling and tight coupling is that the signaling traffic goes through the UMTS core network but not the data traffic. GSN simulator has an interface to connect with UMTS core network for signal and an interface to connect with IP networks for signal and data.

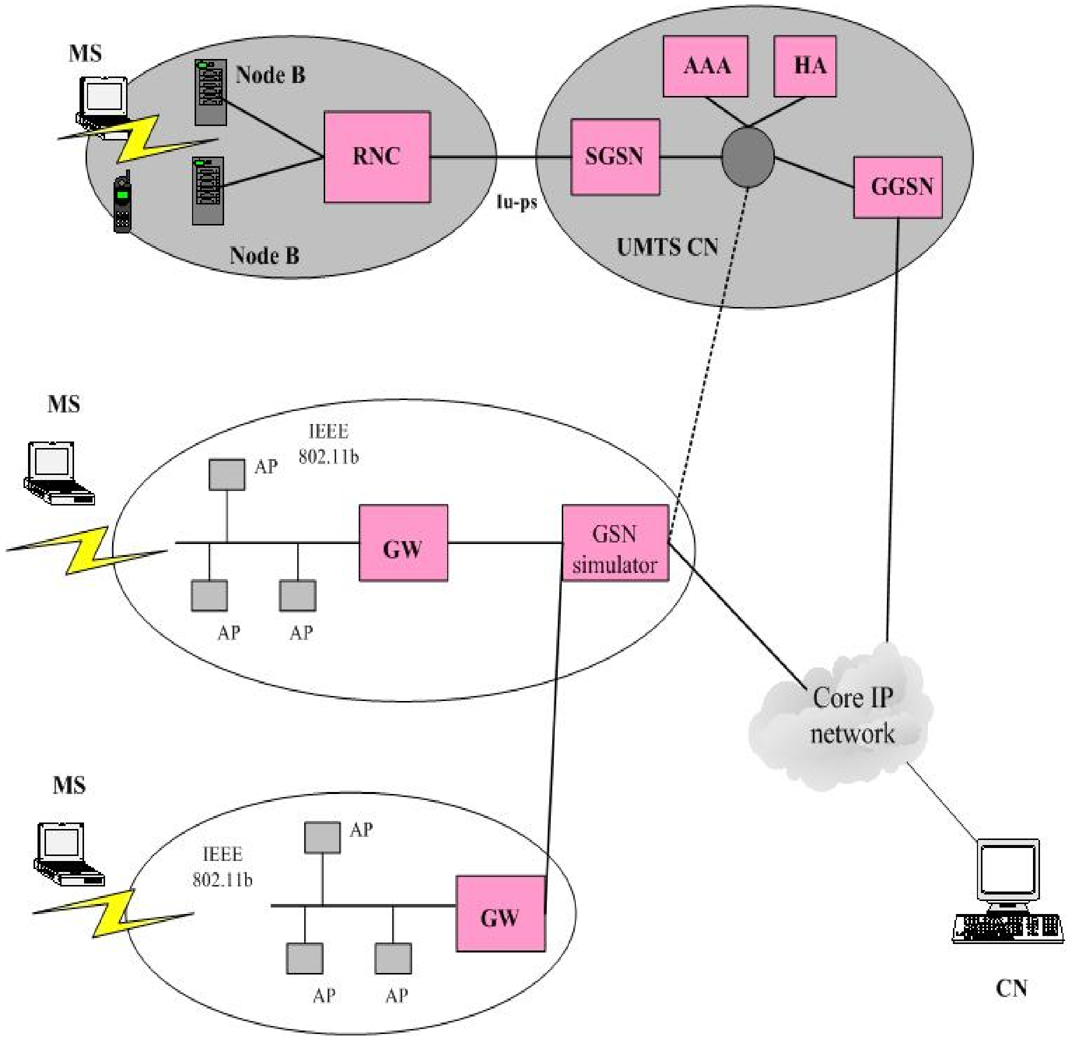


Figure 2.13: Loose Coupling Architecture

2.6 Differentiated Services Architecture (Diffserv)

Differentiated service (Diffserv) **[22-23]** provides QoS guarantees in the Internet, instead of per-flow guarantees that IntServ **[24]** provides, it maps multiple flows into an aggregate class of service which denoted by 6 bits differentiated services code point (DSCP) in the IP packet header. Those marked DSCP packets will be recognized and have a particular treatment or per-hop behavior (PHB) according to the class priority at each network nodes along the path. The **Figure 2.14** shows the Diffserv architecture overview. When the sender initiates a flow to the receiver, ingress node will perform marking, policing, shaping and forming the traffic into a specific behavior aggregate. After the traffic enters the core network, each interior nodes will forward those packets based on their marked DSCP. These behaviors are called per-hop behaviors (PHBs). There are two PHBs have defined: Assured Forwarding (AF) **[25]** and Expedited Forwarding (EF) **[26-27].** The last egress node performs remarking, shaping or even dropping the incoming traffic.

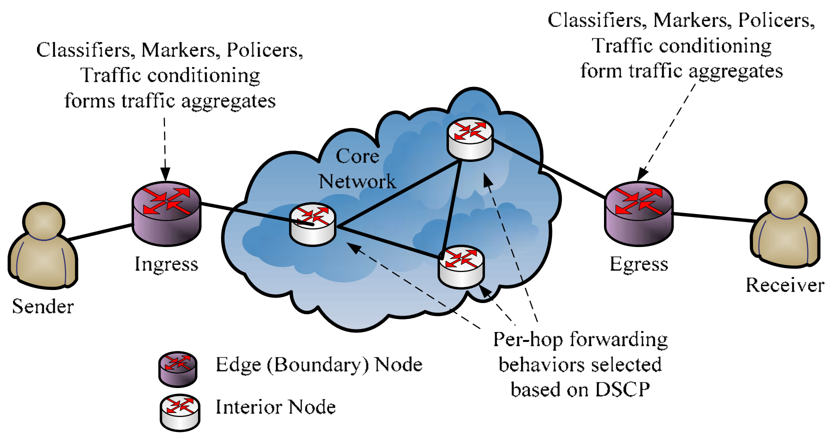


Figure 2.14: Diffserv Architecture

One operator's network could be a Diffserv domain, maybe it can be divided into several non-overlapped domains depicted as **Figure 2.14**. The traffic inside the domain is under control by the operator belong the domain, while the traffic entering the domain is not. Therefore, the edge nodes need to police and shape the traffic to prevent that external customer or operator would misuse resources of the domain. To enlarge the Diffserv domains, it will be a Diffserv region **(see Figure 2.15)** in which many operators provide different services to its customers by having a service level agreement (SLA) **[28-29]**.

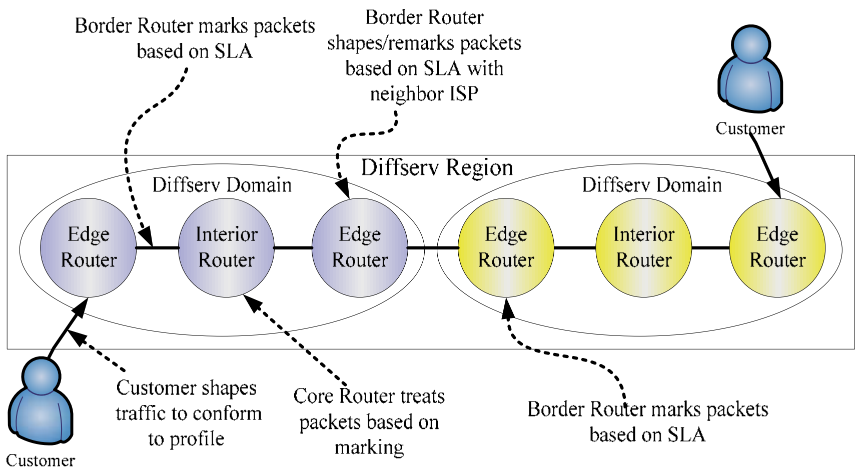


Figure 2.15: Diffserv Region-wide Architecture

Diffserv definitely solve the scalability problem of Intserv, since each interior node needs to perform scheduling based on the DSCP, instead of storing each flow's information. However, Diffserv is impossible to reserve resources in domain, because no resource management mechanisms are supported in Diffserv domain. For example, a customer uses the premium traffic class which is expected very low delay and few lost packets. Unfortunately, it could be happened that many users may request the service at the same time, there will be congestion in the domain, and the original premium class could be experienced as a degraded service with lots of lost packets.

2.7 Intserv over Diffserv

This scheme described in RFC2998 **[30]** provides end-to-end QoS for applications by using the integrated services mechanism over Diffserv domains. The network consists of some Intserv nodes which classify each flow and perform admission control, the Diffserv nodes that process only aggregate traffic control per PHB. Routers in the mixture network may or may not participate in RSVP signaling. However, there is a problem of how to do resource management in the Diffserv domain. The end user may request a quantitative reservation using Intserv and expects that some guarantee could be kept in the intermediate nodes, which the Diffserv architecture does not support. RFC2998 proposed three possible solutions are as follows:

Static provisioned resources: The amount of traffic that each edge node allowed into the network will be configured statically by the operator. It could bring the resource waste due to the over resource provisioning.

Resources dynamically provisioned by RSVP: In this solution, some RSVP traffic sent into the network and some interior nodes are RSVP-aware. This could raise up scalability problem that the RSVP origin had since the per-flow states still exist in the core network, which will be solved by the use of aggregation of RSVP flows **[31]**. However, RSVP is a very complex protocol and the aggregation is not an easy job, it also will make the RSVP aggregation more complex.

Resources dynamically provisioned by other means: A bandwidth broker **[32]** or another signaling protocol can be alternatives of RSVP.

2.8 Resource Management in Diffserv (RMD)

The RMD protocol is based on standardized Diffserv principles for traffic differentiation and performs an edge-to-edge resource management scheme. RMD extends the Diffserv principles to provide dynamic resource management and admission control in Diffserv domains. In the RMD framework, there are two types of protocols defined: Per Domain Reservation (PDR) protocol and the Per Hop Reservation (PHR) protocol. The PDR protocol is used for resource management in the whole Diffserv domain, while the PHR protocol is used for managing resource for each node, on per hop basis. The PDR can either be a newly defined protocol or an existing one, such as RSVP or RSVP aggregation, while the PHR is a newly defined protocol. So far there is only one PHR protocol specified, called RMD on-demand (RODA) PHR **[33]** protocol. **Figure 2.16** shows the RMD architecture that mix up the PHR and PDR protocols.

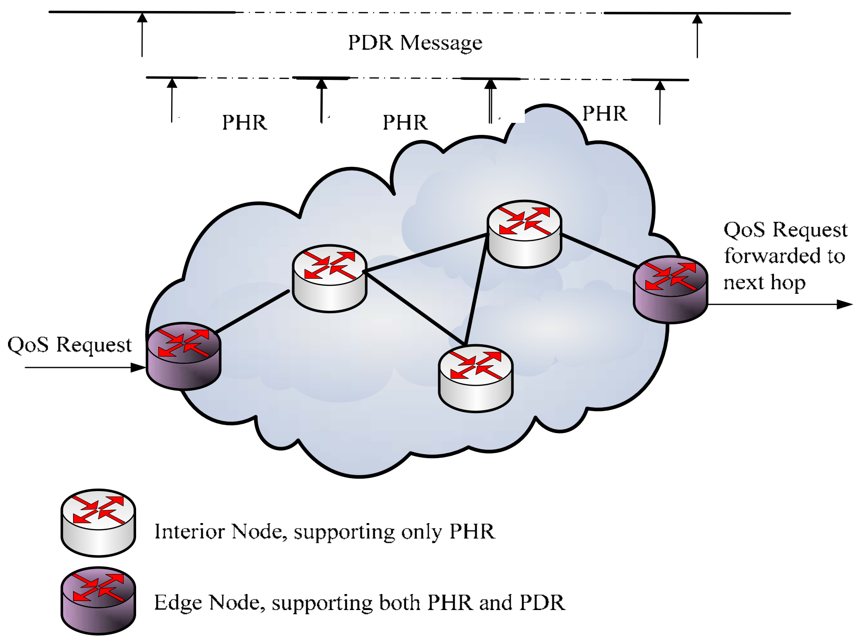


Figure 2.16: RMD Architecture

2.8.1 Per Domain Reservation – PDR protocol

PDR protocol is only implemented in the edge nodes of the domain. It handles the interoperation with external resource reservation protocols, the mapping of external QoS requests to DSCP, the translation of the parameters of these requests to parameters used by the PHR protocol (e.g., the amount of resource units to reserve on each hop along the path). Moreover, the PDR handles severe congestion once it has been signaled by the PHR protocol.

2.8.2 Per Hop Reservation – PHR protocol

The RMD framework defines two PHR groups:

The Measurement-based Admission Control (MBAC) PHR: The availability of resources is checked by means of measurements before any reservation requests are admitted, without maintaining any reservation state in the nodes along the path. These measurements are done on the average real traffic data load.

The Reservation-based PHR: It enables dynamic resource reservation per PHB in each node along the path. All the nodes maintain one state per PHB and no per flow states. The reservation is done in terms of resource units (e.g., bandwidth)

2.8.3 RMD Normal Operation

The RMD signaling messages are categorized into RODA PHR and PDR protocol messages. These signaling messages and their description are depicted in **Table 2.2**.

Table 2.2: RODA PHR and PDR Signaling Messages

	Signaling Message	Signaling Message Description
RODA PHR	**PHR_Resource_Request**	Initiate the PHB reservation state on all nodes located on the communication path between the ingress and egress nodes according to the external reservation request.
	PHR_Refresh_Update	Refreshes the PHB reservation soft state on all nodes located on the communication path between the ingress and egress nodes according to the resource reservation request that successfully processed by the PHR during a previous refresh period. If this reservation state doesn't receive a "PHR_Refresh_Update" message within a refresh period, reserved resources associated to this PHR message will be released automatically.
	PHR_Release_Request	Explicitly releases the reserved resources for a particular flow from a PHB reservation state.
PDR	**PDR_Reservation_Request**	Initiates or updates the PDR state in the egress. It's generated by ingress node.
	PDR_Refresh_Request	Refreshes the PDR states located in the egress. It's generated by the ingress node.
	PDR_Release_Request	Explicitly release the PDR state. It's generated by the ingress node.
	PDR_Reservation _Report	Reports that a "PHR_Resource_Request"/ "PDR_Reservation_Request" has been received and that the request has been admitted or rejected. It's sent by the egress node to the ingress node.
	PDR_Refresh_ Report	Reports that a "PHR_Refresh_Update"/ "PDR_ Refresh _Request" has been received and processed. It's sent by the egress node to the ingress node.
	PDR_Congestion_ Report	Used for severe congestion notification and it sent by egress to ingress.
	PDR_Request_Info	Contains the information that is required by the egress node to associate the PHR signaling message. It's generated by the ingress node.

When external QoS request arrives at the ingress node **(see Figure 2.17)**, the PDR protocol, after classifying it into appropriate PHB, will calculate the requested resource unit and create the PDR state. If the request is satisfied locally, then the ingress node will generate the "PHR_Resource_Request" and the "PDR_Reservation_Request" signaling message, which will be encapsulated in the

"PHR_Resource_Request" signaling message. The encapsulated PDR signaling message will be decapsulated and only processed by the egress node. The interior nodes receiving the "PHR_Resource_Request" must identify the DSCP type of the PHR signaling message and, if possible, reserve the requested resources. The egress node, after processing the "PHR_Resource_Request" message, decapsulates the PDR signaling message, then report to the ingress node.

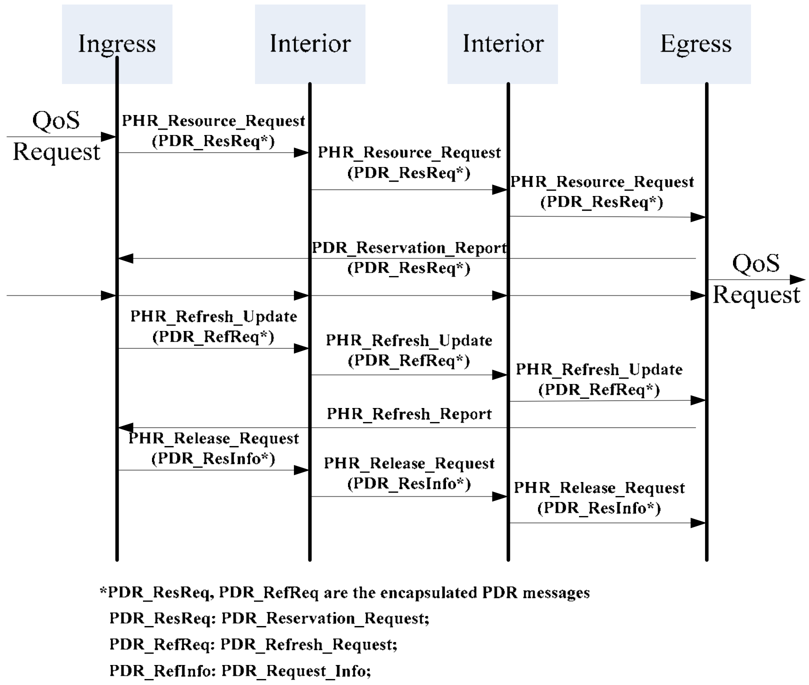
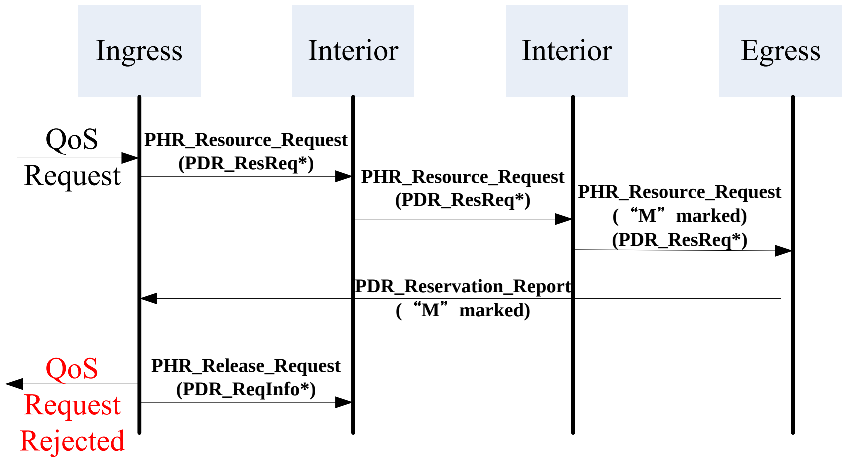


Figure 2.17: RMD Functional Operation for a Successful Reservation

After receiving this report message, the ingress node will inform the external source of the successful reservation, which will in turn send user data.

If there were no resource available in one of the interior nodes (see **Figure 2.18**), the "PHR_Resource_Request" will be "M" marked and, as a result, the reservation request will be rejected.



***PDR_ResReq, PDR_RefReq are the encapsulated PDR messages**
PDR_ResReq: PDR_Reservation_Request;
PDR_RefReq: PDR_Refresh_Request;
PDR_RefInfo: PDR_Request_Info;

Figure 2.18: RMD Functional Operation for a Failed Reservation

Chapter 3
QoS Management in Heterogeneous Home Networks

This chapter introduces a QoS mechanism to manage heterogeneous home networks. Based on an OSGi (Open Services Gateway Initiative) home gateway system, the mechanism can manage the integration of heterogeneous home networks protocols and devices to develop ubiquitous applications. It is an adaptive QoS management mechanism based on the RMD (Resource Management in Diffserv) architecture that can monitor real-time network traffic, and adaptively controls the bandwidth, to satisfy the minimum requirement for each application in home network congestion.

3.1 Introduction

Figure 3.1 shows the home network environment. The home network system consists of a residential gateway and a variety of networked appliances, and includes both wired and wireless sub-networks. Users can control these digital home devices by either wired or wireless links. Each sub-network is intended as a UPnP (Universal Plug and Play) sub domain in which home appliances are discovered by each other when they plug into the UPnP domain. This chapter adopts three real-time applications, namely Voice over IP (VoIP), High Definition (HD) video stream and Standard Definition (SD) video stream. The real-time traffic can be delivered by IP-based UPnP devices in home networks. A service provider supplies home network applications to in-home users across the OSGi home gateway. Service providers offer various services outside the OSGi home gateway including the following:

Energy Metering: The service provider manages the peaks and valleys in energy usage in the home network. Additionally, this service cooperates with automation to provide automated metering, remote control and home energy usage optimization.

Automation Service: The service provider provides a centralized point in the OSGi home gateway server to automate the home entertainment and utility systems. The gateway provides an integration and management point, allowing JINI and/or HAVi devices to be deployed and managed.

AV: The service provider provides on-demand Audio/Video multimedia content to in-home users.

Telecom: Some European operators provide triple-play services providing voice, data and broadcasting services using DSL (digital subscriber line) technologies.

QoS: Except for the outside home gateway QoS brokers, the service provider also negotiates with the residential QoS mechanism, which is built into the OSGi home gateway server.

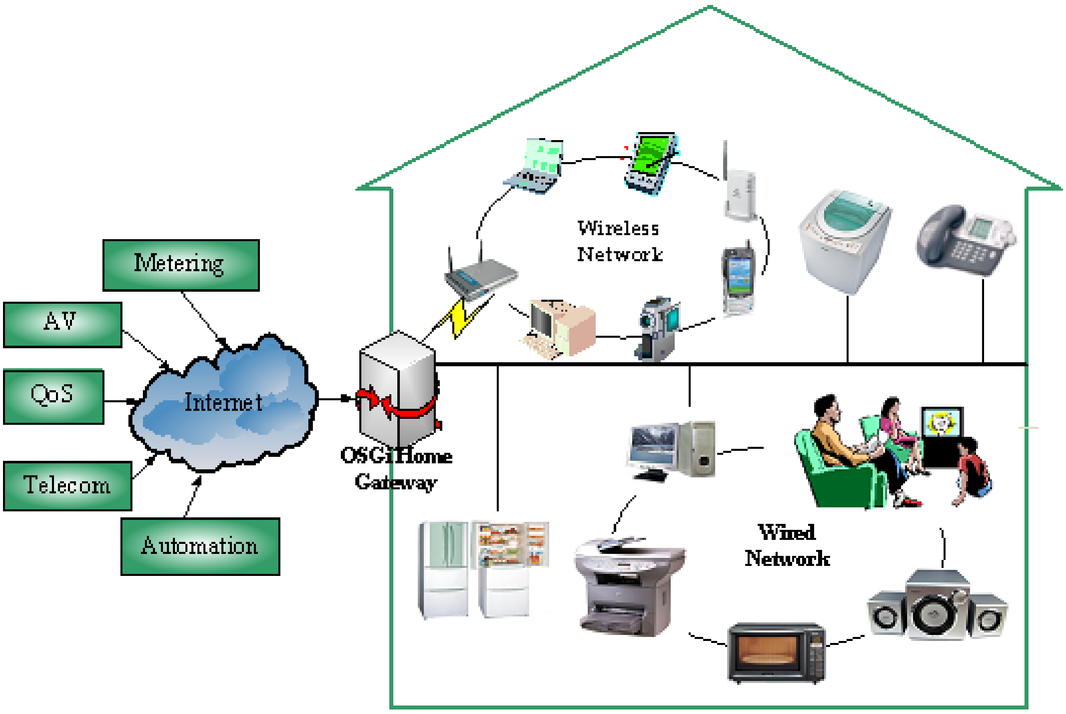


Figure 3.1: Home Network Environment

The service provider can pack the adaptive QoS management daemon into a java-based bundle, and upload it to the OSGi home gateway. The service provider or home users can monitor and remotely control the OSGi-related in-home devices by any handheld mobile device after these devices are deployed. This module minimizes the cost of existing system maintenance and functional enhancement. **Figure 3.2** shows an example of OSGi QoS deployment and control.

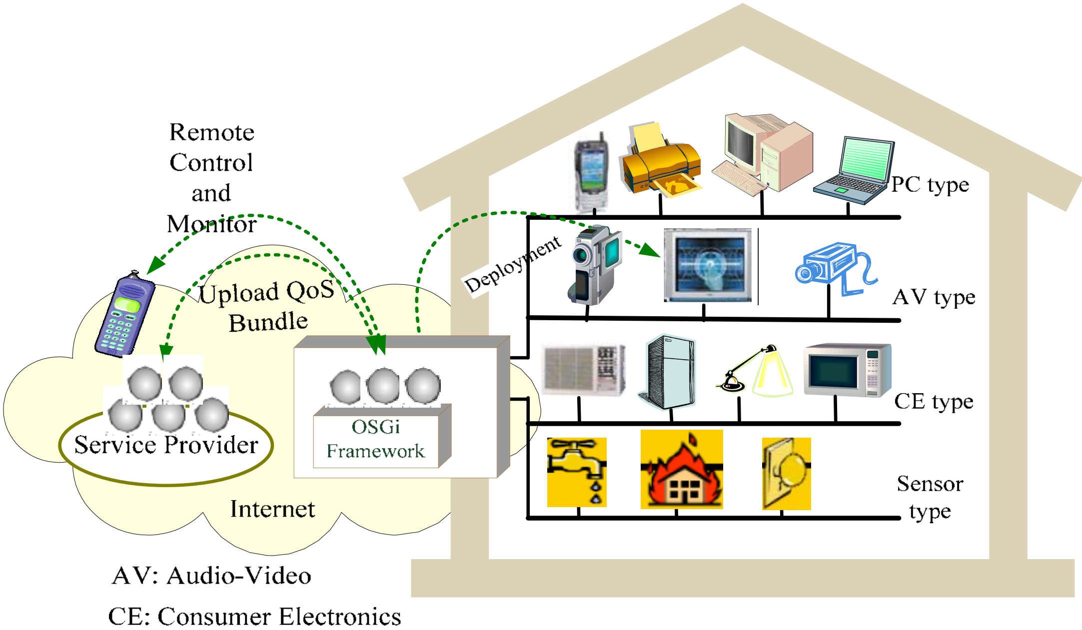


Figure 3.2: OSGi QoS Deployment and Control

3.2 QoS-aware OSGi Home Gateway in RMD Mechanism

The proposed OSGi home network is built on the RMD network architecture. As mentioned earlier, RMD has two resource reservation protocols, namely PHR and PDR. PHR protocol enables the reservation of resources per PHB in each node within a Diffserv domain. The nodes that implement PHR do not have per-flow responsibilities. PDR is responsible for resource reservation within the complete Diffserv domain. The PDR is adopted by edge nodes (ingress and egress), but not by the interior nodes. This chapter focuses on the last mile between the service provider and the home network, and assumes that the QoS profile of the service provider is the same as the home network. Therefore, the edge-to-edge resource reservation (PDR) in the service provider is the same as that in the home network. That is, both the home gateway and the service provider have the same Service Level Agreements (SLA) and Service Level Specifications (SLS). The PHR protocols extend the PHB in Diffserv by adding resource reservation, thus enabling reservation of resources per Diffserv class PHB per-hop in each node within a Diffserv domain. **Figure 3.3** depicts the peers in the communication in the PDR and PHR between the home network and the service provider.

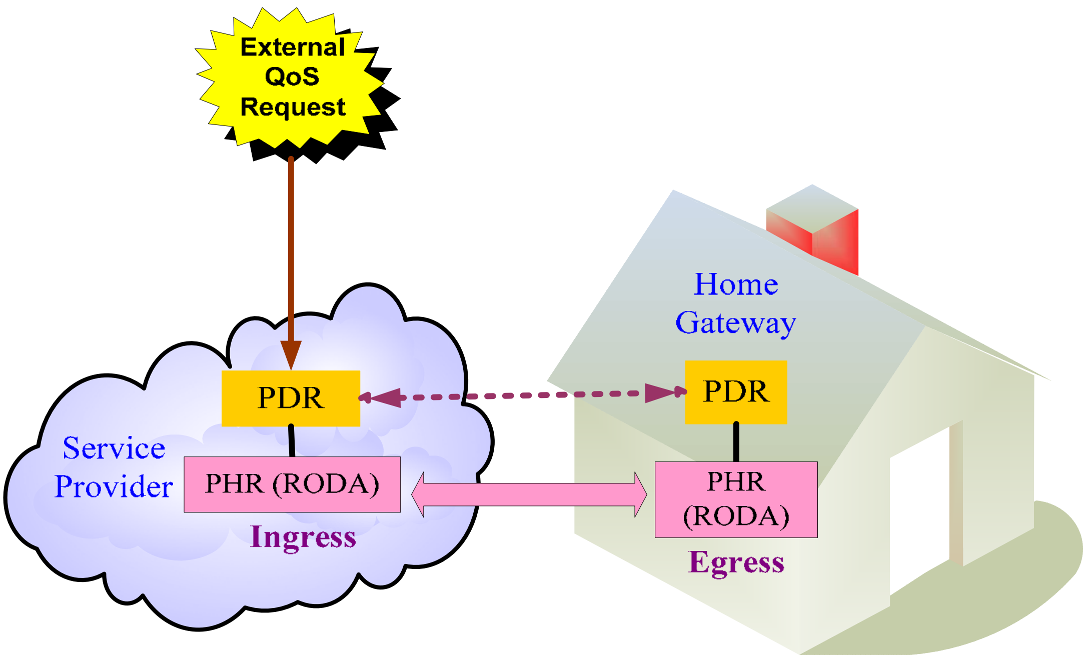


Figure 3.3: Peer Communications in PDR and PHR Protocol

3.2.1 Resource Reservation in RMD Mechanism

The PDR protocol adopts the PHR protocol or any underlying protocol to transport PDR messages. RMD is a sender initiative protocol. Only one PHR protocol, namely RODA, is currently specified. PHR can be adopted to make the service provider (Ingress) and home gateway (Egress) in the Diffserv domain utilize certain message types to request and maintain the reserved resources for the flows going through the Diffserv domain. Each flow occupies a certain number of resource units assigned to a particular Diffserv class. **Figure 3.4** shows the resource reservation scheme.

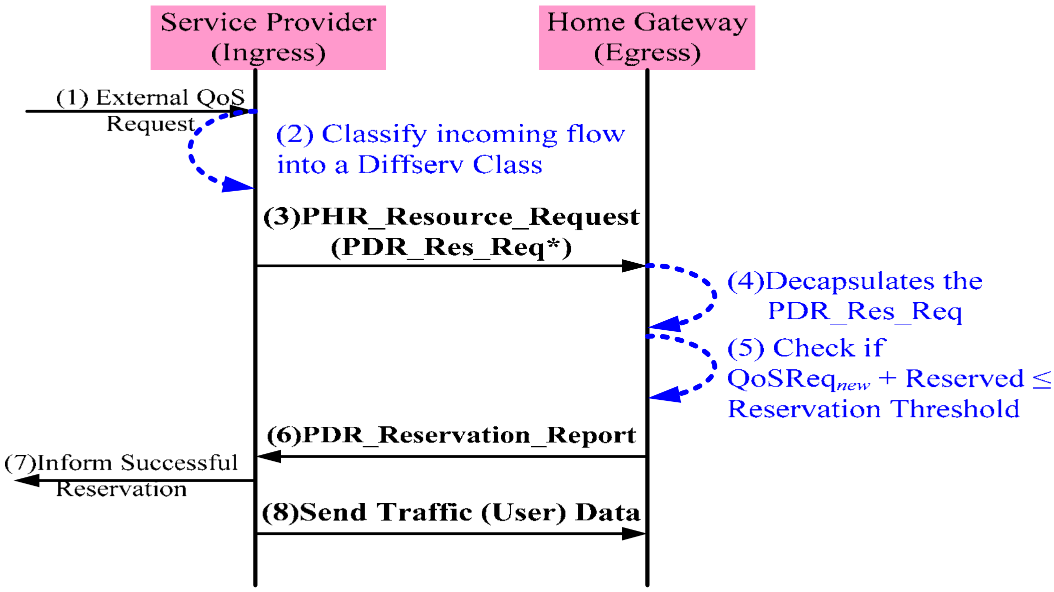


Figure 3.4: Resource Reservation Scheme

The process shown in **Figure 3.5** can be performed when the ingress node (Service Provider) receives an external QoS request to modify the number of reserved resources. Because the reservation-based and soft state RODA PHR are adopted, the aggregated reservation state related to the Diffserv class PHB (DSCP) must be periodically refreshed or updated. This external QoS request in the proposed scenario could be initiated by the in-home user, or by the service provider. If the modified request requires additional reserved resources, then the ingress node subtracts the old and already reserved resources from the resources included in the new modified request. If the modified request requires a decrease in reserved resources, then the ingress node subtracts from the old and already reserved number of resources.

$$
\begin{array}{l}
if\ (\mathbf{QoSReq_{new}} > \mathbf{Reserved_{old}}) \\
\quad \mathrm{PHR\_Resource\_Request} = \mathrm{QoSReq_{new}} - \mathrm{Reserved_{old}} \\
else\ if\ (\mathbf{QoSReq_{new}} < \mathbf{Reserved_{old}}) \\
\quad \mathrm{PHR\_Release\_Request} = \mathrm{Reserved_{old}} - \mathrm{QoSReq_{new}} \\
endif
\end{array}
$$

Figure 3.5: Modified Version of a Reservation State

3.2.2 Severe Congestion Detection and Notification

Severe congestion may occur as a result of link failure or route changes. In RMD, severe congestion is signalled to the edges by the interior nodes, which report the severe congestion occurrence to the edges by PHR signalling messages. However, since the interior node does not maintain any flow-related information flow ID, and the IP address of the ingress node cannot be identified, the interior node cannot notify the ingress node of severe congestion. The PHR in the interior nodes detects severe congestion, and the PDR protocol informs the edge nodes of it.

3.2.3 Adaptive QoS Manager

This chapter classifies the home traffic, and defines the key QoS concern for each home network applications, as shown in **Table 3.1**.

Security Class: This class includes home care and health care data. Traffic in this class is urgent and must not be lost. This class needs a small bandwidth, but also needs a high-priority channel.

OSGi Web Control Class: This class includes OSGi remote control signals for home appliances, and appears at the user request. The class only occupies a small resource. To reduce the round-trip time delay during remote control, this class is assigned a high priority but a small bandwidth reservation.

Table 3.1: Traffic Classes

Traffic Class	Diffserv Setting	QoS Concern	Resource Reservation
Security	EF	Delay and Reliable	3%
OSGi Web Control	AF	Reliable	2%
VoIP	AF	Delay and Jitter	15%
Video Streaming	AF	Packet Loss	80%
Best Effort	—	Throughput	—

VoIP Class: Traffic in this class is adopted in Voice over Internet Protocol (VoIP), and requires a large bandwidth, small delay and jitter. VoIP can endure some dropping during congested period, because congestion does not degrade the integrity of the whole conversation.

Video Streaming Class: Traffic in this class is adopted in applications such as digital TV streaming and teleconferencing. Like the VoIP class, it requires large bandwidth, but can tolerate a small delay and jitter. However, packet loss is an important concern, since it affects the quality of definition.

Best Effort Class: This class is transmitted with best effort only, and is given no resource reservation. Traffic in this class is adopted in applications including FTP and Email. The only concern for FTP traffic is its throughput.

Figure 3.6 shows the interaction of the adaptive QoS Manager between the OSGi home gateway and the service provider. QoS Manager negotiates with the lower layers of both the service provider and the OSGi home gateway, which are basic Diffserv components. The QoS Manager monitors the specified traffic to perform adaptive control according to the QoS priority mapping table in layers 2 and 3. Finally, the RODA PHR is adopted to transport the controlling message to the edge node.

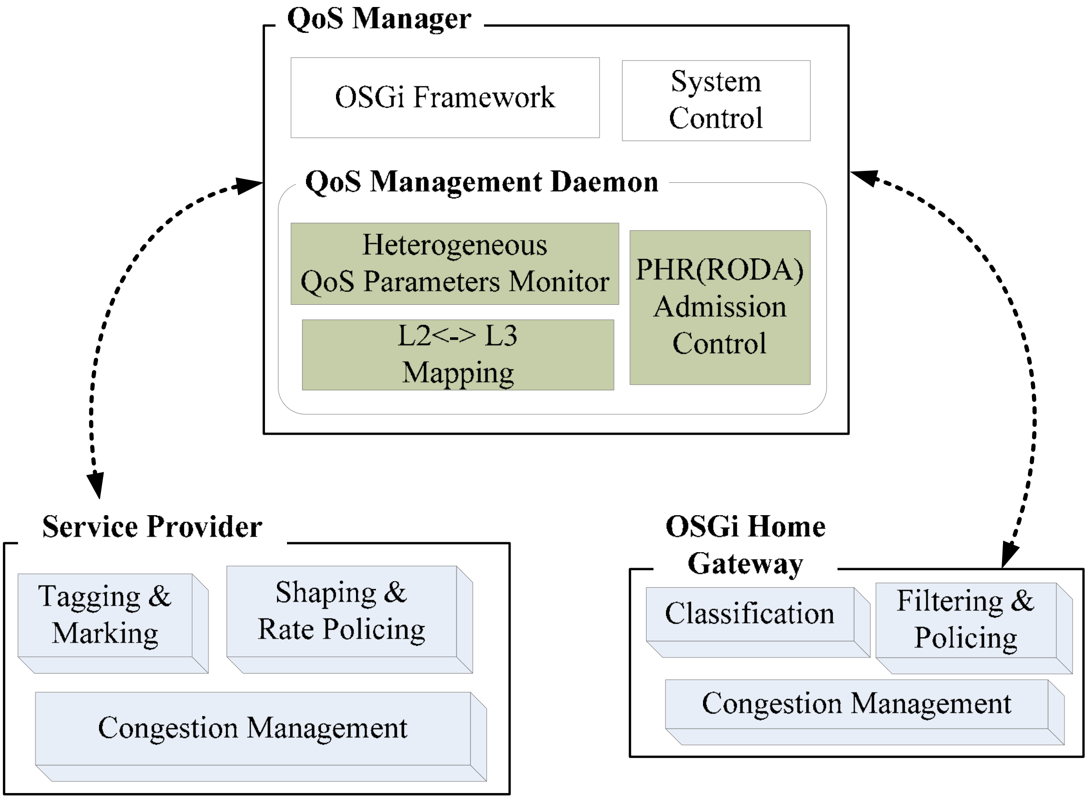


Figure 3.6: Adaptive QoS Manager

Our previous works have indicated that some home network applications are related to the UPnP QoS Architecture **[34].** This chapter does not seek to map priorities between layers 2 and 3, but instead adopts an offline mapping table, shown in **Table 3.2**, which refers to the 802.1D/802.1P (MAC-based) protocol in the UPnP QoS Architecture Version 1.0. The edge node performs on-line classification or filtering according to the mapping table, and the QoS Manager catches the marked packets to process adaptive control.

Table 3.2: QoS Mapping with DSCP and 802.1D/802.1P

Traffic Class in UPnP	802.1D Traffic Type	Traffic Importance (Priority)	DSCP	Application Traffic Class
Network Control	NC	7	EF	Security Traffic
Streaming Control	NC	7	EF	OSGi Web Control
Voice	VO	6	AF	VoIP
Gaming	VO	6	AF	HD Video Stream
AV	VI	5	AF	SD Video Stream
Audio	VI	5	AF	Audio
Image	EE	3	AF	—
Data	BE	0	BE	FTP
Other	BE	0	BE	—

3.2.4 Adaptive QoS Control Mechanism

The proposed adaptive QoS control scheme, which is based on the RMD admission control, performs a reengineered process. The traffic is monitored and adjusted periodically after it is admitted by the PHR in RODA. The dropping probability is measured at every time interval T to check whether the values are below the target level, while maximizing the network utilization. **Table 3.3** shows the notation adopted throughout this chapter.

Table 3.3: Notation and Description

Notation	Description
B	available bandwidth of the bottleneck link
$B_{remainder}$	remainder capacity of the bottleneck link
$B(dscp)_{RESV}$	reserved bandwidth of the specified class
$B(dscp)_{REQ}$	request bandwidth of the specified class
$\hat{B}(dscp)_{REQ}$	new request bandwidth of the specified class
$\varepsilon\,(dscp)_{drop}$	packet dropping threshold of the specified class
$\varepsilon\,(dscp)_{jitter}$	jitter threshold of the specified class

Figure 3.7 shows the adaptive QoS control algorithm. As mentioned earlier, the dropping ratio of each PHB class is monitored at each interval time to ensure that it remains below the threshold limit. Increasing the dropping ratio is likely to raise the jitter of the VoIP. The flow is adjusted when the dropping ratio is above the threshold value. A flow over the threshold value is permitted when the amount of available resources increases. However, if no additional resources are available, then the subclass flow is also checked, and the dropping ratio of the subclass is measured to ensure that it is also below the dropping threshold. A signalling message is then sent to the sender to block the subclass flow that has most recently requested a reservation. If no more adjustment is needed in the time interval, then a signalling message is sent to the sender to unblock the flow in first-in-first-out order. If no subclass flow or resources are available, then a signalling message is sent to the sender to block the flow.

DESCRIPTION

The adaptive QoS manager measures the accumulated dropping ratio of the HD video stream, and calculates the accumulated jitter of VoIP, at every estimate time interval T to verify that the performance of the specified application is below the threshold.

ASSUMPTION

The utilization threshold measuring the congestion condition is the same for each class in the bottleneck link. The available bandwidth of the bottleneck link B is given by $\sum B_{dscp}$, for each class, B_{dscp} = Link_Capacity(dscp) × Threshold(dscp). Traffic can be borrowed from other classes.

For every T seconds do

$\hat{B}(dscp)_{REQ} = (B(dscp)_{REQ} / B) \times B_{remainder}$

if ((DSCP == DSCP_HD && $D(dscp) > \varepsilon(dscp)_{\mathrm{drop}}$) ||

(DSCP == DSCP_VoIP && $J(dscp) > \varepsilon(dscp)_{jitter}$)) *then* {

if ($B_{remainder} \geq \hat{B}(dscp)_{REQ}$) *then* {

Increase resource reservation for the specified class

$B(dscp)_{RESV} += \hat{B}(dscp)_{REQ}$

$B_{remainder} -= \hat{B}(dscp)_{REQ}$

} *else* {

if (**subclass exists** && $D(subdscp) \leq \varepsilon(subdscp)_{\mathrm{drop}}$) *then* {

Signal the sender to block the flow of the subclass

Record the blocked flow ID in the queue Q

} *else* {

Signal the sender to block the flow of the class

Record the blocked flow ID in the queue Q

} *endif*

} *endif*

} *else if* (Q **is not empty**) *then*

Unblock the flow in FIFO order

endif

endif

Figure 3.7: Adaptive QoS Algorithm

3.3 Simulation Results and Performance Analysis

Network Simulator 2 (NS2) was adopted to evaluate the performance of the proposed QoS scheme. In our simulation scenario as shown in **Fig. 3.8**, there are three classes of real-time application in the home network. Internet service providers initiate High Definition (HD) video streams, Voice over IP (VoIP) streams, and Standard Definition (SD) video streams to in-home users.

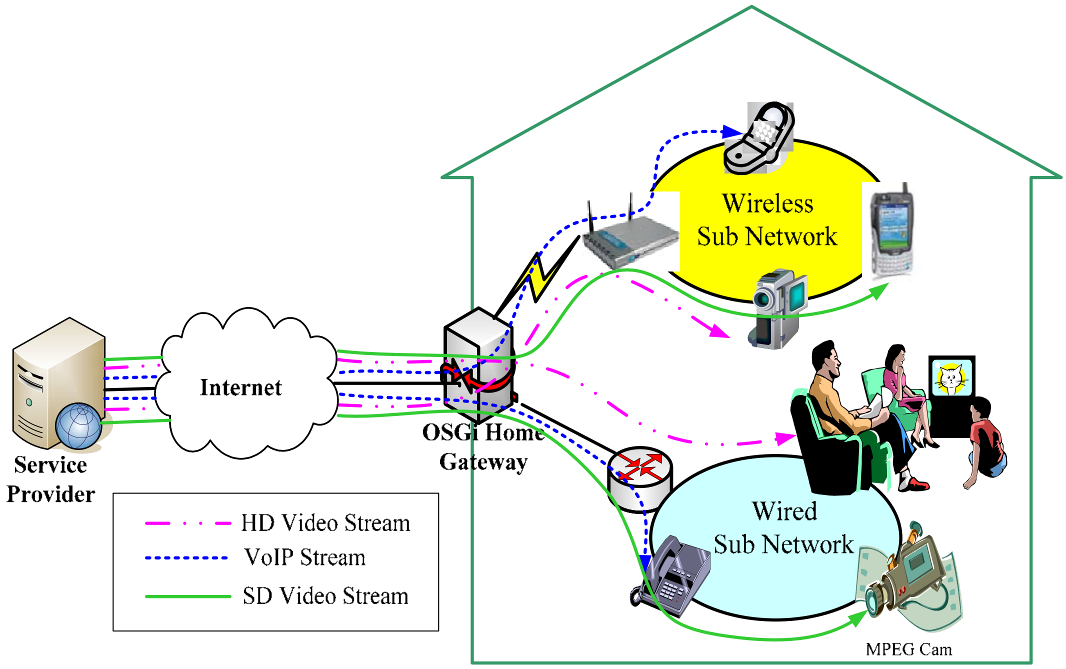


Figure 3.8: Simulation Scenario

3.3.1 Comparison of QoS Strategies

The proposed simulation scenario was divided into two parts, one was running in the wired homogeneous testbed, as shown in **Fig. 3.9**, another was running in the wired/wireless heterogeneous testbed, as shown in **Fig. 3.10.** The performance of Diffserv, RSVP, and RMD schemes in the wired and wired/wireless home networks was compared.

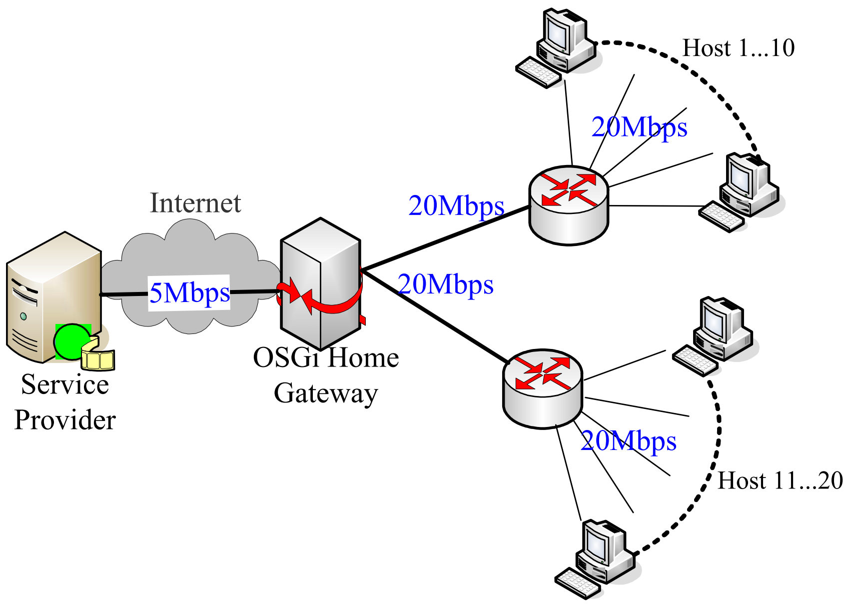


Figure 3.9: Homogeneous Wired Testbed

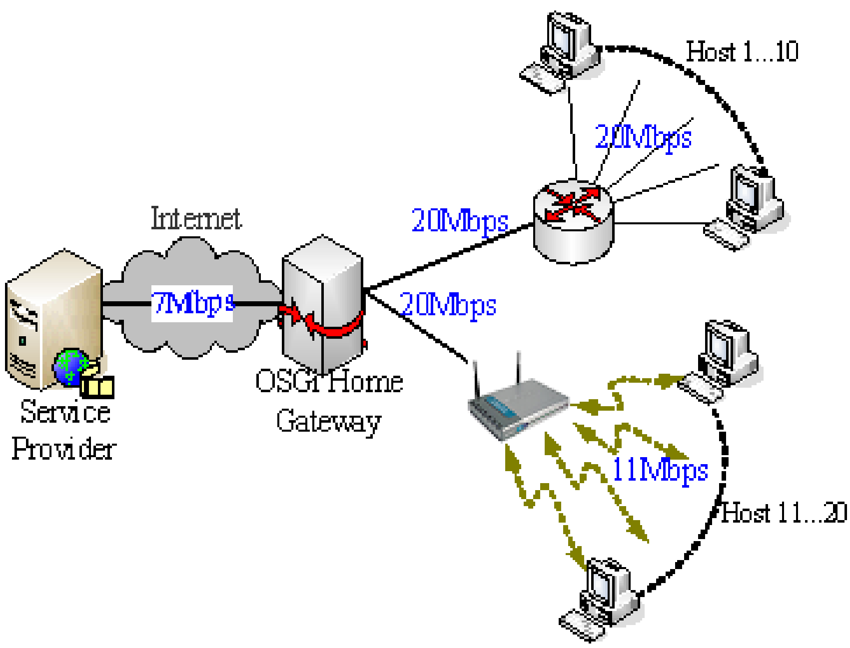


Figure 3.10: Heterogeneous Wired/Wireless Testbed

Table 3.4 shows the traffic characteristics of both the homogeneous and heterogeneous home networks. To prevent additional traffic streams from affecting the existing traffic load, applications were started at different times. All of the applications were kept running until the end of the simulation. The simulation end time of the heterogeneous network was set to 150 seconds, while that of the homogeneous network was set to 100 seconds.

Table 3.4: Traffic Characteristics

Flow Type	Traffic Type	Packet Size (Bytes)	Data Rate (kbps)	Arrival Time (sec)	Stop Time (sec)
VoIP	VBR	200	16	2.0+ exponential	100 (or 150)
HD Video	VBR	1500	256	5.0+ exponential	100 (or 150)
SD Video	VBR	1500	64	10.0+ exponential	100 (or 150)

The comparison results of the RSVP, RMD and Diffserv mechanisms are shown below. Individual flow performance was evaluated at each simulation time interval (5 seconds). **Figure 3.11** indicates Diffserv mechanism had a higher packet loss ratio than RSVP and RMD mechanisms without admission control. RMD caused no packet loss in the wired link network. **Figure 3.12** shows the average end-to-end delays, indicating that Diffserv and RSVP mechanisms had a higher latency than RMD mechanism due to the unlimited waiting resulting from packet loss. RMD had an average delay of only 0.0054 seconds. Although RMD performed admission control, its throughput was almost as high as that of Diffserv.

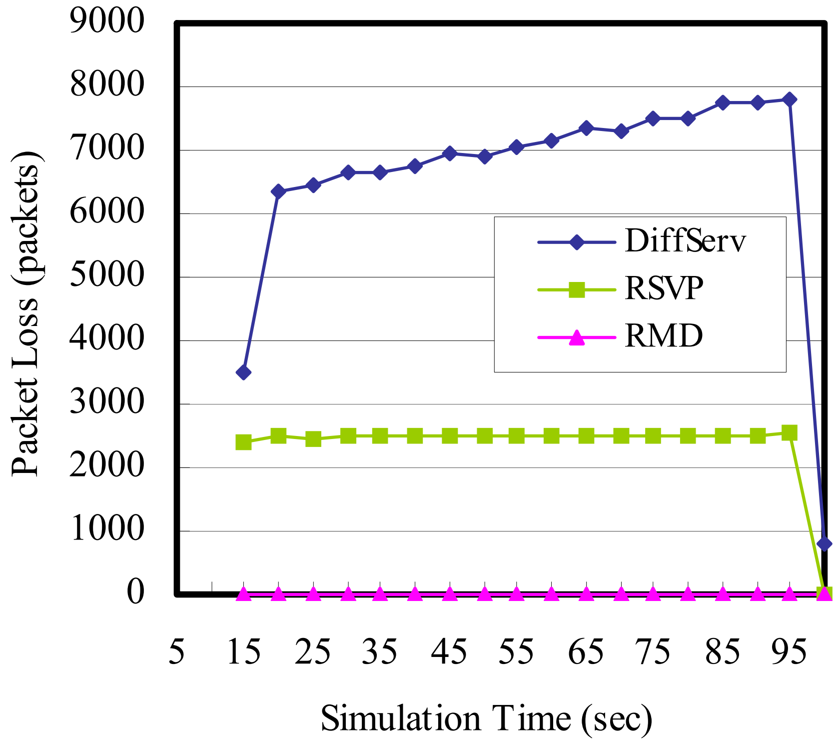


Figure 3.11: Average Packet Loss Ratio

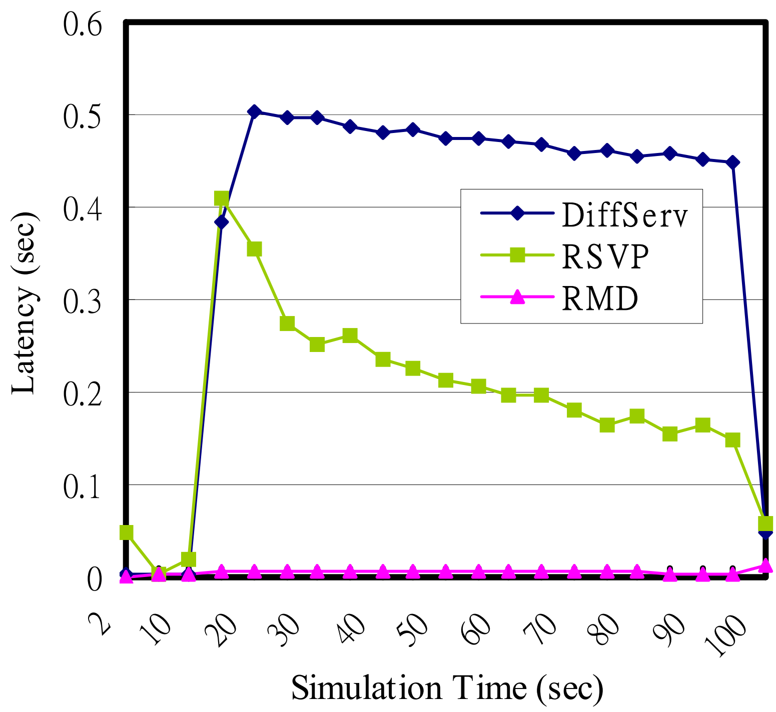


Figure 3.12: Average End-to-End Delay

The exponential arrival time was adopted to prevent large amounts of traffic from congregating at the same time, since that phenomenon would reduce the overall performance, aggravating the CPU loading for admission control processing. **Figure 3.13** indicates that the throughput of RMD mechanism rose stepwise due to the need to perform admission control. The throughput in Diffserv mechanism burst faster than in both RSVP and RMD mechanisms because Diffserv has no prepared admission control time.

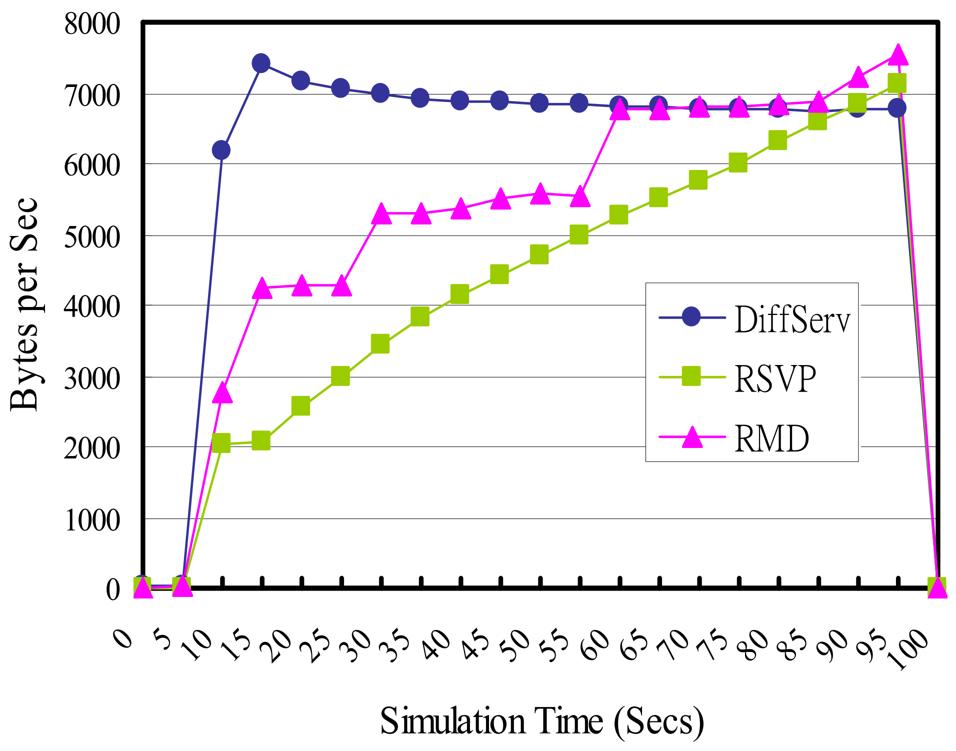


Figure 3.13: Throughput in Bottleneck Link

3.3.2 Performance Analysis

As mentioned above, the proposed adaptive QoS manager is based on the RMD mechanism, and performs a reengineered process. The dropping probability (which may cause jitter) was kept below a target level while maximizing network utilization in an attempt to meet different QoS requirements. That is, VoIP should have the minimum delay and jitter, while HD video stream should have the minimum packet loss. In this experiment, SD had a lower priority than HD, so the loss ratio of SD was bigger than that of HD when the network increased in size.

The simulation results are shown below. **Figure 3.14** indicates that the average latency of the proposed adaptive admission control (ADC) mechanism was almost equal to that of RMD. For individual applications, **Figure 3.15** indicates that the end-to-end latency of VoIP applications in adaptive ADC scheme was less than that in RMD mechanism. Additionally, **Figure 3.16** indicates that the average jitter of the adaptive ADC scheme was slightly less than that of RMD. For VoIP applications, the jitter of adaptive ADC scheme was less than that in RMD mechanism, especially when transmitting consecutive video streams during periods of 50-100 seconds, as shown in **Figure 3.17**. **Figure 3.18** indicates that the average loss ratio in adaptive ADC scheme was below that of RMD mechanism. Finally, the loss ratio of HD video stream in adaptive ADC scheme was less than that in RMD mechanism, as revealed in **Fig. 3.19**.

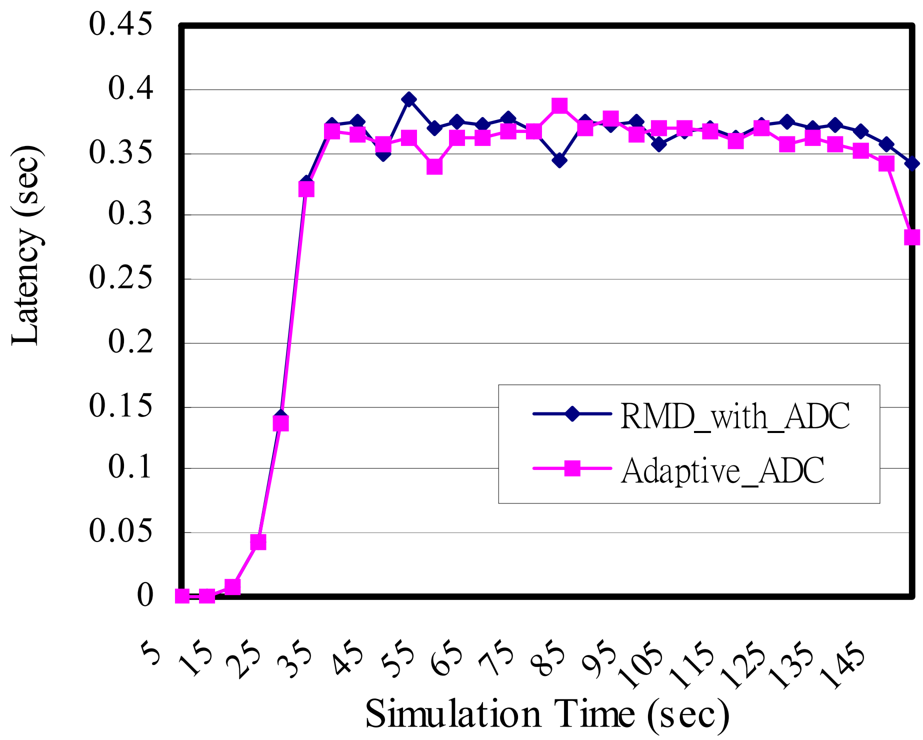


Figure 3.14: Average Latency

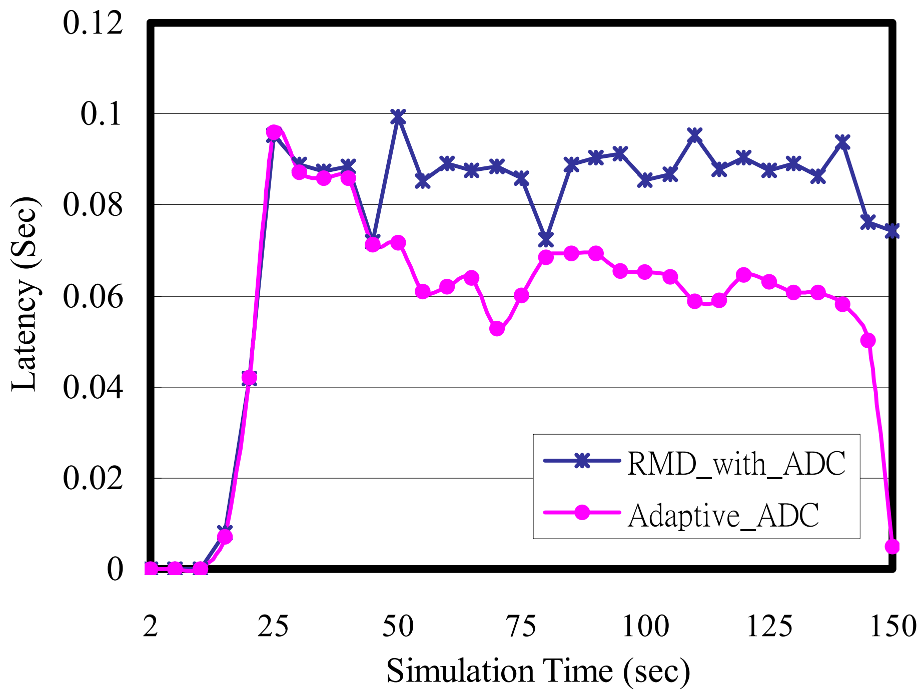


Figure 3.15: End-to-End Delay in VoIP Applications

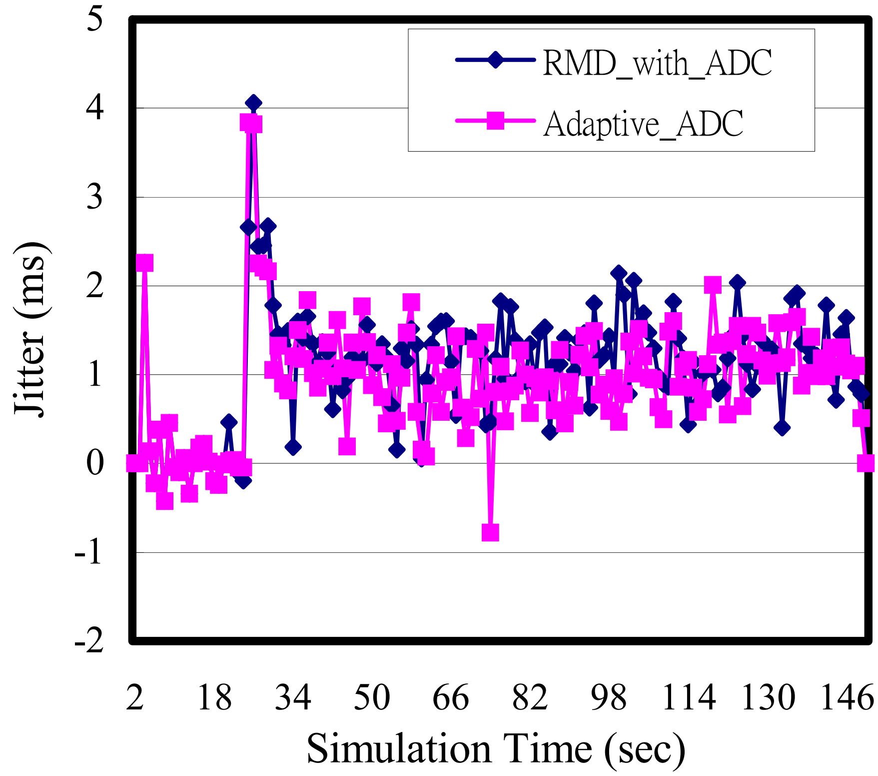


Figure 3.16: Average Jitter

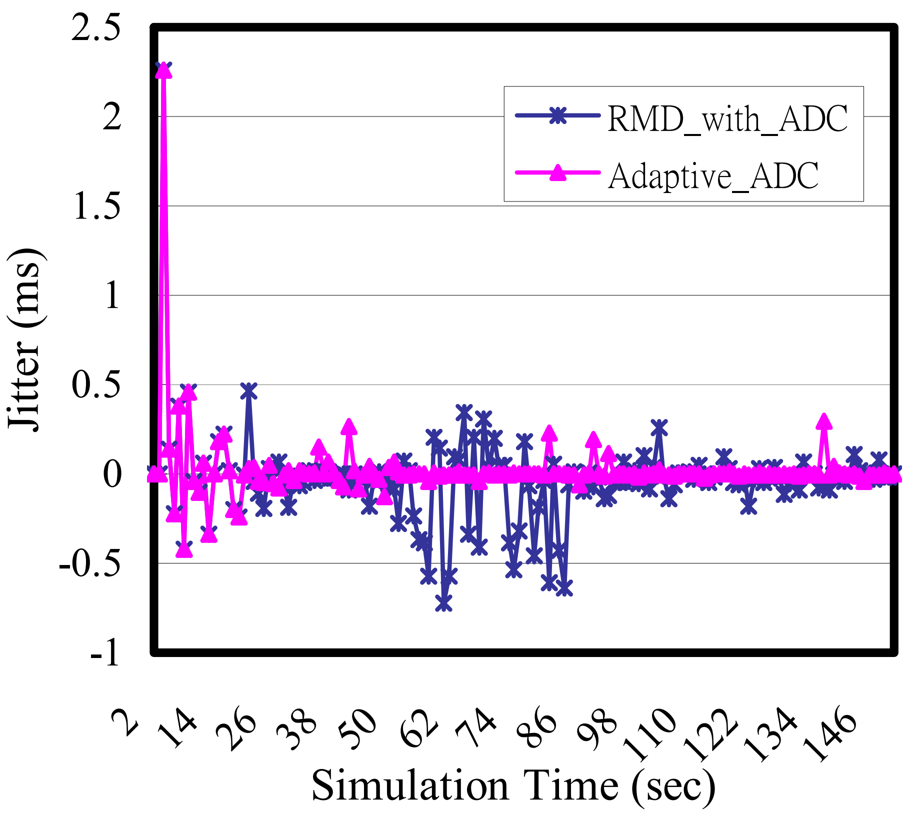


Figure 3.17: VoIP Jitter

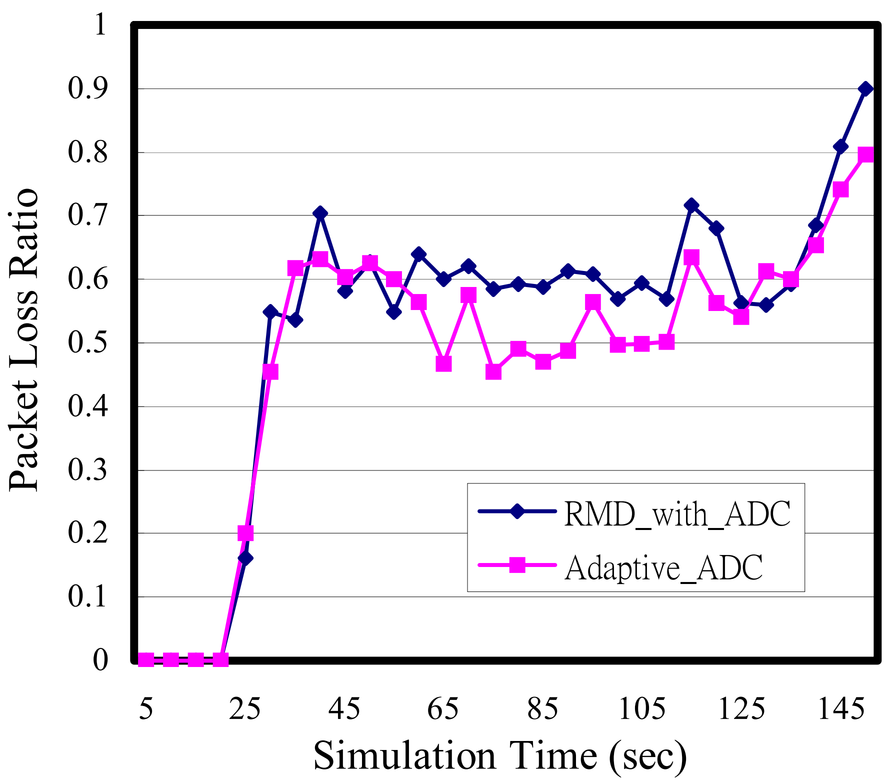


Figure 3.18: Average Loss Ratio

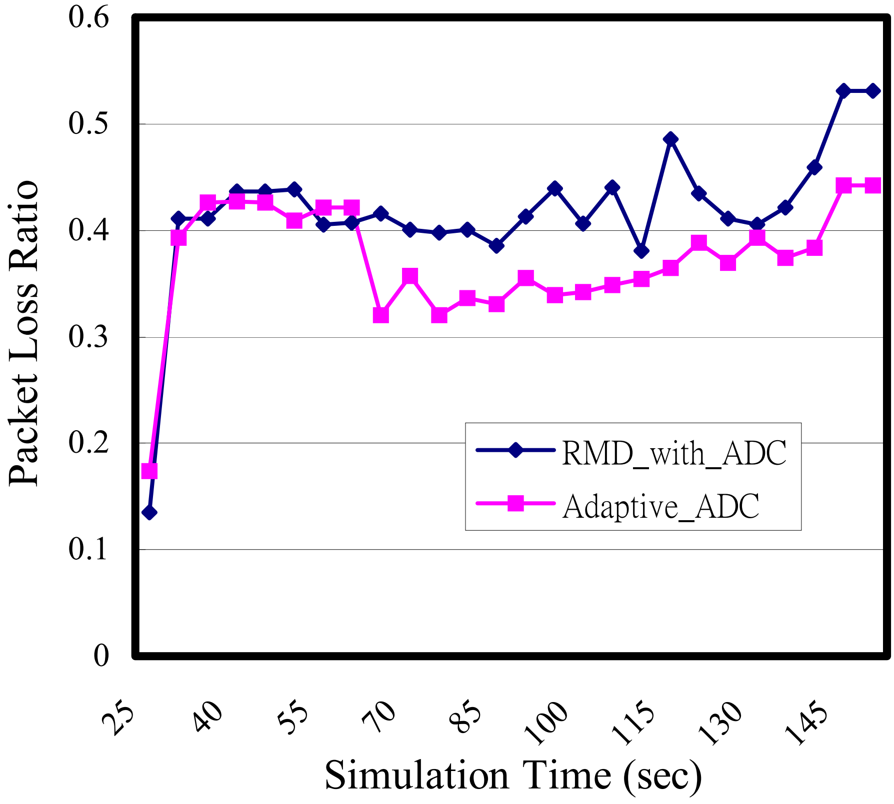


Figure 3.19: Loss Ratio in High Definition Video Applications

The proposed adaptive ADC scheme sacrifices a small amount of the throughput in order to address the QoS requirements of each application in a congested home network. Therefore, the proposed scheme not only minimizes the end-to-end delay of all real-time applications, but also reduces the packet loss ratio in HD video stream. **Figure 3.20** plots the throughput.

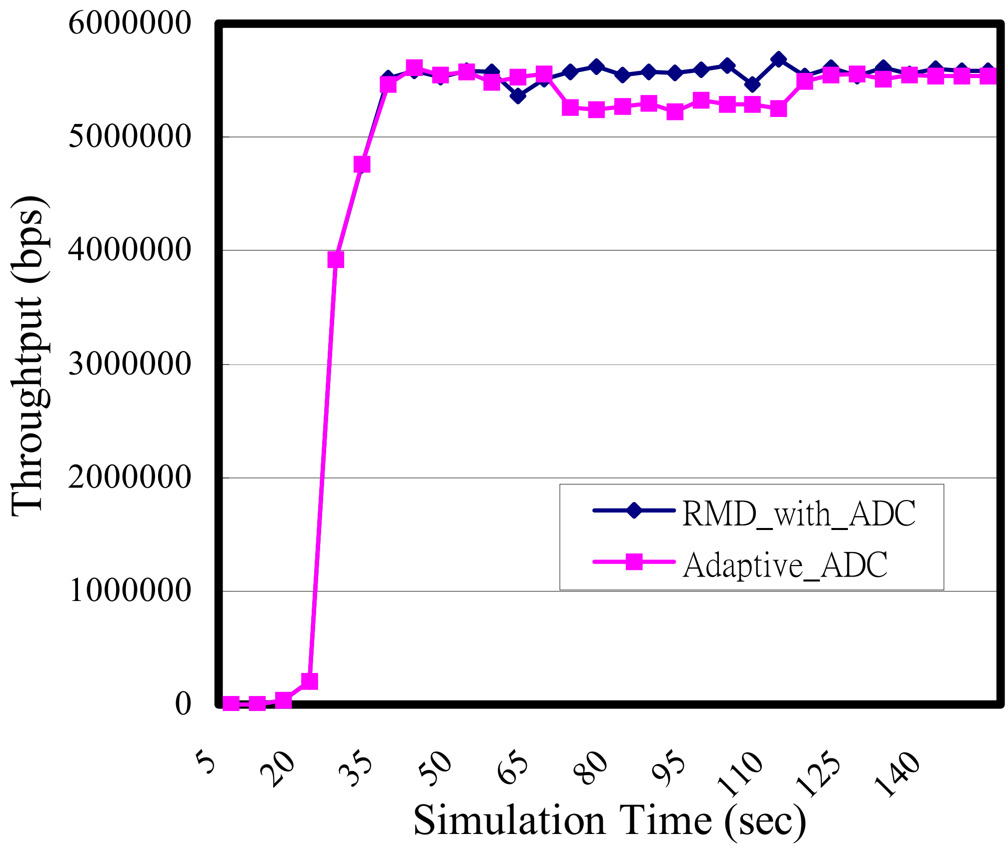


Figure 3.20: Throughput in Bottleneck Link

In summary, the proposed adaptive QoS mechanism achieved the best end-to-end latency, packet loss ratio and the jitter ratio among the schemes tested, as indicated in **Table 3.5**. **Table 3.6** shows that the proposed adaptive QoS mechanism satisfies the specified concerned QoS requirements of each application more effectively than other mechanisms.

Table 3.5: Average Performance Improvement

QoS Parameter	Decrease/Increase	Improvement (Average)
Jitter	Decrease	0.1391 (ms)
Latency	Decrease	6.6 (sec)
Loss Ratio	Decrease	5.43%

Table 3.6: Performance Improvement in each Application

Flow Type	Decrease / Increase	QoS Parameter	Improvement (Average)
HD	Decrease	Loss Ratio	4.53%
HD	Increase	Throughput	1.2%
SD	Decrease	Loss Ratio	1.89%
VoIP	Decrease	Latency	20.9 (ms)
VoIP	Decrease	Jitter	0.0407 (ms)

3.4 Summary

The proposed adaptive QoS manager combines the wired and wireless QoS concerns into admission control. Different home applications have different key QoS requirements in congested home networks. For instance, VoIP applications are concerned about the jitter, but can tolerate some packet loss, which does not significantly harm the integrity of the conversation. High-definition video stream is mainly concerned with packet loss, but can tolerate a small time delay and jitter. The QoS attributes are integrated to perform adaptive admission control in wired-wireless heterogeneous home networks. Simulation results indicate that although some throughput is sacrificed in low-priority application like Standard Definition (SD) video stream, each application is guaranteed to have the minimum but the most important QoS requirement. Simulation results also indicate that the proposed mechanism reduces the average jitter, latency and packet loss by 0.1391 ms, 6.6 ms and 5.43%, respectively. The proposed mechanism reduced the packet loss ratio by 4.53%, and increased the throughput by 1.2%, in high definition video stream. Additionally, it decreased the jitter and latency to 0.0407 ms and 20.9 ms, respectively, in VoIP applications.

Chapter 4
Adaptive Radio Resource Management in GPRS/UMTS Service Networks

This chapter presents a novel resource management based on timeslot utilization in an integrated GPRS/UMTS service network. This approach gives UMTS high priority for sending data services, it also makes voice services available over GSM. UMTS is the interference limited system. More subscribers will increase the probability of error bits. To keep a clear spectrum, CAC (Call Admission Control) will not allow all subscribers to access the cell. Some subscribers will be blocked to reduce the interference. To consider of the continuity of data services, we can handoff some subscribers with dual stacks from UMTS to GSM/GPRS if the traffic in UMTS is in heavy loading. In GSM/GPRS, one timeslot can be used by GSM or GPRS. GSM and GPRS cannot use the same timeslot simultaneously. To keep the more available timeslots, we can handoff some voice traffic from GSM to UMTS in order to allocate for GPRS. It can increase the capacity of the integrated system and does not drop voice services. This chapter presents a radio resource scheduling between GSM/GPRS and UMTS, combining timeslot-limited and interference-limited systems. The proposed approach significantly increases capacity, and contains the existing UMTS call admission control mechanism.

4.1 Introduction

Third-generation (3G) networking technology has taken many years to be deployed worldwide. The ability to operate seamlessly between existing 2/2.5G GSM/GPRS (Global System for Mobile Communications/General Packet Radio Service) and developing 3G UMTS (Universal Mobile Telecommunications System) networks is critical for its widespread adoption. To reduce construction cost, most operators generally constructed new 3G base stations in the same buildings as previous generation base stations. Under the scenario of the same base station with 3G system and GSM/GPRS system, GSM/GPRS can compensate the coverage hold caused by the limitation of 3G transmission power. An integrated GPRS and UMTS system should effectively adopt of UMTS radio resources for high data rate applications, and ensure continued service when UMTS coverage is unavailable. Hence, a major challenge is managing different radio resources among co-sited base stations.

In GSM systems, each cell supports dedicated timeslots for voice connections and data connections based on circuit switching. Each cell has several channels, which are generally frequency and timeslot pairs. Each voice connection adopts exactly one channel. New voice connections (or handoff connections) are admitted if the cell has any free channels, and no further calls are admitted when all channels are occupied. The radio resource management (RRM) algorithm of GSM is simple and straightforward. To enhance data connection services, standard GPRS over GSM supports multi-timeslot data access, in which several data connections share particular timeslots **[35-36]**. Namely, GSM and GPRS offer particular timeslots for voice and data connections, and RRM algorithms for GSM/GPRS are based on the timeslot system. The new generation UMTS consumes fewer air interface resources for the same data rate than GSM/GPRS. In UMTS, power control techniques determine the power required to sustain a given data rate. In either of these two cases, stations with poor channel conditions need require more air interface resources to sustain a given data rate than stations with good channel conditions. Qualities of air interface resource depend on radio interference and transmission power. The RRM algorithms in UMTS are limited by interference and transmission power.

Simplistic RRM algorithms **[37-41]**, including those that merely consider the number of subscribers attached in the system, and block new calls as a threshold is reached, cannot use the flexibility provided by the integrated and co-sited system. The integrated RRM is designed to optimize the resource usage both within and across systems. Such an RRM is obtained by assigning the radio resource to each service provided by the network, considering both system loading and resource availability, and allowing a subscriber to be served using the most efficient system, retaining the connections-requested quality. To select the best network to access, Filippo **[42]** investigated cost-based traffic control mechanisms in a scenario comprising a GPRS and a UMTS network providing cellular coverage in the same area. Filippo's algorithm provides the best radio access network selection to increase the cell capacity and decrease the cost. However, GPRS and GSM share the same radio resources. The distribution of GSM traffic directly influences the amount of GPRS resource. Filippo's cost-based radio resource management only considers GPRS and UMTS. In summary, GSM/GPRS form a heterogeneous network sharing the same radio resource pool. Radio resource management of an integrated network must consider heterogeneous network resources, and take advantage of the benefits of the heterogeneous network resource.

This chapter emphasizes on continuous coverage of all services, especially in heavy loading in UMTS. The effect of inter-cell interference is not considered. This study presents a novel radio resource scheduling between GSM/GPRS and UMTS, combining timeslot-limited and interference-limited systems. The proposed approach significantly increases capacity, and contains the existing UMTS call admission control mechanism.

4.2 GSM/GPRS and UMTS Service Handoff

GSM/GPRS is the currently developed system, and UMTS is the system under development. Hence, integration between UMTS and GSM/GPRS systems probably involves three scenarios **[43-44]**. First, the subscriber is covered by UMTS and GSM/GPRS, and the subscriber is approved into UMTS to improve service quality. UMTS can be adopted to multimedia data services, as shown in **Fig. 4.1(a)**. Depending on the number of subscribers, the network provider can arrange for subscribers in heavy serving cells to hand off to other cells. Subscribers in a standalone system can access other light intra-system cells based on network decisions if the serving cell is overloaded. The network provider has many traffic load balancing mechanisms in an integrated system, as shown in **Fig. 4.1(b)**. GSM/GPRS provides wider coverage than UMTS when initially deploying UMTS. Data communication may be dropped if a subscriber moves along the weak UMTS area. To prevent the existing service from dropping, the subscriber is handed off from UMTS to GSM/GPRS if the UMTS coverage is insufficient, as shown in **Fig. 4.1(c)**.

Adopting UMTS instead of GPRS can increase the data rate for high-speed data applications in integrated cellular networks, but may produce the “breathing effect”, shrinking the UMTS convergence radius and leading to serious connection dropping, as shown in **Fig. 4.1(d)**. One solution to this problem is to install new Base Stations, and specify the antenna configuration carefully according to quality constraints. The other solution applies the mechanism of Call Admission Control (CAC) to ensure UMTS radio convergence. Restated, the traffic property can be set in terms of the maximum and minimum required capacities.

The proposed resource management is based on timeslot usage. UMTS has a higher potential resource for data connections than GPRS. Some data connections must be handed off to GPRS to guarantee a clear radio environment if UMTS is overloaded. Voice connections in GSM are forced to handoff to UMTS in order to vacate more timeslots for GPRS. Although GPRS cannot support high-speed data connections, it can support low-speed data connections. Therefore, GPRS can share the traffic loading from UMTS, and reduce UMTS interference.

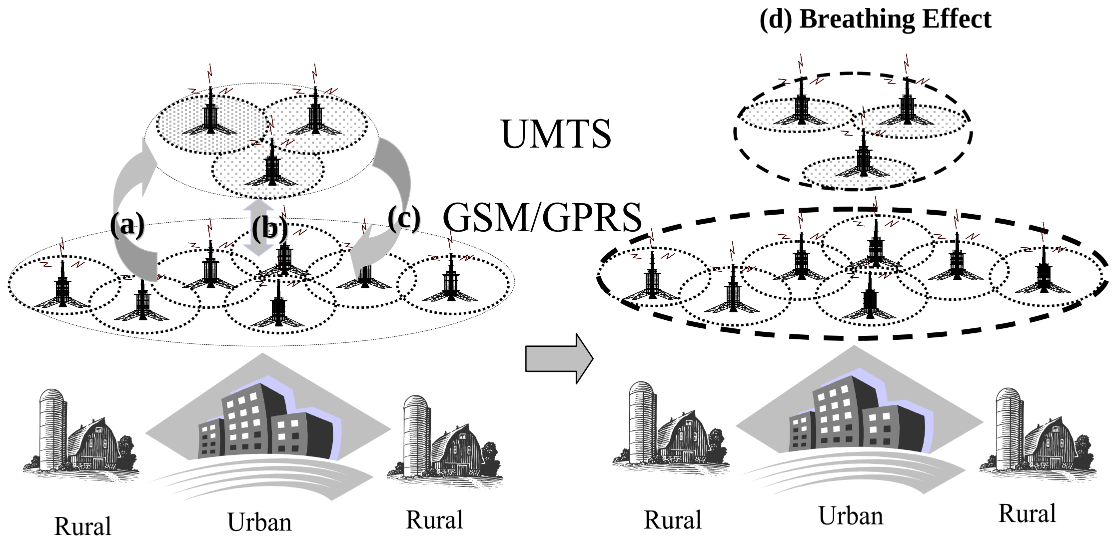


Figure 4.1: (a)Handoff from GSM/GPRS to UMTS to Obtain Increased Bandwidth; (b)Handoff between GSM/GPRS and UMTS to Balance the Traffic Loading; (c)Handoff from UMTS to GSM/GPRS to Ensure Service Continuity, and (d)Breathing Effect

4.3 Adaptive Radio Resource Scheduling

The UMTS capacity depends on the data throughput, and the GPRS capacity depends on the number of timeslots. The AMR (Adaptive Multi-Rate) transcoding technology was implemented to reduce the bandwidth usage of voice services in UMTS **[43-47]**. The voice service was only 12.2 kbps. However, in the GSM, the voice service was given a single timeslot that can transmit data at 21.4 kbps.

To ensure a clear spectrum, the CAC (Call Admission Control) mechanism is allowed to restrict access air resources for all services. In an integrated system involving GSM/GPRS and UMTS, some mobile services can be handed off from UMTS to GSM/GPRS under heavy traffic loading conditions. In GSM/GPRS system, GSM and GPRS can share one timeslot, but cannot use it simultaneously. To maintain the number of available timeslots, some voice traffic can be handed off from GSM to UMTS to increase the available bandwidth for GPRS services. The capacity of the integrated system can be increased without reducing the level of voice traffic. **Figure 4.2** shows this approach during handoff from UMTS to GPRS. The mobile data service starts in UMTS. Scheduling forces the existing GSM voice services to handoff to UMTS until UMTS becomes heavily loaded. The mobile data service is then transferred to GPRS until the service is blocked.

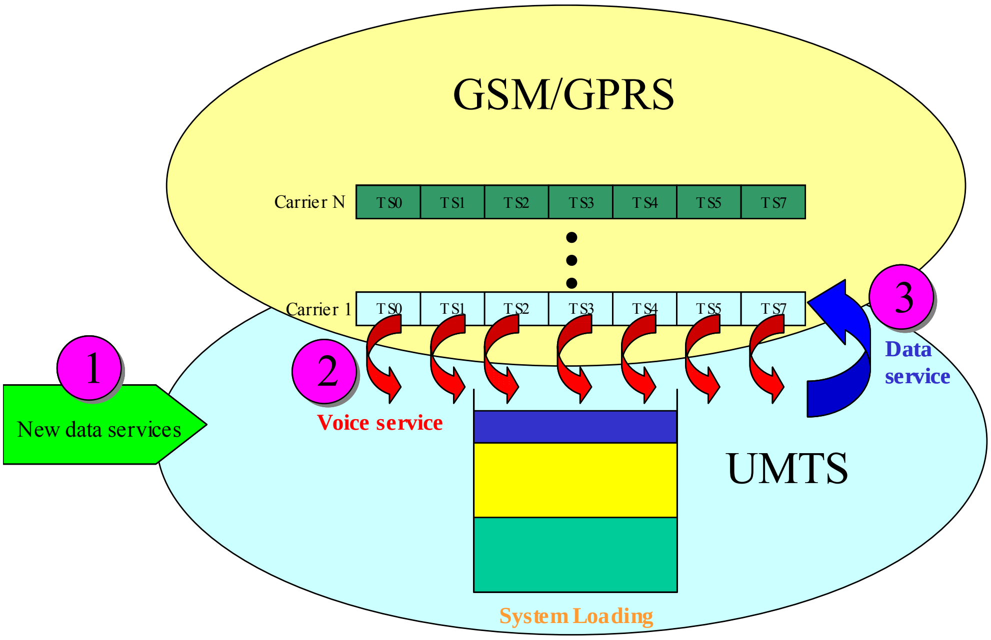


Figure 4.2: Service Handoff Operation

This chapter examines a cost-based traffic control approach in a scenario comprising GSM/GPRS and UMTS networks, both providing cellular coverage in the same region. The approach assigns cellular services to the RAN (Radio Access Network), which minimizes the resource consumption and reduces the capacity. In the GSM/GPRS network, the shared resource S is represented by physical channels, namely the pair (Fc,Ts), where Fc denotes the number of frequency carriers, and Ts denotes the number of timeslots. The number of timeslots is the resource consumption of each service supported by the GSM/GPRS network. The cost of GSM/GPRS is given by

$$\eta_{GPRS} = \frac{\sum_{j=1}^{N} S_j}{S_{\max}}$$

where S_j denotes the estimated timeslot consumption of the *j*th service, and S_{max} represents the total number of timeslots in (F_c,T_s).

The shared resources in the UMTS network represent the downlink power at the base station. Therefore, the resource consumption is represented by the power required to provide each service adequately, and depends on the relationship among E_b, N_o and the actual traffic loading, while the available resources are represented by the amount of power not yet used. The performance indicator E_b/N_o is always related to a quality target. Notably, unlike in the GSM/GPRS network, resource consumption is not known a priori, and hence must be estimated by the network for cost calculation. In the downlink, E_b/N_o is calculated by the model **[48-49]**:

$$\frac{E_b}{N_o} = \frac{W}{R} \cdot \frac{P_{rx}}{I_{own}(1-\alpha) + I_{oth} + P_N}$$

where I_{own} denotes the total power received from the service cell; I_{oth} denotes the total power received from the surrounding cell; P_N denotes the noise power; W denotes the chip rate (3.84Mcps), and R denotes the data rate. The factor α is called the orthogonality factor, and depends on the multipath conditions. The codes are fully orthogonal, and are therefore cancelled with α = 1 when the serving cell has no multipath interference. This chapter found that R and P_{rx} need to be proportional to maintain an appropriate E_b/N_o value. The cost of UMTS is given by

$$\eta_{umts} = \frac{\sum_{j=1}^{N} P_{rx_j}}{P_{\max}}$$

where P_{max} denotes the maximum of the received power in the worst case.

For user j, $\frac{E_b}{N_o} = \frac{W}{R_j} \cdot \frac{P_j^{rx}}{I_j^{own}(1-\alpha) + I_{oth} + P_j^N}$ (1)

Hence, $P_j^{rx} = \frac{E_b}{N_o} \cdot \frac{R_j}{W} \cdot \left(I_j^{own}(1-\alpha) + I_j^{oth} + P_j^N\right)$ (2)

The total receiver power of n users is $\sum_{j=1}^{n} P_j^{rx} = P_1^{rx} + P_2^{rx} + . \;. \;. + P_n^{rx}$ (3)

Then (1) substitutes (3), $\sum_{j=1}^{n} P_j^{rx} = \frac{E_b}{N_o} \cdot \frac{1}{W} \cdot \sum_{j=1}^{n} R_j \cdot \left(I_j^{own}(1-\alpha) + I_j^{oth} + P_j^N\right)$ (4)

In order to define the interference limitation of the UMTS system, we suppose that the channel condition is the worst case and a constant σ to present the maximum interference $\sigma \geq \left(I_j^{own}(1-\alpha) + I_j^{oth} + P_j^N\right), \quad j \in \{1,2,3...,n\}$. (5)

If all of channel conditions are the poorest, the maximum total received power $P_{\max}$ in the worst case is $P_{\max} = \max\left\{\sum_{j=1}^{n} P_j^{rx}\right\} = \frac{E_b}{N_o} \cdot \frac{1}{W} \cdot \sigma \cdot \sum_{j=1}^{n} R_j$ (6)

Then, we define that the UMTS cost η_{umts} is

$$\frac{\sum_{j=1}^{n} P_j^{rx}}{P_{\max}} = \frac{\frac{E_b}{N_o} \cdot \frac{1}{W} \cdot \sum_{j=1}^{n} R_j \cdot \left(I_j^{own}(1-\alpha) + I_j^{oth} + P_j^N\right)}{\frac{E_b}{N_o} \cdot \frac{1}{W} \cdot \sigma \cdot \sum_{j=1}^{n} R_j}. \quad (7)$$

The data rate maximum is $R_{\max} = \sigma \cdot \sum_{j=1}^{n} R_j$. (8)

To summarize equations (5) and (6), we can get $\sum_{j=1}^{n} R_j \cdot \left(I_j^{own}(1-\alpha) + I_j^{oth} + P_j^N\right) \leq \sigma \cdot \sum_{j=1}^{n} R_j$ and

$$\eta_{umts} = \frac{\sum_{j=1}^{n} P_j^{rx}}{P_{\max}} = \frac{\sum_{j=1}^{n} R_j \cdot \left(I_j^{own}(1-\alpha) + I_j^{oth} + P_j^N\right)}{\sigma \cdot \sum_{j=1}^{n} R_j} \leq 1 .$$

The proposed management assigns radio resource according to the loading cost between the UMTS and GSM/GPRS systems, and thus increases the capacity by reducing the interference. **Figure 4.3** shows the proposed radio resource management. The integrated system allocates the radio resources by comparing η_{GPRS} with η_{UMTS}. To minimize the cost, the GSM/GPRS system allocates resource to satisfy new and handoff connections while η_{UMTS} exceeds η_{GPRS}. Otherwise, the UMTS system allocates resource to satisfy new and handoff connections. The proposed management allocates the radio resources of UMTS for services, except that the UMTS system has a heavy loading ($\eta_{umts} \geq 90\%$). Under heavy loading, new and handoff connections can be assigned GSM/GPRS resources if GSM/GPRS system has free resources. If the GSM/GPRS resources is insufficient for new and handoff data connections, the GSM/GPRS system forces existing voice connections to move to the UMTS system, and vacates GSM/GPRS resources for new and handoff data connections. New and handoff data connections then obtain the GSM/GPRS resources. New and handoff connections are blocked if GSM/GPRS and UMTS cannot satisfy the requirements of new and handoff connections.

Figure 4.4 summarizes the radio resource allocation algorithm with the pseudo code. To estimate the interference cost, η_{UMTS_REQ} is defined as the required resource of UMTS network,

$$\eta_{UMTS\_REQ} = \frac{\sum_{j=1}^{N} P_{rx_j} + \sigma \cdot R_{NEW\_CONNECTION}}{P_{\max} + \sigma \cdot R_{NEW\_CONNECTION}}$$

where $R_{NEW\_CONNECTION}$ denotes the UMTS connections requirements of new and handoff connections, and σ denotes the maximum tolerable interference. Moreover, $\eta_{GPRS\_GSM\_REQ}$ denotes the required resource of GSM/GPRS network, and $\eta_{GPRS\_GSM\_REQ}$ depends on timeslot requirements of new and handoff connections,

$$\eta_{GPRS\_GSM\_REQ} = \frac{\sum_{j=1}^{N} S_j + S_{NEW\_CONNECTION}}{S_{\max}}$$

where $S_{NEW\_CONNECTION}$ denotes the number of timeslots for requirements of new and handoff connections; $S_{NEW\_CONNECTION} = 1$ when it receives a voice connections.

$\eta_{UMTS\_THRESHOLD}$ is the threshold of UMTS heavily loading, is 90% here. $\eta_{GPRS\_GSM\_OCCUPIED}$ is the resource currently occupied by a GSM voice call.

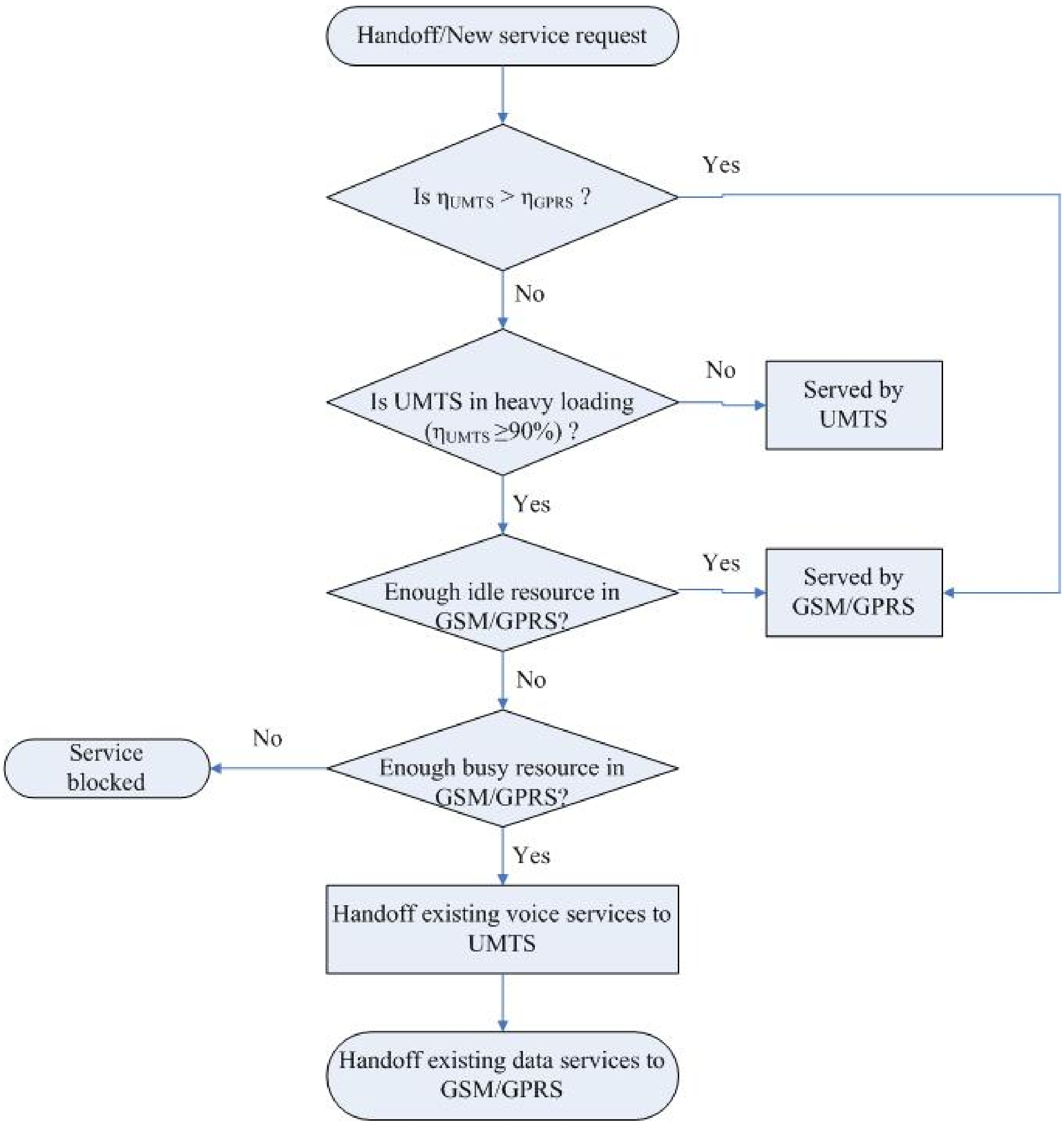


Figure 4.3: Proposed Radio Resource Management

01 *Initialize* $\eta_{UMTS} = 0$ *and* $\eta_{GPRS} = 0$

02 *Wait for connection request arrival*

03 *IF new voice connection request arrives*

04   *IF* $(\eta_{UMTS} + \eta_{UMTS\_REQ}) > (\eta_{GPRS} + \eta_{GPRS\_GSM\_REQ})$

05   *IF* $(\eta_{GPRS} + \eta_{GPRS\_GSM\_REQ}) \le 1$

06   *Admit new voice connection with GSM resource*

07   *ELSE*

08   Re*ject voice conncetion request*

09   *ELSE*

10   *Admit new voice conneciton with UMTS resource*

11 *IF new data connection request arrives*

12   *IF* $\eta_{UMTS} < \eta_{UMTS\_THRESHOLD}$

13   *Admit new data connection with UMTS resource*

14   *ELSE*

15   *IF* $(\eta_{GPRS} + \eta_{GPRS\_GSM\_REQ}) \le 1$

16   *Admit new data connection with GPRS resource*

17   *ELSE*

18   *IF* $(\eta_{GPRS} - \eta_{GPRS\_GSM\_OCCUPIED} + \eta_{GRPS\_GSM\_REQ}) \le 1$

19   *Handoff a voice connection to UMTS network*

20   *GOTO* 15

21   *ELSE*

22   Re*ject new data connection request*

Figure 4.4: Resource Allocation Algorithm

4.4 Performance Analysis

The proposed scheme was examined by the UMTS and GSM/GPRS models in an NS2 simulator with the traffic source set at a constant bit rate. Two different connections, speech and high-speed data, were available to subscribers. The speech connection was provided with GSM, while the high-speed data connection was offered with UMTS. A subscriber was allowed to transmit data regardless of whether it specified UMTS or GPRS.

Moreover, each subscriber could obtain up to eight data transmission timeslots. The C/I was assumed to be guaranteed at the cell border for all GPRS connections, and satisfied once a subscriber was admitted. Each GPRS cell contained six 200 kHz carriers. The simplification comes from the reuse of frequency in the specified layout. Slow fading was modeled, and the path-loss was calculated using the Okumura-Hata formula. CS-4 code technology was implemented in GPRS. The UMTS used one frequency block of 5 MHz, and the shared resource was the downlink Base Station power. The system was assumed to be interference-limited but not code-limited. An ideal power control loop was implemented. **Table 4.1** lists the parameters of the connections used in the simulation. The experimental system comprised two systems. The base stations of both systems were co-sited. A set of uniformly distributed subscribers was created at each iteration of the simulation. The simulations were static, meaning that each iteration corresponded to a traffic situation uncorrelated with that of the previous iteration. Each subscriber was assigned one data session at a time after the loading was updated.

The simulated environment included two data rates, 64 kbps and 128 kbps. A set of simulations was performed to evaluate the capacity gain of the investigated traffic balance schemes. The simulations of UMTS, GSM and GPRS were performed to estimate the throughput, packet loss ratio and delay time. The traffic source was set at a constant bit rate.

Table 4.1: Simulation Parameters

Parameters	Value
Activity Factor of Subscribers	Speech: 0.8
	Data: 1
E_b/N_o Signal Energy	Speech: 12.2db
	RT 64k: 5.5db
	RT 128k: 6.5db
UMTS Chip Rate	3.84 Mcps
Data Rate	64 kbps
	128 kbps
α Orthogonality Factor (1 fully orthogonality; 0 no orthogonality)	1
Maximum interference σ	Speech: -19.4db
	RT 64k: -17.8db
	RT 128k: -11.6db
The number of GSM/GPRS Carriers F_c	6
The number of GSM/GPRS timeslots T_s	8
The number of total timeslots S_{max}	48
Requirements of timeslots for data connections	RT 64k: 4 timeslots
	RT 128k: 8 timeslots

4.4.1 Average allocated bandwidth

Figure 4.5 indicates that the UMTS capacity at an average allocated bandwidth of 64kbit/s was 26 subscribers, while the GSM/GPRS capacity was 12 subscribers. The capacity of the proposed scheme is 40 subscribers. Experimental results show that the proposed scheme with traffic balancing clearly outperforms other scheme for data rates up to 120 kbps. At 128 kbps, six subscribers achieved up to 120 kbps in standalone GPRS, while 14 subscribers achieved the same speed in standalone UMTS. The sum of both standalone systems was 20 subscribers. Notably, 24 subscribers achieved up to 120 kbps in the integrated system, increasing the capacity to serve four additional subscribers.

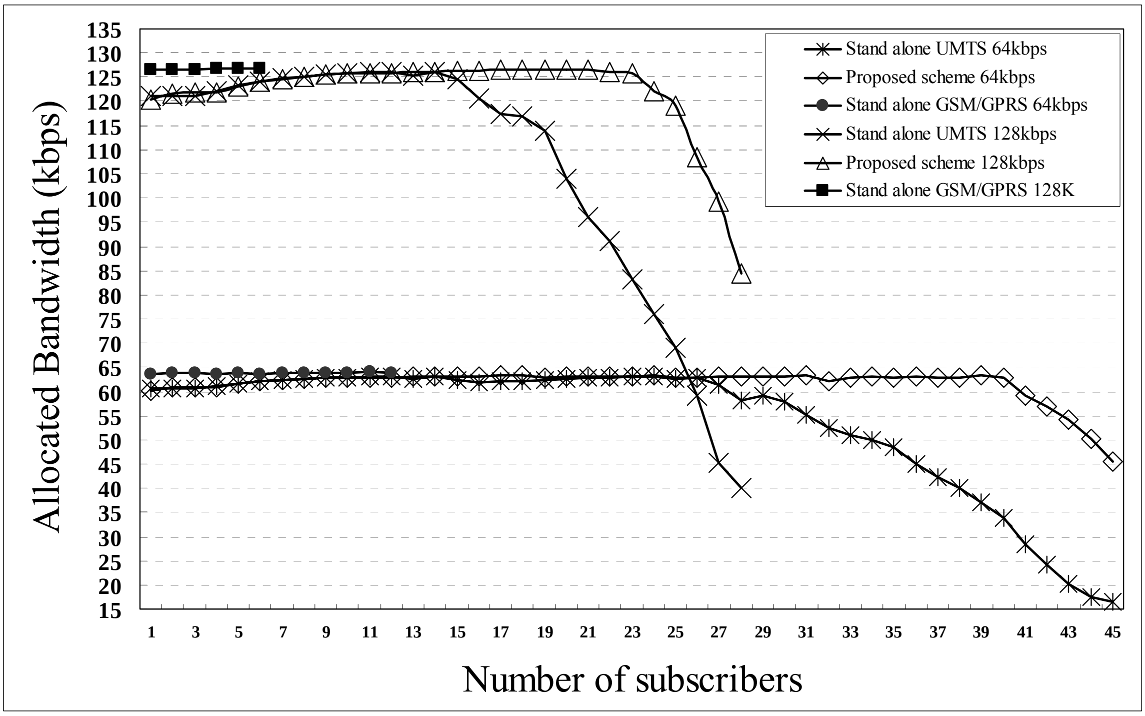


Figure 4.5: Average Allocated Bandwidth

4.4.2 Average packet loss ratio

Figure 4.6 shows that noise from other subscribers did not affect the GPRS packet loss ratio. The packet loss ratio in GPRS was below 10%. The packet loss ratio in UMTS was almost 1%, while the UMTS network has a light loading. However, the packet loss ratio in UMTS increased exponentially, while the UMTS had heavy loading. The packet loss ratio of the integrated system was similar to that of UMTS. The packet loss ratio in UMTS was influenced by the subscriber number. Therefore, the packet loss ratio of the integrated system depends on numbers of subscribers. This phenomenon occurs because UMTS is an interference-limit system, and co-interference increases with rising data flows.

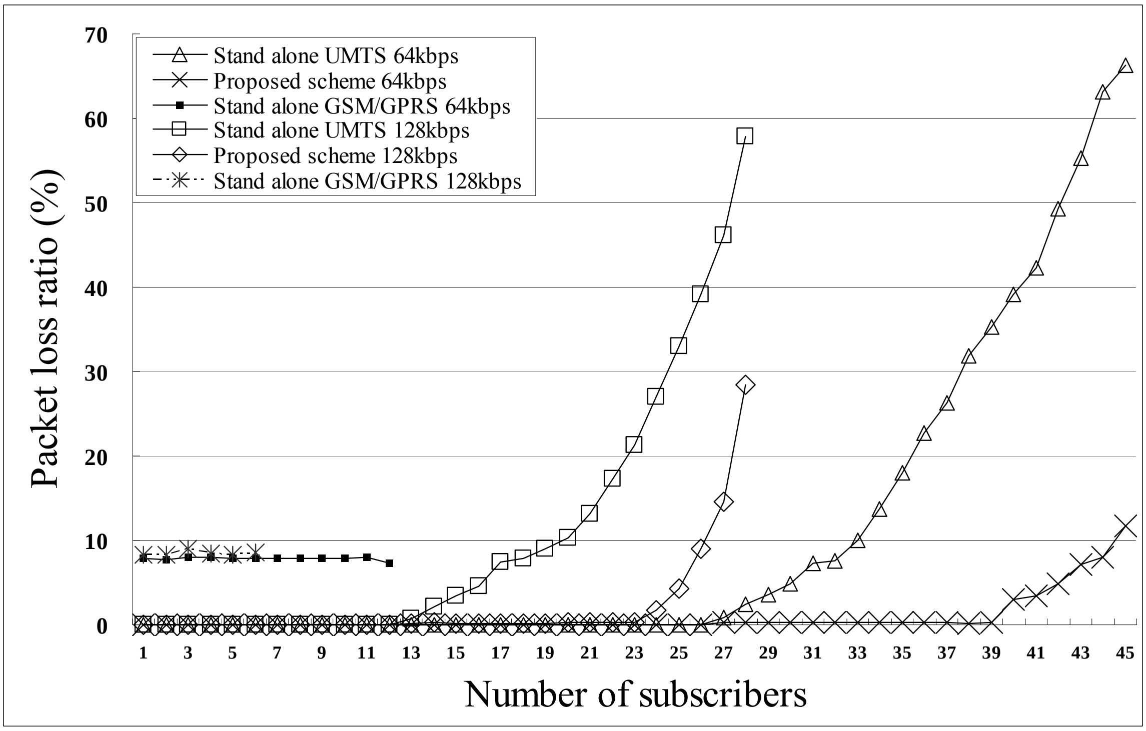


Figure 4.6: Average Packet Loss Ratio

4.4.3 Average delay time

According to **Fig. 4.7**, the average delay time was approximately 1400ms in GPRS and less than 100ms in UMTS. Therefore, GPRS is not appropriate for providing real time connections, since it normally raises the delay time when data connections are handed off from UMTS to GPRS. Raising the delay time degrades the performance of real time connections, and is likely to result in echoes and periods of silence. Significantly, the delay time in UMTS was found to be less than that in GPRS. The average delay time in the integrated system was above 100ms with the traffic balancing scheme. The average delay time of real time connections has to be less than 400ms, as indicated in the rules of 3GPP **[50]**. The data connections in the integrated system can support real-time connections except for data connections in GSM/GPRS. GPRS cannot support real-time data connections.

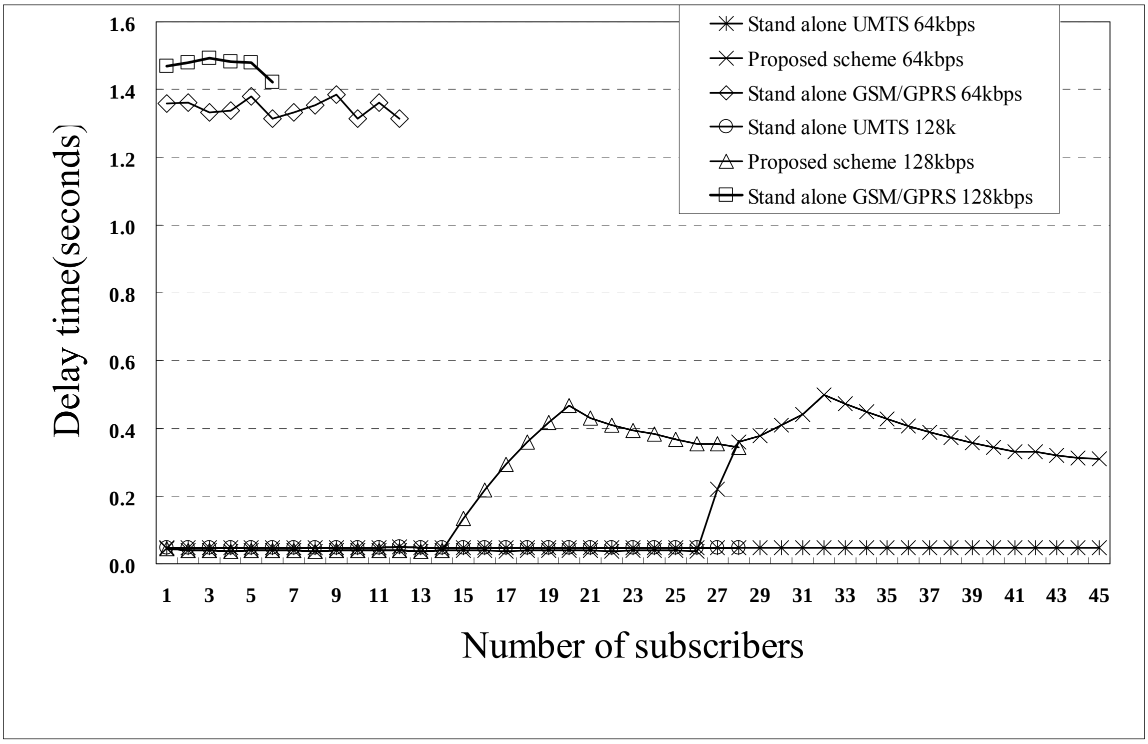


Figure 4.7: Average Delay Time

4.4 Summary

Resource schedule attempts to achieve efficient provision of resources. This chapter presents an integrated resource scheduling scheme comprising GSM/GPRS and UMTS. Based on the definition of the traffic loading threshold, the scheme assigns subscribers to the most appropriate service network, thus raising the spectrum efficiency. A handoff operation of voice connections was implemented in order to avoid dropping connections, reducing the UMTS traffic loading and raising the utilization capacity of UMTS and GSM/GPRS.

Chapter 5
Enhancing WLAN/UMTS Dual-Mode Services Using Distributed Multi-Agent Scheduling Scheme

This chapter proposes a distributed multi-agent scheduling scheme to enhance WLAN/UMTS dual-mode services. WLAN services are inexpensive and have a high bandwidth, while UMTS services provide wider coverage area and high mobility. Based on intelligent deduction, this chapter presents a novel service scheduling scheme for WLAN/UMTS dual-mode networks (see **Fig. 5.1**). The proposed system, Distributed Multi-Agent System (DMAS), consists of a set of problem-solving agents that autonomously process their own tasks and interoperate with one another by a shared database to reach a suitable schedule for dual-mode network services. A two-level control mechanism comprising local-control and meta-control is presented to achieve a high degree of goodness in service scheduling. Simulation results indicate that the quality of service (QoS) of the proposed discipline in terms of average delay and jitter is better than that of the pure UMTS network by 25% and 10%, respectively. The scheduling discipline can improve the service quality in dual-mode networks.

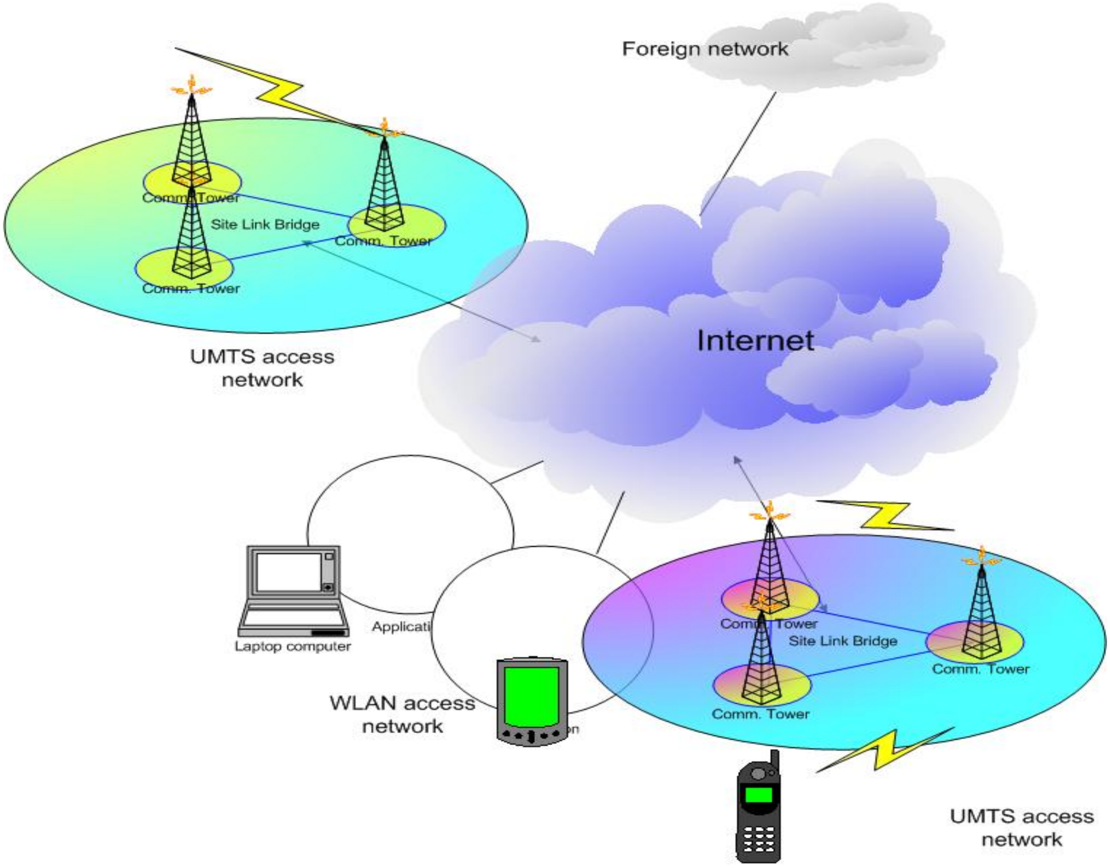


Figure 5.1: Dual-Mode Access Networks

5.1 Proposed DMAS Scheme

The network includes information about delay, packet loss ratio, jitter and free resource. These parameters affect the quality of transmission, especially for real-time services. Real-time services require strict conditions for delay, jitter and packet loss ratio, while non-real time services are less sensitive to these factors. In practice, conversational and streaming services are real-time services, interactive and background services are non-real time services. **Table 5.1** shows the characteristics of different service classes.

Table 5.1: Service Characteristics

Service Class	Mapping Criteria
Conversational Class	Low Delay, Low Jitter
Streaming Class	Low Delay
Interactive Class	Low Loss
Background Class	Best Effort

The network situation influences the networking choice made by the mobile node. Additionally, the service fare and the movement of mobile node are also significant factors in choosing a suitable access network. If a mobile node moves, then it can leave between the WLAN and UMTS radio coverage areas. Each cell in a UMTS network has a wide coverage area, and each base station deploys in high density. Moreover, UMTS has had good handoff mechanisms for a long time. Conversely, WLANs still do not have any good handoff mechanisms, wide radio coverage area or high-density development. Thus, a UMTS network is more suitable than a WLAN for mobility. However, a UMTS service is expensive, since it charges by the number of packets transmitted. The WLAN service is cheaper than UMTS, making it suitable for users who consider cost as a significant factor.

A distributed multi-agent scheme, or DMAS, consists of a collection of interacted agents connected to form a communication network. In the proposed mechanism, the DMAS is added in two-tier architecture (see **Fig. 5.2**). DMAS has several functions, including monitoring the network situation in real time, and acting as QoS broker, determining the level of free resources currently available in the network. DMAS should transmit small packets to probe the network situation on a regular time schedule. The DMAS adopts the information related to the network situation, as variables in a decision function. The result of decision function is then sent to the mobile node. After receiving the result from the DMAS, the mobile node compares them based on the network, and chooses the best network to access. Each agent is a problem solver that can solve the service scheduling issue in a dual-mode network. Such an environment must interoperate to share knowledge and data, and cooperate to solve problems.

Each agent in such an environment can compute autonomously and cooperate with other agents to obtain a satisfactory solution for service scheduling. The three major modules in this system are the knowledge source, the blackboard module and the control engine. **Figure 5.3** shows the multi-agent paradigm and the relationship among different modules. Each problem-solving module is introduced in the following section.

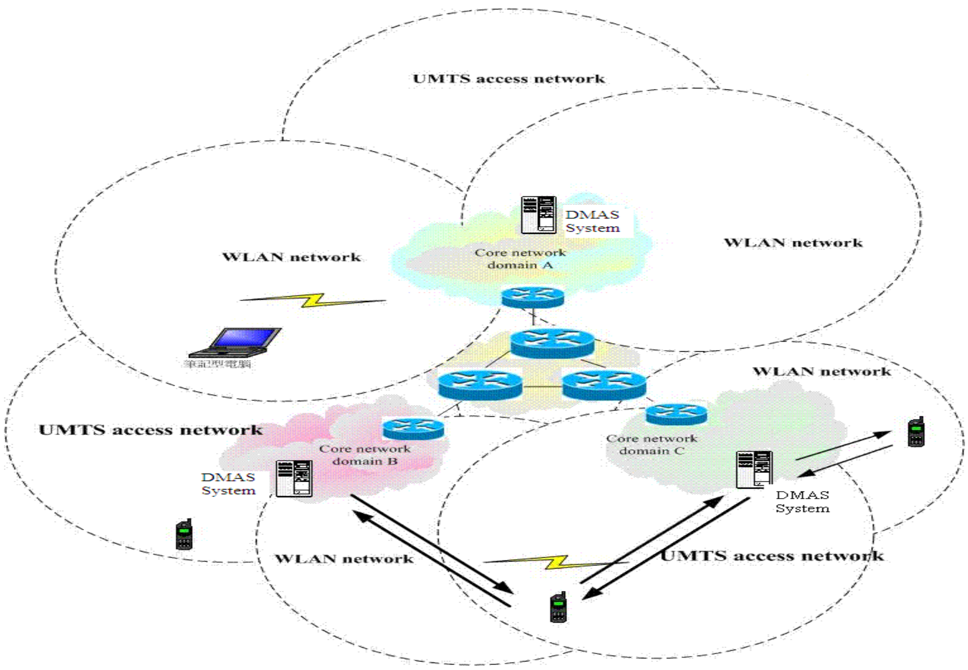


Figure 5.2: DMAS System Operation

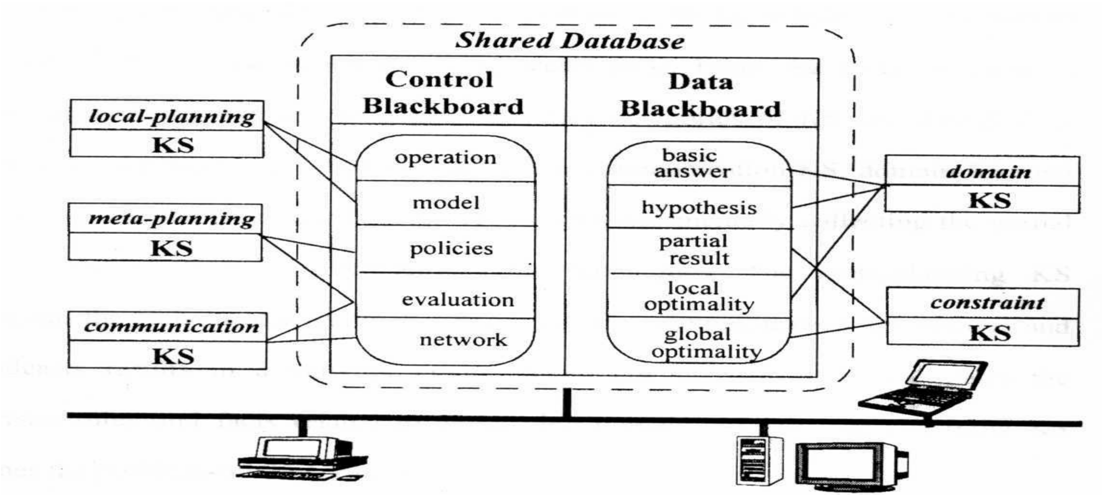


Figure 5.3: Proposed DMAS Paradigm

5.1.1 Knowledge Sources (KSs)

A large-scale application should consist of many agents (problem solvers) to assist the analyst with problem resolution. Each particular agent, KS, has knowledge and reasoning philosophy. From the point of view in industrial applications and this research focus, the five categories of KS are classified as local-planning KS, meta-planning KS, communication KS, domain KS and constraint KS. The local-planning KS focuses on incrementally collecting the partial results derived from the problem-solving agents. The meta-planning KS maintains the global status and makes decisions. The communication KS receives and broadcasts results in a network environment. The domain KS manipulates the heuristic rules and facts related to service scheduling issue. Finally, the constraint KS defines the required criteria in the application domain.

5.1.2 Blackboard Modules

The blackboard system is shared by KSs cooperating to achieve the problem-solving result. The blackboard module, which is a shared database in the physical view, consists of a data blackboard and a control blackboard. For parallelism, community and plurality, both blackboards are structured into several layers. The data blackboard, which synthesizes the results, is separated into five layers: basic answer, hypothesis, partial result, local optimality and global optimality. Additionally, to enhance the efficiency of the inference process, the control blackboard is composed of operation, model, policies, evaluation and network layers.

5.1.3 Control Engine

Two basic issues should be addressed when designing a multi-agent cooperative paradigm. One issue is the optimal control of each problem-solving agent and the other is the obtaining a global optimal from the partial results. Figure 5.4 shows a control engine to resolve the two issues. The engine is described as follows.

Action 1: Both domain knowledge and constraint knowledge are broadcasted to every agent assigned to participate in the problem-solving tasks.

Action 2: Each agent executes the local control based on the blackboard system.

Action 3: Partial result or local optimal results are broadcasted to the idle agent for handling conflict resolution.

Action 4: The processing executes repeatedly until a global optimal result is reached.

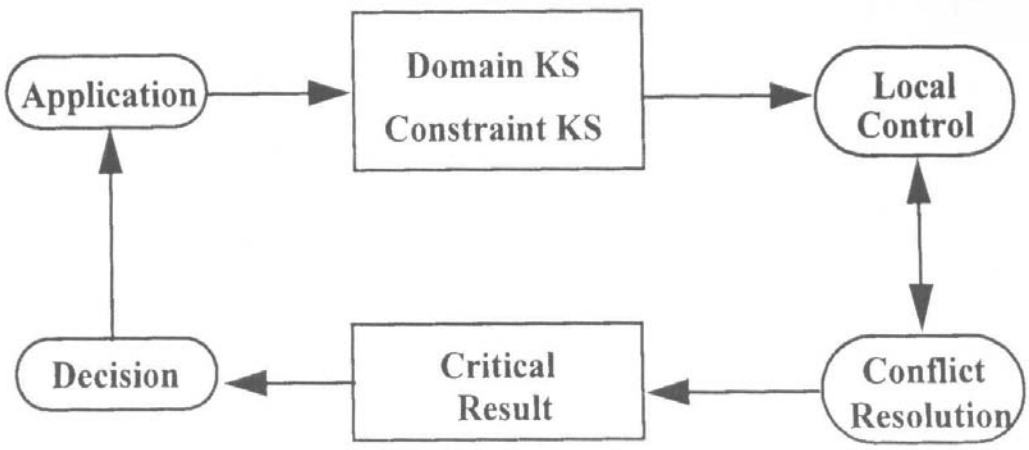


Figure 5.4: Proposed Control Engine

5.1.4 Decision Function

This chapter focuses on optimally allocating service networks to mobile users to maximize the degree of satisfaction for users. This section considers the factors that influence the QoS of the services and the cost of mobile users. The state of the network can be estimated directly from the delay, jitter and loss ratio. The other factors are also introduced to be considered: free resource, mobility capability and service fare. A cost function is proposed as a decision function that simply combines the above weighted factors. The cost function is formulated as:

$$D(R, M, D', J, L, C) = R \times (a \times M + b \times D' + c \times J + d \times L + e \times C) \qquad (5\text{-}1)$$

R: “Resource Parameter”

If enough resources are available in the service network, then R=1, else R=0.

M: “Mobility Activity Parameter”

If the mobile users move frequently, then the user has the opportunity to leave the coverage area of this service network. The action and location of users can be recorded in the system database. The probability of moving can be estimated in advance based on the historical data. The value of M is calculated from the time of moves divided by the total time in the record.

M = $Time_{(moves)}/Time_{(total)}$

D’: “Delay Parameter”

$$1 - \frac{Delay_{UMTS/WLAN}}{Delay_{UMTS} + Dealy_{WLAN}}$$

J: “Jitter Parameter”

$$1 - \frac{Jitter_{UMTS/WLAN}}{Jitter_{UMTS} + Jitter_{WLAN}}$$

L: “Loss Ratio Parameter”

$$1 - \frac{Loss\ Ratio_{UMTS/WLAN}}{Loss\ Ratio_{UMTS} + Loss\ Ratio_{WLAN}}$$

C: “Service Fare Parameter”

C is a constant. The value is set to 1 in the UMTS networks and to 0 in the WLAN networks.

Each network has one set parameter to represent it at each moment. Equation (5-2) is the decision function of UMTS, and Eq. (5-3) is the decision function of WLAN. The result is calculated from Eq. (5-4), as the UMTS decision function minus the WLAN decision function.

$$D(R,M,D',J,L,C)_{UMTS} = R_{UMTS} \times (a \times M_{UMTS} + b \times D'_{UMTS} + c \times J_{UMTS} + d \times L_{UMTS} + e \times C_{UMTS}) \tag{5-2}$$

$$D(R,M,D',J,L,C)_{WLAN} = R_{WLAN} \times (a \times M_{WLAN} + b \times D'_{WLAN} + c \times J_{WLAN} + d \times L_{WLAN} + e \times C_{WLAN}) \tag{5-3}$$

$$\begin{aligned} D(R,M,D',J,L,C)_{UMTS} - D(R,M,D',J,L,C)_{WLAN} = \\ R_{UMTS} \times (a \times M_{UMTS} + b \times D'_{UMTS} + c \times J_{UMTS} + d \times L_{UMTS} + e \times C_{UMTS}) - \\ R_{WLAN} \times (a \times M_{WLAN} + b \times D'_{WLAN} + c \times J_{WLAN} + d \times L_{WLAN} + e \times C_{WLAN}) \end{aligned} \tag{5-4}$$

R_{UMTS} and R_{WLAN} cannot both equal to zero, since each mobile node must be in either the UMTS or WLAN network, and have its resources while decision function is calculated. The system operator can adjust parameters *a*, *b*, *c*, *d* and *e*. If some mobile nodes intend to pay much to use UMTS and ensure that the connection is stable during movement, then parameter *e* can be enlarged. Parameters *b*, *c* and *d* can be increased to ensure a reasonable quality of service of conversational class service. Finally, if the service fare issue is very important for some users, then parameters *a* and *e* can be adjusted separately to suitable values.

Besides the difference in bandwidth, different QoS standards present another problem. The service in UMTS can be classified into four different classes by its characteristics, namely conversational, streaming, interactive and background **[51]**. Defining and distinguishing services by UMTS class after the traffic made handoff into the WLAN network is a popular issue. In an interworking system, the proposed mechanism allows users to select a suitable network must work with a fine QoS mechanism together to obtain the best communication quality and eliminate serious packet loss in important messages.

Diffserv **[52]** is specified as the QoS mechanism in WLAN, and a mapping table was introduced between different QoS levels in UMTS and WLAN. The traffic is mapped into different QoS classes when the traffic traverses from UMTS to WLAN or vice versa. For instance, all the packets from UMTS to WLAN are reclassified in the WLAN edge router. The edge router must support the Diffserv function. For conversational class traffic, the packets are classed into a higher degree. For streaming and interactive class traffic packets belong to the lower degree. Data in the background class are transmitted by way of best effort. The router supporting the Diffserv function first drops lower-level packets when congestion occurs. Some related works have presented mapping tables **[53-54]**. **Table 5.2** presents the QoS mapping between UMTS QoS classes and Diffserv classes.

Table 5.2: QoS Mapping of UMTS Traffic Classes

UMTS QoS Classes	Diffserv Classes
Conversational Class	EF only
Streaming Class	AF1x, AF2x
Interactive Class	AF3x, AF4x
Background Class	BE only

Figure 5.5 shows the flowchart of deciding whether to select WLAN or UMTS networks. If the connection between mobile node and access network is being established for the first time, then the mobile node uses the WLAN as the candidate. Otherwise, the mobile node may use the most appropriate network at that moment. The node receives the information, and calculates the difference between the values derived from the decision function of UMTS and WLAN. If the difference is less than zero, then the mobile node should use the WLAN radio resource, because it implies that the cost in the WLAN is greater than that in UMTS. Conversely, the mobile node should use the UMTS radio resource to communicate when the difference between the UMTS and WLAN is more than zero. If the difference value is zero, then the cost of accessing the UMTS network is the same as that of accessing the WLAN network, and the mobile device can stay in the original network.

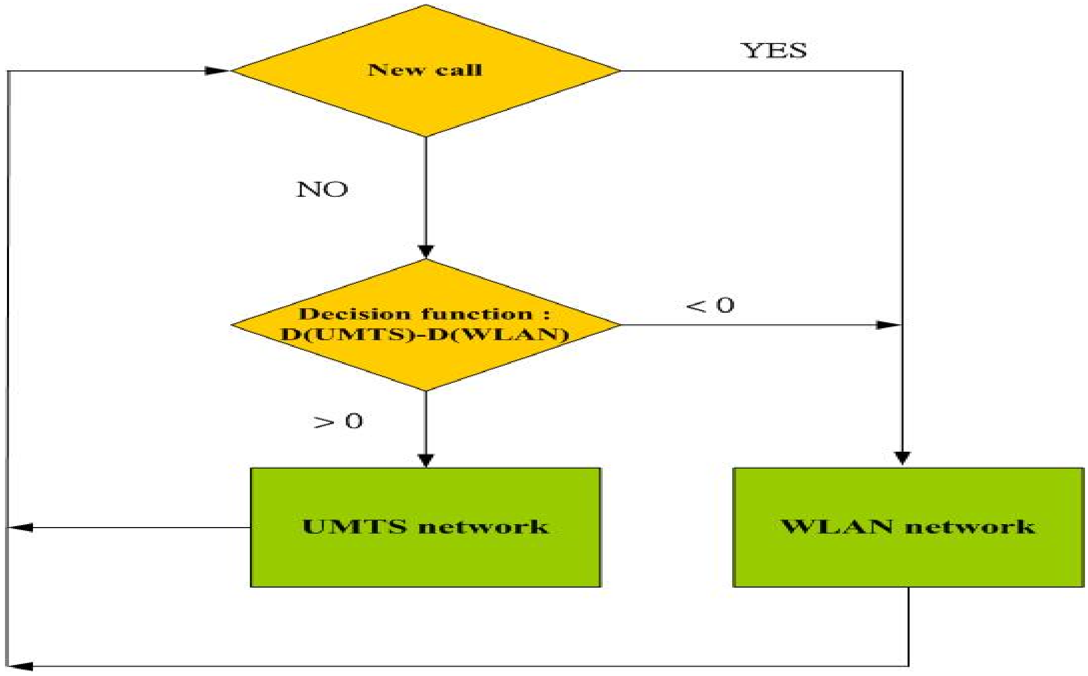


Figure 5.5: Decision Flowchart

5.1.5 Implementation

The prototype of multi-agent paradigm is implemented on Jess toolkit. Jess is a clone of the popular expert system shell CLIPS, written entirely in Java (http:// herzberg.ca.sandia.gov/jess/). Since CLIPS is based on the C programming language, and its Java-based user interface is very flexible, Jess was selected for developing the autonomous functions of agents. The interoperability functions are implemented by distributively processing the Jess, which is the source code of CLIPS, by adding the RPC function call facility.

5.2 Performance Evaluation

The Network Simulator (ns2 toolkit) and the UMTS module EURANE (Enhanced UMTS Radio Access Network Extensions for ns-2) were used to simulate the WLAN/UMTS environment. The following subsections describe the simulation results.

5.2.1 Knowledge Sources

The domain KS and constraint KS are two major components of the DMAS scheme. According to the experimental data and service features of each network, the domain KS, including the rule base and data base, can be generated and placed into the programming interface. Examples of a basic rule in the rule base, and a datum in the data base, are given below.

```
(defrule      WLAN-Scheme
        (> (service-data-rate    2))
        (wlan-rssi high)
        (service fare low)
=>
          (save-facts WLAN-Scheme)
(printout t “The WLAN scheme is suitable for service network” crlf))

(deffacts UMTS-Scheme
          (service-data-rate low)
(high-mobility yes
(umts-delay-time 0.8)
(umts-delay-time-0.5 0.4))
```

When the scheme is enabled and the current status is input into the test environment, the domain KS is then fed into the scheme, thus inferring the feasible schedules that satisfy the basic rules. Section 5.1.3 introduces a key component, the control engine, to obtain the optimal solution. Some criteria in the constraint KS must also be defined to receive the optimum solution.

5.2.2 Delay Analysis

Mobile nodes can obtain more available bandwidth in WLAN environment, since that a WLAN system provides much more bandwidth than a cellular system in a wireless environment. The estimated transmission delay time is shorter in WLAN than in UMTS when the number of users on the network is small and the network is running (see **Fig. 5.6**). In the interworking system of integrated UMTS/WLAN, a mobile node can adopt either the WLAN resource or the UMTS resource, according to the access duration of the resource after handoff. The delay time in interworking system varies with the period in different networks, possibly due to the combination of UMTS and WLAN delay times.

The delay time is shorter if mobile users spend longer on a WLAN. The mean delay time is longer when mobile users spend more time in UMTS than in WLAN. However, the duration in different network is linked to the usage habits of mobile users and the geographical environment.

Figure 5.6 shows the delay time of the pure WLAN, pure UMTS, and DMAS approaches. To represent the distributions in different networks, five different assumptions were simulated by the DMAS approach. They are (10%UMTS, 90%WLAN), (25%UMTS, 75%WLAN), (50%UMTS, 50%WLAN), (75%UMTS, 25%WLAN), and (90%UMTS, 10%WLAN). The notation (10%UMTS, 90%WLAN) means a 10% possibility of a mobile node moving to the UMTS network and a 90% possibility of it moving to a WLAN network. Similarly, (50%UMTS, 50%WLAN) indicates that the probability of using UMTS is the same as that of using WLAN. Each user in each network may have been the user of that network (UMTS/WLAN), or have come from other networks (WLAN/UMTS) after handoff.

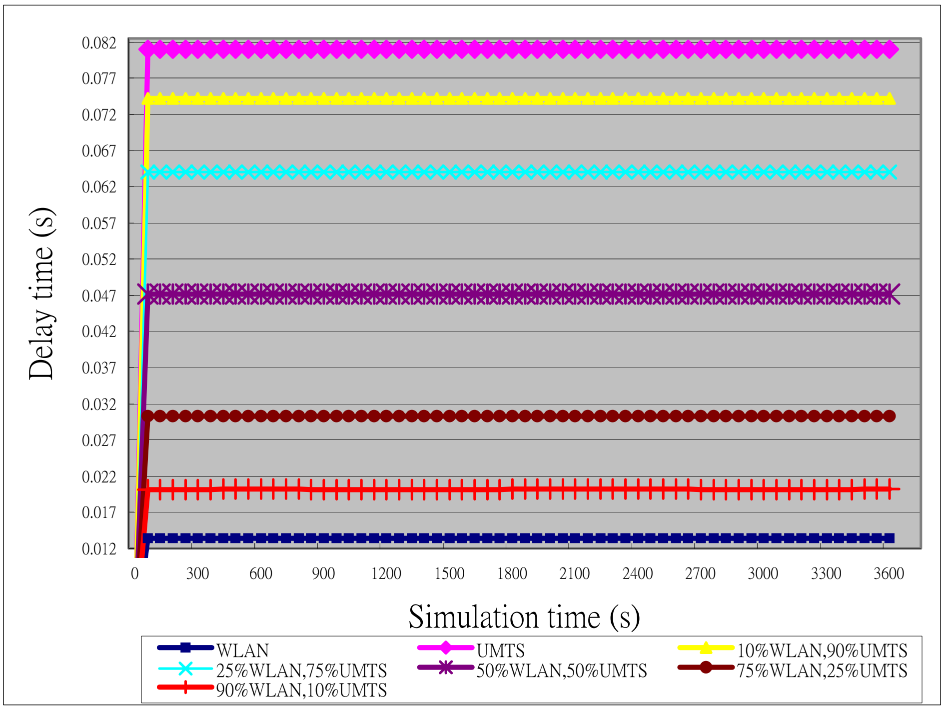


Figure 5.6: Delay Time of Streaming Service

5.2.3 Jitter Analysis

Jitter significantly influences the quality of interoperating services. For instance, the variance of delay time must be minimized in video conferencing, audio streaming and general conversation. **Table 5.3** shows the jitters of different service types in a WLAN and UMTS network.

Table 5.3: Jitter of Different Service Types (s: second)

Service Type	UMTS	WLAN
Conversational service	0.004922s	0.000011s
Streaming service	0.033248s	0.000027s
Interactive service	0.051013s	0.000613s
Background service	Don’t care	0.000538s

Users in integrated WLAN/UMTS networks may adopt either UMTS or WLAN, so the variance of delay time can also be greater than that in a pure WLAN or pure UMTS environment. The result is shown in **Table 5.4**.

Table 5.4: Jitter of Different Service Types in Integrated WLAN/UMTS Networks

	(0.1,0.9)	(0.25,0.75)	(0.5,0.5)	(0.75,0.25)	(0.9,0.1)
Conversational Service	0.070018s	0.033502s	0.033684s	0.033502s	0.026538s
Streaming Service	0.020785s	0.029986s	0.030731s	0.030397s	0.021721s
Interactive Service	0.02121s	0.027950s	0.026352s	0.018636s	0.007078s
Background Service	Don’t care	Don’t care	Don’t care	Don’t care	Don’t care

5.2.4 Mobility Analysis

UMTS clearly has better mobility than WLAN. A UMTS is the most appropriate network to use when comparing pure UMTS and WLAN networks. However, a UMTS is more convenient and beneficial to use when mobility is needed, while a WLAN is preferable when mobility is not requested. Thus, the system service can be provided to the people who urgently need mobility while saving cost. This scenario can be achieved by the proposed decision rule in the interworking system of integrated UMTS/WLAN. Mobility is clearly not the only factor in the proposed decision rule. The decision result is significantly improved when the parameters of other factors are much greater than the mobility parameter. If operators consider mobility as a very important factor influencing the QoS and customer satisfaction, then the mobility parameter can be set to a high value (value=1). Additionally, the decision result is not to use UMTS when mobile users are in the mobile state, since two networks are available, and the analytical results indicate that WLAN is more useful than UMTS. However, it always conforms to benefit in use and network performance based on what the operators concerned about, and the parameters that they set. An integrated UMTS/WLAN has a “medium” grade mobile property.

5.2.5 Integrated UMTS/WLAN System with/without Decision Rule

The analytical results, presented in Subsections 5.2.1-5.2.4, indicate that an integrated UMTS/WLAN is a feasible system. However, if no good mechanism is available to balance the traffic loading between the two networks, UMTS and WLAN, when the interworking system works, then each may be a bottleneck. Almost all traffic is centered on only one of the networks when no suitable decision rule is available to inform a mobile device of the most suitable network to access.

The proposed decision rule can solve the problem of traffic balance. The system can probe the network performance, such as delay, loss and jitter, before accessing any network. An appropriate rule is activated based on the probed result. The load of a network with poor performance cannot be increased. This could prevent deterioration in transmission quality.

A simulation environment is described herein to explain the proposed system. One environment was created with different traffic loadings, and the variation was examined in different networks (WLAN and UMTS). **Figure 5.7** shows the delay time at different loadings. The transmission delay in the UMTS was about 0.04 seconds. The delay time in the WLAN varied according to different loadings. The transmission delay in UMTS was lower than that in WLAN when the traffic loading was more than 3076kbits. In this case, the decision rule can inform heavily loaded mobile users to adopt the UMTS network.

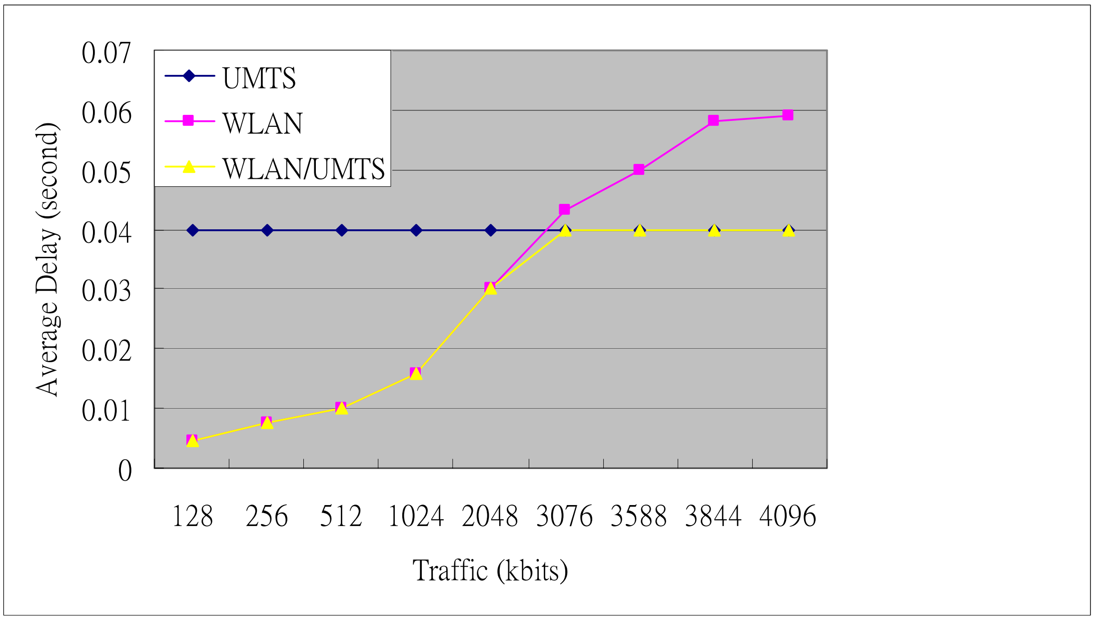


Figure 5.7: Delay Time with Different Loadings

Figure 5.8 shows the influence of jitter on the network. The jitter in UMTS was about 0.2544 seconds, and the variance of delay time was more turbulent in WLAN with rising loading variance. The jitter in UMTS network was low when the traffic loading was greater than 2048kbits. Hence, UMTS is a suitable network for users when the traffic is above 2048kbits. This case can be one reference for a decision rule. The decision rule lets a mobile user make a handoff to solve the jitter problem in networks.

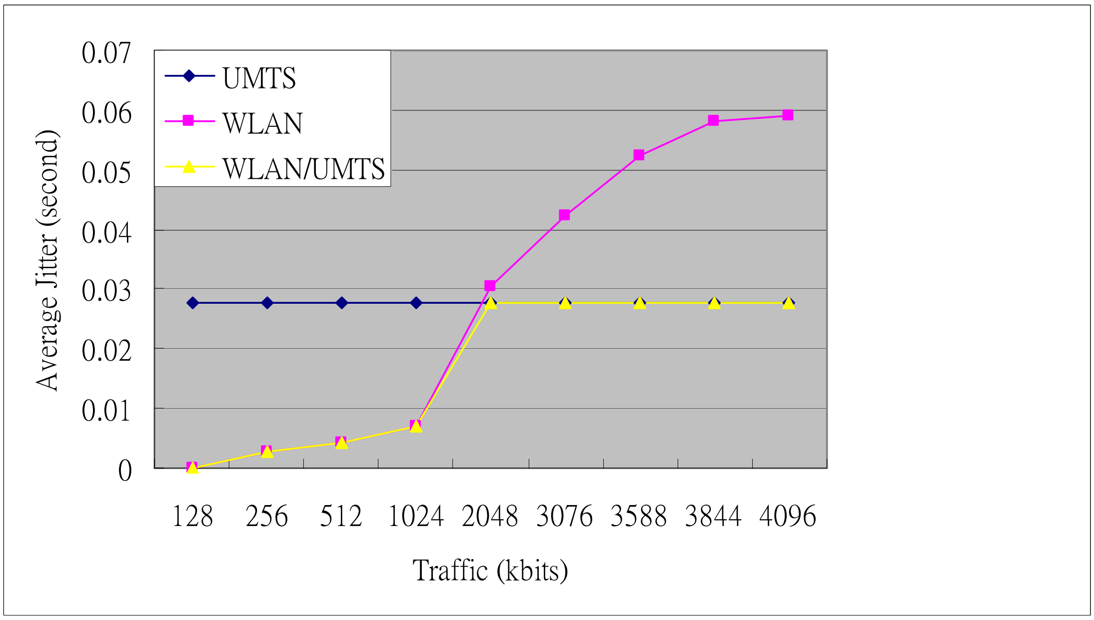


Figure 5.8: Jitter with Different Loadings

5.3 Summary

This chapter has presented a distributed multi-agent scheme to enhance service scheduling in WLAN/UMTS dual-mode networks. The developed system is implemented in the Jess expert system. Simulation results clearly indicate that this scheme works very well. Additionally, the distributed multi-agent scheme can satisfy all heuristic rules by scheduling plan, and is an effective means of enhancing service scheduling. A similar method can be applied to the design of large-scale data networks in which a variety of high QoS services are available.

Chapter 6
Conclusion and Future Works

6.1 Conclusion

The proposed adaptive QoS manager combines the wired and wireless QoS concerns into admission control. Different home applications have different key QoS requirements in congested home networks. For instance, VoIP applications are concerned about the jitter, but can tolerate some packet loss, which does not significantly harm the integrity of the conversation. High-definition video stream is mainly concerned with packet loss, but can tolerate a small time delay and jitter. The QoS attributes are integrated to perform adaptive admission control in wired-wireless heterogeneous home networks. Simulation results indicate that although some throughput is sacrificed in low-priority application like Standard Definition (SD) video stream, each application is guaranteed to have the minimum but the most important QoS requirement.

Resource scheduling scheme attempts to achieve efficient provision of resources. This dissertation presents an integrated resource scheduling scheme comprising GSM/GPRS and UMTS. Based on the definition of the traffic loading threshold, the scheme assigns subscribers to the most appropriate service network, thus raising the spectrum efficiency. A handoff operation of voice connections was implemented in order to avoid dropping connections, reducing the UMTS traffic loading and raising the utilization capacity of UMTS and GSM/GPRS.

This dissertation has presented a distributed multi-agent scheme to enhance service scheduling in WLAN/UMTS dual-mode networks. The developed system is implemented in the Jess expert system. Simulation results clearly indicate that this scheme works very well. Additionally, the distributed multi-agent scheme can satisfy all heuristic rules by scheduling plan, and is an effective means of enhancing service scheduling. A similar method can be applied to the design of large-scale data networks in which a variety of high QoS services are available.

6.2 Future Works

Based on the research results of my Ph.D dissertation, several issues on QoS can be further investigated as listed blow.

Study on other QoS related concerns in home networks

With the rapid growth on home networking technology, more QoS parameters should be taken into consideration, such as mobility [63,64], security [65] and handoff. If a QoS manager has a negotiation mechanism to communicate with outside gateway server such as a QoS broker, it will involve more resources to manage those large scale of heterogeneous integration problems, not only wired and wireless QoS integration in home.

Study on resolving overhead of dual-mode service networks

The overhead of handoff may exist when we select a proper network. Too many times of handoff makes the system busy in registration, location area update, routing area update, and so on. It is a vital issue to solve the problem of overhead and to avoid the high cost of handoff in the future.

Study on End-to-End QoS guarantees

Modern wireless networks are essentially composed of different portions and technologies. Each single portion may implement a different QoS solution. The challenge is to offer end-to-end QoS guarantees over such heterogeneous networks to the users. QoS requests should traverse the overall network from the source to the destination through portions that implement different technologies and different protocols. There are still several issues on QoS of heterogeneous wireless networks can be investigated further, such as *Cross-Layer Design* [55-57], *Cognitive Network* [58-60] and *Cooperative Network* [61-62]. More QoS mechanisms for other wireless network technologies such as WPAN and WMAN are also required to develop.

References

[1] K. Wongthavarawat, "Resource Management for QoS Support in Wireless Networks," *Ph.D Dissertation*, University of Massachusetts Amherst, 2004.

[2] P. A. Wasan, "Quality of Service Support in IEEE 802.11 Wireless Lan," *Ph.D Dissertation*, University of Pittsburgh, 2005.

[3] M. S. Gast, "802.11 Wireless Networks: The Definitive Guide," O,REILLY & Associates, Inc., 2002.

[4] Nokia Network, "A History of Third Generation Mobile," http://www.nokia.com/BaseProject/Sites/NOKIA_MAIN_18022/CDA/Categories/Networks/Technologies/Network/GSM/_Content/_Static_Files/3g_history.pdf, March 2003.

[5] De Vriendt Johan, Lainé Philippe, Lerouge Christophe, and Xiaofeng Xu, "Mobile Network Evolution: A Revolution on the Move," *IEEE Communications Magazine*, vol. 40, no. 4, pp. 104-111, April 2002.

[6] GSM 05.05, "European digital cellular telecommunications systems (Phase 2): Radio Transmission and Reception."

[7] Jingao Wang, Qing-An Zeng, Agrawal and Dharma P., "Performance analysis of a preemptive and priority reservation handoff scheme for integrated service-based wireless mobile networks," *IEEE Transactions on Mobile Computing*, vol. 2 , no. 1, pp. 65–75, January-March 2003.

[8] GSM 03.02, "Network architecture."

[9] Jatuporn Pliancharoen, Punyapat Chofabunthit and Suthichai Noppanakeepong, "Optimum traffic channel in GSM network by using alternative call routing (international roaming analytic)," *Proceedings of International Symposium on Parallel Architectures and Algorithms and Networks*, pp. 404–407, May 2004.

[10] GSM 02.60, "GPRS, Service description, Stage 1."

[11] GSM 09.60, "GPRS, GPRS Tunneling Protocol (GTP) across the Gn and Gp interface."

[12] GSM 04.65, "GPRS, MS-SGSN, Subnetwork Dependent Convergence Protocol (SNDCP)."

[13] GSM 04.64, "GPRS, MS-SGSN, Logical Link Control (LLC) layer."

[14] GSM 08.18, "Digital cellular telecommunications system (Phase 2+); General Packet Radio Service (GPRS); Base Station System (BSS) Serving GPRS Support Node (SGSN); BSS GPRS Protocol (BSSGP)."

[15] GSM 08.16, "Digital cellular telecommunications system (Phase 2+); General Packet Radio Service (GPRS); Base Station System (BSS) Serving GPRS Support Node (SGSN) interface; Network Service."

[16] GSM 04.60, "GPRS, MS-BSS interface, Radio Link Control /Medium Access Control (RLC/MAC) protocol."

[17] GSM 09.61, "GPRS, Interworking between the PLMN supporting GPRS and Packet Data Networks (PDN)."

[18] Josef F. Huber, "Mobile next-generation networks," *Proceedings of IEEE Multimedia*, vol.11, no. 1, pp. 72–83, January-February 2004.

[19] 3GPP TS23.121, "Architectural Requirements for Release 1999."

[20] Aftelak, S.and Bhatoolaul, D., "Rate matching attribute settings and error rate performance sensitivity for selected UMTS FDD services," *Proceedings of IEEE*

58th Vehicular Technology Conference, vol. 3, pp. 1558–1562, 6-9 October 2003.
[21] F. Fitzek, M. Munari, V. Pastesini, S. Rossi, and L. Badia, “Security and authentication concepts for UMTS/WLAN convergence,” *Proceedings of IEEE 58th Vehicular Technology Conference*, vol.4, pp.2343-2347, 2003.
[22] S. Blake, D. Black, M. Carlson, E. Davies, Z. Wang and W. Weiss, “An Architecture for Differentiated Services,” *IETF RFC2475*, December 1998.
[23] S.Brim, B. Carpenter, F. Le Faucheur, “Per Hop Behavior Identification Codes,” *IETF RFC2836*, May 2000.
[24] J. Wroclawski, “The Use of RSVP with IETF Integrated Services,” *RFC 2210*, September 1997.
[25] J. Heinanen, Telia Finland, F. Baker, W. Weiss and J. Wroclawski, “Assured Forwarding PHB Group,” *IETF RFC 2597*, June 1999.
[26] V. Jacobson, K. Nichols and K. Poduri, “An Expedited Forwarding PHB,” *IETF RFC 2598*, June 1999.
[27] B. Davie, A. Charny and J.C.R. Bennett, “An expedited forwarding PHB (Per-Hop behavior),” *IETF RFC 3246*, March 2002.
[28] K. Kilkki, “Differentiated Services for the Internet,” Macmillan Technical Publishing, 1999.
[29] C. Bouras, M. Campanela and A. Sevasti, “SLA definition for the provision of an EF-based service,” *Proceedings of 16th International Workshop on Communications Quality and Reliability (CQR 2002)*, pp.17-21, May 2002.
[30] Y. Bernet, P. Ford and R. Yavatkar, “A Framework for Integrated Services Operation over Diffserv Networks,” *IETF RFC 2998*, November 2000.
[31] F. Baker, C. Iturralde, F. Le Faucheur and B. Davie, “Aggregation of RSVP for IPv4 and IPv6 Reservations,” Internet Draft draft-ietf-issll-rsvp-aggr-04.txt, Work in Progress, April 2001.
[32] K. Nichols, V. Jacobson and L. Zhang, "A Two-bit Differentiated Services Architecture for the Internet," *RFC 2638*, July 1999.
[33] L. Westberg, G. Partain, S. D. Oosthoek, M. Jacobsson and V. Rexhepi, “Resource Management in Diffserv On DemAnd (RODA) PHR,” Internet Draft draftwestberg -rmd-od-phr-00.txt, Work in Progress, April 2001.
[34] H.F. Lu, S.H. Chang, B.J. Huang, J.L. Chen, B. Huang and M.C. Yang, “Design of Home Network Diagnosis System,” *Proceedings of International Conference on Internet and Multimedia Systems and Applications,* pp.101-106, August 14-16, 2006.
[35] ETSI TS 101 113 V7.5.0 (2000-07), Technical Specification, “Digital cellular telecommunications system (Phase 2+); General Packet Radio Service (GPRS); Service description; Stage 1; (GSM 02.60 version 7.5.0 Release 1998)”
[36] 3GPP TS 03.60 V7.9.0 (2002-09), Technical Specification, “3rd Generation Partnership Project; Technical Specification Group Services and System Aspects; Digital cellular telecommunications system (Phase 2+); General Packet Radio Service (GPRS); Service description; Stage 2 (Release 1998)”
[37] G. Cerri, R.D. Leo, D. Micheli and P. Russo, “Base-station Network Planning Including Environmental Impact Control,” *Proceeding of IEE Communications,* vol. 151, no. 3, pp. 197-203, 25 June 2004.
[38] J.-S. Chen and C.-R. Dow, “Fixed Channel Allocation Scheme Performance Enhancement for Cellular Mobile Systems,” *Proceeding of IEE Communications,* vol. 149, no. 4, pp. 232-236, Aug. 2002

[39] J. Singh, “Performance Aspects of Enhanced Radio Resource Management with QoS Differentiation in UMTS Network,” *Proceedings of 2005 6th International Conference on 3G and Beyond*, pp. 1-5, Nov. 2005.

[40] S. Oriol, P. Jordi, A. Ramon and C. Ferran, “Provisioning Multimedia Wireless Networks for Better QoS: RRM Strategies for 3G W-CDMA,” *IEEE Communication Magazine*, vol. 41, no. 2, pp. 100-106, Feb. 2003.

[41] R. R. Collmannn, “Evaluation of Methods for Determining the Mobile Traffic Distribution in Cellular Radio Networks,” *IEEE Transactions on Vehicular Technology,* vol. 50, no. 6, pp. 1629-1635, Nov. 2001

[42] M. Filippo, M. Breveglieri, L. Vignali, P. Leaves, J. Huschke, “Traffic control algorithms for a multi access network scenario comprising GPRS and UMTS,” *Proceedings of 57th IEEE Vehicular Technology Conference*, vol. 1, pp.145-149, 22-25 April 2003.

[43] T. Ozugur, A. Bellary, and F. Sarkar, “Multiobjective Hierarchical 2G/3G Mobility Management Optimization: Niched Pareto Genetic Algorithm,” *Proceedings of IEEE Global Telecommunications Conference*, vol.6, pp.3681-3685, Nov. 2001.

[44] M. Benson, and H.J. Thomas “Investigation of the UMTS to GSM handover procedure,” *Proceedings of IEEE 55th Vehicular Technology Conference,* vol. 4, pp. 1829-1833, May 2002

[45] 3GPP TS 26.103 v6.1.0 (2004-12), “Speech codec list for GSM and UMTS (Release 6)”

[46] 3GPP TS 26.171 v6.0.0 (2004-12), “Adaptive Multi-Rate – Wideband (AMR-WB) speech codec; General description (Release 6)”

[47] ITU-T Recommendation G.722.2, “Wideband coding of speech at around 16 kbit/s using Adaptive Multi-Rate Wideband (AMR-WB),” 2003.

[48] S. Ariyavisitakul, “SIR-based power control in a CDMA systems,” *Proceedings of GLOBECOM 92nd*, vol. 2, pp. 868-873, Dec 1992.

[49] K. Sipia, Z.C. Honkasalo, S.J. Laiho and A. Wacker, “Estimation of Capacity and Required Transmission Power of WCDMA Downlink based on a Downlink Pole Equation,” *Proceedings of 51st IEEE Vehicular Technology Conference*, vol. 2, pp.1002-1005, May 2000.

[50] 3GPP TS23.107, “Quality of Service (QoS) Concept and Architecture,” June 2001.

[51] 3GPP, “Quality of Service (QoS) Concept and Architecture (Release 5),” *Technical Specification*, TS 23.107, V5.13.0, December 2004.

[52] S. Blake, D. Black, M. Carlson, E. Davies, Z. Wang and W. Weiss, “An Architecture for Differentiated Services,” *IETF RFC 2475*, December 1998.

[53] S. Y. Park, D. Kim, K. Kim, S. Choi, and S. Hong, “Collaborative QoS Architecture between DiffServ and 802.11e Wireless LAN,” *Proceedings of the 57th IEEE Semiannual Vehicular Technology Conference (VTC'03-Spring)*, vol. 2, pp. 945-949, April 2003.

[54] F. Agharebparast and V.C.M. Leung, "QoS Support in the UMTS/GPRS Backbone Network using DiffServ", *Proceedings of Global Telecommunications Conference (GLOBECOM’02)*, vol. 2, pp. 1440-1444, November 2002.

[55] M. V. Schaar and S. Shankar, “Cross-Layer Wireless Multimedia Transmission: Challenges, Principles, and New Paradigms,” *IEEE Wireless Communications*,

vol. 12, no. 4, pp. 50-58, Aug. 2005.
[56] M. Johansson and L. Xiao, "Cross-Layer Optimization of Wireless Networks Using Nonlinear Column Generation," *IEEE Transactions on Wireless Communications*, vol. 5, no. 2, Feb. 2006.
[57] K. A. Noordin and G. Markarian, "Cross-Layer Optimization Architecture for WiMAX Systems," *Proceedings of the 18th Annual IEEE International Symposium on Personal, Indoor and Mobile Radio Communications (PIMRC'07)*, pp. 1-4, Sept. 2007.
[58] R. W. Thomas, L. A. Dasilva, M V. Marathe and K. N. Wood, "Critical Design Decisions for Cognitive Networks," *Proceedings of IEEE International Conference on Communication (ICC'07)*, pp. 3993-3998, June 2007.
[59] P. Houze, D. Ruiz, S. B. Jemaa and P. Cordier, "Dynamic Spectrum Allocation Algorithm for Cognitive Networks," *Proceedings of Third International Conference on Wireless and Mobile Communications (ICWMC'07)*, pp. 25-29, March 2007.
[60] Z. Huang, H. Huang, F. Lin, G. Yue and D. Xu, "Dynamic Channel Allocation Scheme for Hybrid Cognitive Network," *Proceedings of IEEE International Workshop on Anti-Counterfeiting, Security, Identification*, pp. 179-183, April 2007.
[61] I. Krikidis and J. C. Belfiore, "Scheduling for Amplify-and-Forward Cooperative Networks," *IEEE Transactions on Vehicular Technology*, vol. 56, no. 6, Nov. 2007.
[62] Y. W. Hong, W. J. Huang, F. H. Chiu and C. C. Kuo, "Cooperative Communications in Resource-Constrained Wireless Networks," *IEEE Signal Processing Magazine*, vol. 24, no. 3, pp. 47-57, May 2007.
[63] S. Ben Rejeb, Z. Choukair and S. Tabbane, "QoS Management and Real Time Agreement Protocols for Resource Reservation for Multimedia Mobile Radio Network," *Computer Systems and Applications*, pp.45-53, March 2006.
[64] H. Chih-Shun, S Jang-Ping and T. Shen-Chien, "An On-Demand Bandwidth Reservation QoS Routing Protocol for Mobile Ad Hoc Networks," *Proceedings of the Sensor Networks, Ubiquitous, and Trustworthy Computing Conference*, pp.198-207, June 2006.
[65] X. ZhengYou and H. YunAn, "Extending RSVP for Quality of Security Service," *Internet Computing*, Vol. 10, No. 2, pp.51-57, March-April 2006.

Curriculum Vitae

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Publication List

Journal Paper

1. J. L. Chen, **M. C. Chen** and Y. R. Chian, "QoS Management in Heterogeneous Home Networks," *Computer Networks*, vol. 51, no. 12, pp. 3368-3379, Aug. 2007. (SCI, IF=0.631)
2. J. L. Chen, **M. C. Chen**, C. W. Chen and Y. C. Chang, "Architecture Design and Performance Evaluation of RFID Object Tracking Systems," *Computer Communications*, vol. 30, no. 9, pp. 2070-2086, June 2007. (SCI, IF=0.444)
3. J. L. Chen, **M. C. Chen** and Y. C. Chang, "Enhancing WLAN/UMTS Dual-Mode Services Using A Novel Distributed Multi-Agent Scheduling Scheme," accepted by *Computers & Electrical Engineering*, Dec. 2007. (SCI)
4. J. L. Chen, S. W. Pan, and **M. C. Chen**, "Adaptive Radio Resource Management in an Integrated GPRS/UMTS Service Network," under revision in *European Transactions on Telecommunications (ETT-06-0051.R1)*, Oct. 2007. (SCI)

Conference Paper

1. J. L. Chen, **M. C. Chen**, T. L. Chiang, Y. C. Chang and F. Shih, "Distributed Fermat-Point Location Estimation for Wireless Sensor Network Applications," *Proceedings of 2007 Sarnoff Symposium,* New Jersey, USA, May 2007.
2. J. L. Chen, **M. C. Chen**, P. Y. Huang and Y. C. Chang, "Cluster-Grid Structure Routing Protocol for Sensor Mobility Management," *Proceedings of 2007 Sarnoff Symposium,* New Jersey, USA, May 2007.
3. J. L. Chen, **M. C. Chen**, C. W. Chen and Y. C. Chang, "Architecture Design and Performance Evaluation of RFID Object Tracking Systems," *Proceedings of the International Computer Symposium 2006 (ICS 2006)*, pp. 765-770, Taipei, Taiwan, December 04-06, 2006.
4. **M.C. Chen**, J.L. Chen, Y.C. Chang and Y.R. Chian, "Adaptive QoS Management in Heterogeneous Home Network Applications," *Proceedings of 18th International Conference on Parallel and Distributed Computing and Systems (PDCS 2006)*, pp. 271-276, Dallas, USA, November 13-15, 2006.
5. J. L. Chen, Y. C. Chang and **M. C. Chen**, "Enhancing WLAN/UMTS Dual-Mode Services Using a Novel Distributed Multi-Agent Scheduling Scheme," *Proceedings of the Eleventh IEEE Symposium on Computers and Communications (ISCC 2006)*, pp. 851-856, Santa Margherita di Pula, Cagliari, Italy, June 2006.
6. **M. C. Chen**, J. L. Chen and P. C. Yao, "Efficient Handoff Algorithm for Bluetooth Networks," *Proceedings of 2005 IEEE International Conference on Systems, Man and Cybernetics (SMC 2005)*, pp. 3884-3889, the Big Island, Hawaii, USA, October 2005.
7. **M. C. Chen**, H. F. Lu, J. L. Chen and P. C. Yao, "Efficient Handoff Scheme for Bluetooth Network Services," *Proceedings of the IASTED International Conference on Wireless Networks and Emerging Technologies (WNET 2005)*, Banff, Alberta, Canada, July 2005.

8. H. F. Lu, H. W. Tzeng, **M. C. Chen** and J. L. Chen, "Design of a Residential Gateway for Tele-Homecare Systems," *Proceedings of the Ninth International Symposium on Consumer Electronics (ISCE2005)*, pp. 291-295, Macau, June 2005.
9. W. C. Chan, **M. C. Chen**, J. L. Chen, F. Y. Lo and P. C. Yao, "Seamless Handoff Mechanisms for Bluetooth-IP Network Services," *Proceedings of the 8th World Multi-Conference on Systemics, Cybernetics and Informatics (SCI 2004)*, Vol. XI, pp.1-5, Orland, Florida, USA, July 2004.