# QUALITY OF SERVICE AWARE DYNAMIC ADMISSION CONTROL IN IEEE 802.16J NON-TRANSPARENT RELAY NETWORKS

A THESIS SUBMITTED TO
THE GRADUATE SCHOOL OF NATURAL AND APPLIED SCIENCES
OF
MIDDLE EAST TECHNICAL UNIVERSITY

BY

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IN PARTIAL FULFILLMENT OF THE REQUIREMENTS
FOR
THE DEGREE OF MASTER OF SCIENCE
IN
COMPUTER ENGINEERING

FEBRUARY 2010

Approval of the thesis:

**QUALITY OF SERVICE AWARE DYNAMIC ADMISSION CONTROL IN IEEE 802.16J NON-TRANSPARENT RELAY NETWORKS**

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# ABSTRACT

QUALITY OF SERVICE AWARE DYNAMIC ADMISSION CONTROL IN IEEE 802.16J NON-TRANSPARENT RELAY NETWORKS

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February 2010, 76 pages

Today, telecommunication is improving rapidly. People are online anywhere anytime. Due to increasing demand in communication, wireless technologies are progressing quickly trying to provide more services in a wide range. In order to address mobility and connectivity requirements of users in wide areas, Worldwide Interoperability for Microwave Access (Wimax) has been introduced as a forth generation telecommunication technology.

Wimax, which is also called Metropolitan Area Network (MAN), is based on IEEE 802.16 standard where a Base Station (BS) provides last mile broadband wireless access to the end users known as Mobile Stations (MS). However, in places where high constructions exist, the signal rate between MS and BS decreases or even the signal can be lost completely due to shadow fading. As a response to this issue, recently an intermediate node specification, namely Relay Station, has been defined in IEEE 802.16j standard for relaying, which provides both throughput enhancement and coverage extension. However, this update has introduced a new problem; call admission control in non-transparent relay networks that support coverage extension.

In this thesis, a Quality of Service (QoS) aware dynamic admission control algorithm for IEEE 802.16j non-transparent relay networks is introduced. Our objectives are admitting more service flows, utilizing the bandwidth, giving individual control to each relay station (RS) on call acceptance and rejection, and finally not affecting ongoing service flow quality in an RS due to the dense population of service flows in other RSs. The simulation results show that the proposed algorithm outperforms the other existing call admission control algorithms. Moreover, this algorithm can be interpreted as pioneer call admission control algorithm in IEEE 802.16j non-transparent networks.

Keywords: Wimax, IEEE 802.16j, Quality of Service (QoS), Call Admission Control, Non-Transparent Relay Station

# ÖZ

TRANSPARAN OLMAYAN IEEE 802.16J RÖLE AĞLARDA SERVİS KALİTESİ FARKINDA DİNAMİK KABUL DENETİMİ

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Şubat 2010, 76 sayfa

Günümüzde, kişiler arasısanal iletişim hızla gelişmektedir. İnsanlar her zaman ve her yerde çevrimiçi olmak isterler. Bu talep karşısında, kablosuz teknolojiler gelişmeye ve daha geniş alanlarda daha çok kişiye servis vermeye başladılar. Bunun sonucunda, geniş bant kablosuz erişim olarak da bilinen mikro dalga erişim için dünyada yaygın birlikte çalışabilirlik (Wimax) teknolojisi dördüncü nesil iletişim geliştirildi.

Kent çapında ağ olarak da adlandırılan Wimax, IEEE 802.16 standardına bağlı olarak oluşturulmuştur. Bu standartta bir baz istasyonu (BI) mobil istansyonlara (MI) kablosuz servis sağlamaktadır. Fakat bazı yüksek yapıların bulunduğu yerleşkelerde, BI ve MI arasındaki radyo sinyalleri gölgelemeden dolayı çok düşmekte ve hatta kaybolmaktadır. Bu yüzden arada taşıyıcı görevi gören yeni bir istasyon IEEE 802.16j standardı kapsamında geliştirilmiştir. Bu röle istasyonları hem kapsama alanını genişletmeye hem de taşınan veri miktarını arttırmaya yarar. Bu iyileştirme sonucu, çağrı kabul denetimi konusu kapsama alanı genişletme özelliğine sahip transparan olmayan röle ağlarda açık olarak kalmıştır.

Bu tezde, servis kalitesi farkında IEEE 802.16j destekli transparan olmayan röle ağlarda dinamik kabul denetimi için bir algoritma öne sürülmüştür. Amacımız daha fazla kullanıcıya servis sağlamak, kullanıcılara sağlanan bant genişliğini verimli hale getirmek, transparan olmayan röle istasyonuna kabul denetimi hakkı vermek ve son olarak, bir röle istasyonuna bağlı mobil istasyonların diğer istasyonlardaki yoğunluktan çok fazla etkilenmemesini sağlamaktır. Similasyon sonuçları da gösteriyor ki önerilen algoritma mevcutta bulunanlardan çok daha iyi sonuç vermektedir. Aynı zamanda bu algoritma transparan olmayan IEEE 802.16j röle ağlarda uygulanan dinamik kabul denetimi algoritmaları arasında öncü niteliği taşımaktadır.

Anahtar Kelimeler: Wimax, IEEE 802.16j, Servis Kalitesi, Çağrı Kabul Denetimi, Transparan Olmayan Röle İstasyonu

*To my family...*

# ACKNOWLEDGMENTS

I would like to express my sincere gratitude and appreciation to Dr. Cevat Şener for his encouragement and support throughout this study.

I would also like to convey thanks to jury members for their valuable comments on this thesis.

I am deeply grateful to my family; Ersin Kılıç, Asuman Kılıç, Emel Saat, Esen Aktürk, Bülent Saat, Asım Aktürk and Melih Saat for their love and support. Without them, this work could not have been completed.

I am also grateful to my dear friend Mustafa Yüksel for his love, continued motivating support and welcomed presence.

I am deeply indebted to my friends, Selcan Yavuz, Engin Ezer, Rıdvan Tekdoğan, Mustafa Gökçe Baydoğan and all colleagues at Siemens Enterprise Communication, whose stimulating suggestions and encouragement helped me in all the time of research for and writing of this thesis.

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4G 4th Generation
AES Advanced Encryption Standard
AMC Adaptive Modulation and Coding
ARQ Automatic Repeat Request
ATM Asynchronous Transfer Mode
BE Best Effort
BPSK Binary Phase-Shift Keying
BS Base Station
BWA Broadband Wireless Access
CDMA Code Division Multiple Access
CID Connection Identifier
MAC CPS MAC Common Part Sublayer
CRC Cyclic Redundancy Check
MAC CS MAC Convergence Sublayer
D/A Digital/Analog
DCD DL channel descriptor
DES Data Encryption Standard
DL Downlink
DL-MAP Downlink Map
DSL Digital Subscriber Line
FCH Frame Control Header
FDD Frequency Division Duplexing
FEC Forward Error Correction
FFT Fast Fourier Transform
FTP File Transfer Protocol
GUI Graphical User Interface
HMAC Hashed Message Authentication Code
IE Information Element
IEEE Institute of Electrical and Electronics Engineers
IMT International Mobile Telecommunications
LAN Local Area Network
LDPC Low-Density Parity-Check
LOS Line of Sight
MAC Media Access Control
MAN Metropolitan Network Area
MIB Management Information Base
MIMO Multiple input Multiple Output
MPEG Moving Picture Experts Group
MR-BS Multihop Relay Base Station
MS Mobile Station
NCTUns National Chiao Tung University Network Simulator
NLOS Non-Line of Sight
nrtPS Non-Real-Time Polling Service
NS2 Network Simulator 2
NT-RS Non-Transparent Relay Station
OFDM Orthogonal Frequency Division Multiplexing
OFDMA Orthogonal Frequency-Division Multiple Access
OTcl Object Oriented Tool Command Language
PDU Protocol Data Unit
PHS Packet Header Suppression
PHY Physical Layer
PKM Privacy Key Management
PMP Point to Multipoint
QAM Quadrature Amplitude Modulation
QOS Quality of Service
QPSK Quadrature Phase-Shift Keying
RNG-REQ Ranging Request
RNG-RSP Ranging Response
RS Relay Station
RSA Rivest Shamir Adleman
rtPS Real-Time Polling Service

SCa Single-Carrier

SDUs Service Data Units

SFID Service Flow Identifier

SNR Signal to Noise Ratio

SOFDMA Scalable OFDMA

SS Subscriber Station

MAC SS MAC Security Sublayer

Std Standard

TCP Transmission Control Protocol

TDD Time Division Duplexing

TLV Type Length Value

T-RS Transparent Relay Station

UCD UL channel descriptor

UDP User Datagram Protocol

UGS Unsolicited Grant Service

UL Uplink

UL PUSC Uplink Partially Used Sub-Carrier

UL-MAP Uplink Map

VoIP Voice over Internet Protocol

WIMAX Worldwide Interoperability for Microwave Access

WMAN Wireless Metropolitan Area Network

# CHAPTER 1

# INTRODUCTION

In the last decades, the telecommunication technology has improved very rapidly. There is no doubt that the demand for mobility has increased in parallel to the development of telecommunication technologies. Mobile communication is a part of our lives now. In such a demanding field, wireless technologies are still being developed continuously. Today, the focus of the literature is more on providing services with better Quality of Service (QoS) in terms of data rate, coverage, robustness, latency, availability and reliability.

Wimax (*Worldwide Interoperability for Microwave Access*) is one of the 4G (4th Generation) telecommunication technologies that supplies wireless communication of data via different transmission links like point-to-multipoint [4]. Wimax, which is also known as WMAN (Wireless Metropolitan Area Network), provides "broadband wireless to the masses" [5]. The IEEE 802.16 Working Group has published 802.16 air interface specification for Broadband Wireless Access (BWA).

BWA is a cellular system where base stations serve connection to mobile, nomadic or fixed subscriber stations in a certain radius of several miles/kilometers. Base stations are located on the top of tall buildings or other high constructions. The signal coming from the subscriber stations are then routed to backbone via backhaul link. The connection between Base Station (BS) and core network could be via standard Ethernet cable either directly to a single computer, or to an 802.11 hot spot or a wired Ethernet LAN.

Initially, IEEE 802.16 air interface provided service in a range of 10 to 66 GHz spectrum with line of sight (LOS) configuration and fixed network support. However, LOS configuration is not convenient for the settlements where high constructions are placed, due to signal loss. Therefore, modified version (IEEE 802.16d) with non-line of sight (NLOS) property is defined in the 2-11 GHz frequency range for fixed networks. The biggest deficiency of this protocol was the mobility. Hence, the most frequently used version, namely IEEE 802.16e which operates in the 2-6 GHz range with mobility support is specified.

Although IEEE 802.16e has NLOS and mobility reinforcement, the signal-to-noise ratio (SNR) may be difficult to achieve in the edge of the BS coverage area, also known as cell. Moreover, one of the aims on introducing the BWA air interface is to fill the coverage gaps and provide service in an area that is as wide as possible. In order to achieve these goals, more BS deployment is necessary. However, due to the exceeding cost of BSs and the difficulty of constructing a backhaul link from BS to the backbone, this is not easily achievable. Therefore, the new amendment IEEE 802.16j is introduced for relaying to enhance the coverage extension and throughput. Relay Stations (RSs) are deployed more easily and economically. Moreover, this intermediate relay node increases the signal rate between a Base Station and Mobile Station (MS) which can be decayed by shadow fading, as shown in 1.1. There are two types of RSs that are developed for different purposes. While transparent relay stations (T-RS) are increasing the throughput, non-transparent relay stations (NT-RS) extend the coverage range of the cell. As a result, IEEE 802.16j relay networks became a popular investment area for investors and a popular research area for researchers.

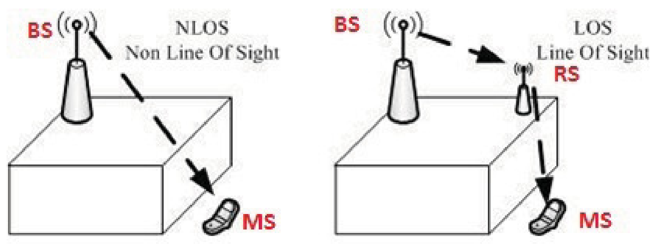


Figure 1.1: NLOS and LOS transmission

Increasing the SNR ratio is not enough to increase number of the service flows to which base station serves. The relay and base stations have limited resources to be allocated. Therefore, resource sharing among mobile users and the admission of the received requests are also significant issues to be considered. Admission control allows a new service flow coming from an MS to create an end-to-end uplink connection with BS. Once the connection request reaches to the BS, it decides to admit or reject the request and the amount of bandwidth that should be reserved for MS's entire transmission duration. This is known as call admission control mechanism in IEEE 802.16 specification. MS's service flow request should be one of five types of service classes, which are called Quality of Service (QoS) classes. These service classes can be listed as Unsolicited Grant Service (UGS), Real-Time Polling Service (rtPS), Non-Real-Time Polling Service (nrtPS), Extended-Real-Time Polling Service (ertPS) and Best Effort (BE). The acceptance and rejection of a call is determined according to the QoS requirements of the service flow request.

Like other IEEE 802.16 specifications, resource allocation and call admission control is not addressed in IEEE 802.16j relay air interface protocol. In our work, a QoS aware dynamic admission control mechanism in non-transparent relay IEEE 802.16j networks is proposed to achieve the following goals:

- Increasing the admitted calls in a cell. In this way, service availability, which is the one of the significant requirements of a network, is pursued.
- Increasing the bandwidth utilization for each service flow.
- Giving each NT-RS individual the right to accept or reject the received request and allowing each of them to execute its own admission control mechanism. Hereby, minimum damage is made on ongoing calls in other RSs when a new call request arrives to an NT-RS and sustainability of available connections in other NT-RSs is provided.

The main contribution of this study is the introduction of dynamic admission control in IEEE 802.16j non-transparent relay networks. Furthermore, number of admitted calls are analysed for UGS, rtPS and nrtPS QoS class connections. rtPS service flow admission is increased and service availability is enhanced. Besides, allocated resources for existing connections are maximized for nrtPS calls. Eventually, the effect of the density on an NT-RS to other NT-RSs' admission is minimized.

Three types of QoS class service flows which are UGS, rtPS and nrtPS are studied. In order to achieve our goals, the proposed call admission control algorithm was implemented in NCTUns tool which is developed by the members of Network and System Laboratory [6] in National Chiao Tung University, in Taiwan. rtPS and nrtPS service classes were introduced which are not supported in the latest version of ncTUNS. Moreover, round robin scheduling was implemented for these classes to run our algoritm on this simulator. Same network topologies, where density different service flows on different relay stations varies, are used. All these simulations were compared with the simulations where the admission control algorithm proposed by Wang et. al. [7] was applied. The results show that there is an increase in the number of admitted rtPS calls in our algorithm. Moreover, the density of admitted calls connected to an NT-RS has less effect to the reserved bandwidth for ongoing service flows in other NT-RSs compared to the algorithm by Wang et. al.. Eventually, the fluctuation in reserved bandwidth for existing nrtPS is less in our algorithm.

The rest of this thesis is organized as follows. Section 2 summarizes the enabling technologies and standards of IEEE 802.16. The related work on admission control is explained briefly in Section 3. Section 4 consists of the proposed QoS aware dynamic admission control algorithm in IEEE 802.16j supported non-transparent relay network. The simulation tool, implementation details, simulation results and observations are presented in Section 5. Finally, the thesis is concluded and the future works to be done are mentioned in Section 6.

# CHAPTER 2

# BACKGROUND ON ENABLING TECHNOLOGIES AND STANDARDS OF IEEE 802.16

There are two different working groups which play role in the development of Wimax. One of them is the IEEE 802.16 Working Group [8] on Broadband Wireless Access Standards who develops standards and recommended practices to support the development and deployment of broadband Wireless Metropolitan Area Networks. IEEE 802.16 is a unit of the IEEE 802 LAN/MAN Standards Committee.

The second one is Wimax Forum [9] which is a non-profit and industry-focused organization to certify and promote broadband wireless products based on IEEE 802.16 standard. Wimax Forum speeds up to locate the Wimax products which are completely interoperable into the market.

Wimax Forum has a Certified Program which is for bringing the BWA products and solutions to a certain level. In this manner, interoperability and standardization for all vendors are provided. Thus, the officers and the board of directors of the Wimax Forum which are mainly Alcatel-Lucent, Intel Corporation, Fujitsu, Nokia, Motorola, ZTE Corporation, and Samsung are responsible for leading the Broadband Wireless Access (BWA) market adoption of IEEE 802.16-based BWA systems through promotional activity, certification and interoperability testing. Wimax forum has 18 board members, over 100 principal members and over 300 regular members.

In order to establish the standardization, several working groups were constructed. They are listed as follows:

- *Technical Steering Committee*: Ensure the consistency of Working Group activities. Control technical planning, specification, and certification are compatible with the Wimax Forum specifications. Specify roadmaps and decisions by the Wimax Forum Membership.

- *Applications Working Group*: Illustrate the best practice applications and the achievement of the Wimax applications with competitive technologies.

- *Certification Working Group*: Manage the Wimax Forum Certification Program and ensure the BWA products and solutions are under ethical and technical standards for vendors.

- *Global Roaming Working Group*: Guarantee the roaming service for Wimax technology.

- *Marketing Working Group*: Concentrate on the adoption of the Wimax based product to marketplace anytime and anywhere.

- *Network Working Group*: Specify the end-to-end networking standards for fixed, nomadic, portable and mobile Wimax systems within the scope of IEEE 802.16 protocols. Thus, interoperability and compatibility are supplied for Wimax products and solutions.

- *Regulatory Working Group*: The central authority on spectrum and regulatory matters.

- *Service Provider Working Group*: Deal with the coordination of service providers with other working groups and the Wimax Forum Board.

- *Technical Working Group*: Enhance technical product specifications and certification test suites for the air interface based on OFDMA (Orthogonal frequency-division multiple access) PHY. The main aim is to ensure interoperability and certification of mobile stations, subscriber stations and base stations based upon IEEE 802.16 standards.

The remaining of this section consists of general background information about IEEE 802.16 technology.

## 2.1 Overview of Wireless Metropolitan Area Network

This wireless broadband access standard supplies the missing link for the “last mile” connection in wireless metropolitan area networks. The purpose of Wimax is to promote deployment of broadband wireless access networks by using a global standard and certifying interoperability of products and technologies. Due to the growing demand on various fields of the BWA, several standards have been published by IEEE 802.16 working group that satisfies the user requirements [10]. These are listed as follows:

- IEEE 802.16a [11]: The standard specifies the operation from 2GHz to 11GHz, both licensed and unlicensed exempts. Because the signals at lower frequency can penetrate barriers and thus a line-of-sight connection between the transceiver and receiver is not required, most commercial interests have focused mainly on the lower frequency ranges. Under this premise, IEEE 802.16a standard was thus completed in January 2001. It enables the Wimax implementations with better flexibility while maintaining the data rate and transmission range. IEEE 802.16a also supports mesh deployment, which can extend the network coverage and increase the overall throughput.

- IEEE 802.16c [12]: As the Work Group’s initial interest, IEEE 802.16c defines a 10 to 66 GHz system profile that standardizes more details of the technology. These high frequency bands have more available bandwidth, but the signals cannot diffract the obstacles and require line-of-sight deployment.

- IEEE 802.16d [13]: Approved in June 2004, IEEE 802.16d upgrades the 802.16a standard. This extension aims to improve performance for 802.16, especially in the uplink traffic.

- IEEE 802.16e [14]: This technology standardizes networking between fixed base stations (BSs) and mobile base stations (MSs), rather than just between base stations and fixed recipients. IEEE 802.16e enables the high-speed signal handoffs necessary for communications with users moving in vehicles. It promises to support mobility up to speeds of 70-80mi/h. The subscriber stations (SSs) could be personal communication devices such as mobile phones and laptops.

- IEEE 802.16f [15]: In this standard, Management Information Base (MIB) for IEEE 802.16d protocol is defined. IEEE 802.16f standard was accepted in 2005.

- IEEE 802.16g [16]: This standard defines management plane procedures as enhancements to the IEEE 802.16 air interface standard for fixed and mobile broadband wireless systems. It specifies the management functions, interfaces and protocol procedures.
- IEEE 802.16h [17]: This amendment specifies improved mechanisms, as policies and medium access control enhancements, to enable coexistence among license-exempt systems based on IEEE Standard 802.16 and to facilitate the coexistence of such systems with primary users.
- IEEE 802.16i [18]: This standard is introduced to enhance the Management Information Base (MIB) for mobile networks.
- IEEE 802.16j [19]: This standard specifies OFDMA physical layer and medium access control layer enhancements to IEEE Std. 802.16 for licensed bands to enable the operation of relay stations.
- IEEE 802.16k [20]: This standard amends IEEE Std 802.16d, as previously amended by IEEE Std 802.17aTM-2004, to support the bridging of the IEEE 802.16 medium access control.
- IEEE 802.16m [21]: This standard amends the IEEE 802.16 WMAN-OFDMA specification to provide an advanced air interface for operation in licensed bands. It meets the cellular layer requirements of IMT-Advanced next generation mobile networks. This amendment provides continuing support for legacy WMAN-OFDMA equipment.
- IEEE 802.20 [22]: This standard is for Mobile broadband wireless access. Initially formed as a standards group within the 802.16 Working Group, it consisted of a group of individuals who wished to develop a new technology focused solely on mobility. There is no other relation to Wimax, except for competition perhaps.

IEEE 802.16 task group provides specifications for media access service layer (MAC) and physical (PHY) layer as depicted in Figure 2.1. The MAC layer is divided into three fundamental sublayers: the convergence sublayer (CS), the common part sublayer (CPS), and the security sublayer (SS). The task of CS is to transform and map the data from upper layers into appropriate MAC service data units (SDUs) for MAC CPS. This means that classification of external data with the proper MAC service flow identifier (SFID) and connection identifier (CID) in MAC CPS is the basic of MAC CS layer. The MAC CPS has the core functionality of the 802.16 MAC specifications. The provided functionalities of MAC CPS are duplexing, network entry and initialization, framing, quality of service (QoS), channel access, bandwidth allocation, admission control, connection establishment, connection maintenance and packet scheduling. The security sublayer is responsible for authentication, secure key exchange and encryption. The next operational layer of 802.16 standard, physical layer (PHY), is based on orthogonal frequency division multiplexing (OFDM). OFDM is an elegant and efficient scheme for high data rate transmission in a non-line-of-sight or multipath radio environment. Besides this, various adaptive modulation and coding (AMC) schemes, forward error correction (FEC) and automatic repeat request (ARQ) features are defined. In the following sections, more information is available about the infrastructure of MAC and PHY layers in 802.16 protocol.

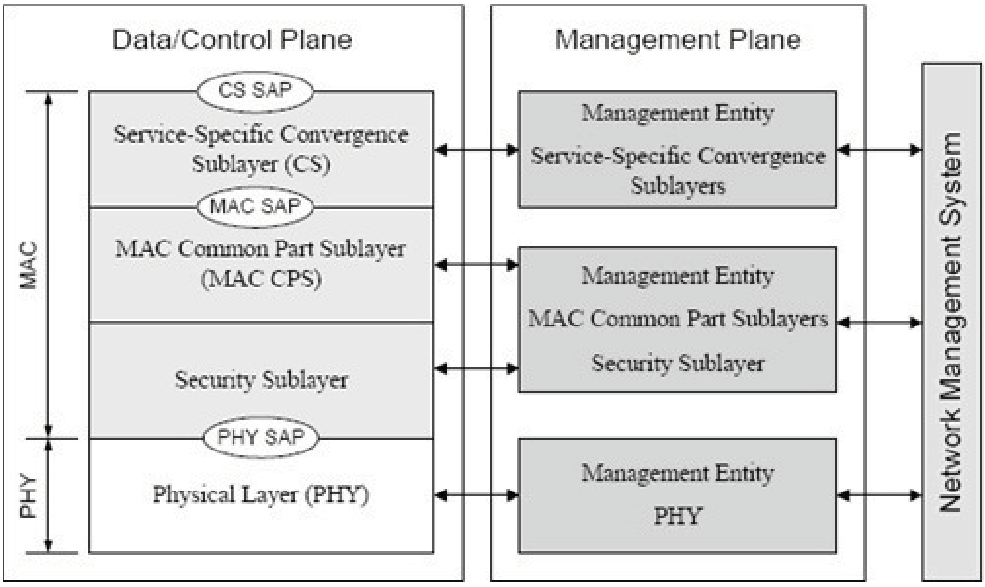


Figure 2.1: The logical architecture of IEEE 802.16

The networking of 802.16 in this thesis is point to multipoint (PMP) mode. Each 802.16 coverage area consists of one base station (BS) and one or more subscriber stations (SSs). BS is connected to a core network which is also defined as the Internet backbone. The backbone connectivity is generally through wired ethernet IEEE 802.3. The PMP network topology is illustrated in Figure 2.2.

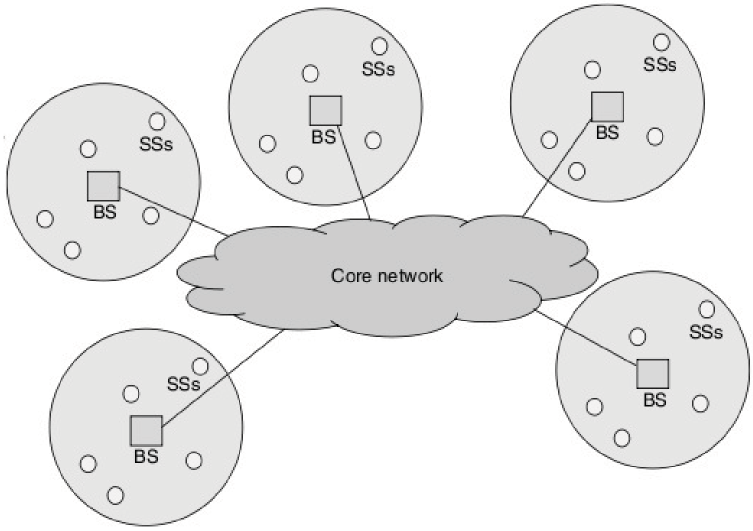


Figure 2.2: PMP Topology in Wimax Network

## 2.2 PHY Layer in IEEE 802.16

Initially, the frequency band of 802.16 protocol was identified from 10 Ghz to 66 Ghz in the scope of 802.16a-2001. Because of the higher frequency, Line-of-Sight (LOS) propagation is necessary. For the settlements, subscriber stations may not have clear line of sight to base stations. Thus, frequency band has been addressed as 2-11 GHz for NLOS operations. The PHY layer modulation schemes defined in IEEE 802.16 family are listed as [23]:

- *WMAN SCa*, A single-carrier modulated air interface for frequencies beyond 11GHz requiring a LOS condition for point-to-multipoint operations. This PHY layer is part of the original 802.16 specifications.

- *WMAN OFDM*, a 256-carrier FFT-based OFDM PHY layer for point-to-multipoint operations in NLOS conditions at frequencies between 2GHz and 11GHz. This PHY layer, finalized in the IEEE 802.16-2004 specifications, has been introduced for fixed operations and is often referred to as fixed Wimax.

- *WMAN OFDMA*, a 2,048-carrier FFT-based OFDMA PHY for point-to-multipoint operations in NLOS conditions at frequencies between 2GHz and 11GHz. This PHY layer is used in the IEEE 802.16e-2005 specifications. Variable sizes of FFT, i.e. 128, 512, 1,024, and 2,048, have been introduced and OFDMA has been modified to Scalable OFDMA (SOFDMA). The variable FFT size yield optimum operation and implementation of the system over a wide range of channel bandwidths and radio conditions. This PHY layer has been accepted by Wimax for mobile and portable operations and is also referred to as mobile Wimax.

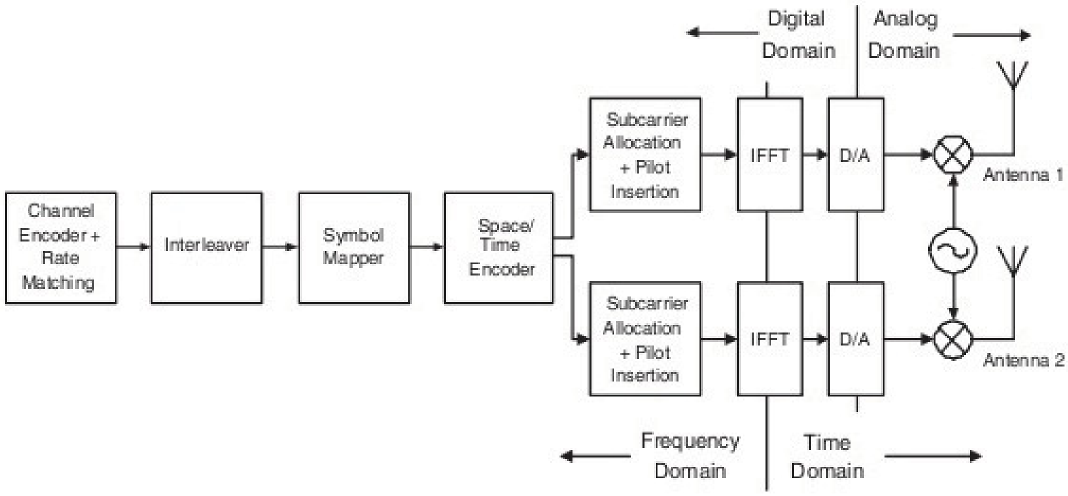


Figure 2.3: Functional Stages of 802.16 PHY

In this section, OFDMA and OFDM basics will be mentioned as modulation scheme. Figure 2.3 illustrates the functional stages of 802.16 specification. First set of task in PHY layer includes forward error correction (FEC), channel encoding, rate matching (puncturing or repeating), interleaving, and symbol mapping. The next stage is construction of the OFDM symbol in the frequency domain. In this stage, data is mapped onto the convenient subchannels and subcarriers. Space-time encoding for transmit diversity or MIMO functionalities can also be implemented in this stage. The final set of functions is responsible for the conversion of the OFDM symbol from the frequency domain to the time domain and eventually the signal is exchanged to digital to analog and transmitted over the air interface. Figure 2.3 is based on the transmitter. The receiver is in the reverse order and has reverse functionalities. In the next sections, the operation sequence is explained according to the transmitter.

### 2.2.1 The First Functional Stage in IEEE 802.16 PHY Interface

Figure 2.4 demonstrates the operations and their dependencies to each other in the first stage of PHY.

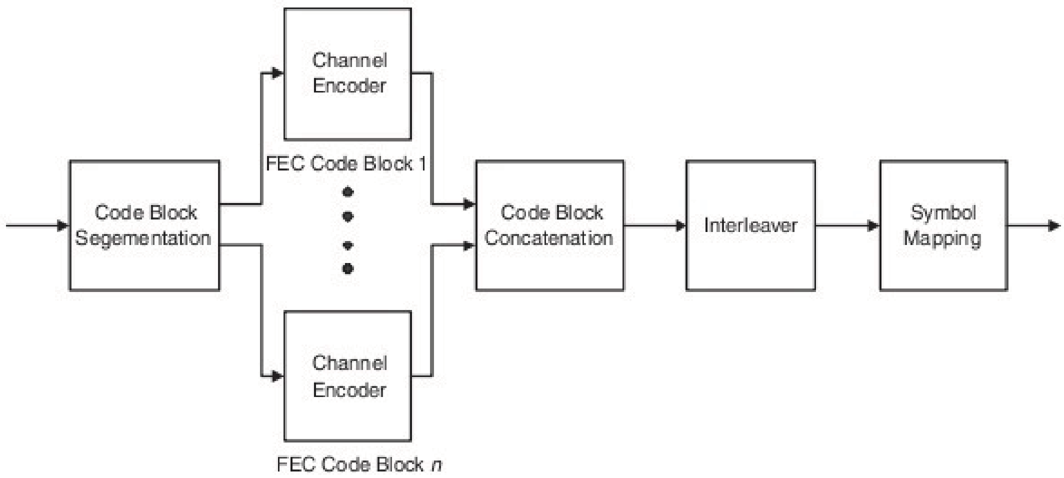


Figure 2.4: Code Block Segmentation (1st Stage of PHY Layer)

#### 2.2.1.1 Channel Coding

In the channel coding, following steps are performed in IEEE 802.16 standard respectively:

- *Data randomization*: Randomization is to avoid long sequences of consecutive ones or consecutive zeros in data bursts. It is also useful for preventing to transmit non-centered data sequences and provide layer 1 encryption. Data randomization is applied to each downlink and uplink data bursts. If the amount of data ready to transmit does not correspond exactly the amount of allocated data, 0xFF ("ones" only) is added to the end of the transmission block [24].

- *Forward error correction (FEC)*: FEC is a mechanism for error control where transmitter adds irrelevant data, also called as error-correction code, to the original data blocks. Receiver can detect and correct the error occurred during transmission without additional data [25].

- *Rate matching*: The FEC code block is matched to any data rate with respect to Mbps according to its modulation. Interleaving: Interleaving is used to protect the transmission against long sequences of consecutive errors, which are very difficult to correct. The encoded data bits are interleaved by a block inter-leaver with a block size corresponding to the number of coded bits per allocated subchannels per OFDM symbol [13].

- *ARQ/HARQ*: Repetition was added by the 16e amendment for OFDMA PHY. Automatic repeat request (ARQ) (or automatic repeat-query) is an error-control method for data transmission which uses acknowledgements and timeouts to achieve reliable data transmission over an unreliable service. An acknowledgement is a message sent by the receiver to the transmitter to indicate that it has correctly received a data frame or packet. A timeout is a reasonable point in time after the sender sends the frame/packet; if the sender does not receive an acknowledgement before the timeout, it usually re-transmits the frame/packet until it receives an acknowledgement or exceeds a predefined number of re-transmissions [26].

#### 2.2.1.2 Symbol Mapping

During the symbol mapping stage, the sequence of binary bits is converted to a sequence of complex valued symbols. The symbol conversion and complexity differ with respect to the adaptive a modulation code (AMC) type [27] [28]. AMC is used to refer the matching of the modulation, coding, other signal and protocol parameters to the conditions on the radio link. The radio link is affected from pathloss, interference with any other signal sent by other transmitters and sensitivity of the receiver. Due to these conditions, Modulation is changed according to the received signal rate and the symbol mapping and coding is performed according to the determined modulation by base station. The modulation and coding types supported in IEEE 802.16 specification are listed in Table 2.1.

### 2.2.2 The Second Functional Stage in IEEE 802.16 PHY Interface

OFDMA symbol construction is the main task of the second functional stage of IEEE 802.16 PHY Interface.

Table 2.1: Modulation and Coding Types Supported in IEEE 802.16 [1]

| | **Downlink** | **Uplink** |
|---|---|---|
| **Modulation** | BPSK, QPSK, 16 QAM, 64 QAM; BPSK optional for OFDMA-PHY | BPSK, QPSK, 16 QAM; 64 QAM optional |
| **Coding** | Mandatory: convolutional codes at rate 1/2, 2/3, 3/4, 5/6<br>Optional: convolutional turbo codes at rate 1/2, 2/3, 3/4, 5/6; repetition codes at rate 1/2, 1/3, 1/6, LDPC, RS-Codes for OFDM-PHY | Mandatory: convolutional codes at rate 1/2, 2/3, 3/4, 5/6<br>Optional: convolutional turbo codes at rate 1/2, 2/3, 3/4, 5/6; repetition codes at rate 1/2, 1/3, 1/6, LDPC |

#### 2.2.2.1 OFDMA Symbol Construction

OFDM is a very powerful transmission technique which is based on the principle of transmitting simultaneously many narrow-band orthogonal frequencies, often also called OFDM subcarriers [29]. A large number of closely-spaced OFDM subcarriers are used to carry data as illustrated in Figure 2.5. Several data and subcarrier information are located in subchannel which is the basic unit of resource allocation in PHY layer. In this stage, high data rate sequence of symbols is split into multiple parallel low data rate sequences, each of which is used to modulate an orthogonal subcarrier according to the modulation types. The mapping between modulation type and allocated OFDM symbol is illustrated in Table 2.2.

IEEE 802.16d (fixed service) uses Orthogonal Frequency Division Multiplexing (OFDM). IEEE 802.16e (mobile) uses Orthogonal Frequency Division Multiple Access (OFDMA). The difference between OFDM and OFDMA is that OFDMA is a multi-user OFDM that allows multiple access on the same channel which is a group of subcarriers. OFDMA distributes subcarriers among users so all users can transmit and receive at the same time within a subchannel [30], Figure 2.6. In Figure 2.6, each different dashed cell indicates a different user [31].

Table 2.2: Modulation and coding

| Data Rate (Mbps) | Modulation scheme | Coding rate | Coded Bits per sub carrier | Code bits per OFDM symbol | Data bits per OFDM symbol |
|---|---|---|---|---|---|
| 6 | BPSK | 1/2 | 1 | 48 | 24 |
| 9 | BPSK | 3/4 | 1 | 48 | 36 |
| 12 | QPSK | 1/2 | 2 | 96 | 48 |
| 18 | QPSK | 3/4 | 2 | 96 | 72 |
| 24 | 16-QAM | 1/2 | 4 | 192 | 96 |
| 36 | 16-QAM | 3/4 | 4 | 192 | 144 |
| 48 | 64-QAM | 2/3 | 6 | 288 | 192 |
| 54 | 64-QAM | 3/4 | 6 | 288 | 216 |

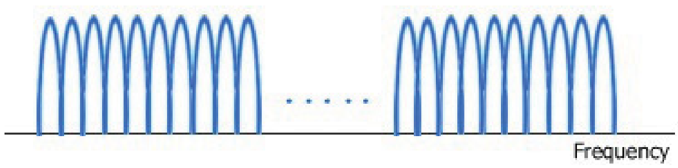


Figure 2.5: Orthogonal Division Multiplexing Frequency illustration

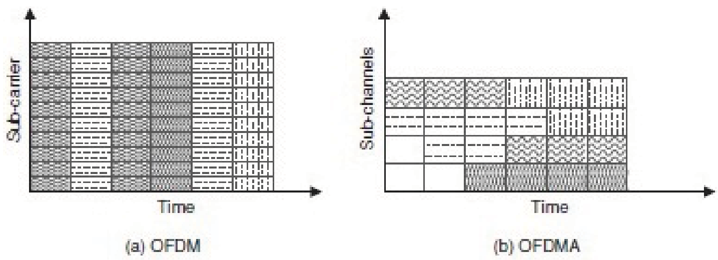


Figure 2.6: OFDM and OFDMA channel allocation

### 2.2.3 The Third Functional Stage in IEEE 802.16 PHY Interface

In this stage, constructed OFDM symbols are transformed digital signals with Fast Fourier Transform (FFT). These digital signals are converted into analog signals which can be transfered by RF transmitter by D/A converter.

## 2.3 MAC Layer in IEEE 802.16

As previously defined, there are three main functional part of MAC layer of IEEE 802.16. They are Convergence Sublayer, Common Part Sublayer and Security Sublayer.

### 2.3.1 Convergence Sublayer (CS)

CS is the most upper sublayer of Wimax MAC. The CS is responsible for accepting higher-layer PDUs from the higher layers and transmits them to the MAC CPS where classical type MAC procedures are applied. Each received SDU is mapped to a connection identifier (CID), which indicates a logical connection between BS and SS, based on destination address, the quality of service (QoS) requirements, source destination and service flow identifier (SFID). Then, the repetitive part of the payload headers is eliminated by the Packet Header Suppression (PHS) mechanism. At the end, the MAC SDU is sent to the lower sublayer, CS. Unlike the other two sublayers, different CSs are defined in the MAC layer for different network layer protocols. ATM CS for ATM connection and Packet CS for ethernet connection are supported by IEEE 802.16 protocol, currently.

### 2.3.2 Common Part Sublayer (CPS)

The main tasks of MAC CPS are duplexing, network entry and initialization, framing, quality of service (QoS), channel access, bandwidth allocation, admission control, connection establishment and maintenance and packet scheduling. The details of the functionalities are explained in the following sections:

- *Mac PDU construction and transmission*: In CPS sublayer, incoming SDUs from CS are restructured as MAC PDUs which are the basic payload unit handled by MAC and PHY layer. Multiple SDUs can be unified in a single MAC PDU, also referred as or a single SDU can be separated into multiple MAC PDUs. The operation differs according to the length of SDU. Fragmentation and packing of SDUs [32] are illustrated in Figure 2.7. Each MAC PDU consists of a header followed by a payload and a cyclic redundancy check (CRC). IEEE 802.16 protocol supports two types of PDUs which have different header contents. One is the generic MAC PDU for carrying data and MAC-layer signaling messages. This PDU has generic packet header followed a payload and CRC. The other is the bandwidth request PDU used by SS to request more bandwith from BS for Uplink (UL). This type of PDU does not contain payload or CRC. Only bandwidth request header is available.

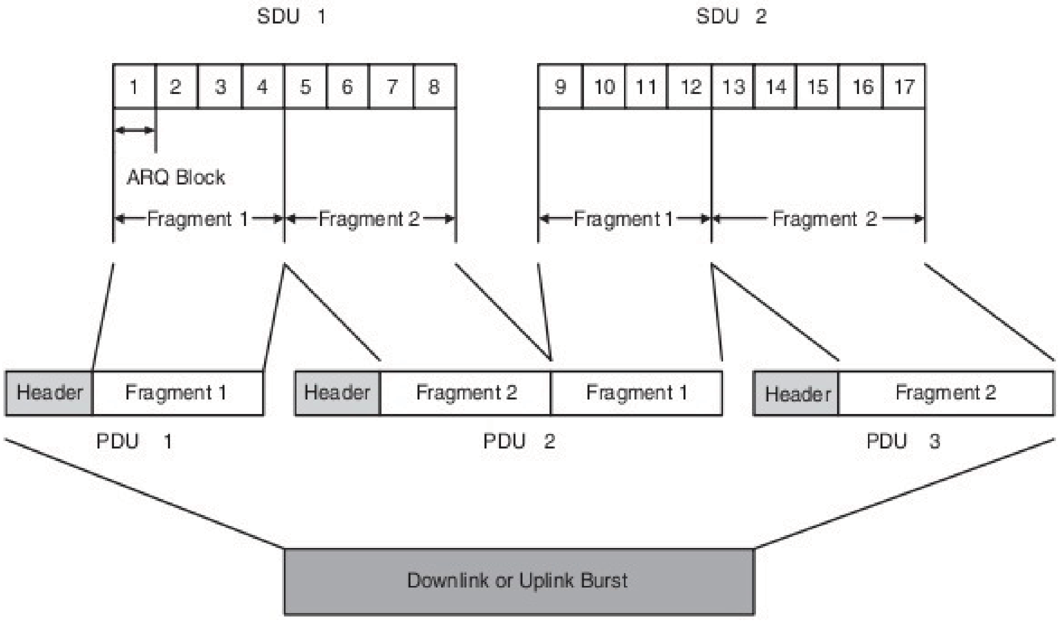


Figure 2.7: Constuction of MAC PDUs from SDUs

After MAC PDU construction, these packets are sent to the scheduler which is responsible for scheduling MAC PDU over the available PHY resources. The scheduler checks the SFID and CID of PDU to measure the QoS requirements of the packet. Then, it determines the optimum PHY resource to be allocated for all PDUs based on predefined QoS requirements. The scheduling procedure can be implemented differently according to the equipment manufacturers. Therefore, each product may have different performance and capacity and scheduling algorithm becomes distinguishing feature among manufacturers.

- *Quality of Service (QOS)*: The key functionality of MAC layer of IEEE 802.16 is to provide quality of service (QoS) constraint for MAC PDUs [33]. This means that the latency, jitter, data rate, packet error rate and system availability should be met for all service flow. Based to these QoS requirement, PDUs are scheduled and PHY resources are utilized efficiently. Due to the different data service presence, five distinct scheduling services in Table 2.3 are introduced as follows:

  - *Unsolicited grant service (UGS)*: This scheduling service is defined for real-time service flows which generate fixed-size data packets on a periodic basis, like VoIP and T1/E1 [34]. UGS does not need SS to send bandwidth request.
  - *Real-time polling service (rtPS)*: This type is designed for real-time service flows which generate variable-size data packets on a periodic basis, like MPEG video [35]. The BS provides unicast polling opportunities for the SS to request bandwidth.
  - *Non-real-time polling service (nrtPS)*: This is similar with rtPS, but BS provides contention-based polling in the uplink for SS to request bandwidth [36]. FTP is a nice example of this service type.
  - *Best-effort service (BE)*: Best Effort services do not have QoS constraint which means no guarantee to deliver data. Data is sent when resource is available [37]. The SS uses only the contention-based polling opportunity for bandwidth request. Web Browsing is a BE service which is supported by Wimax.
  - *Extended real-time polling service (ertPS)*: This scheduling service is a sort of combination of UGS and rtPS. It supports real-time applications, such as VoIP with silence suppression [38], that have variable data rates but require guaranteed data rate and delay. It is introduced in IEEE 802.16e.

- *Bandwidth allocation and call admission control*: Bandwidth allocation is responsible for allocating scarce radio resource, bandwidth, to ongoing and incoming connections condidering their QoS requirements [39]. In the downlink, all decisions related to the allocation of bandwidth for MSs are made by the BS on a per CID basis. When MAC PDUs arrive for each CID, the BS schedules them for the physical resources, according to their QoS requirements. If reserved physical resources have been allocated for the transmission of data, the BS indicates this allocation to the MS via DL-MAP message. In the uplink, the MS requests resources by either using a stand-alone bandwidth-request MAC PDU or piggybacking bandwidth requests on a generic MAC PDU. In both cases, grant-management subheader is used. All resource requests are made in terms of bytes of information, rather than physical layer resources, such as number of OFDM symbols and number of subchannels. Because, the burst profile associated with a CID can change dynamically.

Table 2.3: Quality of Service Classes and Requirements

| Service Class | Key Qos Parameters | Applications |
|---|---|---|
| **Unsolicited Grant Service (UGS)** | Maximum Sustained Traffic Rate<br>Maximum Latency<br>Jitter Tolerance | VoIP |
| **Real-time Pooling Service (rtPS)** | Minimum Reserved Traffic Rate<br>Maximum Sustained Traffic Rate<br>Maximum Latency | Streaming Audio or Video |
| **Extended Real-time Pooling Service (ertPS)** | Minimum Reserved Traffic Rate<br>Maximum Sustained Traffic Rate<br>Maximum Latency<br>Jitter Tolerance | VoIP with Silent Suppression |
| **Non Real-time Pooling Service (nrtPS)** | Minimum Reserved Traffic Rate<br>Maximum Sustained Traffic Rate | FTP (File Transfer) |
| **Best Effort (BE)** | Maximum Sustained Traffic Rate | Web Browsing |

Call admission control is applied to avoid overwhelming limited radio resource due to accepting too many connections [39]. There is not any call admission control algorithm in the specification. However, the proposed algorithm in the literature is explained in related work section.

- *Network entry and Initialization [40]*: When SS is powered on, it starts to scan the available downlink frequency channel within the coverage of the present Wimax network. SS listens the downlink frame preambles. If it detects an appropriate preamble, SS begins to synchronize itself with the BS where preamble is sent by sending management messages, like DCD, UCD, DL-MAP, UL-MAP, FCH. According to the UL parameters in management messages, SS determines if the channel is appropriate for its purpose. If it decides the channel is convenient, SS listens UL-MAP messages to extract the information to initiate ranging. Initial ranging is to gather the timing and power-level adjustments to maintain UL connection within BS. In periodic basis, the ranging process is repeated to fix the UL connection which can be distorted because of mobility, fast fading, shadow fading etc. This is also referred as periodic ranging. After initial ranging, capabilities, which are the supported modulation level, coding scheme and rates and duplexing methods are exchanged between BS and SS. Then, the registration of SS to the network is performed and ip connectivity is supplied. At the end, service flow establishment is initiated by SS or BS according to the service flow type and aim.

### 2.3.3 Security Sublayer (MAC SS)

Security sublayer provides authentication, secure key exchange, encryption and exchange of encryption keys between the BS and the SS to prevent the system from all known security attacks. The main functionalities are data encryption and authentication.

In the IEEE 802.16 standard, encrypting connections between the SS and the BS is made with a data encryption protocol applied on MAC PDUs for both ways. The encryption algorithms included in IEEE 802.16 are RSA (Rivest Shamir Adleman), DES (Data Encryption Standard), AES (Advanced Encryption Standard), HMAC (Hashed Message Authentication Code).

An authentication protocol, the Privacy Key Management (PKM) protocol is used to provide the secure distribution of keying data from the BS to the SS. Encryption keys are exchanged between BS and SS in MAC management messages during network initialization phase. Through this secure key exchange, due to the key management protocol the SS and the BS synchronize keying data. The basic privacy mechanisms are strengthened by adding digital-certificate-based SS authentication to the key management protocol. In addition, the BS uses the PKM protocol to guarantee conditional access to network services.

## 2.4 MAC Support of PHY

The MAC layer allocates PHY resources in terms of time or frequency for all users in units of slots which are the smallest amount of PHY resource allocated for a single user in time or frequency domain [41]. As indicated before, slot allocation is based on either time or frequency. Therefore, IEEE 802.16 specification supports both time division duplexing (TDD) and frequency division duplexing (FDD). TDD is duplexing scheme that requires only one channel for transmitting downlink and uplink sub-frames at two distinct time slots. FDD requires two distinct channels for transmitting downlink sub-frame and uplink sub-frame at the same time slot. Figure 2.8 illustrates the frame structure of TDD mode which is mostly used in duplexing mode in Wimax.

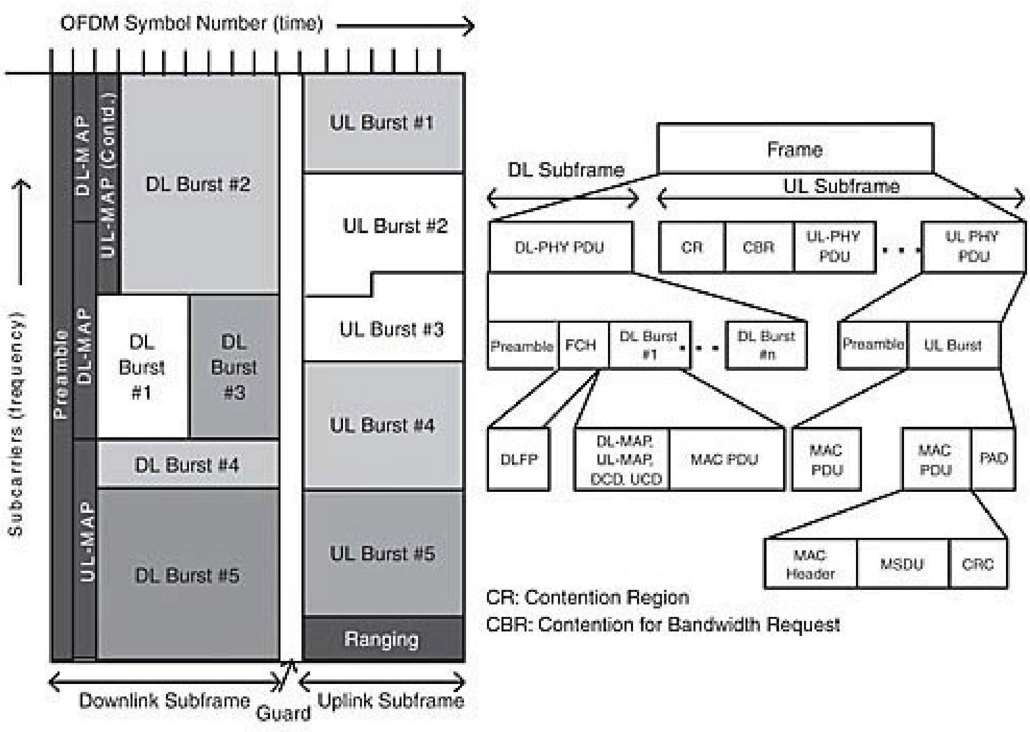


Figure 2.8: TDD Frame structure

## 2.5 IEEE 802.16j Specification

IEEE 802.16j protocol is the amendment to enhance coverage, throughput and system capacity of 802.16 networks by introducing new 802.16 multihop relay capabilities and functionalities of interoperable relay stations and base stations. Therefore, OFDMA PHY and MAC layer enhancements to IEEE 802.16 for license bands are specified in the scope of this protocol for relaying operations. Subscriber station specifications are not modified.

### 2.5.1 PHY Modifications

Former IEEE 802.16 specifications supports only single hop which performed between SS and BS in PMP networks as illustrated in Figure 2.2. However, in relaying mode (IEEE 802.16j) demonstrated in Figure 2.9, multi-hop functionality is defined and PHY layer specifications are changed according to the multi-hop operations. When BS transmits data, it uses downlink connection. In case of data transmission from SS, uplink connection is active. But, relay station (RS) should receive and transmit in both uplink and downlink. Therefore, the frame structure of IEEE 802.16j is separated into two zone, access zone and relay zone.

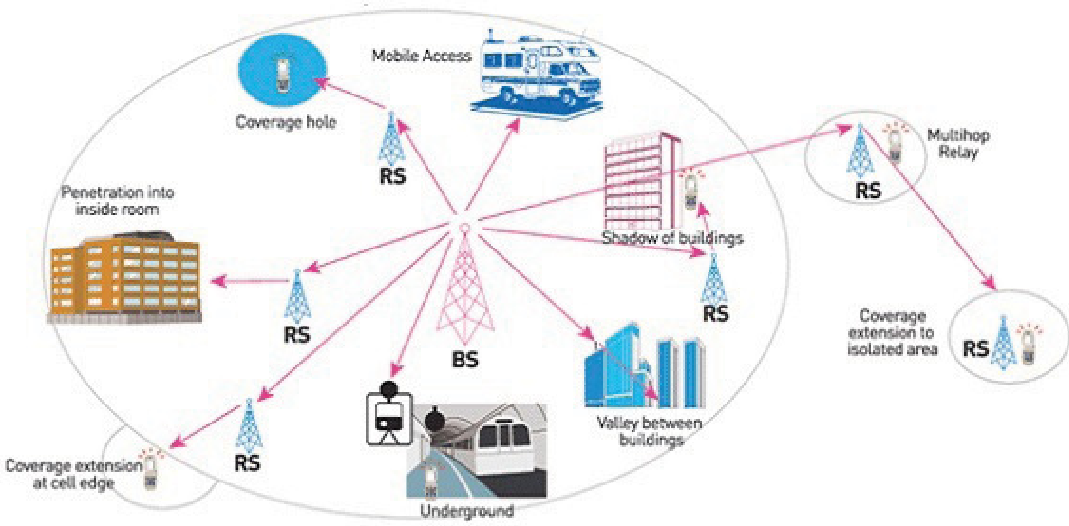


Figure 2.9: Sample Network Topology of MMR Network [3]

There are also two different relay frame structures for two different relay mode: Transparent mode and Non-Transparent mode.

- *Transparent Mode*: This relay mode is introduced to increase the throughput which means that it facilitates capacity increases within the BS coverage area. It has no support for coverage extension because it does not forward framing information to BS. This type of relay mode has lower complexity and only operational in two-hop network topology with centralized scheduling. The frame structure of transparent mode is shown in Figure 2.10.

- *Non-Transparent Mode*: The main purpose to define non-transparent relay mode is to increase the coverage extension of BS by generating RS's own framing information or forwarding the framing information from BS to SS. Due to the high interference between signals from different RSs, the throughput enhancement is limited. Non-transparent relays can operate in the network topologies which have more that two-hop with centralized or distributed scheduling. Therefore, the complexity of this type of

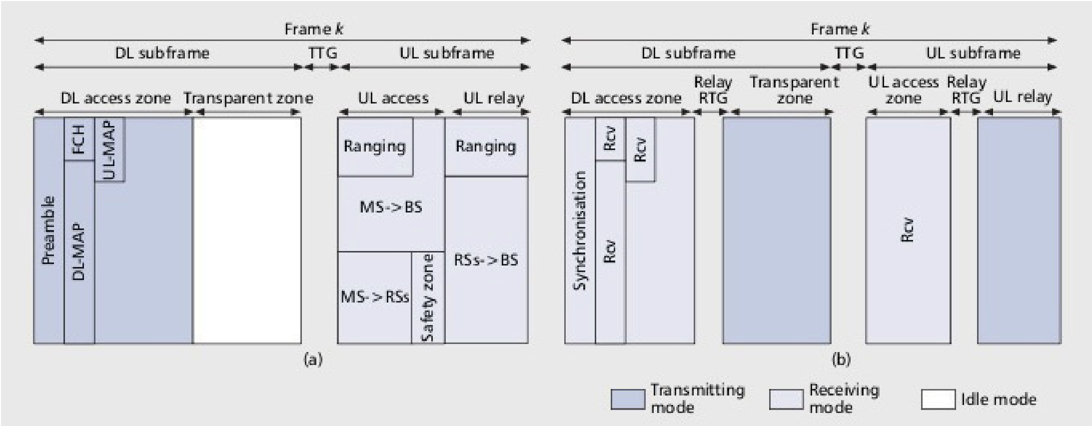


Figure 2.10: Transparent mode frame structure: a) the frame structure as viewed at the BS b) the frame structure as viewed at the RS [2]

RS is higher than the transparent RS. The frame structure also differs from the one in transparent mode, as shown in Figure 2.11.

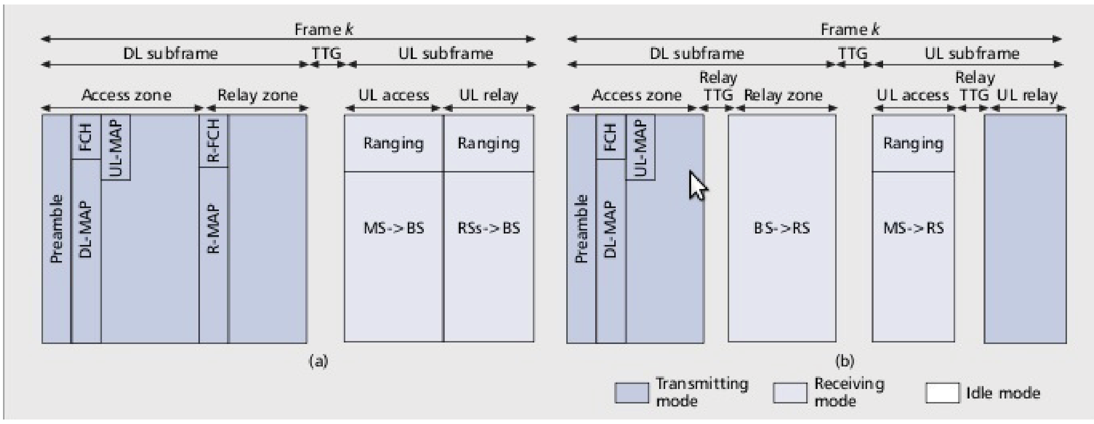


Figure 2.11: Non-transparent mode frame structure (two-hop case): a) the frame structure as viewed at the BS b) the frame structure as viewed at the RS [2]

In Table 2.4, the difference between transparent relay mode and non-transparent relay mode is illustrated.

### 2.5.2 MAC Modifications

There are three branches in MAC layer modification for multi-hop relay networks. These are forwarding scheme, routing and path selection, initial ranging and network entry.

Table 2.4: Difference between TP and Non-TP Relay modes [2]

| | **Transparent RS** | **Non-transparent RS** |
|---|---|---|
| **Scheduling** | Centralized | Centralized/Distributed |
| **Number of Hops** | 2 | 2 or more |
| **Coverage Extension** | No | Yes |
| **RS Cost** | Low | High |
| **Performance** | In BS coverage: High<br>Out of the BS coverage : None | In BS coverage: Normal (No improvement)<br>Out of the BS coverage : Medium |
| **Inter RS cells interference** | None | High |
| **Hops** | 2 | 2 or more |
| **Forwarding Scheme** | CID-based | CID-based/Tunnel-based |
| **Channels** | 1 | 1 or 2 |
| **Path Management** | Embedded/Explicit mode | Embedded/Explicit mode |
| **RRM** | Less Complex | More Complex |

- *Forwarding schemes*: Two different schemes which are tunnel-based scheme and CID-based scheme is available in the current specification to enhance the system efficiency. The tunnel-based scheme provides support for explicit tunnels characterized by a unique CID, two specific endpoints, and quality of service (QoS) requirements .In the tunnel-based approach tunnels are used to aggregate traffic from disparate MSs on the BS-RS connection for either management or transport connections with similar QoS requirements. The CID-based scheme has no such tunnels and does not explicitly support traffic aggregation, but requires less complexity. The CID-based scheme, on the other hand, supports only the legacy management and transport connections as defined in the 802.16e-2005 standard.

- *Routing and path selection*: Since this specification is defined for multi-hop networks, routing and path management become a crucial issue to be handled. Generally, tree based routing is used in this system where same metrics, such as radio resource availability, radio link quality, and traffic load at the RSs , provided by specification is used. The standard specifies two path management approaches, embedded and explicit. Other path selection algorithms are left to the vendors.

- *Initial Ranging and Network Entry*: In IEEE 802.16j, initial ranging and network entry was modified for SS and new procedure is introduced for RS. They are listed as follows:
    - MS initial ranging in transparent mode relay
    - MS initial ranging in non-transparent mode relay
    - RS initial ranging
    - RS network entry

# CHAPTER 3

# RELATED WORK

IEEE 802.16 Task Group specified the bandwidth request management message and the bandwidth request PDUs. These are also explained in the section where protocol overview is expressed. However, bandwidth allocation and admission control algorithms applied to the Wimax products are left to the vendors. In order to help manufacturers to develop optimum bandwidth allocation algorithms, the task group also standardizes some measurements and defines Quality of Service (QoS) classes. Table 3.1 summarizes the QoS classes and their requirements in IEEE 802.16 protocol.

Table 3.1: Quality of Service Classes and Their Properties

| **Service Class** | **Key QoS Parameters** | **Applications** |
|---|---|---|
| **Unsolicited Grant Service (UGS)** | Maximum Sustained Traffic Rate<br>Maximum Latency<br>Jitter Tolerance | VoI P |
| **Real-time Pooling Service (rtPS)** | Minimum Reserved Traffic Rate<br>Maximum Sustained Traffic Rate<br>Maximum Latency | Streaming Audio or Video |
| **Extended Real-time Pooling Service (ertPS)** | Minimum Reserved Traffic Rate<br>Maximum Sustained Traffic Rate<br>Maximum Latency<br>Jitter Tolerance | VoIP with Silent Suppression |
| **Non Real-time Pooling Service (nrtPS)** | Minimum Reserved Traffic Rate<br>Maximum Sustained Traffic Rate | FTP (File Transfer) |
| **Best Effort (BE)** | Maximum Sustained Traffic Rate | Web Browsing |

Due to the absence of the specification on bandwidth utilization and admission control, various algorithms were introduced to increase the performance of their own Wimax supported products. In this way, vendors provide the customers, in both last mile and backhaul connections, with service availability and service connectivity which are the main concerns in a network.

In the research performed by Guo et. al. in [42], a dynamic bandwidth allocation and admission control scheme is proposed for IEEE 802.16e system. The proposed scheme is based on reserving resource as admission guard bandwidth, which is dynamically adjusted according to the bandwidth request of on-going rtPS connections in addition to guaranteeing the resource requirement of potential handoff services. The first problem to be resolved is data loss in rtPS traffic. In order to prevent data loss, some amount of bandwidth in the range of [0, Maximum Sustained Traffic Rate - Minimum Reserved Traffic Rate] is added to its minimum the reserved traffic rate, and the total bandwidth is reserved for variable-bit-rate connections. For the handoff calls which constitute the second problem, some amount of bandwidth is reserved. However, the reserved bandwidth lowers the utilization of the system. Therefore, the maximum one is selected and is assigned as reserved bandwidth. When a new call arrives, the proposed algorithm checks whether the sum of its minimal required resource (Breserved) and the overall reserved bandwidth (R) calculated before is less than the available resource; if this is the case, then the call is admitted, else it is rejected. For handoff calls, only Breserved is compared to the available bandwidth, and the call is admitted if the value is less than available one. Breserved is different for all service types. It is maximum sustained traffic rate for UGS, minimum reserved traffic rate for rtPS, nrtPS and ertPS, and zero for BE. Therefore, rtPS, nrtPS and ertPS service flows cannot utilize the overall bandwidth, even if there is available bandwidth for maximum sustained rate. In our algorithm, variable bit rate service flows, especially rtPS and nrtPS, can utilize the available bandwidth as much as they can. Besides, this scheme is specified for only IEEE 802.16e protocol and has no relay support. But, this study is performed in Wimax relay networks.

A similar dynamic admission control mechanism based on dynamic guard channel is proposed by Chaudhry and Guha [43]. The bandwidth is reserved for new calls with respect to a threshold which is adjusted dynamically based on admission of the arriving handoffs and termination of existing handoffs. The benefit of this algorithm is to utilize the reserved bandwidth for new connections and prevent new calls to be rejected in a system where handoffs are higher priority. However, minimum reserved rate is used for nrtPS calls in the algorithm of Chaudhry and Guha [43]. Moreover, this algorithm is applied in network which supports

IEEE 802.16e protocol. However, the bandwidth is utilized for nrtPS connections in IEEE 802.16j.

There is also one basic approach on call admission control proposed by Wongthavarawat and Ganz [44], Jiang and Tsai [45]. In these algorithms, the service calls are admitted according to the available bandwidth in the system. Available bandwidth for a new call is calculated by structuring the reserved bandwidth for all ongoing calls from the total radio resource. No adaptation is applied on the allocated bandwidth for ongoing calls. However, our algorithm modifies the reserved resources for existing calls by satisfying QoS requirements. Like other admission controls algorithms, these were not applied in IEEE 802.16j supported networks.

The algorithm proposed by Wang et. al. [7] is one of the QoS aware call admission control algorithm for IEEE 802.16e supported network. The reserved bandwidth for ongoing service flows is dynamically adjusted according to the incoming call QoS requirements. The adaption is performed by reducing and degrading the reserved bandwidth of the nrtPS connections step by step, if there is no more resource for new call. In this way, the required bandwidth for new call is constituted by preventing the existing nrtPS calls minimum requirements. This degradation is applied only on nrtPS calls. However, rtPS QoS class service flows has variable length bit rate property like nrtPS. In case of rtPS service flow demand increase, call dropping probability of rtPS calls rise, because there is no degradation procedure applied on ongoing rtPS calls. Hence, the rtPS call degradation in our algorithm was carried out in Wimax relay networks.

# CHAPTER 4

# PROPOSED QUALITY OF SERVICE AWARE DYNAMIC ADMISSION CONTROL MECHANISM

One of the important features which provide manufacturers of Wimax network adapters a crucial advantage among other manufacturers is the quality of the implemented admission control mechanism in their systems. The important cause why admission control makes a difference for vendors is that there is no certain specification in all versions of IEEE 802.16 protocols. As it is mentioned in Chapter 3, many admission control algorithms are proposed, introduced and implemented to fulfill the requirements in this area. In this chapter, primarily, the interaction between our call admission control mechanism to other functional module of MAC and PHY layers is explained in detail. Then, it is stated in which part of network entry stage our algorithm takes place and is applied. Finally, our algorithm is expressed in detail with pseudocode.

## 4.1 Call Admission Control Mechanism

The purpose of this section is to explain in which stage of network entry [46], call admission control is in progress in non-transparent relay station. Admission control is a QoS procedure which controls and determines how the resources of the system are allocated by the available connections with various requirements. In this study, the slots for the relay link are the main resource for admission control algorithm. Slot is the minimum time frequency resource allocated by Wimax system. A series of slots are assigned to a given user which is called as the user's data region. The scheduling algorithm implemented in Wimax system allocates data regions for different users, according to demand, QoS requirements and channel conditions.

For this work, the tasks performed during admission control are performed by different modules as illustrated in Figure 4.1. The tasks of the used modules in Figure 4.1 can be listed as follows:

*MAC Management Entity:* This is the main module which is responsible for processing all MAC management messages, like ranging request-response, bandwidth request-response, and handover request-response. In our case, the fundamental role of this module in NT-RS is to extract the RNG-REQ message and forward it to the Call Admission Control module. The MAC Management Entity module in MS is used to process the RNG-RSP in the scope of this work.

*Scheduler:* The MAC PDUs are scheduled in this module. The NT-RS should schedule all MAC PDUs in UL and DL connections for all MSs already registered to NT-RS. MS is only responsible for its own PDUs.

*Connection Clasifier:* The task of this module is to classify connection identifiers (CIDs) for Service Data Units (SDUs) from upper layer.

*Call Admission Control:* This is the main module where our algorithm is implemented. New incoming requests are decided to be accepted or rejected in this module. The result is sent to the MAC Management Entity and then the information is forwarded to the Ranging module to complete the request.

*DL/UL Map Generator:* It is responsible for generating DL and UL Maps.

*Burst Preperation/Resource Allocation:* This module has a close connection with Scheduler. The MAC PDUs are fragmented, concatenated based on the send and receive activity. At the end, data burst are constructed for PHY layer.

In this thesis, admission control is performed during initial ranging in MAC CPS layer. When MS sends ranging request to NT-RS, it is processed by the Management Entity of NT-RS. After the message classification, the incoming RNG-REQ management message is forwarded to the Call Admission Control module. The call is accepted or rejected based on slots available for relay link by applying the proposed QoS aware dynamic admission control algorithm, which will be explained in more detail in the following sections, in this region of MAC CPS. If the call is accepted, the RNG-REQ is relayed by ranging module to MR-BS. In case of a rejection, the failure ranging response is sent to MS which demands a connection from NT-RS.

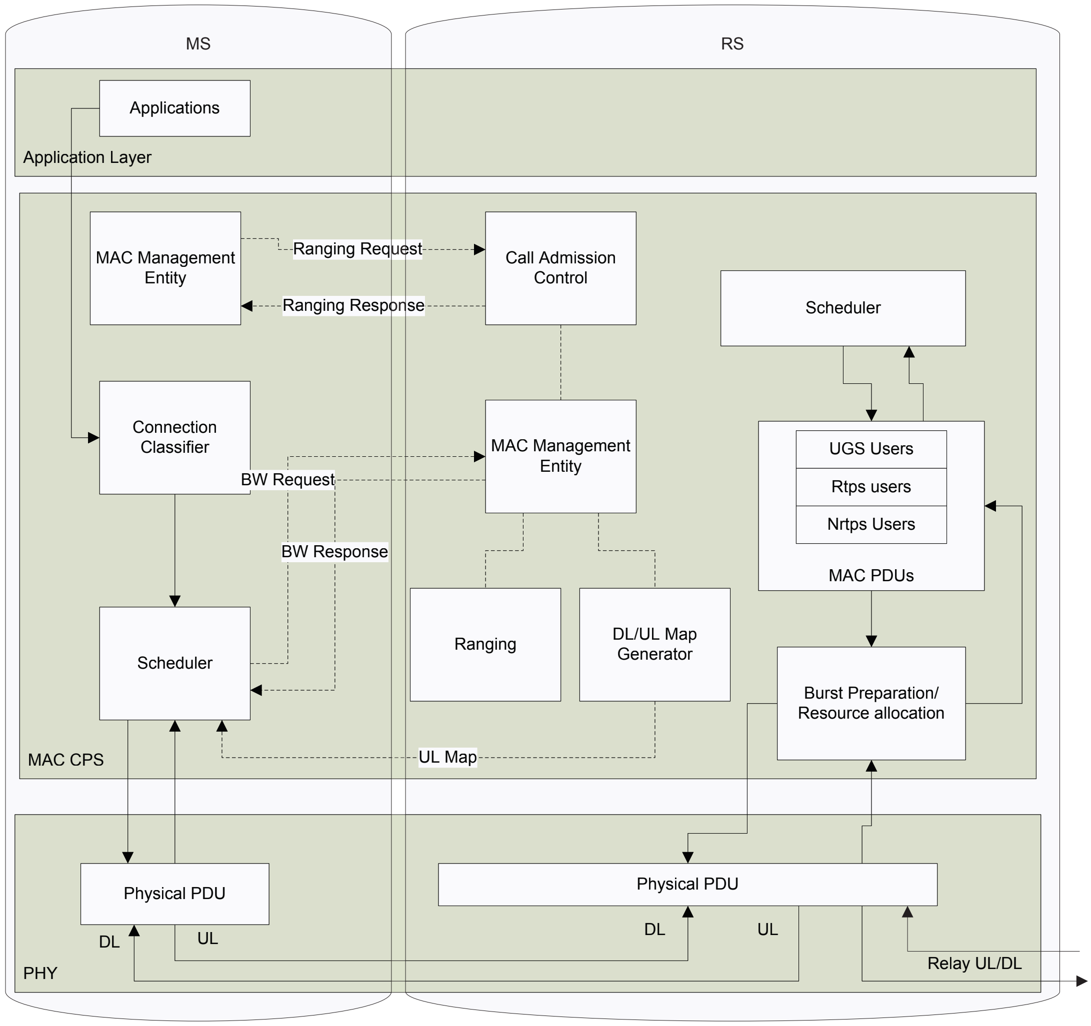


Figure 4.1: Call Admission System Architecture Modules

## 4.2 Initial Ranging of MS to RS

As it is mentioned in the previous section, connection admission control is performed during initial ranging of MS. In the current IEEE 802.16j specification, initial ranging of an MS is performed as follows and demonstrated in Figure 4.2 [19]:

1. RS sends the initial ranging opportunity and UL map containing Initial Ranging IE with

a broadcast Connection ID.

2. MS transmits randomly selected initial ranging code in a randomly selected ranging slot from available ranging region.

3. RS sends RNG-RSP with time and power corrections, original ranging code and ranging slot status as “Continue”.

4. MS receives RNG-RSP message with ranging code and ranging slot matching the sent values. It adjusts time and power parameters.

5. RS sends initial ranging opportunity in RS access link and map containing initial ranging IE with a broadcast CID.

6. MS transmits randomly selected initial ranging code in a randomly selected ranging slot from available ranging region.

7. RS receives ranging code and sets the status as “Success”.

8. If ranging status is success in RNG-RSP, RS sends RNG-RSP containing success status with ranging CID.

9. MS receives RNG-RSP message with ranging code and ranging slot matching the sent values.

10. RS sends UL-MAP containing CDMA Allocation IE.

11. MS transmits RNG-REQ containing MS MAC address and continues with regular initial network entry.

12. If RNG-REQ coming into RS has ranging CID and MS MAC address, then RS relays it to BS with RS Basic CID.

13. BS receives RNG-REQ relayed by RS, identifies MS and its connecting RS. Then, it sends RNG-RSP containing management CIDs with RS Basic CID.

14. RS receives RNG-RSP and relays the RNG-RSP to MS RNG-RSP with ranging CID.

15. MS receives RNG-RSP message with matching MS MAC address.

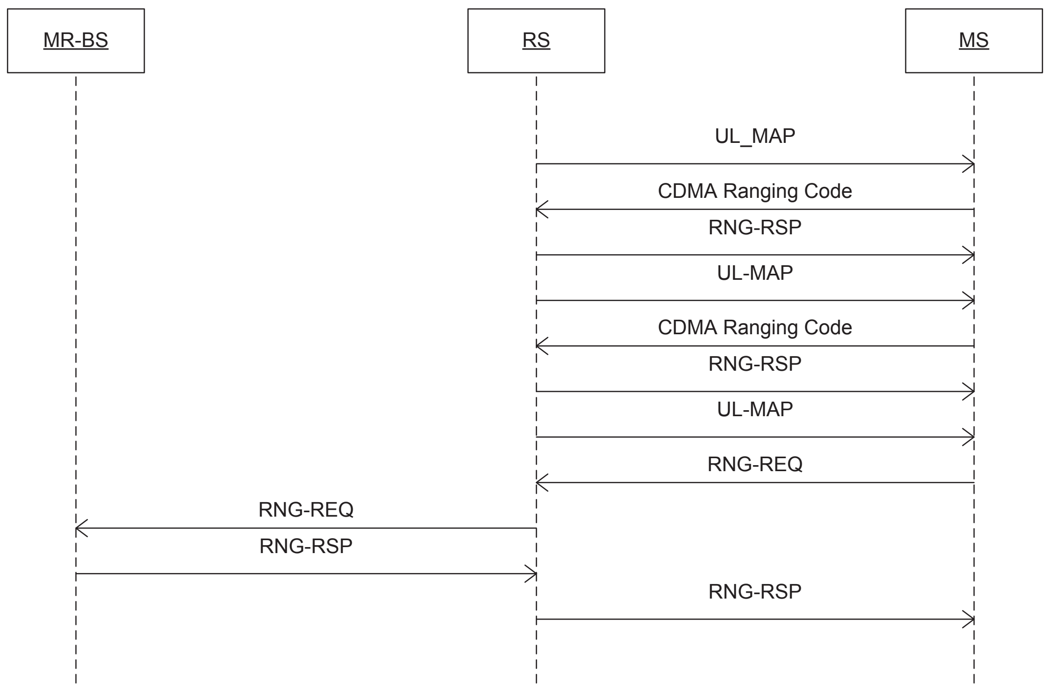


Figure 4.2: Ranging Process in IEEE 802.16j Specification

In our proposed solution for admission control, initial ranging procedure defined in the specification is modified as illustrated in Figure 4.3 and Call Admission Control mechanism is applied just after the last ranging request sent by MS to RS. All the CDMA ranging procedures are remained as they are.

When the last ranging request has arrived with MS MAC address and QoS information of the MS in TLV packet [47] of the RNG-REQ message, the proposed dynamic admission control on RS, which will be explained in detail in the following section, is applied onto the new incoming call in NT-RS MAC CPS layer. According to calculations in NT-RS, the call is either accepted or rejected.

Besides the current RNG-REQ TLV encoding, “Service Class Name”, “Maximum Sustained Traffic Rate” and “Minimum Reserved Traffic Rate” are included in the scope of this thesis.

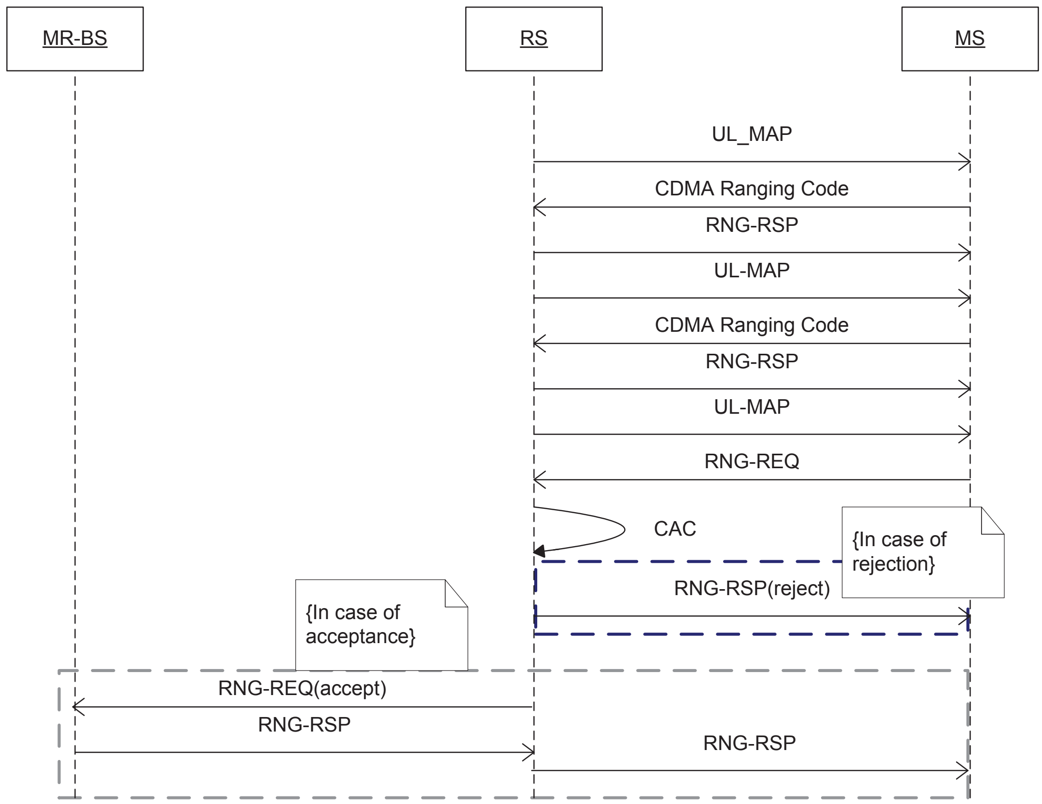


Figure 4.3: Modified Ranging Process

## 4.3 Proposed Dynamic Call Admission Control

In this section the proposed QoS aware dynamic admission control is explained. First, the QoS Class types and mappings examined in this work are clarified.

Three main QoS Classes are assigned to the mobile stations. These are UGS, rtPS and NRTS. BE and rtPS classes are not used in this study. The reason why BE class is not in the scope of this thesis is that it has no bandwidth requirement unlike the other QoS classes. BE service classes only transfer when bandwidth is available. Therefore, they are always accepted by the system. rtPS connection is designed for real-time traffic with variable data rate such as VOIP service with silent suppression. It is introduced to utilize the UGS and rtPS service class properties in one type of service class. Thus, it is included in this work.

In the simulation, the UGS service calls are classified as GOLD, rtPS calls as SILVER and finally nrtPS calls as BRONZE connection. The following table illustrates the QoS classes and their properties used in this work.

Table 4.1: QoS Classes and Properties Used in This Study

| **QOS Classes** | **Properties** |
|---|---|
| UGS (GOLD) | Maximum Sustained Rate |
| rtPS (SILVER) | Maximum Sustained Rate<br>Minimum Reserved Rate<br>Level |
| nrtPS (BRONZE) | Maximum Sustained Rate<br>Minimum Reserved Rate<br>Level |

The following declarations used in the pseudo code of the algoritm are explained to clarify the algoritm and calculations.

### 4.3.1 Assumptions

In this work, it is assumed that each MS has just one type of service flow defined before.

### 4.3.2 Declarations

This section includes some declarations in pseudo code of the algorithm.

| | |
|---|---|
| $NC_{ugs}$ | New UGS call |
| $NC_{rtPS}$ | New rtPS call |
| $NC_{nrtPS}$ | New rtPS call |
| $AS_{sys}$ | All allocated slots in the subnet |
| $AS_{ugs}$ | All allocated slots for UGS calls in the subnet |
| $AS_{rtPS}$ | All allocated slots for rtPS calls in the subnet |
| $AS_{nrtPS}$ | All allocated slots for nrtPS calls in the subnet |
| $AS_{relay}$ | All allocated slots for relay messaging in the subnet |

| | |
|---|---|
| $RS_{av}$ | Available total relay slots |
| $SN_{NCugs}$ | Slots needed for new UGS call |
| $MinSN_{NCrtPS}$ | Minumum slots needed for new rtPS call |
| $MaxSN_{NCrtPS}$ | Maximum slots needed for new rtPS call |
| $MinSN_{NCnrtPS}$ | Minimum slots needed for new nrtPS call |
| $MaxSN_{NCnrtPS}$ | Maximum slots needed for new nrtPS call |
| $R_C$ | Current Relay Station which retrieves RNG-REQ from new call |
| $rtPS_{RC}$ | All admitted rtPS calls in the current Relay station |
| $nrtPS_{RC}$ | All admitted nrtPS calls in the current Relay station |
| $rtPS_{AR}$ | All admitted rtPS calls in the other Relay stations |
| $nrtPS_{AR}$ | All admitted nrtPS calls in the other Relay stations |

There are three use cases where the call admission control mechanism is applied. One of them is the state where new UGS call is attempting to subscribe to the relay station. The other is the new rtPS call admission and the final use case is for the new nrtPS call acceptance to the system. Although the final decision for the new call is “Accept” or “Reject” on RS side, there is a major difference on how it proceeds during admission control.

### 4.3.3 Use Case 1: New UGS Call Admission Control

The UGS calls are interpreted as the highest priority QoS class in most of the Wimax systems. It is also referred as GOLD connection in this study. Therefore, the resources are rearranged to accept the UGS call into the subnet.

In our algorithm in Figure 4.4, the purpose is the same: to admit the new UGS call to the extent that the system resources can permit. After RS gets the ranging request form MS, it extracts the QoS data from TLV information restored in RNG-REQ message. The QoS data for new call is “Maximum Sustained Rate” and “QoS Class Name”. If it is a UGS call, the call admission control algorithm is performed as follows.

```
AS_sys  = AS_ugs + AS_rtPS + AS_nrtPS + AS_relay
if  SN_NCugs + AS_sys <= RS_av
      return ACCEPT NC_ugs
else
      emptySlots = 0;
      while 1
            for i < sizeof(nrtPS_RC)
                  emptySlots += DegradeLevel1(nrtPS_RC[i])
                  if emptySlots >= SN_NCugs
                        return ACCEPT NC_ugs
              endif
            endfor
            if the levels of nrtPS_RC equals 0
                  break
         endif
      endwhile
      while 1
            for i < sizeof(rtPS_RC)
                  emptySlots += DegradeLevel1(rtPS_RC[i])
                  if emptySlots >= SN_NCugs
                        return ACCEPT NC_ugs
               endif
            endfor
            if the levels of rtPS_RC equals 0
                  break
          endif
      endwhile
     while 1
            for i < sizeof(nrtPS_AR)
                  emptySlots += DegradeLevel1(nrtPS_AR[i])
                  if emptySlots >= SN_NCugs
                        return ACCEPT NC_ugs
               endif
            endfor
            if the levels of nrtPS_AR equals 0
                  break
          endif
      endwhile
      while 1
            for i < sizeof(rtPS_AR)
                  emptySlots += DegradeLevel1(rtPS_AR[i])
                  if emptySlots >= SN_NCugs
                        return ACCEPT NC_ugs
               endif
            endfor
            if the levels of rtPS_AR equals 0
                  break
          endif
      endwhile
      return REJECT NC_ugs
```

Figure 4.4: Admission control for new UGS call algorithm

Initially, all allocated slots in the system, which can be also refered as subnet, are calculated. This is the sum of all existing UGS, rtPS, nrtPS connections and the slots reserved for each relay station for passing the management messages between RS and MR-BS or any other RS. Then, the necessary slots for incoming UGS call is calculated by using the code in Figure 4.5.

```
bitsPerFrame = maximum sustained rate / 1000.0 * frame duration code
of NT-RS
if ((bitsPerFrame / 8) % fec == 0)
      SN_NCugs = ((bitsPerFrame / 8) / fec)
else
      SN_NCugs =((bitsPerFrame / 8) / fec) + 1
```

Figure 4.5: Necessary slots for a UGS call

If the sum of the allocated slots in the system and slots required by the new UGS call do not exceed the overall relay slots, the call is accepted and RNG-REQ message is relayed to the MR-BS. The MS is also subscribed to the NT-RS which means the MS is admitted by NT-RS.

```
bitsPerFrame = (maximum sustained rate - minimum reserved rate) *
level / 1000.0 * frame duration code of NT-RS
if ((bitsPerFrame / 8) % fec == 0)
      freedSlots = ((bitsPerFrame / 8) / fec)
else
      freedSlots =((bitsPerFrame / 8) / fec) + 1
```

Figure 4.6: Slots freed after degradation by 1 level

However, if the sum exceeds the overall sytem relay slot capacity, dynamic admission control algorithm takes place and some slots are tried to be freed for the new incoming UGS call. The first step is to borrow some reserved slots for admitted nrtPS call in the system. The reason why the nrtPS call adaptation is performed primarily is that nrtPS is designed to support non real-time service flows that require variable size data grant like FTP applications. Therefore, this type of service flow has lowest priority and is referred as BRONZE connection. Initially, all nrtPS flows in the RS which receives the RNG-REQ from MS are degraded 1 level, one at a time. As it is mentioned previously, nrtPS service flows can maintain the connection and continue to serve in a range of “maximum sustained rate” and “minimum reserved rate”. Therefore, there is minimum reserved rate limit where the reserved slot for nrtPS calls can be decreased. The degradation process is conducted as in Figure 4.6.

In our case, “level” is 0.1, because the aim is to reduce the reserved slot for indicated nrtPS flow by 1 level. The maximum level assigned to the flow is 10 where the number of reserved slots for this flow is equal to “maximum sustained rate” and the minimum is 0 where the number of reserved slots is same as “minimum reserved rate”. After degradation of each nrtPS flow in RS, if the number of freed slots is equal to or greater than the required value of pending call, then the call is accepted and ranging process is successfully completed. If the retrieved empty slots are not enough, reduction continues till all nrtPS service levels reach to 0. Still, if the freed slots are not sufficient, then the second step is activated. When the next step is passed to, the necessary slots do not have to be equal to $SN_{NCugs}$. The freed slots in this step are preserved and the remaining slots migrated to the next step. This rule is valid for all steps.

In the second step, the purpose is to cut down adequate slots for the new UGS call from rtPS flows already admitted in NT-RS which retrieves new incoming request. rtPS flows are real-time variable size services which are classified as SILVER connection. Thus, the slot reduction of rtPS flows is established in the second stage. The rtPS flows are serviceable between the “maximum sustained rate” and “minimum reserved rate” like nrtPS calls. Therefore, the same degradation by 1 level is applied to available rtPS calls. If newly freed slots together with the freed slots inherited from the previous step satisfy the new call requirements, the call is accepted. If the freed slots are still not enough even when all rtPS service levels are set to 0, dynamic admission control algorithm passes to the third step.

In the third step, the nrtPS admitted calls in other Relay Stations are exposed to the reduction of reserved slots procedure. If still the emptied slots are not enough, then the last step is performed; else the call is accepted.

As the final step, rtPS available service flows in other Relay Stations are degraded till the levels of these flows are 0 or there is adequate number of freed slots required for the new UGS call. If the former case comprises, the new incoming UGS service call is rejected. In the latter case, the new call is accepted and usual ranging procedure is carried out. The received ranging request is relayed by NT-RS. The algoritm is explained in Figure 4.7 as a flow chart.

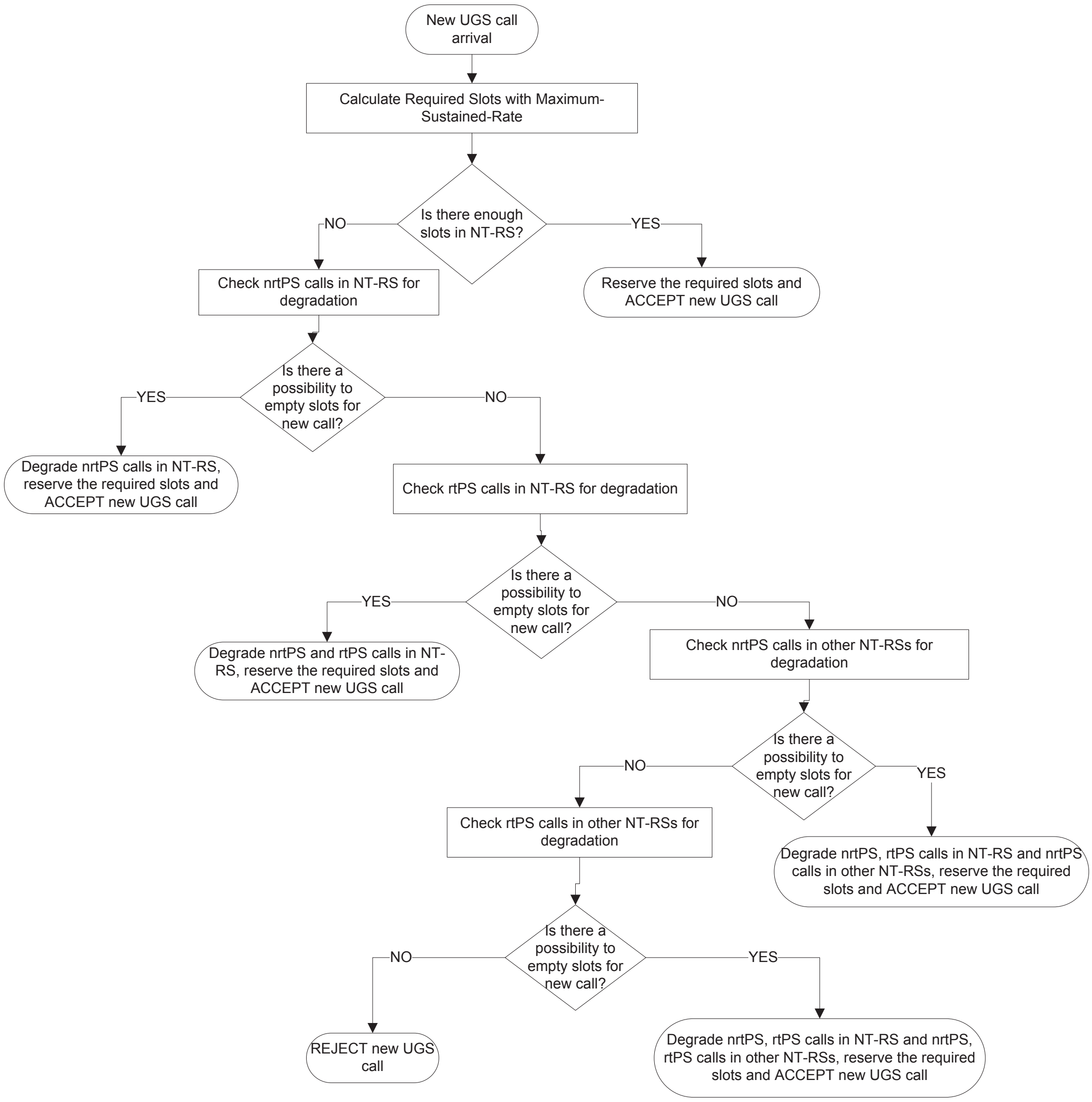


Figure 4.7: Flow chart of UGS call admission contol

```
bitsPerFrame = minimum reserved rate / 1000.0 * frame duration code
of NT-RS
if ((bitsPerFrame / 8) % fec == 0)
      MinSN_NCrtPS = ((bitsPerFrame / 8) / fec)
else
      MinSN_NCrtPS =((bitsPerFrame / 8) / fec) + 1
```

Figure 4.8: Minimum necessary slots for an rtPS call

### 4.3.4 Use Case 2: New rtPS Call Admission Control

The admission of rtPS calls in Figure 4.9 into an NT-RS has similar steps like in UGS call admission control. However, there is one difference on the determination of the sufficient slots for new rtPS call.

Primarily, the "maximum sustained rate" is handled in the calculation of necassary slots for new rtPS service flow, because one of the purposes of this algoritm is to utilize and maximize the slot allocation. However, if the system does not have enough slots, the "minimum reserved rate" takes place in the evaluation of required slots for incoming rtPS call as illustrated in Figure 4.8.

If the number of unused slots in the system is still less than $\mathrm{MinSN}_{NCrtPS}$, the same degradation procedure in UGS call admission control is performed based on $\mathrm{MinSN}_{NCrtPS}$ value. At the end, the acceptance or rejection decision is made by the dynamic admission control algorithm depending on the number of freed slots.

The main reason why $\mathrm{MinSN}_{NCrtPS}$ is used is to keep all available calls with minimum distortion while trying to realize service availability for the new rtPS call. The algoritm is explained in Figure 4.10 as a flow chart.

### 4.3.5 Use Case 3: New nrtPS Call Admission Control

The last QoS class type is nrtPS which has the lowest priory among all call types. The difference of this case from the previous two is the application of degradation on only admitted nrtPS calls in the whole system. This means that nrtPS Call Admission Control has no effect on available UGS and rtPS calls.

Necessary slot calculation procedure for the new nrtPS call is the same with the procedure in rtPS call admission control. Besides, same degradation procedure for accepted nrtPS service flows occurs to free some slots to mee the new nrtPS call requirements. If there are enough empty slots for the newly received nrtPS call, the call is accepted; else it is rejected. The whole pseudocode for nrtPS call admission call is illustrated in Figure 4.11 and explained in Figure 4.12 as flow chart.

```
AS_sys  = AS_ugs + AS_rtPS + AS_nrtPS + AS_relay
if  MaxSN_NCrtPS + AS_sys <= RS_av
      return ACCEPT NC_rtPS
else if  MinSN_NCrtPS + AS_sys <= RS_av
     set level of NC_rtPS to 0
      return ACCEPT NC_rtPS
else
      emptySlots = 0;
      while 1
            for i < sizeof(nrtPS_RC)
                  emptySlots += DegradeLevel1(nrtPS_RC[i])
                  if emptySlots >= MinSN_NCrtPS
                    set level of NC_rtPS to 0
                        return ACCEPT NC_rtPS
               endif
            endfor
            if the levels of nrtPS_RC equals 0
                  break
          endif
      endwhile
      while 1
            for i < sizeof(rtPS_RC)
                  emptySlots += DegradeLevel1(rtPS_RC[i])
                  if emptySlots >= MinSN_NCrtPS
                    set level of NC_rtPS to 0
                        return ACCEPT NC_rtPS
               endif
            endfor
            if the levels of rtPS_RC equals 0
                  break
          endif
      endwhile
     while 1
            for i < sizeof(nrtPS_AR)
                  emptySlots += DegradeLevel1(nrtPS_AR[i])
                  if emptySlots >= MinSN_NCrtPS
                    set level of NC_rtPS to 0
                        return ACCEPT NC_rtPS
               endif
            endfor
            if the levels of nrtPS_AR equals 0
                  break
          endif
      endwhile
      while 1
            for i < sizeof(rtPS_AR)
                  emptySlots += DegradeLevel1(rtPS_AR[i])
                  if emptySlots >= MinSN_NCrtPS
                    set level of NC_rtPS to 0
                        return ACCEPT NC_rtPS
               endif
            endfor
            if the levels of rtPS_AR equals 0
                  break
          endif
      endwhile
      return REJECT NC_rtPS
```

Figure 4.9: Admission control for new rtPS call algorithm

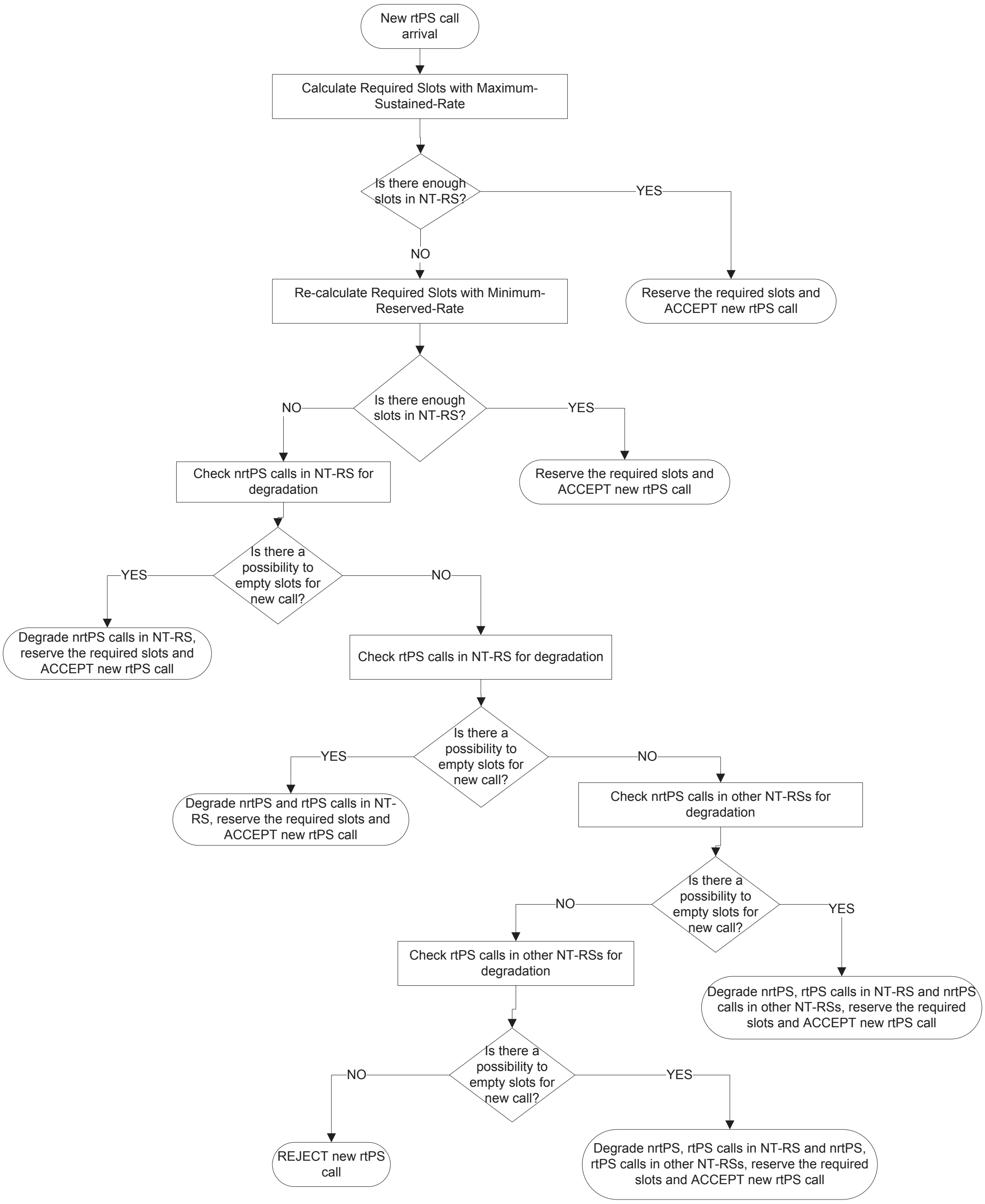


Figure 4.10: Flow chart of rtPS call admission contol

```
AS_sys  = AS_ugs + AS_rtPS + AS_nrtPS + AS_relay
if  MaxSN_NCnrtPS + AS_sys <= RS_av
      return ACCEPT NC_nrtPS
else if  MinSN_NCnrtPS + AS_sys <= RS_av
     set level of NC_nrtPS to 0
      return ACCEPT NC_nrtPS
else
      emptySlots = 0;
      while 1
            for i < sizeof(nrtPS_RC)
                  emptySlots += DegradeLevel1(nrtPS_RC[i])
                  if emptySlots >= MinSN_NCnrtPS
                   set level of NC_nrtPS to 0
                        return ACCEPT NC_nrtPS
               endif
            endfor
            if the levels of nrtPS_RC equals 0
                  break
          endif
      endwhile
     while 1
            for i < sizeof(nrtPS_AR)
                  emptySlots += DegradeLevel1(nrtPS_AR[i])
                  if emptySlots >= MinSN_NCnrtPS
                   set level of NC_nrtPS to 0
                        return ACCEPT NC_nrtPS
               endif
            endfor
            if the levels of nrtPS_AR equals 0
                  break
          endif
      endwhile
      return REJECT NC_rtPS
endif
```

Figure 4.11: Admission control for new nrtPS call algorithm

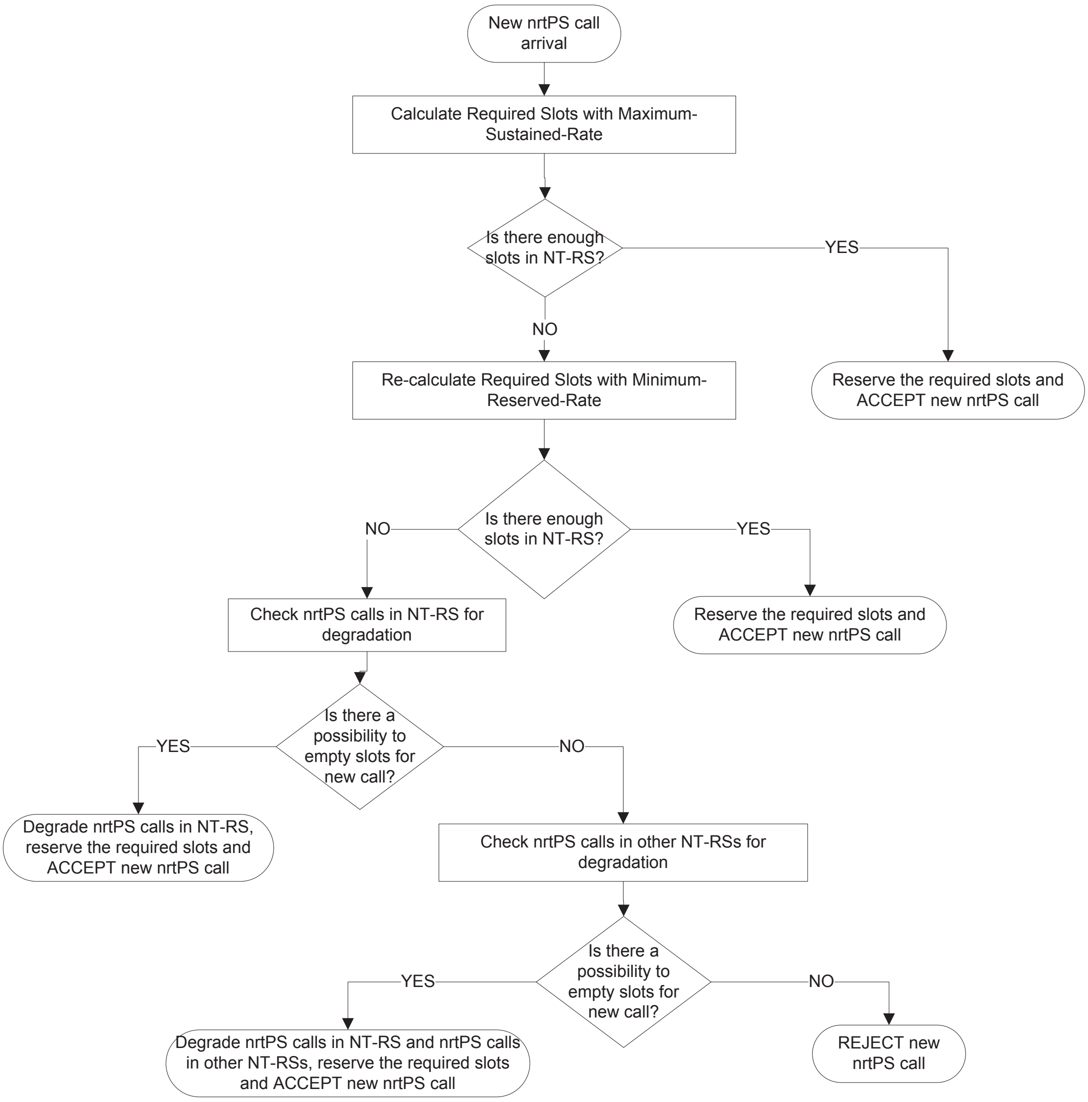


Figure 4.12: Flow chart of nrtPS call admission contol

# CHAPTER 5

# SIMULATION RESULTS AND OBSERVATIONS

This chapter consists of the detailed information about simulation results, network topologies and values used during simulation, observations, implementation and the studied tools for this study.

## 5.1 Studied Simulation Tools

For this thesis, various network simulators were investigated and experimented with. The most frequently used network simulators are Network Simulator 2 (NS2), Opnet, Omnet, Qualnet and NCTUns. Qualnet and Opnet are the commercial ones which supply academic certificate for research purpose.

Especially, Opnet has an advanced graphical user interface which provides users with convenience on creating network topology, assigning values to the network items, creating their own nodes, processes, frame packets and retrieving the analysis of a network simulation. Besides these properties, users can change and edit source code of nodes, processes and states in some restriction for further development as it is illustrated in Figure 5.1. However, it has only IEEE 802.16e support. Therefore, it is not feasible to use Opnet as a simulation tool for this study.

The Network Simulator 2 (NS2) is the most popular open source discrete event simulator target in networking research. NS2 is implemented in C++. It provides a simulation interface through OTcl [48], an object-oriented version of Tcl. The network topology is written by OTcl scripts and then the main NS program binds the previously defined OTcl objects dynamically and simulates the topology with specified parameters. The simulation output can be given to

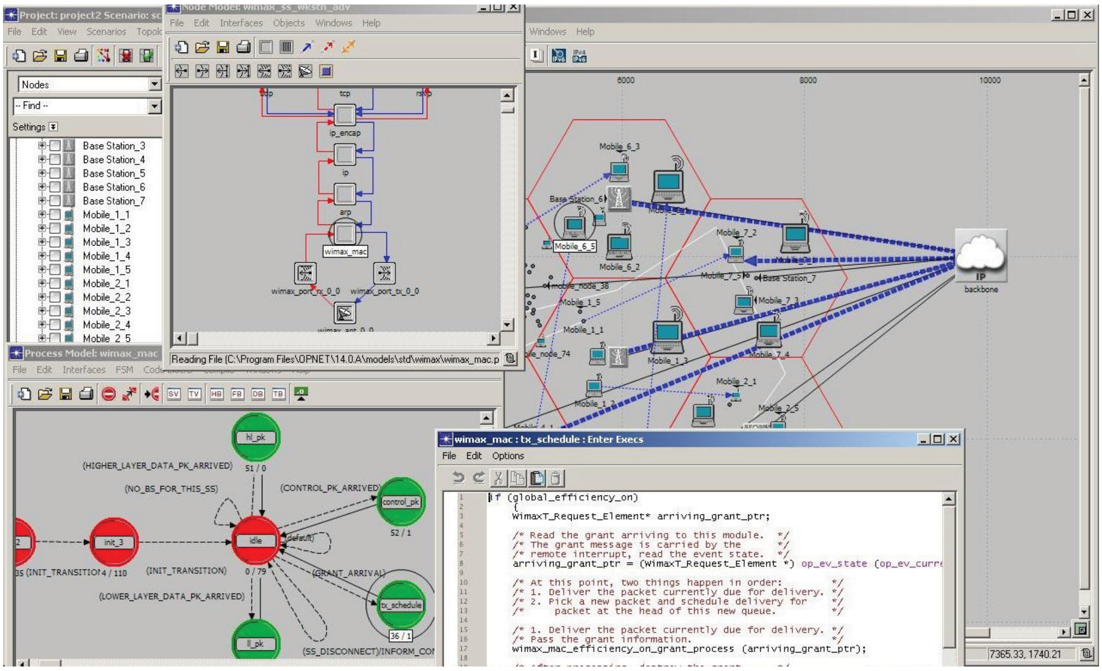


Figure 5.1: Opnet Modeler

the NS/Nam, which is a tool where NS2 simulation results are interpreted and graphically displayed. A display for a basic wireless by Nam is demonstrated in Figure 5.2. Wimax module [49] which can be integrated into NS2 was developed by members of the Computer Networks Laboratory (LRC), Institute of Computing, UNICAMP. But, there is no IEEE 802.16j support of this module either. Therefore, NS2 was not used as the simulation tool.

The last studied simulator was NCTUns which is a discrete event simulator like NS2 [50]. It was developed by the members of Network and System Laboratory in National Chiao Tung University, in Taiwan. The source codes are written in C++ and the network topology is defined in Tcl scripts. It uses the real-life TCP/IP (or UDP/IP) protocol stack in the Linux kernel to perform simulations and emulations and can run up any real-life application program on simulated nodes during simulation to generate realistic network traffic. These two features make the NCTUns a powerful network simulator. It has also IEEE 802.16j support [51] which is very critical for our study. Hence, the proposed QoS aware dynamic admission control mechanism is implemented, simulated and tested on NCTUns platform. In the following section, more information will be given about the features and the architecture of this simulation tool.

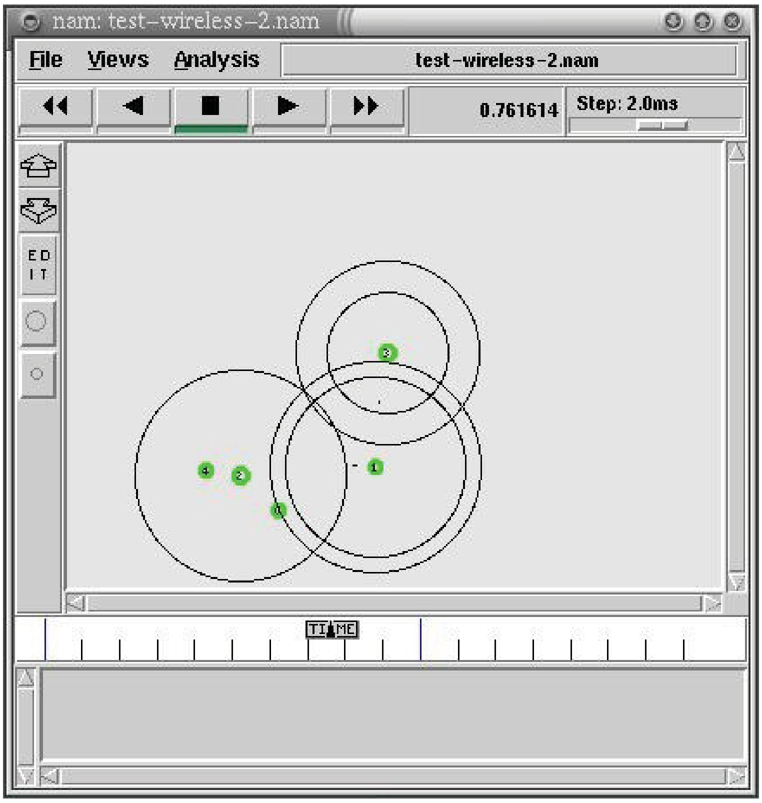


Figure 5.2: NS/Nam Wireless Simulation Screen

## 5.2 NCTUns Architecture

NCTUns is constructed as a distributed architecture to support remote and concurrent simulations [52]. It has four main functional parts where network topology is created and simulated. These are graphical user interface (GUI), dispatcher coordinator and simulation engine.

### 5.2.1 Graphical User Interface (GUI)

GUI side [53], as it is understood from its name, is responsible for creating the network topology; modifying the network nodes, protocol modules, MAC and PHY layer parameters; specifying network traffic; plotting performance curves; playing back animations of logged packet transfers, graphically.

### 5.2.2 Dispatcher

The dispatcher program supports concurrent simulations on multiple simulation machines. It should be executed and kept alive to manage multiple simulation machines. When a user submits a simulation job to the dispatcher, the dispatcher will select an available simulation machine to execute this job. If no machine is available, the submitted job can be queued and managed by the dispatcher as a background job. Later on, when a simulation machine becomes available, the dispatcher will automatically send a background job to it for execution on behalf of the user. When simulation is finished, the simulation logs are transferred to the convenient GUI machines as it is illustrated in Figure 5.3. In our work, the dispatcher is not used. Simulations are performed on a single machine.

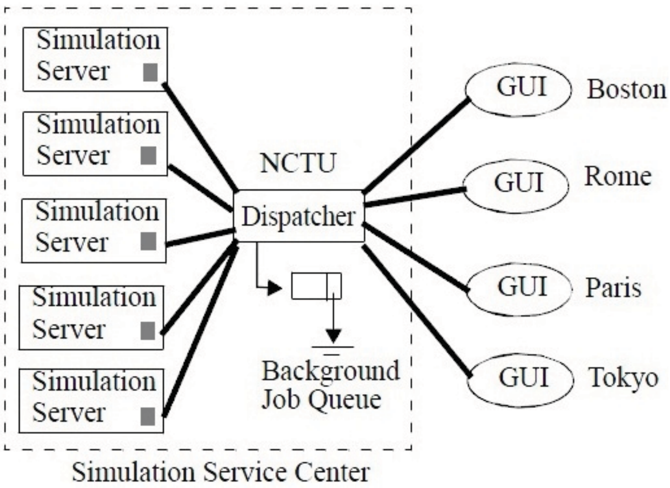


Figure 5.3: NCTUns Dispatcher Functionality

### 5.2.3 Coordinator

Every simulation machine has a coordinator program to communicate with the GUI and the dispatcher. The coordinator is responsible for the following tasks:

- *Forking a simulation engine process to perform a simulation.* When the coordinator receives a simulation job from the job dispatcher, it forks (executes) a simulation engine process to simulate the specified network and protocols. The forked simulation server process will kill itself when its simulation is finished.

- *Reporting the status of the simulation machine to the dispatcher.* The coordinator informs the dispatcher whether this machine is currently busy with running a simulation or not. When executed, it first registers itself with the dispatcher to join the dispatcher's simulation machine farm. Later on, when its status (idle or busy) changes, it notifies the dispatcher of the new status. Based on the machine status information, the dispatcher can choose an available machine from its machine farm to service a job.

- *Communicating with the GUI and dispatcher.* The simulation engine process periodically sends the current simulation time of the simulated network to the coordinator. The coordinator then forwards this information to GUI to inform the GUI user of the simulation progress. During simulation, the user can also set or retrieve an object's value (e.g., to query or set a switch's switch table) online. Message exchanges between the simulation engine process and the GUI program are performed via the coordinator.

### 5.2.4 Simulation Engine

The simulation engine is the core of NCTUns. It is a user-level program that provides a module-based platform for users to develop their protocols and integrate them into the NCTUns simulator. Besides, important services like simulation clock maintenance, timer management, event scheduling, variable registrations are all handled in the simulation engine.

Wimax non-transparent relay MAC and PHY modules of the simulation engine are used within the scope of this work. No modification is done in PHY layer. However, new properties are added in packet handling in MAC layer of both MR-BS and NT-RS.

In PHY layer, OFDMA is implemented for MS, NT-RS and MR-BS by team working in NSL laboratory [6]. Besides, channel coding, channel model and forward error correction (FEC) is included in PHY layer.

In MAC layer, MAC management messages such as ranging request/response, registration ranging/response, DCD/UCD and DL/UL map handling, PDU scheduling, data burst construction, CID classification, UGS QoS class scheduling, timers defined for management message expiries in MS, NT-RS and MR-BS are implemented. However, nrtPS and rtPS QoS classes are not defined in NCTUns 6.0. Moreover, there is no support and scheduling for rtPS and nrtPS service flows. Finally, call admission control is not provided in this version of NCTUns. Therefore, these features are implemented in the scope of this thesis. Moreover, gdb support of NCTUns for debugging is used during implementation phase [54]. The implementation details are presented in the following sections.

## 5.3 Modifications done on the NCTuns Tool

Within the scope of our work, rtPS and nrtPS QoS classes and their relay and access link scheduling are added to the NCTUns network simulation tool. Moreover, our call admission control algorithm and the algorithm suggested by Wang et. al. [7] are introduced as a new module.

### 5.3.1 Module for QoS Support for IEEE 802.16j

In NCTUns, a service flow with QoS class is assigned to each MS. The original version has only "Gold" QoS class type. "Silver" and "Bronze" QoS class types are introduced, as shown in Figure 5.4. Besides, "Min-Reserved-Rate" is written for these two types and handling is a new modification on service flow creation in the simulator. These settings are stored in a configuration file with ".mr_wimax_nt_cfg" extension in the created project and it is read just before the simulation is started to store the service flow information.

### 5.3.2 Module for Scheduling rtPS and nrtPS for IEEE 802.16j

Introduction of two more QoS classes requires adaptation in scheduling UL relay and access channels. In the original version, only UGS service flows are scheduled and slot allocation is performed for them. In order to maintain the tool, round robin scheduling is applied to all QoS class types. For UL access link scheduling, BS scheduler is modified. Initially, slots are allocated for existing UGS service flows directly connected to the BS. Then, reservation is done for rtPS flows in BS. At last, nrtPS service flows in BS are scheduled. For UL relay zone scheduling, slot allocation is applied for each relay station. BS scheduler reserves slots for RS itself. Then, the slots are allocated for UGS, rtPS and nrtPS service flows in the current RS, respectively. This procedure is applied to each RS connected to the BS.

```
Table wmanIfBsServiceClassTable
      Entry
            QoS-Index = 1
            Class-Name = Gold
            Max-Sustained-Rate = 60
      End Entry
      Entry
            QoS-Index = 2
            Class-Name = Silver
            Max-Sustained-Rate = 60
            Min-Reserved-Rate = 30
      End Entry
      Entry
            QoS-Index = 3
            Class-Name = Bronz
            Max-Sustained-Rate = 60
            Min-Reserved-Rate = 30
      End Entry
End Table
```

Figure 5.4: QoS class definitions in the configuration file

### 5.3.3 Module for Call Admission Control for IEEE 802.16j

There is no call admission control support in the original version of NCTUns. Therefore, our algorithm and the algorithm proposed in [7] are implemented as a separate module in the simulation engine.

QoS aware dynamic admission control that is proposed is performed in NT-RS and the decision is made by NT-RS. The incoming ranging request with QoS info from an MS to NT-RS is initially processed and our algorithm is applied by MAC management entity of NT-RS. Finally, the acceptance and rejection decision is made.

The algorithm in [7] is carried out in MAC management entity of BS. The received ranging request to NT-RS is forwarded to the BS. BS runs the call admission control algorithm and makes the decision. Finally, ranging response is sent to the NT-RS in order to be forwarded to the MS itself.

### 5.3.4 Logging and Graphical Display

A macro is introduced to log the slot allocation information in a trace file, periodically. Moreover, the MS call acceptance and rejection information is stored by this macro. The usage is similar with "printf" method defined in standard library of C programming language as shown below:

LOG_MSG("Log Message String parameter: %d \n", parameter1);

The other enhancement in the scope of this study is admitted call display in GUI. During simulation run, all packet transmission and receive information is logged into a file with extention ".ptr". However, there is no indication for admitted calls by RS and BS in graphical display. GUI does not have support to add extra message display items. Therefore, the modification is performed in the simulation engine by using the existing display items. In this way, the user can see the admitted calls and the allocated slots for each MS as it is seen in Figure 5.5.

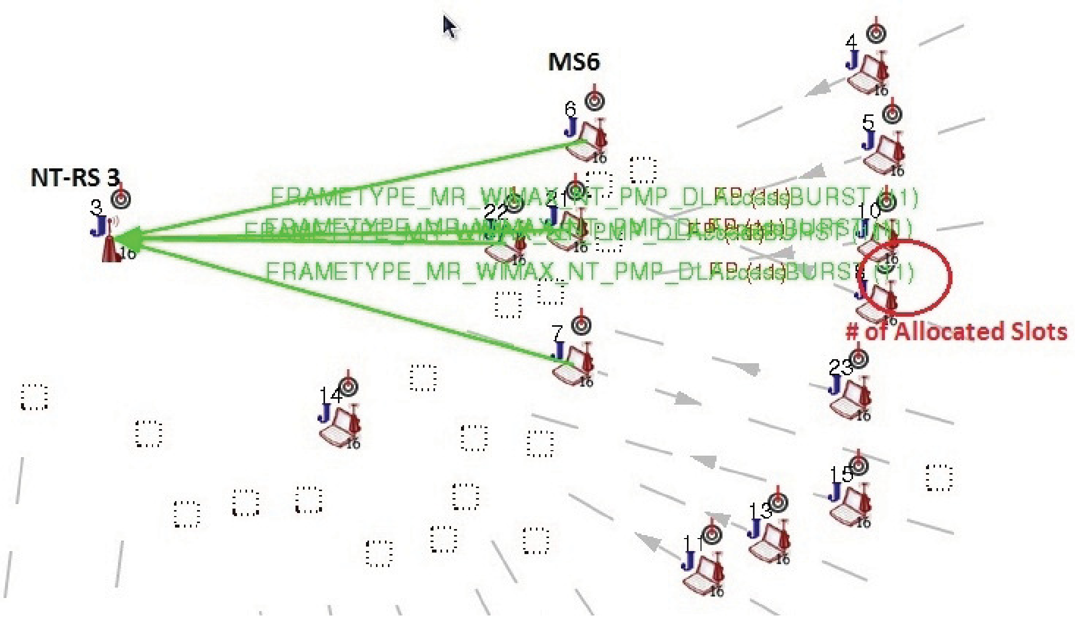


Figure 5.5: Graphical display of admitted calls

## 5.4 Simulation Results

In this section, the topology and the settings used in the simulation, the results and the comparison of the results with the research previously done are explained.

### 5.4.1 Topology

Point-to-multipoint mode (PMP) is used in the scope of this study. Two NT-RS and one MR-BS are located to serve the MSs. The BS has a backbone connection with a host via ethernet interface. In order not to create a direct access link between MR-BS and MS, the transmission power of MR-BS is set to 2dbm which is less than the original deployment parameters in the real life settlement. Because our aim is to benefit from the coverage extension property of the non-transparent relay stations and evaluate the performance of our algorithm in relay stations. Therefore, MSs are out of the coverage range of BS.

The NT-RSs are in the coverage area of the MR-BS. They are connected to MR-BS via IEEE 802.16j air interface. There is a relay link between each NT-RS and MR-BS. In addition, there is an access link between each MS and NT-RS. The transmission power is set to 2 dbm in OFDMA PHY of NT-RSs, too. All MSs send the ranging request to the NT-RSs and contact to the MR-BS and send the PDUs and management messages via relay station.

Approximately 30 MSs are located in this topology. They do not perform the network entry at the same time. Due to a system bug in NCTUns, when 30 MSs send the ranging request in two second intervals, the packets cannot be handled and processed properly in the physical layer. System is crashed with a segmentation fault because of the removal of the unprocessed packets which are also created as events in NCTUns. Thus, a simple path is defined more than half of the MSs in order to register to NT-RS at reasonable intervals. In this way, monitoring of our call admission control and comparison with the other algorithms is more convenient.

As a result, a tree based PMP mode topology is used in simulations as it is illustrated in Figure 5.6.

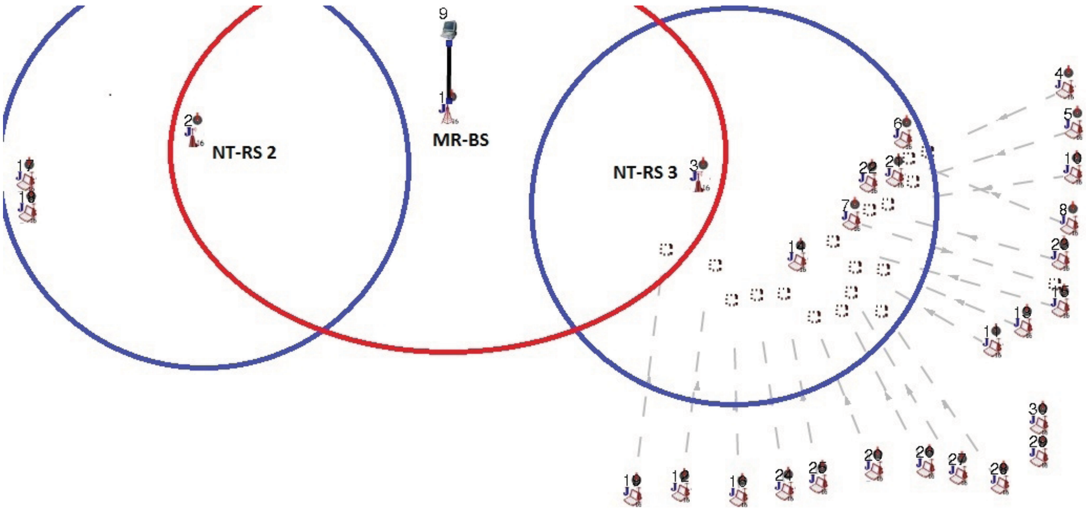


Figure 5.6: Topology used in the simulations

### 5.4.2 Parameters Used in Simulation

All used nodes including MSs, NT-RSs, MR-BS in the simulation have predefined settings. In this section, these parameters are explained.

As it is mentioned in the previous section, the transmission power of the NT-RSs and MR-BS is 2 dbm.

All used MSs have a unique service flow which owns a type of service class among UGS, rtPS and nrtPS. Maximum sustained rate of UGS service class is assigned as 60 kbps. Maximum sustained rate of rtPS and nrtPS is assigned as 60kbps. Finally, minimum reserved rate of rtPS and nrtPS is determined as 30kbps. These are listed in Table 5.1.

The parameters used during slot calculation are listed in Table 5.2. In this study, the slots reserved by an NT-RS has no limitation. This means an NT-RS has the opportunity to use the all slots in reserved for all NT-RSs in BS. Therefore, it can be said that the limit of an NT-RS can be less or equal to the reserved relay slots in BS and the limits of each NT-TS are adjusted automatically by applying our algorithm.

In order to compare the results of our algorithm and the results in previous studies, all these settings and the topology are left as they are and these different algorithms are run on them. At the end, more accurate results are retrieved.

Table 5.1: Parameters for changed for simulation

| **Parameters** | **Value** |
|---|---|
| Transmission Power of MR-BS | 2 dbm |
| Transmission Power of NT-RS | 2 dbm |
| Maximum Sustained Rate of UGS Service Class | 60 kbps |
| Maximum Sustained Rate of rtPS Service Class | 60 kbps |
| Minimum Reserved Rate of rtPS Service Class | 30 kbps |
| Maximum Sustained Rate of nrtPS Service Class | 60 kbps |
| Minimum Reserved Rate of nrtPS Service Class | 30kbps |

Table 5.2: Parameters used in slot calculation

| **Parameters** | **Value** | **Calculation** |
|---|---|---|
| UL Subchannels | 35 | |
| Symbols Per Frame | 48 | |
| Maximum UL Ratio | 0.5 | |
| Maximum UL Relay Ratio | 0.5 | |
| UL_PUSC | 3 | |
| UL Available Symbols | 24 | symbolsPerFrame * Maximum UL Ratio |
| UL Relay Available Symbols | 12 | ULAvailableSymbols * Maximum UL Relay Ratio |
| UL Relay Available Slots | 116 | ( UL Relay Available Symbols / UL_PUSC) * (UL Subchannels - 6) |

### 5.4.3 Simulation Results and Comparison

This section includes the simulation results in various scenarios where different Quality of Service classes are examined. The compared values of all algorithms are total allocated slots, bandwidth utilization, allocated slots for RS2, allocated slots for RS3 and the total admitted calls in simulation. The bandwidth utilization is measured with the code in formula 1.

```
Bandwidth Utilization = Slots Used / Total Available Slots   (1)
```

Before giving the simulation results and comparison, the details of the algorithm in [7] which is compared with ours is explained briefly. In this algorithm, some amount of bandwidth referred as U is reserved for UGS calls initially. Then, the admission control performs as follows:

- When a request for UGS connection arrives at BS, if the allocated bandwidth by ongoing connections plus the necessary bandwidth for incoming call is less than or equal to total bandwidth, then the call is accepted, else it is rejected.
- When a request for rtPS connection arrives at BS, if the allocated bandwidth by ongoing connections plus the necessary bandwidth for incoming call is less than or equal to total bandwidth minus U, then the call is accepted. Else, the ongoing nrtPS connections are degraded by level 1. It is checked again. If it is still not enough, then all nrtPS connections are degraded one level more, till the sufficient bandwidth is extracted or the maximum degradation step is reach. If there is enough bandwidth for new call, the call is accepted, else it is rejected.
- When a request for rtPS connection arrives at BS, if the allocated bandwidth by ongoing connections plus the necessary bandwidth for incoming call is less than or equal to total bandwidth minus U, then the call is accepted.Then, the degradation process is applied to nrtPS ongoing calls like in rtPS call admission control. One difference from the previous state is that the necessary bandwidth for an incoming nrtPS call is adjusted according to the degradation level. This means the required bandwidth for new call is degraded before the start of admission control.

#### 5.4.3.1 nrtPS Call Admission Results

In this scenario, all MSs request bandwidth for nrtPS QoS class service flow. Comparison is done with the study previously conducted on degradation of the ongoing calls in the BS which is compatible with 802.16e. The same algorithm is also implemented in MAC layer of MR-BS which is compatible with IEEE 802.16j. The charts in this section give detailed information about the performance of our QoS aware dynamic call admission call in IEEE 802.16j non-transparent relay networks.

Figure 5.7 consists of the data about the slot usage in the life cycle of entire simulation. The bandwidth utilization of the whole network is illustrated in Figure 5.8. As it seen from the figures, the bandwidth utilization of our algorithm is better than the others. Even if the slots reserved for potential UGS connections are 0 (U=0), there is reduction in the slot allocation. Because, the other algorithm performs degradation to all connected nrtPS calls. However, in our algorithm, degradation is applied until enough slots are available for new nrtPS connection. This enables the other users not to lose their bandwidth unless it is necessary. Moreover, if the assigned value to reserved slots for UGS connections is 30 (U=30), which is one third of the all available relay slots which can be used by all service flows, the bandwidth utilization is less because all service flows in this simulation has nrtPS QoS service class. Therefore, one third of total slots are useless for new incoming nrtPS and rtPS service flows.

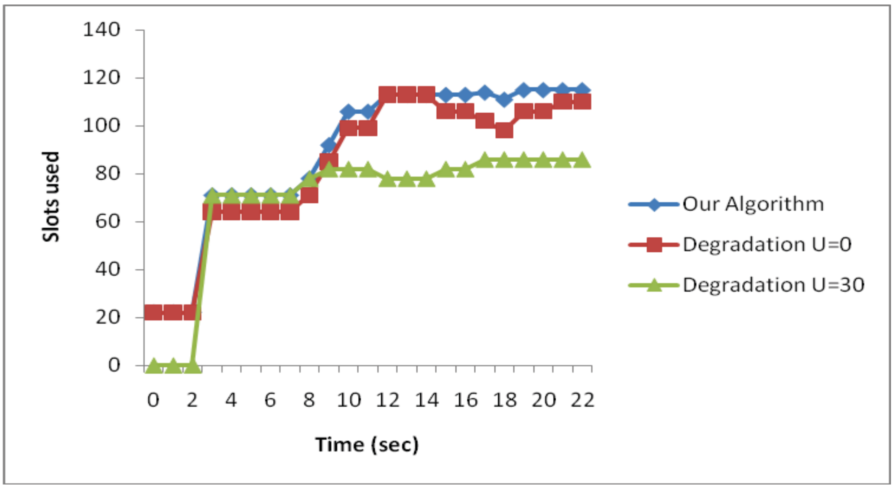


Figure 5.7: Total slots used by nrtPS calls

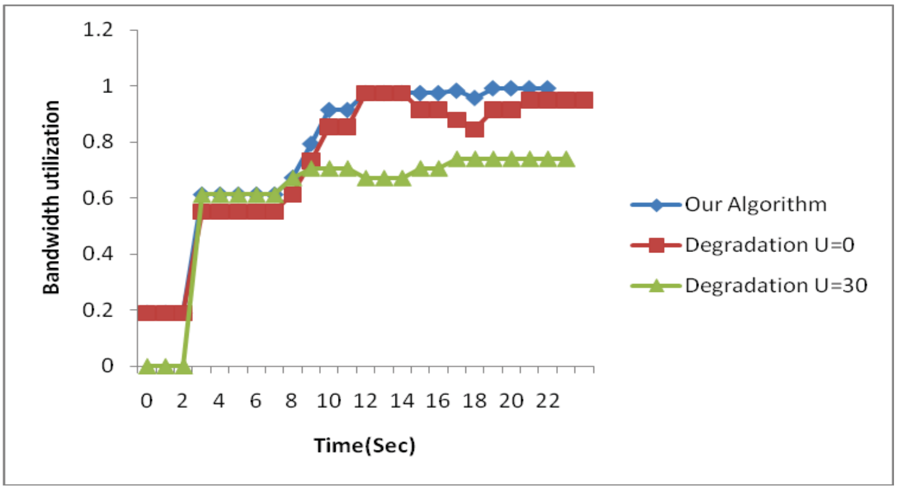


Figure 5.8: Bandwidth utilization of nrtPS calls

As it is seen in Figure 5.9, there is no difference in the number of admitted call when U is set to 0 in the other algorithm, since at the end of the simulation, same degradation level is reached. Both algorithms make use of all bandwidth to admit a new call. If U is set to 30, of course nrtPS admitted call numbers decrease. Because, one third of the total slots are unused and reserved for potential UGS flows.

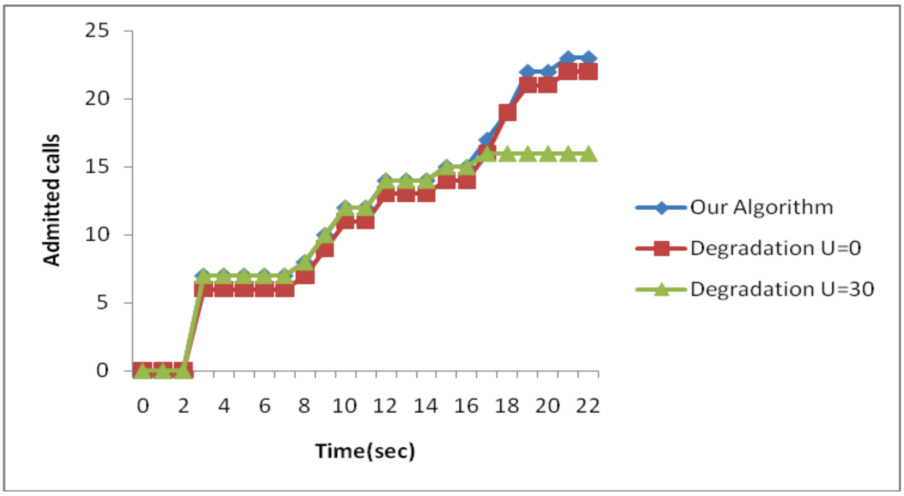


Figure 5.9: Admitted nrtPS calls

Figure 5.10 and Figure 5.11 show the slots usage of the RS2 and RS3. One of the aims of this study is to give individual control to each NT-RS on call admission and slot allocation. The other one is to prevent the slot usage reduction on MSs connected to an NT-RS from the density in the coverage area of the other RSs as possible as it can be. In our simulation represented by Figure 5.10, 2 MSs are connected to RS2. The decrease in allocated slots starts at 14th second when the other algorithms are applied. However, in our algorithm, it starts at 18th second. This indicated that our algorithm preserves the used slots longer. However, this is valid to some extent. In order to preserve the service availability, which is the third aim of our study, the reduction should be applied so that more calls can be admitted.

In Figure 5.11, it is seen that our algorithm ourperforms the other algorithms. Flickering in the curves of other degradation algorithms are more than ours'.

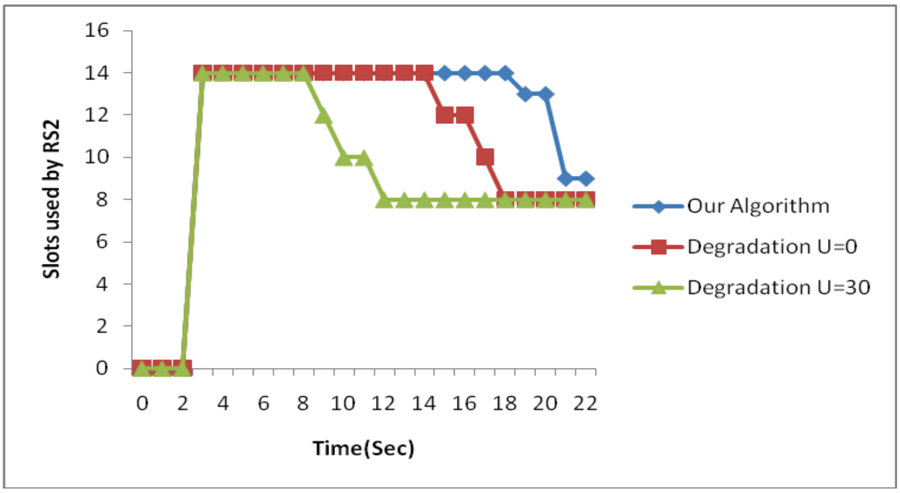


Figure 5.10: Slots used by nrtPS calls in RS2

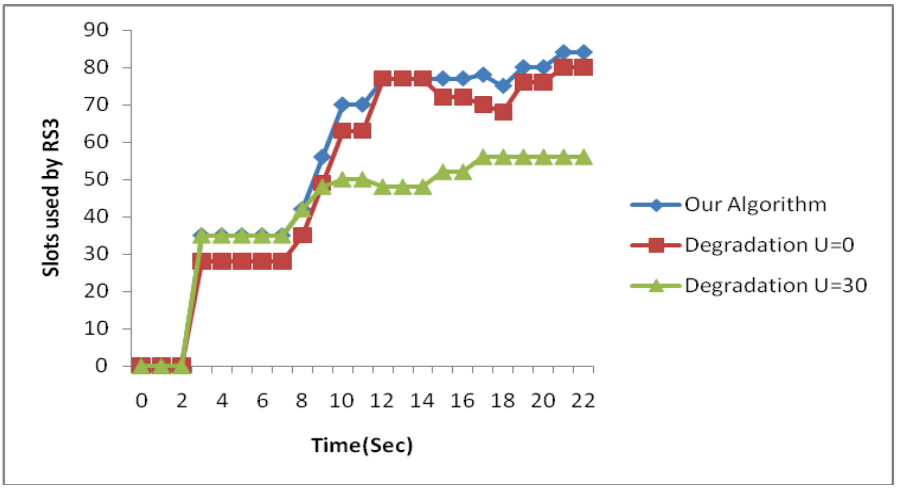


Figure 5.11: Slots used by nrtPS calls in RS3

#### 5.4.3.2 rtPS Call Admission Results

In this scenario, all MSs have rtPS QoS class service flows whose maximum sustained ratio is 60 kbps and minimum reserved rate is 30 kbps. As it is seen in Figure 5.12 and Figure 5.13, bandwidth utilization in our algorithm is a bit better in rtPS call admission control.

The main advantage of our algorithm in a network where all MSs have rtPS QoS class service flows is the number of the admitted rtPS calls. As it is illustrated in Figure 5.14, the number of

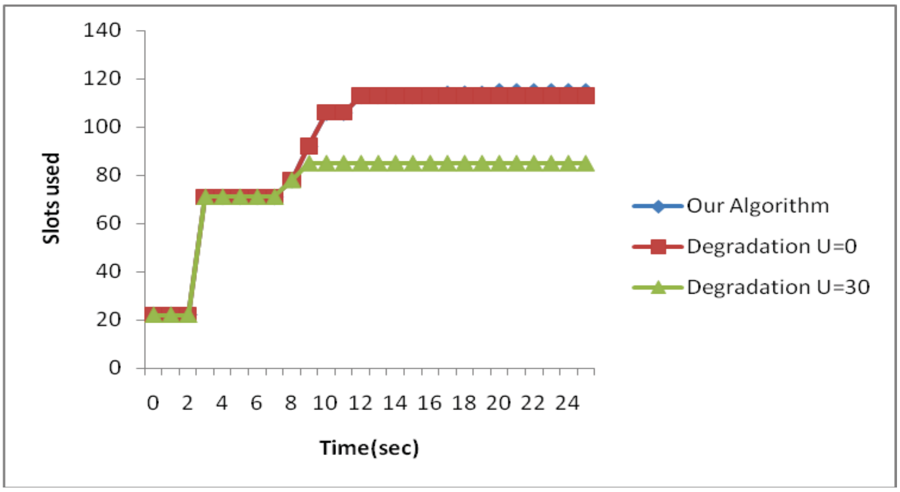


Figure 5.12: Total slots used by rtPS calls

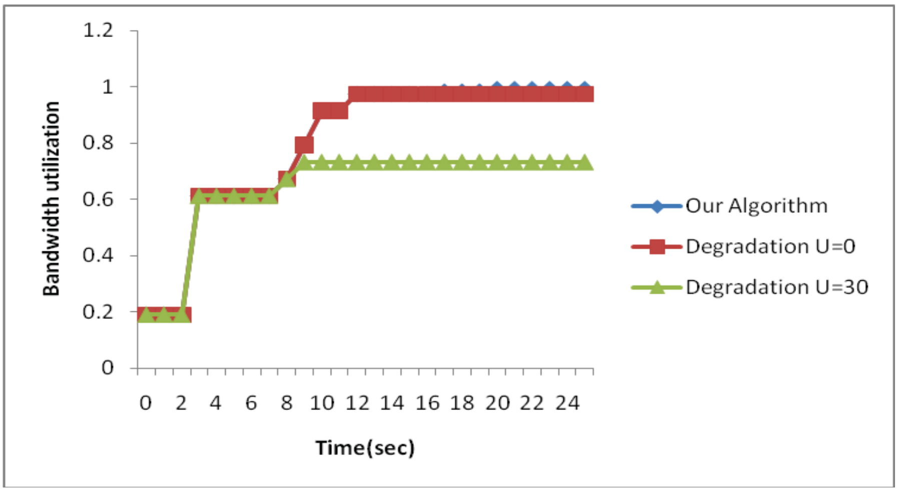


Figure 5.13: Bandwidth utilization of rtPS calls

admitted rtPS calls are nearly twice of the number of admitted calls in other algorithms. The reason is that the degradation is only applied to the ongoing nrtPS calls. The rtPS calls are not degraded till the minimum reserved rate in the other algorithm. However, in our algorithm, if a new call is arrived, existing nrtPS calls are degraded initially. If there is still not enough slots, ongoing rtPS calls are started to be degraded. In this way, the network availability is provided for rtPS calls and more rtPS service flows can be served by NT-RSs.

Figure 5.15 and Figure 5.16 illustrate the slot usage in RS2 and RS3 individually. Due to

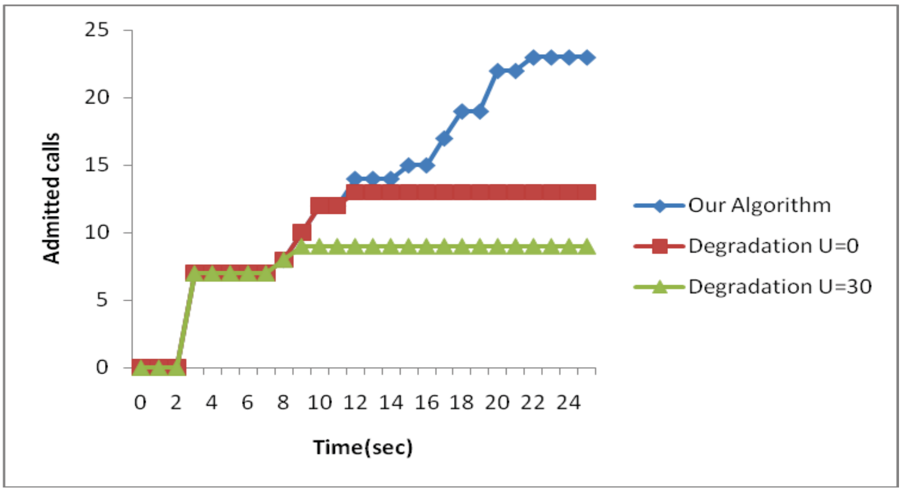


Figure 5.14: Admitted nrtPS calls

the degradation process in our algorithm, the number of used slots start to decrease after 18th second. This means RS3 borrows some slots from RS2. Therefore, the slot usage increases after 18th second in RS3 due to the MS density increase.

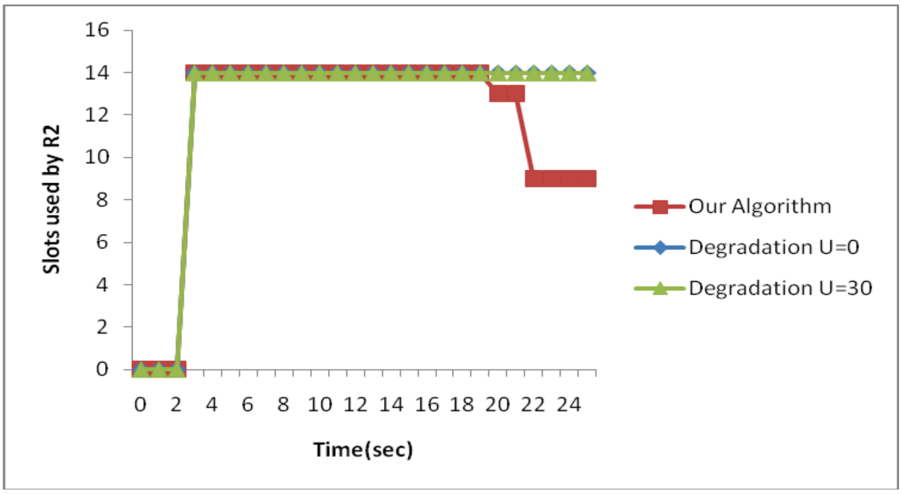


Figure 5.15: Slots used by rtPS calls in RS2

#### 5.4.3.3 UGS Call Admission Results

In this scenario, all calls have UGS QoS class service flow. Due to the constant rate bandwidth requirements of UGS calls, both algorithms have the same bandwidth utility as it is seen in

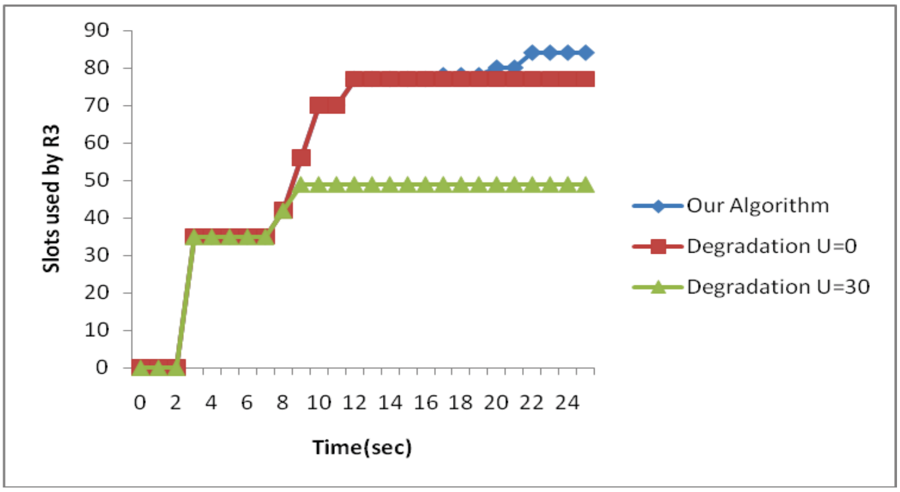


Figure 5.16: Slots used by rtPS calls in RS3

Figure 5.17.

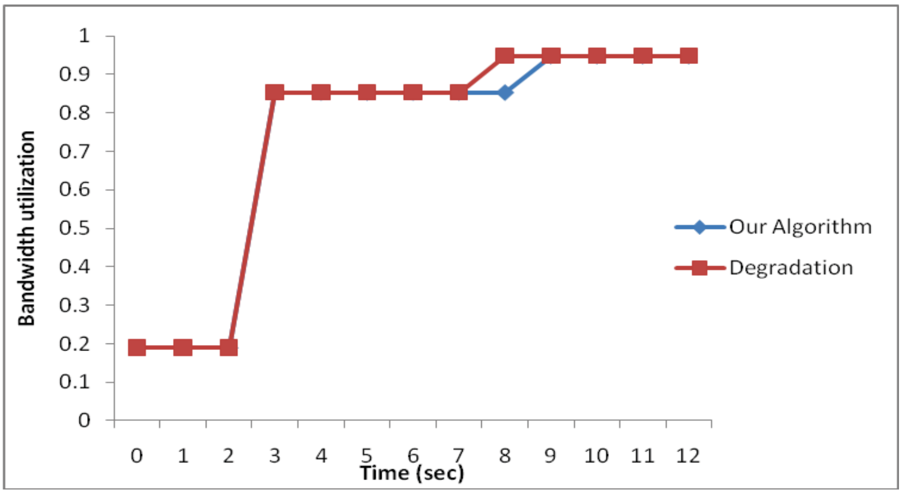


Figure 5.17: Bandwidth utilization of UGS calls

Moreover, the number of admitted calls are same in both algorithms for UGS calls. No degradation can be applied to UGS service flows due to the constant rate necessity. However, admitted calls can be increased with a more advanced scheduling algorithm where jitter tolerance property of UGS QoS class is considered. This is left as a future work.

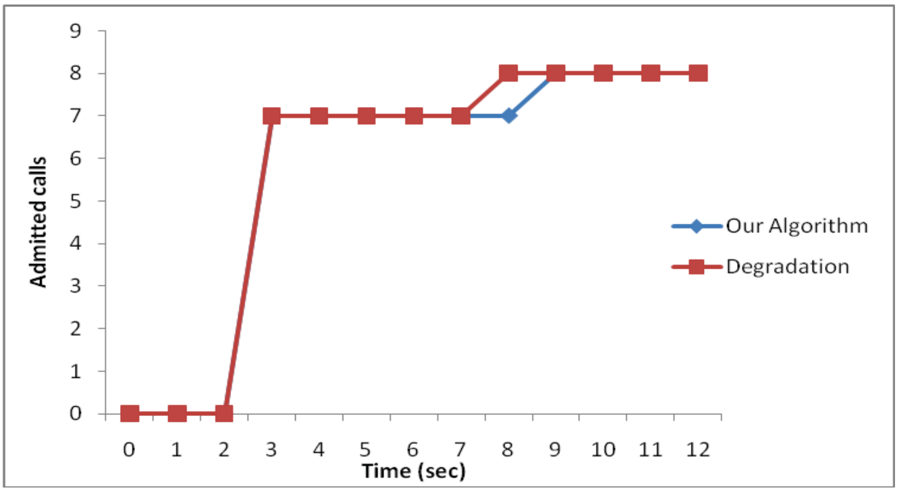


Figure 5.18: Admitted UGS calls

#### 5.4.3.4 rtPS Call Admission Results in Different Topologies

In this section, our algorithm is compared with the three types of topology. One is the topology which is used in Section 5.4.3.2. In this topology, RS2 has very low population which consists of only two mobile subscribers. The charts in this section illustrates our algorithm performance in different topologies where the number of mobile subscribers varies. In the first topology, there is only 2 mobile subscribers connected to the RS2. It is referred as “Low Population in RS2”. In the second one which is also called as “Medium Population in RS2”, 8 mobile subscribers have connection with RS2. In the last one, 14 mobile subscribers are attached to RS2. The number of subscriber is nearly twice as much as the number of subscribers in medium one. So, it is assumed as “High Population in RS2”. In all cases, the mobile stations have the rtPS QoS class service flows.

As it is seen from Figure 5.19 and Figure 5.20, bandwidth utilization and total number of admitted calls in RS2 and RS3 are the same. As it is mentioned in section Section 5.4.2, there is a limit in BS. Therefore, maximum 23 rtPS calls which has the parameters in section Section 5.4.2 can be admitted and total bandwidth can be utilized effectively.

When it is analyzed per RS basis, the allocated slots per each NT-RS change according to the population in RS2. The changes are illustrated in Figure 5.21 and Figure 5.22. In the topology where RS2 has low population consists of 2 subscribers, the degradation of the calls in RS2 starts at 18th second. In the medium population case where 8 mobile stations are subscribed to RS2, the degradation of the ongoing calls in RS2 started at 8th second. In the last high population case where 14 mobile stations are connected in RS2, the degradation of the available calls are performed at startup. Because, the new calls in RS2 at startup needs bandwidth. Therefore, there is no slots left for degradation in order to admit the new call request in RS3. In this case, the ongoing calls in RS3 are degraded to accept the incoming request into RS3. The fairness of this algorithm is seen from these observations. All NT-RSs perform degradation initially on the ongoing service calls connected to each of them.

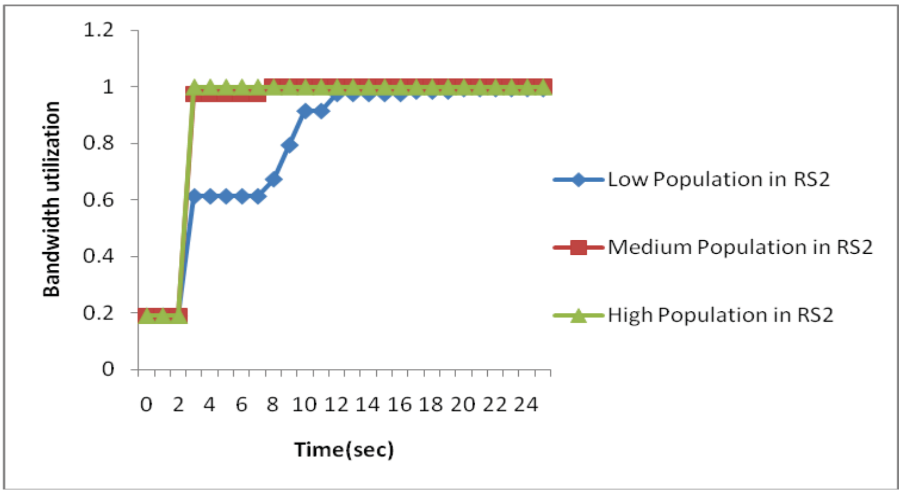


Figure 5.19: Bandwidth utilization of rtPS calls in low, medium and high population of RS2

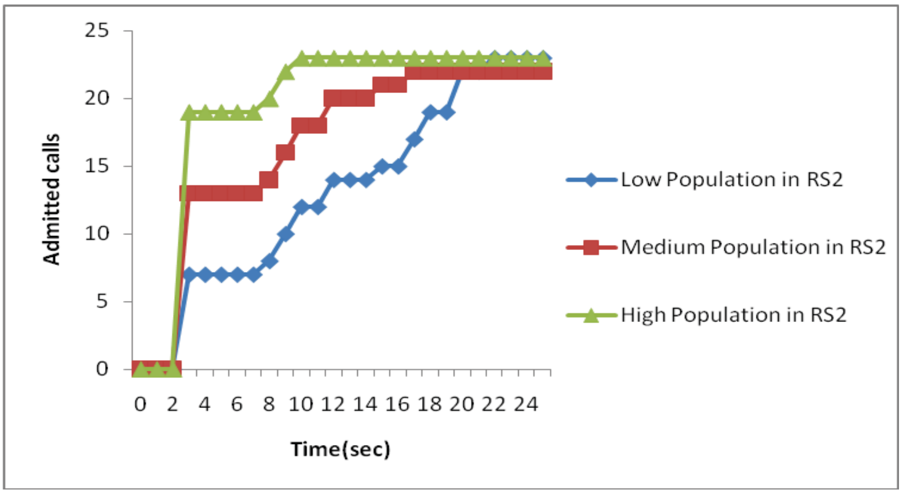


Figure 5.20: Admitted rtPS calls in low, medium and high population of RS2

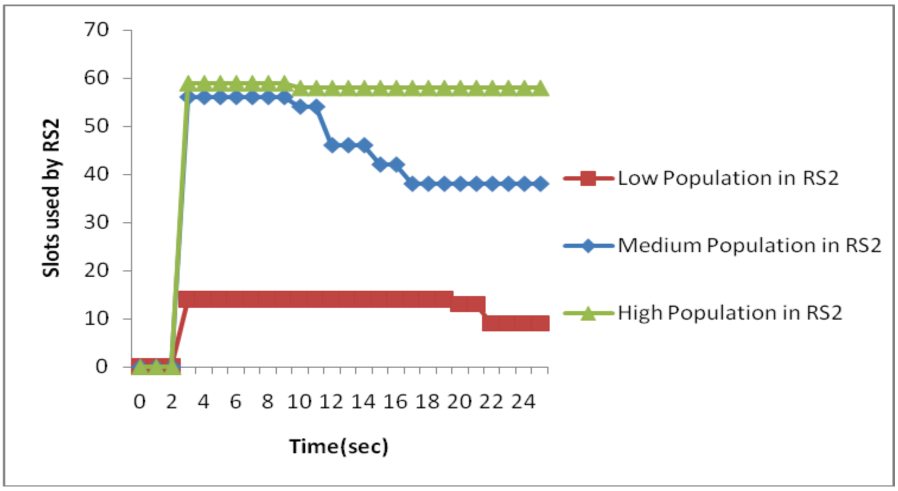


Figure 5.21: Slots used by RS2 in low, medium and high population of RS2

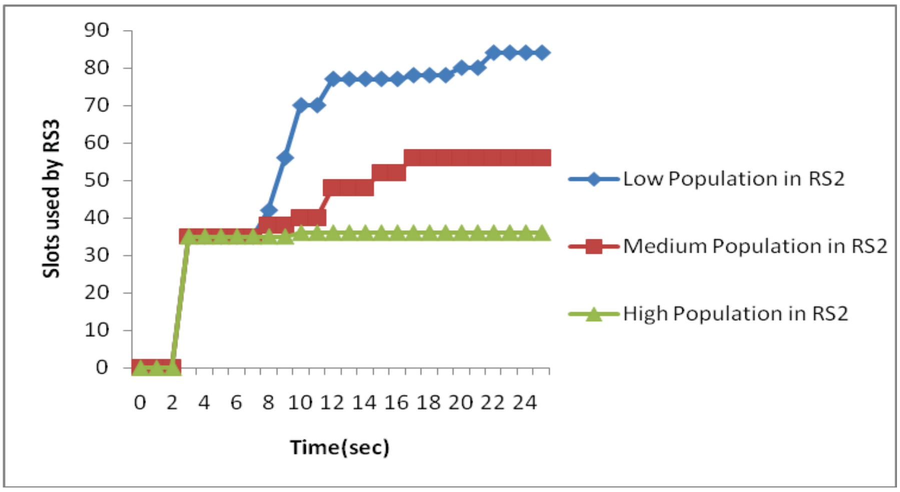


Figure 5.22: Slots used by RS3 in low, medium and high population of RS2

#### 5.4.3.5 Bulk Processing of rtPS Call Admission

In this section, the incoming requests are handled as bulk. The topology in section Section 5.4.3.2 is used. All service flows has rtPS QoS class. All incoming requests in every 5 seconds are stored in a queue in each NT-RS. At the end of the 5 seconds, our degradation algorithm is executed only once and the admission decision of each request is made for each request. The T3 timer which is waiting for RNG-RSP ticks only 30 milliseconds. If MS does not get any RNG-RSP in 30 milliseconds, it sends the RNG-REQ again. Therefore, there could be more than one request for each MS service flow. In order to process multiple RNG-REQ for the same service flow, the existence of the ingoing request in the queue waiting to be processed is checked. If it is already pushed into the stack, no more action is taken. Otherwise, the request is inserted to the queue for bulk admission control process.

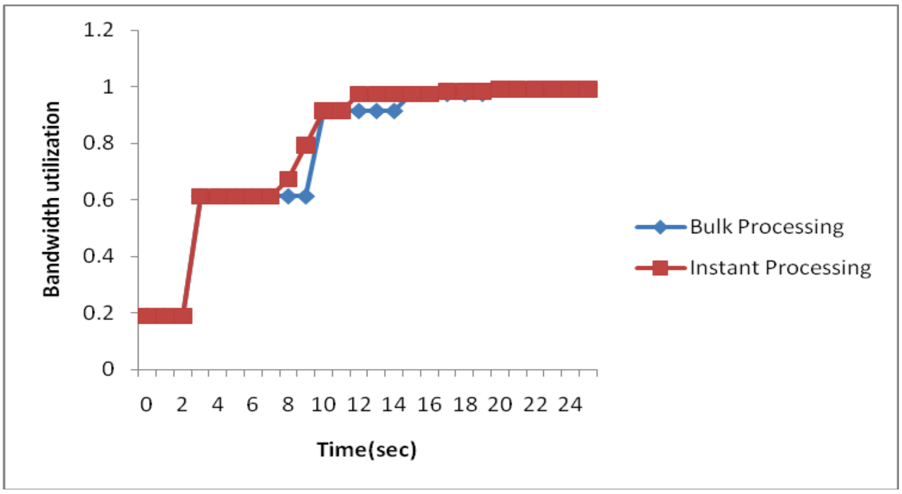


Figure 5.23: Bandwidth utilization of rtPS calls after instant and bulk processing

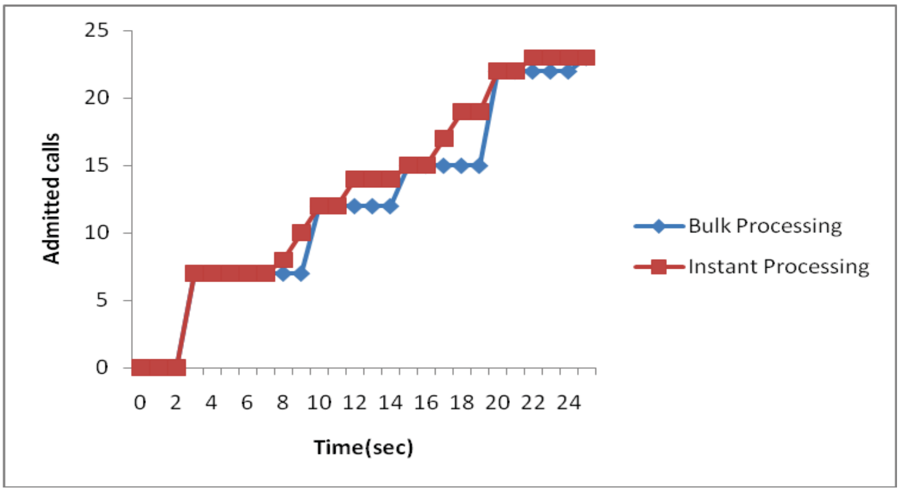


Figure 5.24: Admitted rtPS calls after instant and bulk processing

As it is seen in Figure 5.23, the bandwidth usage increases in every 5 seconds. Moreover, number of admitted calls rises periodically. Besides, the degradation of the ongoing calls in each NT-RS is performed in every 5 seconds due to the bulk process. However, there is no change on total number of admitted calls, the bandwidth utilization, slots allocated by RS2 and RS3 at the end of the simulation. The only benefit of the bulk processing is to decrease the computation time of the call admission control mechanism. However, computational complexity is not one of the concerns and constraints of this study.

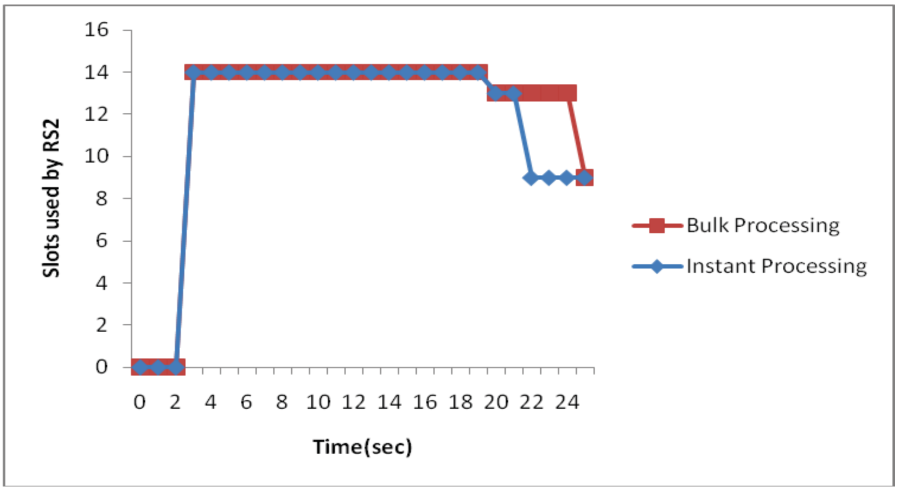


Figure 5.25: Slots used by RS2 after instant and bulk processing

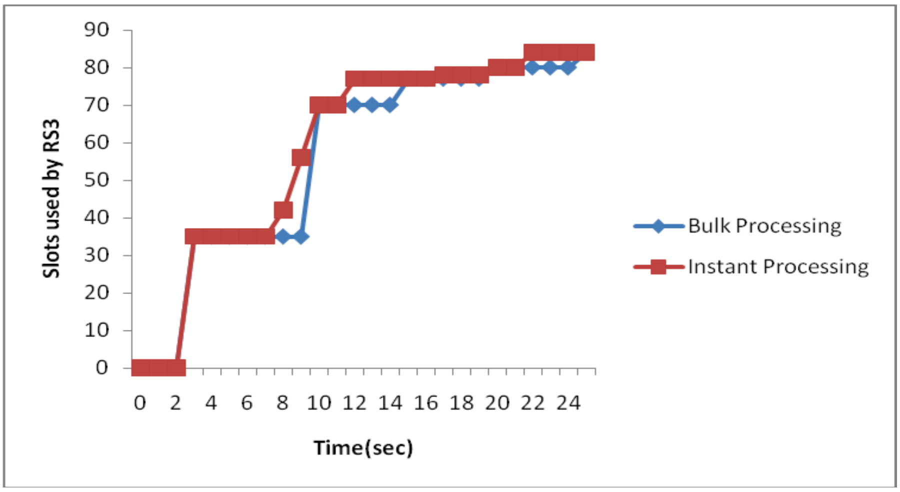


Figure 5.26: Slots used by RS3 after instant and bulk processing

## 5.5 Discussions

As a result of the simulations, it is concluded that our algorithm outperforms the dynamic call admission control algorithm proposed by Wang et. al. [7] in terms of the following aspects:

- nrtPS calls can utilize the total bandwidth.
- More rtPS service flows are admitted.
- Relay stations have self control on call admission and rejection. In case of rejection of a call, BS is not notified and RNG_RSP message is sent directly to the MS. So, overhead of one management message for incoming service flow rejection is eliminated.
- Finally, the crowd of admitted calls in an NT-RS has less effect to the reserved bandwidth for service flows in other NT-RSs. Besides these benefits, the proposed admission control algorithm can be considered as one of the pioneer algorithms applied in IEEE 802.16j non-transparent relay networks.

# CHAPTER 6

# CONCLUSION AND FUTURE WORK

Today, with the increasing demand on telecommunication and mobility, wireless technologies continue to evolve rapidly. Therefore, 4th generation (4G) products are started to be produced as a successor of the 3rd generation products. One of the significant telecommunications technologies of 4G era is Worldwide Interoperability for Microwave Access (Wimax) which is based on IEEE 802.16 standard called Broadband Wireless Access (BWA). Wimax provides the delivery of last mile wireless broadband access as an alternative to cable and Digital Subscriber Line (DSL).

IEEE 802.16 standards started supporting fixed networks, then continued with nomadic networks and finally improved itself to support mobile networks. As a result of this amendment, IEEE 802.16e standard became most frequently used and deployed Wimax standard where a base station provides service to mobile stations. However, in some areas where high buildings or other constructions are located, the signal rate decreases due to shadow fading. Therefore, an intermediate node, which forwards the signal from the base station to mobile station or vice versa, became a necessity. Thus, a new amendment of IEEE 802.16 standard, namely IEEE 802.16j, for this intermediate node called relay station was introduced. Due to easy deployment, cost effectiveness, coverage and throughput enhancements, relay stations are preferred by service providers. As mentioned previously, there are two modes of relay; transparent and non-transparent. In this study, the coverage extension property of the non-tranparent relay stations which extends the service availability was made use of.

Like all other IEEE 802.16 standards, there are some issues which are not specified in IEEE 802.16j protocol. They are left to the product manufacturers. A significant issue handled by vendors is call admission control. Although there are many call admission control proposals for IEEE 802.16e standard, there is not much study on admission control in IEEE 802.16j relay networks due to its recency.

In this thesis, a new quality of service aware dynamic admission control mechanism for non-transparent Wimax relay networks to provide service availability and maintainability was introduced. Service availability is supported by serving the customers at any time to the extent the system resources allow. Service maintainability is performed by keeping the ongoing services' quality regarding QoS requirements while providing service availability for incoming requests.

In accordance with the above-mentioned open issues on call admission control in IEEE 802.16j non-transparent relay networks, in this thesis, significant contributions were made. Our achievements can be summarized as follows:

- In order to increase service availability and maintainability of IEEE 802.16j Non-Transparent networks, a new QoS aware dynamic call admission control algorithm is applied on Non- Transparent Relay Stations.
- One of the achievements of this study is to give the individual control to each NT-RS on call admission and slot allocation.
- The other one is to prevent the slot usage reduction on MSs connected to an NT-RS from the density in the coverage area of the other RSs as possible as it can be.
- One of the advantage of our algorithm in a network where all MSs have rtPS QoS class service flows is the increase in number of the admitted rtPS calls.

Although crucial enhancements in call admission control in non-transparent Wimax relay networks were accomplished, there are some cases which cannot be improved. Bandwidth cannot be utilized for UGS service flows or admitted calls cannot be increased for them too, due to the constant rate bandwidth requirements of UGS calls. In order to increase the number of admitted UGS service calls, a more efficient scheduling algorithm can be used instead of round robin. This enhancement can be done as a future work.

There are some other areas that can benefit from some further enhancements as well. For example; the proposed algorithm is subscriber based which means the admission control and bandwidth allocation is performed per connection. This control can be improved to be service based.

Handover is not handled within the scope of this study, because there is no full handover support in NCTUns tool. Besides, our algorithm is applied during initial ranging. In the future, handoff calls can be handled and it could be supplied that handoff service flows utilize the available bandwidth more effectively in the network which supports IEEE 802.16j specification.

In our simulations, the information about admitted service calls are retrieved from a common management entity. There is no message which requests the currently available services in the whole system introduced from RS to MR-BS. In this case, more modification on management messages between RS and MR-BS which accommodate the total reserved bandwidth and the request from RS to MR-BS for degrading the available nrtPS and rtPS calls in MR-BS should be defined in IEEE 802.16j specification.

Moreover, the algorithm can be enhanced by introducing a learning algorithm where the behavior of the users in a cell can be observed and analyzed. After learning is completed, the call admission decision can be performed automatically without checking the degradation level of each ongoing call. In order to include the learning mechanism in this dynamic admission control, the learning algorithm which is going to be implemented should be determined, such as Q-learning. Moreover, the global parameters which are considered in learning algorithm should be decided to achieve this goal.

As a result, although there are still some topics that can be worked further on to improve this work, quality of service aware dynamic admission control algorithm on IEEE 802.16j non-transparent networks that is proposed outperforms the ones applied in IEEE 802.16e networks. Moreover, there are not so many examples in this area. Therefore, our work can be interpreted as the leading call admission control mechanism in IEEE 802.16j non-transparent networks.

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