國立交通大學

電子工程學系電子研究所碩士班

碩 士 論 文

無線資源管理在行動 WiMAX 系統之研究

Radio Resource Management in Mobile WiMAX System

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中 華 民 國 九 十 五 年 六 月

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摘　　　要

隨著使用者對於高速無線資料傳輸的需求增加，行動 WiMAX 系統能滿足此需求且為 3.5 代行動系統的候選人之一。在本論文中,我們將個別針對即時性和非即時性的服務，在不同的排程控制方法之下去作效能分析，並對允許進入的控制機制提出相關的判斷準則。我們也會針對在混合式服務之下，有使用和沒有使用階層式排程控制其對於系統效能的影響和交互關係。更進一步地，我們調查了不同的頻率重複使用係數對於整各系統效能的衝擊。 為了量化行動 WiMAX 系統的效能，我們發展了一個行動 WiMAX 模擬平台，並且在這平台上建立了包含基礎無線資源管理機制的 MAC 層。根據模擬數據的分析，我們可以得知若使用了考慮通道狀況的排程方法加上頻率正交多重接取技術的子載頻置換，即使細胞站的佈置使用單一頻率，仍然可以提供不錯的系統效能。若又使用階層且合適的排程方法將能夠改進系統在混合式服務之下的使用者容量。

Radio Resource Management in Mobile WiMAX System

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ABSTRACT

With the increasing demand in high-speed wireless transmission, the mobile WiMAX based on IEEE 802.16e standard has been adopted as one of the candidates for beyond 3G cellular technologies. In this thesis, to effectively support real-time and non-real-time services, the system performance of mobile WiMAX with different radio resource managements in scheduler controls and the admission control criterion will be evaluated and analyzed. We also observe the effect and interaction with and without hierarchy scheduling in the mixed traffic. Furthermore, the impacts from different frequency reuse factors will also be investigated. The simulation platform, based on mobile WiMAX, is developed to quantify the mobile WiMAX performance. We also establish the MAC layer with basic resource management in this mobile WiMAX platform. The analytic results show the scheduling control, considered the channel condition and OFDMA permutation, can still offer a good performance in single frequency cell coverage and using hierarchy and appropriate scheduling controls can improve the system capacity with QoS guaranteed in a mixed traffic.

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碩士班兩年的時光匆匆，在新竹求學的六年時間將畫下句點。尤其在碩士班的兩年當中，有很多人跟我因為緣分而相遇，更是我需要去感謝的貴人。也感謝老天讓我如此幸運的遇見你們，一起陪伴著我度過這段難忘的時光。

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Chapter 1

Introduction

The standards of IEEE 802.16 [1][2] family provide fixed and mobile broadband wireless access (BWA) and promise to deliver multiple high data rates services over large areas. An industry group, Worldwide Interoperability for Microwave Access (WiMAX), cleans up and streamlines the implementation of IEEE 802.16 standards. Using the new standard, users in the cell can directly connect to telecommunication networks and Internet via radio links. Based on the complexity and flexibility management of Medium Access Control (MAC) and Physical Layer (PHY), IEEE 802.16 family can offer better QoS performances and an efficient deployment for operators.

The first 802.16 standard approved in 2001 is in the 10-66 GHz range for line of sight wireless broadband services. In order to overcome the disadvantage of line of sight links, the 802.16a completed in 2003 is in the 2-11 GHz band for non-line of sight wireless broadband services. The 802.16d recently approved in 2004 (now named 802.16-2004) upgrades to the 802.16a standard and is the last version of fixed locations by radio links. The new 802.16e standard extends the 802.16d standard and provides the mobility support in cellular deployments [3]. The IEEE 802.16 standards [1][2] define four different PHYs to provide a reliable end-to-end link. They are single carrier (SC) between 10-66 GHz, single carrier release a (SCa) between 2-11 GHz, orthogonal frequency division multiplexing (OFDM), orthogonal frequency division multiple access (OFDMA). The SC and SCa are not used, because they are not suitable for non-line of sight transmissions. The OFDM is less complex than OFDMA and it is adopted by fixed products (802.16d). The OFDMA can offer more system gain performance and scalability than OFDM. Due to these reasons it is more suitable for mobility products (802.16e) [4][5].

The wide bandwidth allocation and QoS mechanisms are provided in the 802.16 standard. But the details of scheduling, admission control and reservation management are left undefined. This thesis focuses on the IEEE 802.16e mobile system level performances. The performance analysis adopts OFDMA technique and different evaluations for various system parameters, such as scheduling algorithms and frequency reuse factors. Detailed system level simulations have been performed in interference limited cellular environment with 6MHz bandwidth. This thesis will present the PHY/MAC throughput, capacity, spectrum efficiency, channel usage, scheduling delay, scheduling jitter, and adaptive modulation and coding scheme (AMC) usage of non-real time services. It will also present the packet loss rate and capacity of real time services. Finally, the interactive of mixed services and admission control criterion will be discussed.

The rest of this thesis is organized as follows: In Chapter 2, the overview of PHY and MAC layers of 802.16e standard are briefly introduced. In Chapter 3, the settings of simulation platform and basic radio resource managements are addressed. Simulation results are presented in Chapter 4. Finally, conclusions and future works are provided in Chapter 5.

Chapter 2

Overview of 802.16e System

In this chapter, the overview of MAC and PHY of 802.16e will be first introduced. In the PHY layer, we present the some details of OFDMA frame structure, PHY slot and data mapping, subcarrier permutation, and adaptive modulation and coding (AMC). We also introduce the advanced features, such as adaptive antenna systems (AAS) and space time coding. In the MAC layer, the details of MAC PDU formats, fragmentation and packing, automatic retransmission request (ARQ), QoS based service classes, handoff / handover, and request-and-grant mechanism are presented. Finally, we use service and data flows to summary the flows of MAC and PHY layers.

2.1 802.16d/e system architecture

IEEE 802.16d architecture consists of two kinds of station: subscriber station (SS) and base station (BS) as shown in Figure 2-1. In IEEE 802.16e standard, the SS is replaced by mobile station (MS). The BS controls all the communicated resources in a cell. The communication path between MS and BS has two directions: downlink channel (from BS to MS) and uplink channel (from MS to BS). Each SS or MS can deliver voice and data using common interfaces with different QoS requirements.

2.2 Overview of Physical Layer

In the first design of the PHY specification for 10-66 GHz, line of sight transmission was deemed a practical necessity. With this condition, single carrier modulation was selected. But the SS or MS may be too low for a clear sight line to the BS antenna due to obstruction by buildings or trees, so the 2-11 GHz PHY is driven by the need for non-line of sight operation, which allows inexpensive and flexible consumer deployment and operation. Therefore, significant multipath propagation must be expected. The three 2-11 GHz air interface specification are:

- **WirelessMAN-SCa:** This uses a single-carrier modulated air interface.
- **WirelessMAN-OFDM:** A 256 carrier orthogonal frequency division multiplexing (OFDM) scheme. Multiple access is time division multiple access (TDMA) based.
- **WirelessMAN-OFDMA:** This uses orthogonal frequency division multiple access. In this system, multiple access is provided by assigning a subset of the carriers to an individual receiver. In 802.16e standard, the total number of subcarriers has four choices: 128, 512, 1024, and 2048 [2].

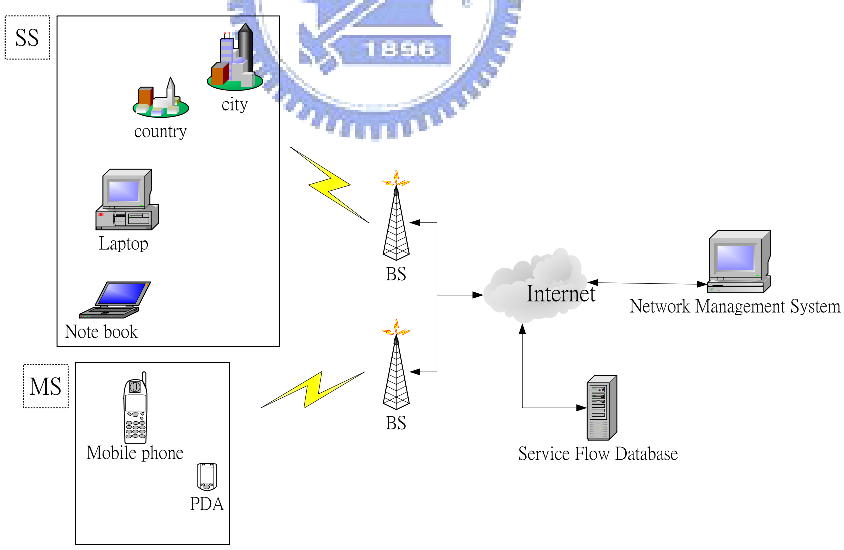

Figure 2-1 IEEE 802.16 system architecture

The two OFDM based systems are more suitable for non-line of sight operation due to the simplicity of the equalization process for multicarrier signals. The OFDM PHY has lower peak to average ration, faster fast Fourier transform calculation, and lesser requirements for frequency synchronization than the OFDMA PHY. But OFDMA's complex subcarrier permutation is aimed at better frequency reuse and easier cell planning. The permutation can minimize the probability of hits between adjacent sectors/cells by reusing subcarriers. The OFDMA enables multipath mitigation without using equalizers and training sequences and can provide single frequency cell coverage, where coverage problem exists and gives excellent coverage. The OFDMA also can offer frequency diversity by spreading the subcarriers all over the used spectrum and time diversity by optional interleaving of carrier groups in time. OFDMA offer high system performance by introducing up and downlink subchannelization and is well suited for full mobile applications [3][5].

In order to ensure global implementation, the standard has been defined with a variable channel bandwidth. The channel bandwidth can be an integer multiple of 1.25 MHz, 1.5 MHz, and 1.75 MHz with a maximum of 20 MHz [1].

2.2.1 PHY Frame Structure

The IEEE 802.16 standard supports two duplex methods: time division duplex (TDD), in which the uplink and downlink share the same bandwidth but do not transmit simultaneously, and frequency division duplex (FDD), in which the uplink and downlink operate on separate bandwidths, sometimes simultaneously. In the TDD mode of PHY, the uplink subframe follows the downlink subframe on the same bandwidth. The system uses a frame of 2, 2.5, 4, 5, 8, 10, 12.5, or 20ms [1]. This frame is divided into physical slots for the purpose of bandwidth allocation and identification of PHY transitions.

In this thesis, we focus on the 802.16e and OFDMA technique. Therefore, we only introduce the OFDMA frame structure [2]. Figure 2-2 shows an example of an OFDMA frame structure in TDD mode. We can see the frame has two dimensions, frequency dimension and time dimension. The frame can be flexibly filled up with two dimensions. The minimum unit of OFDMA frame is a OFDMA symbol. A OFDMA symbol own one subchannel which contains several subcarriers according to the permutation mode. The downlink subframe (DL) consists of the following: Preamble, Frame Control Header (FCH), DL-MAP, and UL-MAP, DL burst. The uplink subframe (UL) consists Ranging subchannel and UL burst.

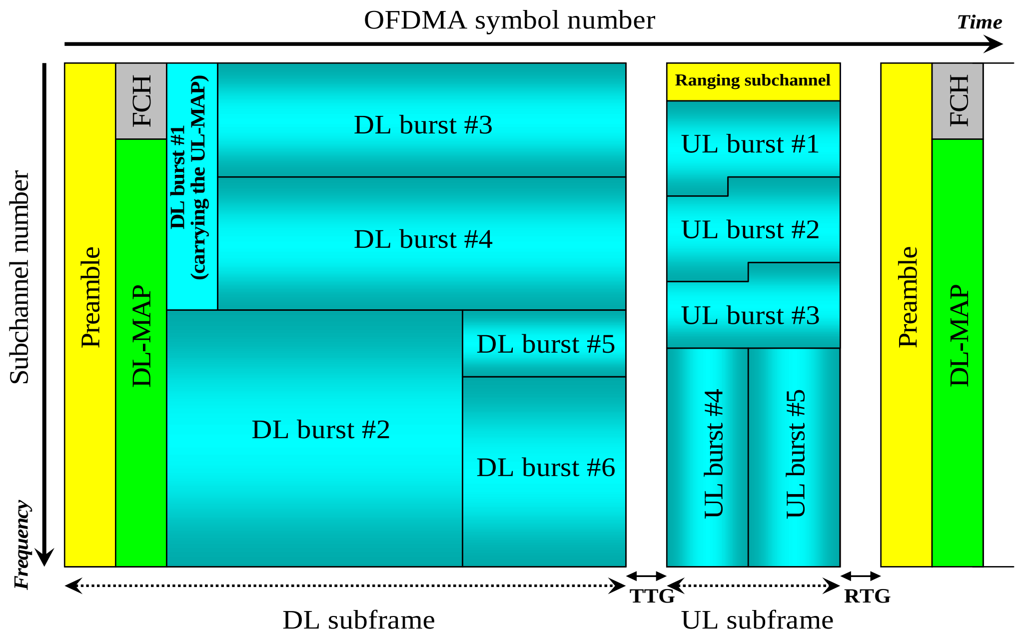

Figure 2-2 Example of an OFDMA frame (with only mandatory zone) in TDD mode

The first symbol is occupied by the preamble. The preamble is used for synchronization. The FCH contains the DL_Frame Prefix, and specifies the length of the DL-MAP message that immediately follows the DL_Frame_Prefix and the repetition coding used for the DL-MAP message. The DL-MAP and UL-MAP completely describe the contents of the DL and UL subframes. The DL-MAP and UL-MAP are used for resource allocation of DL and UL data bursts, include what bursts belong to MSs, modulation and coding schemes per burst, which are indicated by the Downlink Interval Usage Code (DIUC) and uplink Interval Usage Code (UIUC).

The UL-MAP also grants bandwidth to specific MSs. Therefore, the MSs can transmit data in their assigned uplink allocation. The DL bursts and UL bursts are used for data transmission of different users. The ranging subchannel is used for initial ranging, periodic ranging, and bandwidth requests. The initial ranging transmission shall be used by any MS that wants to synchronize to the system channel for the first time. Periodic-ranging transmissions are sent periodically for system time and power update. Bandwidth requests transmissions are for requesting uplink allocations from the BS. Ranging subchannels are dynamically allocated by the MAC layer and indicated in the UL-MAP. The transmit/receive transition gap (TTG) is a gap between the downlink burst and the subsequent uplink burst. It allows time for the BS to switch from transmit to receive mode and MSs to switch from receive to transmit mode. The receive/transmit transition gap (RTG) is a gap between the uplink burst and the subsequent downlink burst. It allows time for the BS to switch from receive to transmit mode and SSs to switch from transmit to receive mode.

2.2.2 PHY Slot and Data Mapping

The OFDMA slot is a minimum unit for data transmissions. One OFDMA slot occupies one subchannel and several OFDMA symbols as shown in Figure 2-3. For downlink Full Usage of Subcarriers (FUSC) using the distributed subcarrier permutation, one slot is one subchannel by one OFDMA symbol. For downlink Partial Usage of Subcarriers (PUSC) using the distributed subcarrier permutation, one slot is one subchannel by two OFDMA symbols. For uplink PUSC using either of the distributed subcarrier permutations, one slot is one subchannel by three OFDMA symbols. For uplink and downlink Band Adaptive modulation and coding (Band AMC) using the adjacent subcarrier permutation, one slot is one subchannel by one, two, three, or six OFDMA symbols. The FUSC, PUSC, Band AMC, and distributed/adjacent subcarrier permutations will be discussed later.

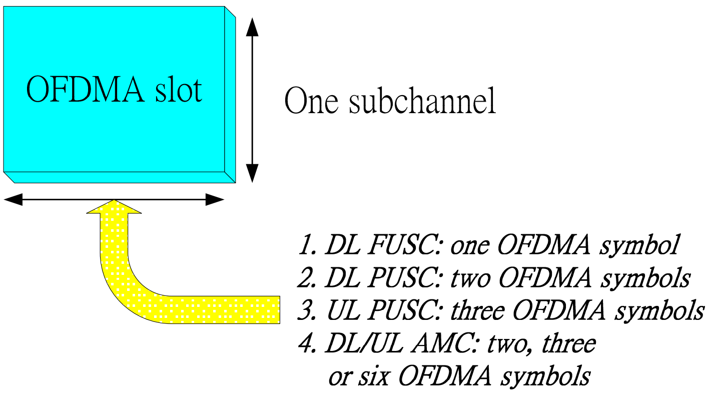

Figure 2-3 OFDMA slot structure

A Data Region is a two-dimensional allocation which contents a group of contiguous subchannels and OFDMA symbols. All the allocations refer to logical subchannels. A two dimensional allocation may be visualized as a rectangle, such as the 4 × 3 rectangle shown in Figure 2-4 [2]. Figure 2-4 also show the principle of data mapping. The minimum unit of data mapping is a OFDMA slot. The basic principles are to segment the data after the modulation block into blocks sized to fit into one OFDMA slot, and map the slots in the subchannels with higher priority than that in the OFDMA symbols. In other words, the data mapping method is frequency first which can meet the delay bound of real time services more easily.

2.2.3 Subcarrier Permutation

In IEEE 802.16 standard, subcarrier permutation is a method for assigning frequency subcarriers to subchannels. The allocation of subcarriers to subcahnnels is accomplished via a permutation rule, include permutation formula and permutation series. The permutation is a function of an IDCell parameter. This parameter is set by the network administrator, and it is different from a sector to other sectors. The resulting subchannels allocations are such the each subchannel overlap subchannels

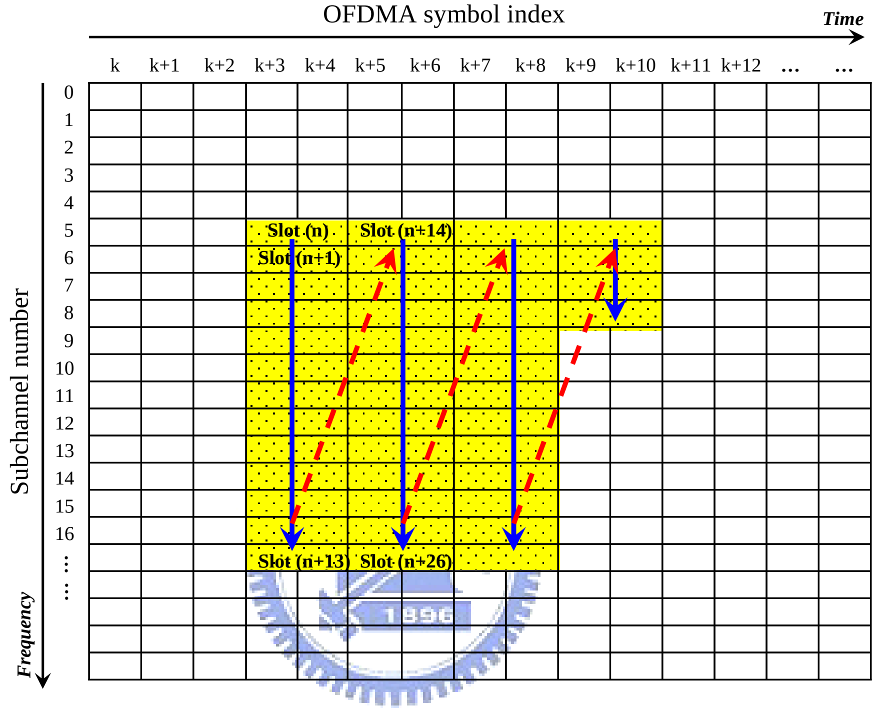

Figure 2-4 Example of mapping OFDMA slots to subchannels and symbols in the downlink PUSC mode

in adjacent subchannels cells only by a fraction of its subcarriers. So, subcarrier permutation can achieve averaging interference from neighboring cells, by using different basic carrier permutations between users in different cells. Interference within the cell is averaged by using allocation with cyclic permutations. There are two kinds of permutation modes: distributed and adjacent subcarrier permutation mode. The distributed subcarrier permutation mode can usefully average intercell interference and avoid fading effect by selecting subcarriers pseudo randomly. Using this characteristic, the burst interference that affect other cells at the same

OFDMA slots will be shared on the all subframe in the statistic, like Figure 2-5. The adjacent subcarrier permutation mode groups adjacent subcarriers into a subset and assign to users. If the adjacent subcarrier permutation is adopted, the frequency selected fading can be used and can get better multiuser frequency diversity as long as the channel does not change significantly during the scheduling process [6]. The IEEE 802.16 has three ways of grouping subcarriers into channels [2]:

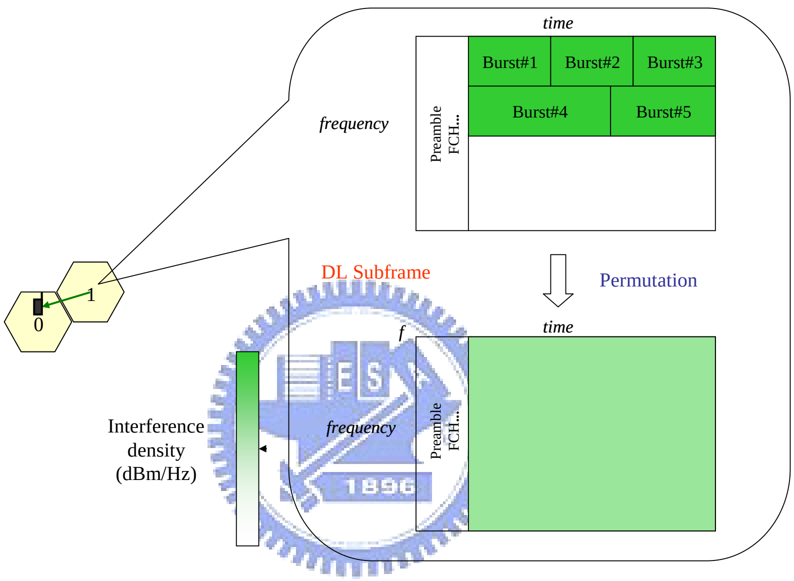

Figure 2-5 Example of the statistic intercell interference in distributed subcarrier permutation mode

- **Full Usage of Subchannels (FUSC):** This mode is downlink only and can use all subcarriers to do permutation for one subchannel. It can achieves the best frequency diversity by spreading subcarriers over entireband.
- **Partial Usage of Subchannels (PUSC):** This method can be used for both uplink and downlink. It groups subcarriers into tiles/clusters to enable fractional frequency re-use. This means the subcarriers of one subchannel only arrange from per groups and a group just offers one subcarrier. It still has distribution of subcarriers across band for each

subchannel.

- **Band Adaptive Modulation and Coding (Band AMC):** It is also suitable for uplink and downlink. In 802.16e standard, if Band AMC is used, all bandwidth will be divided into four sub-bands. Users will measure the transmitted qualities of sub-bands, respectively, and choose better bands for data transmissions. This permutation method is designed for the function of Band AMC and needs the Adaptive Antenna System (AAS). This method uses adjacent subcarriers for each subchannel for use with beamforming.

The PUSC and FUSC belong to distributed subcarrier permutation mode. The Band AMC belongs to adjacent subcarrier permutation mode.

2.2.4 Adaptive Modulation and Coding (AMC)

Adaptive modulation allows the IEEE 802.16 system to adjust the signal modulation scheme depending on the carrier to interference plus noise ratio (CINR) condition of the radio link. IEEE 802.16e OFDMA architecture supports multiple modulation levels: Quadrature Phase Shift Keying (QPSK), 16-state Quadrature Amplitude modulation (16-QAM), and 64-state QAM (64-QAM). When the radio link is good quality, the highest modulation scheme is used. The modulation-level will be dynamically adjusted based on the radio link quality. This feature allows the system to overcome time-selective fading. The OFDMA architecture also supports several coding schemes, like Convolution Code (CC), Low Density Parity Check Code (LDPC), Block Turbo Code (BTC), and Convolution Turbo Code (CTC). The BTC and CTC are optionally supported. Turbo coding can improve the ability of error correction, but increase decoding complexity and latency. The channel quality information channel (CQICH) and UL Sounding help change the modulation and coding scheme. In the distributed subcarrier permutation mode, the MSs use CQICH to report the average downlink CINR of BS preamble. In the adjacent subcarrier permutation mode, the MSs will report the different CINR values of four frequency bands on the CQICH. The BS can use this information to determine the downlink modulation and coding scheme (MCS) level. The UL Sounding, is a kind of uplink pilot signal, can be used to determine the uplink MCS level.

2.2.5 Adaptive Antenna Systems (AAS)

The IEEE 802.16 system enables the usage of intelligent antenna systems. Adaptive Antenna Systems (AAS) is used in the 802.16 specification to describe beam forming techniques. An array of antennas is used at the BS to increase gain to the intended MSs and nulls out interference to and from other MS's interference sources. AAS techniques can be used to enable Spatial Division Multiple Access (SDMA), where multiple MS's that are separated in space can receive and transmit on the same subchannel at the same time. By using beam forming, the BS is able to direct the desired signal to the different MS's and can distinguish between the signals of different MS's even though they are operating on the same subchannels.

2.2.6 Space Time Coding

Space time coding is an optional diversity scheme that can be provided in the downlink. In IEEE 802.16e, this transmit diversity scheme uses two or four antennas at the BS to transmit an STC encoded signal and one reception antenna on the MS side, in order to provide the gains that result from second-order diversity as shown in Figure 2-6. This scheme requires Multiple Input Single Output channel estimation. Decoding is very similar to maximum ratio combining. Each of two (or four) antennas transmits a different symbol in the first symbol time. The two (or four) antennas then transmit the complex conjugate of the symbols of first symbol time in the second symbol time. The resulting data rate is the same as without transmit diversity. If the STC and AAS with SDMA enable, this system can implement the function of Multiple Input, Multiple Output (MIMO).

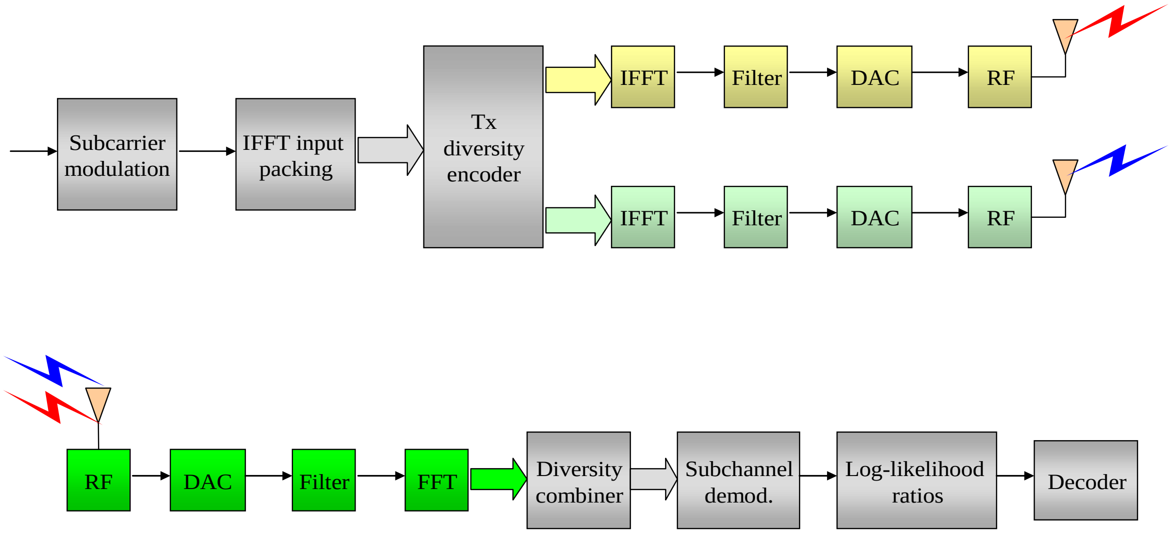

Figure 2-6 Example of Space time coding flow with two transmit antennas.

2.3 Overview of MAC Layer

The MAC layer of IEEE 802.16 was designed for point to multipoint (PMP) broadband wireless access applications and is connection-oriented. It is designed to meet the needs of very high data rate applications with a variety of quality of service (QoS) requirements on both downlink and uplink. The access and bandwidth allocation have been designed for offering a large number of MSs. The MAC layer protocol of IEEE 802.16 system is flexible and efficient over a wide range of different data traffic models. Under the situations of multiple connections per MS, multiple QoS levels per MS, and a large number of statistically multiplexed users, the MAC layer can keep the system efficiency. Because the bandwidth request-and-grant mechanism of MAC layer are designed to be scalable, efficient, and self-correcting. It takes advantage of a wide variety of request mechanisms, balancing the stability of contentionless access with the efficiency of contention-oriented access. In 802.16 system, the MAC layer is divided into three sublayer as shown in Figure 2-7.

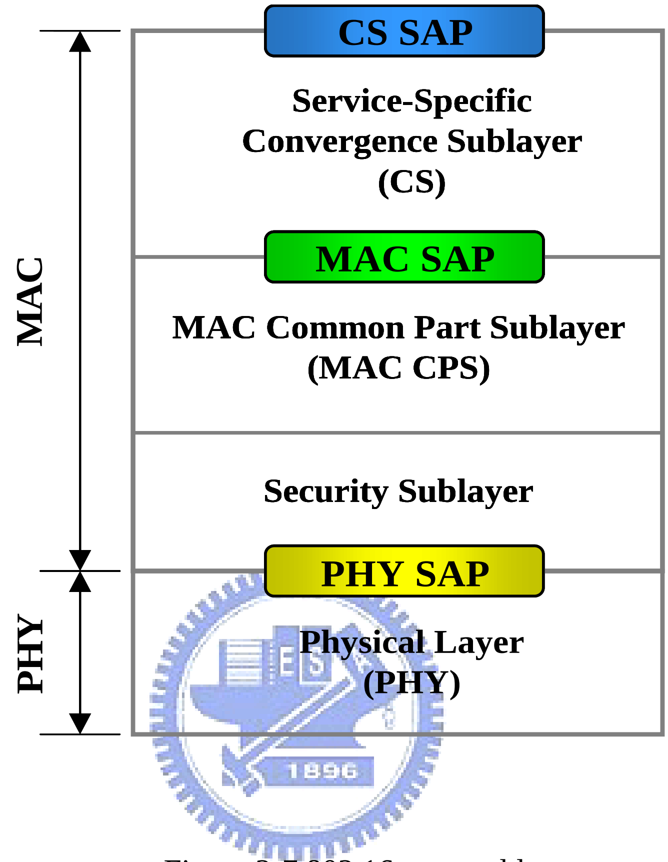

Figure 2-7 802.16 protocol layer

The MAC layer includes service-specific convergence sublayer that interface to higher layers, the core MAC common part sublayer that carries out the key MAC functions, and the security sublayer.

- **Service-Specific Convergence Sublayer (CS):** This sublayer is used to map the transport-layer-specific traffic to MAC layer that is flexible enough to efficiently carry any traffic types. The main function of this sublayer, as shown in Figure 2-8, is to classify service data units (SDUs) to the proper MAC connection and assign connection ID (CID) used for distinguishing different connections and different QoS requirements for each connection. It also enables bandwidth allocation. The detail of packet classifier can refer to [10]. Depending on the type of service, there are different mapping forms. IEEE 802.16 standard defines two types of service-specific convergence sublayers for mapping services to and from MAC connections. The ATM convergence sublayer is defined for ATM services, and the packet convergence sublayer is defined for mapping packet services such as IPv4, IPv6, and Ethernet. The convergence sublayers can also perform more accomplished functions such as payload header suppression and reconstruction to enhance the airlink efficiency.

- **Common Part Sublayer (CPS):** The common part sublayer is independent of the transport mechanism, such as TCP and UDP protocols. The responsibilities of this sublayer are support the main control mechanisms, such as QoS control, fragmentation

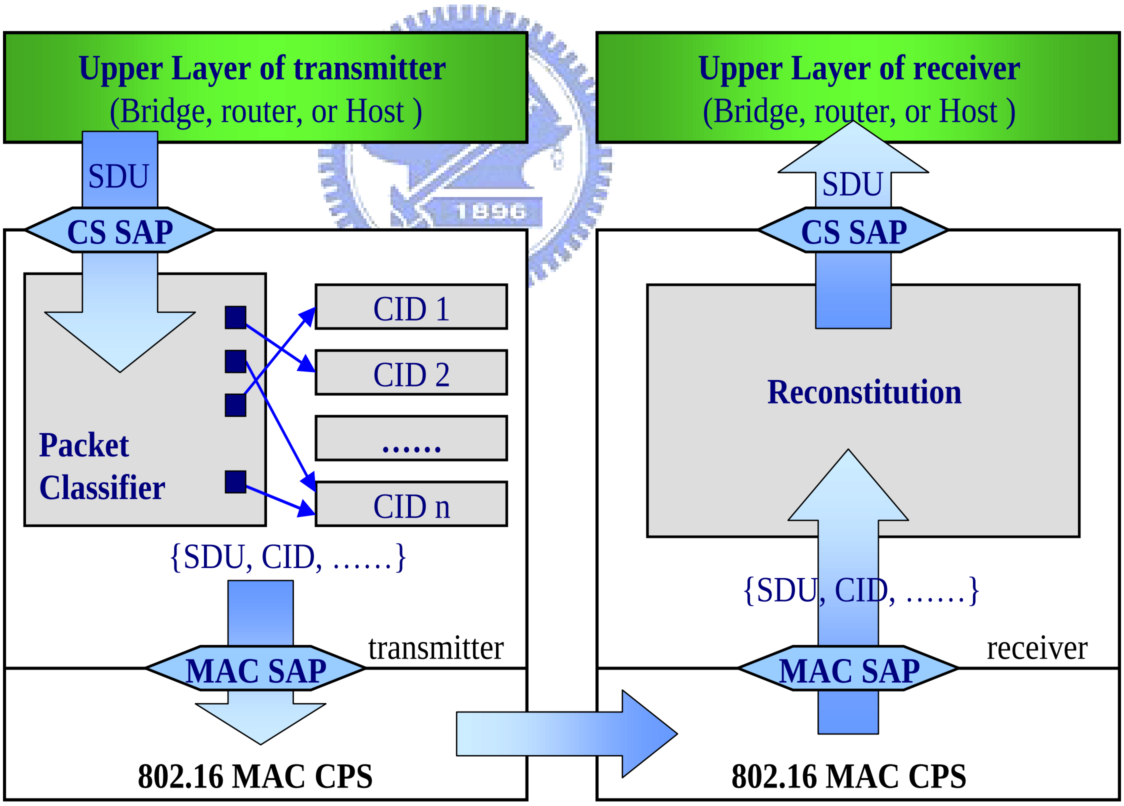

Figure 2-8 The functionality of Service-Specific Convergence Sublayer

and segmentation of MAC service data units (SDUs) into MAC protocol data units (PDUs), scheduling control, bandwidth requests and grants, admission control, ARQ and HARQ retransmission of MAC PDUs, and handoff process. We will introduce these later.

- **Security Sublayer:** This sublayer performs the authentication of network access, registration, key exchange, and encryption of PDUs. Privacy has two component protocols as follows: First, an encapsulation protocol for encrypting packet data across the fixed BWA network. This protocol defines a set of supported cryptographic suites, i.e., pairings of data encryption and authentication algorithms, and the rules for applying those algorithms to a MAC PDU payload. Second, a key management protocol (PKM) providing the secure distribution of keying data from BS to MS. Through this key management protocol, the MS and the BS synchronize keying data. In addition, the BS uses the protocol to enforce conditional access to network services.

In the rest chapter, we will discuss the functionality of CPS.

2.3.1 MAC PDU Formats

The MAC PDU is a data exchanged unit between the MAC layer of the BS and its MSs. A MAC PDU consists of a 48 bits MAC header, a variable-length data payload, and an optional 32 bits Cyclic Redundancy Check (CRC). There are two types of MAC headers: a generic header and a Bandwidth Request (BR) MAC header. The generic header is used to transmit data or MAC messages. The BR header is used by the MS to request more bandwidth on the uplink and bandwidth request MAC PDUs contain no payload. The maximum length of the MAC PDU is 2048 bytes, including header, payload, and CRC. The MAC also defines six types of subheaders which may be present in a MAC PDU with generic MAC header: Mesh, Fragmentation, Packing, Fast-feedback allocation, extended and Grant Management subheader. The subheaders of each PDU may be inserted in MAC PDUs immediately following the Generic MAC header. Grant Management subheader is used to transport bandwidth allocation states between the BS and MS. Fragmentation and Packing subheaders are used to utilize the bandwidth allocation efficiently. The fragmentation subheader indicates the presence and orientation in the payload of any fragments of SDUs. The packing subheader is used to indicate the packing of multiple SDUs into a single PDU. Fast-Feedback allocation is designed for MIMO operation. Mesh subheader is only useful in mesh mode.

2.3.2 Fragmentation and Packing

In the 802.16 system, MAC SDUs from CS will be formatted according to the MAC PDU format in the CPS, possibly with fragmentation and/or packing. After traversing the airlink, MAC PDUs are reconstructed back into the original MAC SDUs. Because airlink resources are very precious, the 802.16 MAC layer performs both fragmentation of MAC SDUs and packing of MAC SDUs in the CS. The 802.16 system takes advantage of incorporating the packing and fragmentation processes with the bandwidth allocation process to maximize the flexibility and efficiency. Fragmentation is the process in which a MAC SDU is divided into one or more MAC SDU fragments. Then, these fragments are packed into the payload field of the PDUs that the payload field is too small to transmit a complete SDU. Figure 2-9 shows the process of fragmentation.

Packing is the process in which multiple MAC SDUs are packed into a single MAC PDU payload in order to fill up the remaining payload field preventing wastage of resources. The CPS allows simultaneous fragmentation and packing for efficient use of the bandwidth as shown in Figure 2-10. Both processes may be initiated by either a BS for a downlink connection or an MS for an uplink connection.

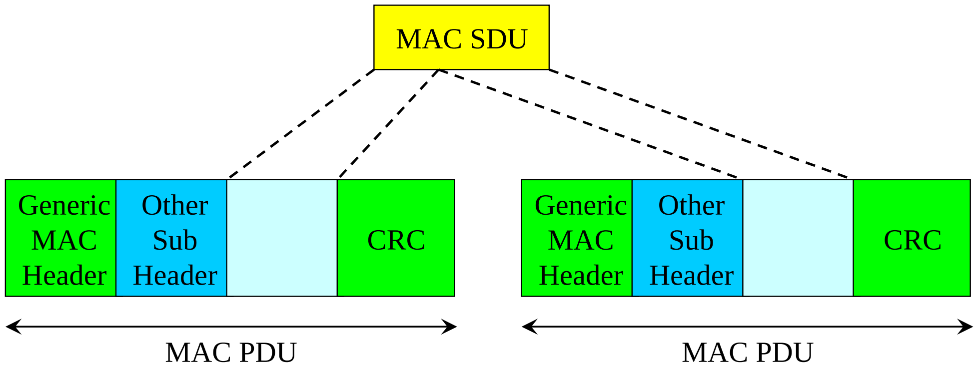

Figure 2-9 Single SDU forms multiple PDUs

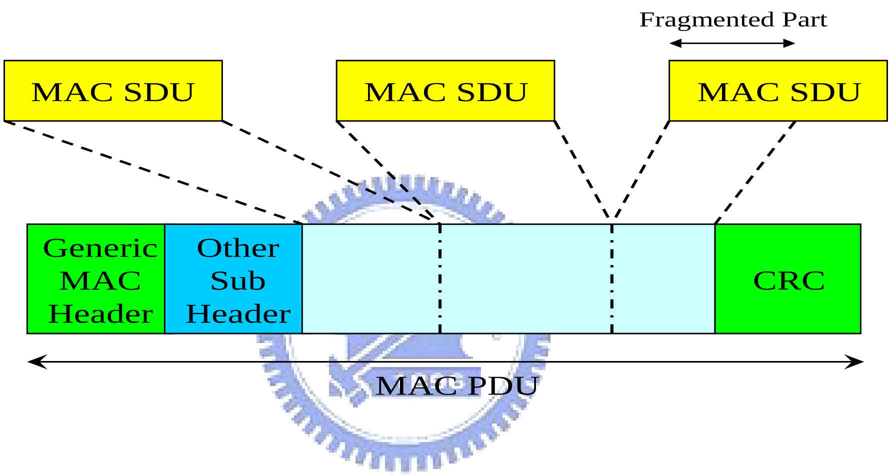

Figure 2-10 Multiple SDUs form single PDU

2.3.3 Automatic Retransmission Request (ARQ)

The automatic retransmission request (ARQ) can request the retransmission of part of a SDU. A MAC SDU is logically partitioned into ARQ blocks whose length is specified by ARQ_BLOCK_SIZE and MAC PDU is formatted by integer number of ARQ blocks as shown in Figure 2-11. Once an SDU is partitioned into a set of blocks, partitioning remains in effect until all blocks of the SDU are successfully delivered to the receiver, or the SDU is discarded by the transmitter state machine. The processing is the process of retransmitting MAC ARQ blocks of one PDU that have been lost or in error. The 802.16 MAC uses a simple

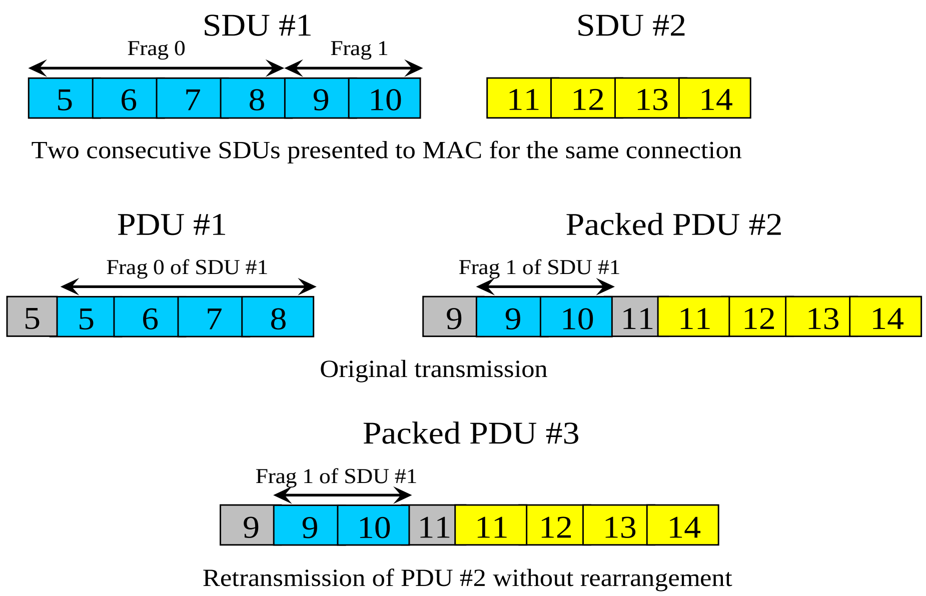

Figure 2-11 Example of ARQ frame structure

sliding window based approach, where the transmitter can transmit up to a negotiated number of blocks without receiving an acknowledgement. The receiver sends acknowledgement or negative acknowledgement messages to indicate to the transmitter which ARQ blocks of the PDUs have successfully been received and which have been lost. If any block of one PDU is lost or in error, the transmitter will retransmit all ARQ blocks of this PDU. When ARQ blocks of a PDU are acknowledged to have been received, the sliding window is moved forward. The 802.16 system also optionally provides the rearrangement of the retransmission blocks. If the ARQ blocks of one PDU need be retransmitted, it can be divided into different PDUs which share the blocks, just like the PDU #3 of Figure 2-11 can be changed to two retransmitted PDUs as shown in Figure 2-12. Hybrid automatic repeat request (HARQ) scheme is an optional part of the MAC. HARQ may be supported only for the OFDMA PHY. Two main variants of HARQ are supported—Chase Combining and Incremental Redundancy (IR). MS may support either Chase Combining or IR. For IR, the PHY layer will encode the HARQ packet generating several versions of encoded subpackets. Each subpacket shall be uniquely

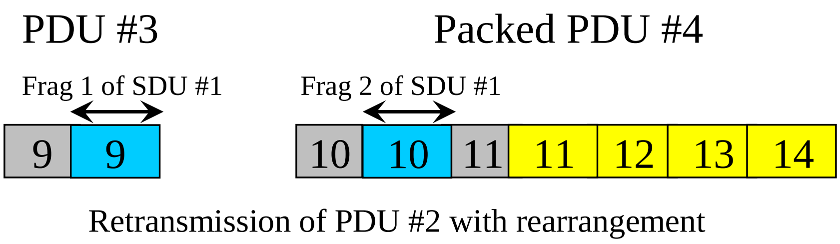

Figure 2-12 Example of rearrangement

identified using a subpacket identifier (SPID). For Chase Combining, the PHY layer shall encode the HARQ packet generating only one version of the encoded packet. As a result, no SPID is required for Chase Combining. For downlink HARQ operation, the BS will send a version of the encoded HARQ packet. The SS will attempt to decode the encoded packet on this first HARQ attempt. If the decoding succeeds, the SS will send an ACK to the BS. If the decoding fails, the SS will send a NAK to the BS. In response, the BS will send another HARQ attempt. The BS may continue to send HARQ attempts until the SS successfully decodes the packet and sends an acknowledgement The utilization of HARQ is on a per-connection basis, that is, it can be enabled on a per CID basis. Two implementations of HARQ are supported: first, per-terminal, that is, HARQ is enabled for all active CIDs for a terminal, and second, per-connection, that is, it can be enabled on a per CID basis. The two implementation methods shall not be employed simultaneously on any terminal. If HARQ is supported, MS shall support per-terminal or per-connection implementation. A burst cannot have a mixture of HARQ and non- HARQ traffic. One or more MAC PDUs can be concatenated and an HARQ packet formed by adding a CRC to the PHY burst. The good overview of HARQ can be found in [7][8] [9].

2.3.4 QoS Based Service Classes

The 802.16e MAC supports QoS differentiation for different types of applications which might transmit on the 802.16e system. The 802.16e standard defines five types of services:

- **Unsolicited Grant Service (UGS):** The UGS is designed to support real-time service flows that generate transport fixed-size data packets on a periodic basis, such as T1/E1. The service offers fixedsize grants on a real-time periodic basis, which eliminate the overhead and latency of MS bandwidth requests in order to meet the delay and delay jitter requirements of the underlying services and assure that grants are available to meet the flow's real-time needs. When used with UGS, the grant management subheader includes the poll-me bit. Connections configured with UGS are not allowed to utilize random access opportunities for bandwidth requests.
- **Real-Time Polling Service (rtPS):** The rtPS is designed to support real-time service flows that generate transport variable size data packets on a periodic basis. The service offers real-time, periodic, unicast request opportunities, which meet the flow's real-time needs and allow the MS to specify the size of the desired grant. This service requires more request overhead and latency than UGS, but supports variable grant sizes for optimum data transport efficiency. The rtPS is well suited for connections carrying services such as VoIP or moving pictures experts group (MPEG) video.
- **Extended Real-Time Polling Service (ertPS):** The ertPS is designed to support real-time service flows that generate variable size data packets on a periodic basis, such as Voice over IP services. Extended rtPS is builds on the efficiency of both UGS and rtPS. The BS shall provide unicast grants in an unsolicited manner like in UGS, thus saving the latency of a bandwidth request. However, whereas UGS allocations are fixed in size, ertPS allocations are dynamic.
- **Non-Real-Time Polling Service (nrtPS):** The nrtPS is designed to support delay-tolerant data streams consisting of variable-sized data packets for which a

minimum data rate is required and may utilize random access transmit opportunities for sending bandwidth requests. In general, services carried on these connections tolerate longer delays and are rather insensitive to delay jitter. The nrtPS is suitable for Internet access with a minimum guaranteed rate, such as FTP and HTTP.

- **Best Effort Service (BE):** The BE service is designed to support data streams for which no minimum service level is required and therefore may be handled on a space-available basis, such as E-mail. Neither throughput nor delay guarantees are provided.

The service classes are distinguished by the Service-Specific Convergence Sublayer (CS). When the packets are classified in the CS, the connection which they are placed into is chosen based on the type of QoS required of applications.

2.3.5 Handoff (Handover)

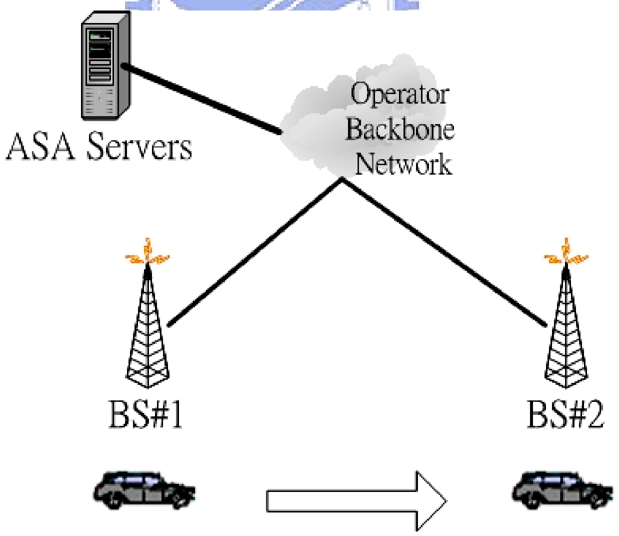

Figure 2-13 Example of Network Model for Handoff Process

Figure 2-13 shows an example where two BSs are connected to operator backbone network. BS1 is the Serving BS for an MS and BS2 is the Neighbor BS. For handoff process, Serving BS broadcasts MOB NBR-ADV message which contains information of Neighbor

BSs and the MSS can scan neighbor BSs to measure the signal power of Neighbor BSs periodically. If the MS moves closer to BS2, BS2 might become a Target BS for a handoff and the MS sends a handover request message to Serving BS. The MS could select several BSs as candidate of Target BS. Then, the Serving BS and Neighbor BSs exchange the handoff information of MS, negotiate handover capability through backbone network and select one Target BS finally. The serving BS sends handover response message to the MSS trying to handover. Then, the MS releases the connection with Serving BS and makes a new connection with Target BS and conducts network re-entry process. The Target BS could get security information of MS from Serving BS or authentication and service authorization (ASA) server. After network re-entry process, which includes synchronization with new downlink, ranging and synchronization with uplink, re-authorization, and re-registration procedures, the MS can send or receive traffic. After releasing the connection with Serving BS, the MS have to synchronize with new downlink of Target BS and obtain downlink parameters with DL-MAP message. After synchronization, the MS shall wait for UCD message from the BS in order to retrieve a set of transmission parameters for a possible uplink channel. The MS conducts ranging process to get correct timing offset and power adjustments. For this ranging process, Target BS can use Fast-Ranging IE to give a non-contention based ranging opportunity to the MSS. If the MSS fails to receive Fast-Ranging IE message, it should conduct contention based ranging process by transmitting the CDMA codes. The remained network re-entry processes are re-authorization and re-registration. If the serving BS transfers all MAC states and security information of the MSS to Target BS by backbone network during the handover process, this MSS could skip re-authorization and re-registration processes. The more detail of handoff process can be founded in [11][12][13].

2.3.6 Request-and-Grant Mechanism

In IEEE 802.16 standard, there are two kinds of transmitting the bandwidth request: contention mode and contention-free mode (polling). The Bandwidth Request (BW-Request) message may be transmitted during any uplink allocation, except during initial ranging intervals. BW-Requests may be incremental or aggregate. When the BS receives an incremental BW-Request, it shall add the quantity of bandwidth requested to its bandwidth. When the BS receives an aggregate BW-Request, it shall replace its bandwidth. In contention mode, MSs send BW-Request during the contention period. Contention is resolved using back-off mechanism. In contention-free mode, BS polls each MS and MSs reply by sending BW-request. Polling is the process by which the BS allocates to the MSs bandwidth specifically for the purpose of making bandwidth requests. There are three kinds of polling: Unicast, Multicast, and Poll-me bit (PM). In the unicast polling, the MS is polled individually and has sufficient bandwidth to respond with a BW-Request. If insufficient bandwidth is available to individually poll many inactive MSs, some MSs may be polled in multicast groups or a broadcast poll using multicast polling. MSs with currently active UGS connections may set the PM bit in a MAC packet of the UGS connection to indicate to the BS that they need to be polled to request bandwidth for non-UGS connections. Contention-free mode is more suitable for real time applications due to the predictable signaling delay of the polling scheme. In the IEEE 802.16 MAC, there are two classes of MS, differentiated by their ability to accept bandwidth grants simply for a connection or for the MS as a whole. With the grant per connection (GPC) class of MS, bandwidth is explicitly granted to a connection, and the MS uses the grant only for that connection. With the grant per SS (GPSS) class, MSs are granted bandwidth aggregated into a single grant to the MS itself. The GPSS MS needs to be more intelligent in its handling of QoS. The two classes of SS allow a trade-off between simplicity and efficiency. GPC is simpler but less efficient than GPSS. Additionally, the ability of the GPSS MS to react more quickly to the needs of the PHY and those of connections enhances system performance. GPSS is the only class of MS allowed with the

10–66 GHz PHY.

2.4 Service and Data Flow

In Figure 2-14, we show the service flow of 802.16 system. IEEE 802.16 standard can support multiple services with different requirements of QoS, such as data, voice, video. The MAC layer defines QoS signaling mechanisms and functions that can control BS and MS data transmissions. The CS layer of downlink and uplink classifies the different QoS services into

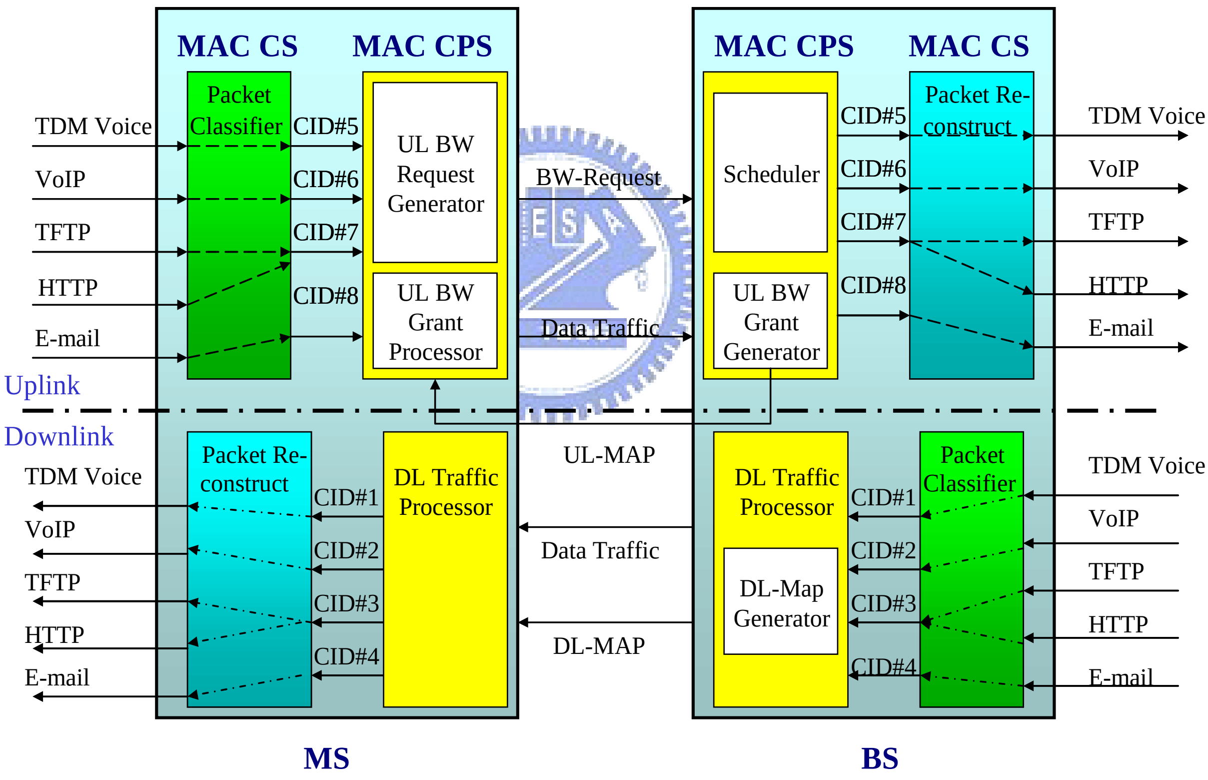

Figure 2-14 Example of 802.16 Service Flow

connections and assigns the connection with a unique connection indicator (CID). Then, the data of the connections are forwarded to the appropriate queues. On the downlink, the transmission is relatively simple because the BS is the only one that transmits during the downlink subframe. The BS schedules the all downlink connections and the data are broadcasted to all MSs and an MS uses the DL-MAP information to picks up the data belong to it. The BS determines the number of slots that each MS will be allowed to transmit in an uplink subframe. This information is broadcasted by the BS through UL-MAP at the beginning of each frame. UL-MAP contains information element (IE), which include the transmission opportunities, i.e. the slots in which the MS can transmit during the uplink subframe. After receiving the UL-MAP, each MS will transmit data in the predefined slots as indicated in IE. The BS uplink-scheduler determines the IEs using BW-request sent from MSs to BS.

In Figure 2-15, we show the data flow of OFDMA TDD mode. The service traffic got into the MAC layer will be mapped into MAC SDUs. A MAC SDU is divided into one or more MAC PDUs or multiple MAC SDUs are packed into a single MAC PDU through the operation of fragmentation or packing. Multiple MAC PDUs may be concatenated into bursts having the same modulation and coding in either uplink or downlink directions. The bursts are mapped into OFDMA frame and transmitted after subcarrier permutation.

In this chapter, we brief introduce the IEEE 802.16e system and you can refer to [3][6][14~18] about those insufficient details. The wide bandwidth allocation and QoS mechanisms are provided in the 802.16 standard. But the details of scheduling, admission control and reservation management are left undefined and provide an important mechanism for vendors to differentiate their equipment. In this thesis, to effectively support real-time and non-real-time services, the system performance of 802.16e with different radio resource managements in scheduler controls and admission control criterion will be evaluated.

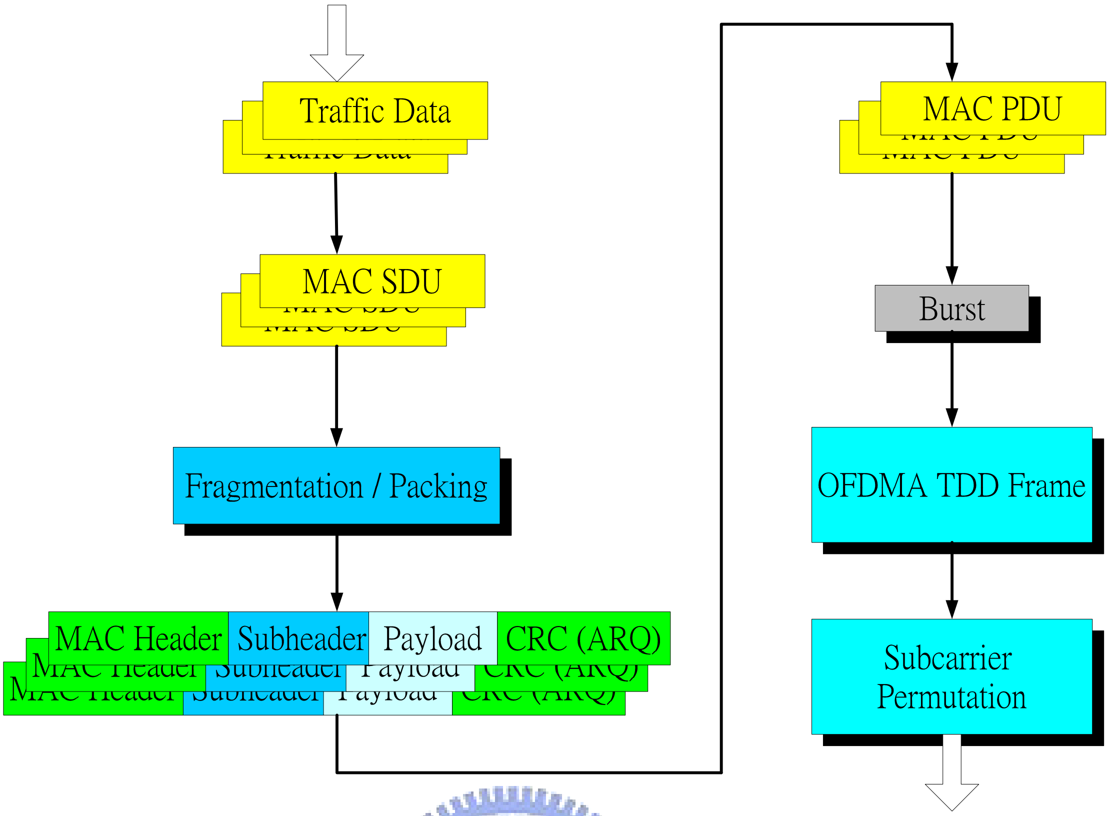

Figure 2-15 Example of Data Flow of OFDMA TDD Mode

Chapter 3

Simulation Setup

In this chapter, the IEEE 802.16e system level simulation platform will be described. The details of system architecture and simulation parameters are going to be presented. Then, the Link Budget, such as path loss and shadow fading, is set to be suitable for IEEE 802.16e standard. The setting of basic radio resource managements, such as power control, rate control (AMC), scheduling controls, and handoff method, is showed in this chapter. Finally, the traffic models of simulation are introduced.

3.1 The Architecture of Mobility Platform

In the simulation of a mobile system, the interference from other cells is an important element that would affect the overall performance of single cell. This effect need to be considered into the simulation. When the number of simulation cells increase, it will cause high load of the simulation time and computation. Between these two tradeoffs of accurate simulation and computing cost, we consider two-tier interference cells. According to approximate the cell coverage with a hexagon, we consider nineteen cells in our simulation as shown in Figure 3-1. From Figure 3-1, we can see that only the central cell completely has two-tier interference cells, the other cells can not find out the symmetric two-tier interference cells. Even if we use nineteen cells to simulate, only the statistic simulation value of the central cell can be referred, causing a lower simulated efficiency. Hence, we adopt a wrap around technique as shown in Figure 3-2. This technique can make any of nineteen cells owns complete two-tier interference cells. The main ideal is the lacks of any two-tier interference

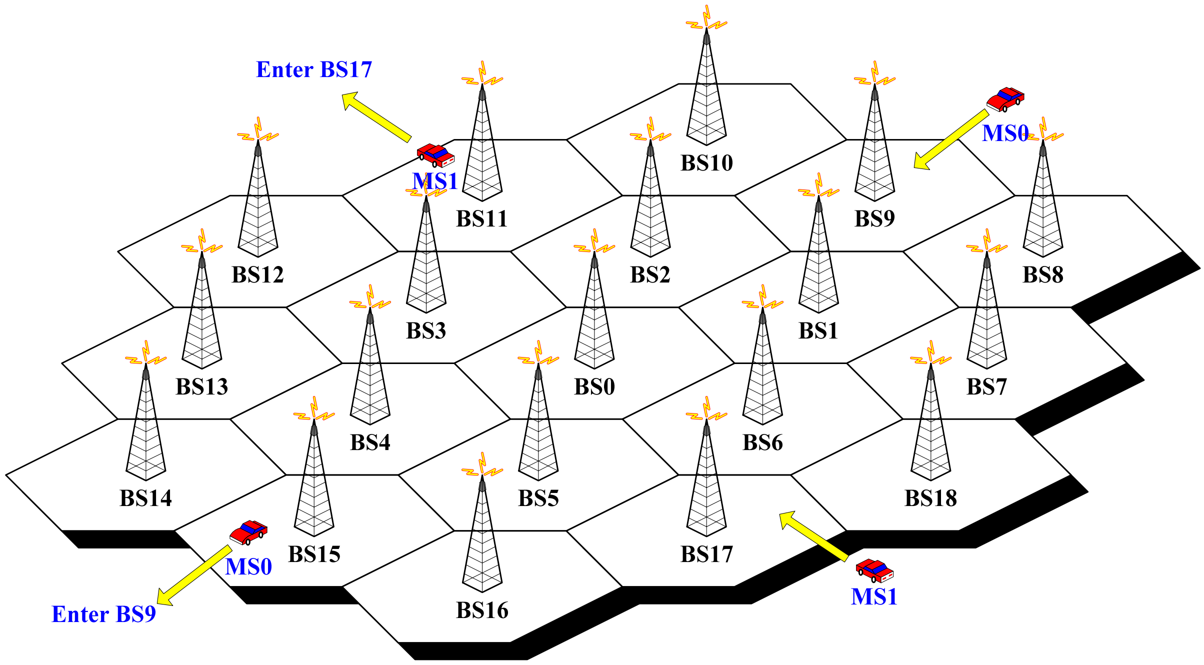

Figure 3-1 Cell Structure of System Simulation

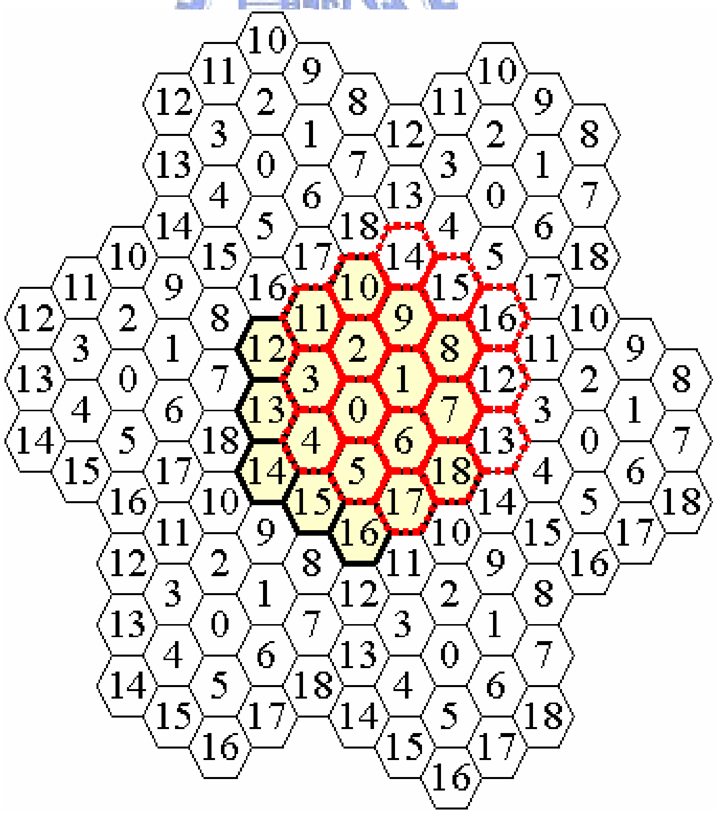

Figure 3-2 Example of Wrap Around

cells of a specific cell are copies from the simulated cells which are besides the two-tier interference cells of the specific cell. Through the clever arrangement, any cell has complete, symmetric, and different two-tier interference cells. Because the cell owns whole interference after wrap around, the statistic simulation value of nineteen cells would be meaningful.

The cell radius which we set is 1 km [19]. This approximate cell coverage is a result from a plan of Link Budget. The total cell bandwidth that we choose is 6 MHz [1]. In our simulation platform, a cell is divided into three sectors as shown in Figure 3-3. Each sector has the different antenna direction and a regular pattern of deployment. The sector architecture in 802.16e system can reduce the transmission power of BS antenna and intercell interference. But it still has a small part of intercell interference in different sectors of distinct BSs due to subcarrier permutation. This characteristic is very difficult to simulate, so we assume a sector would produce interference to other sectors which have the same direction.

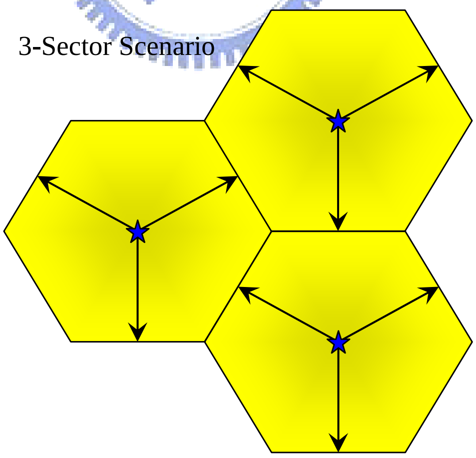

Figure 3-3 Example of sector deployment

In our simulation platform, the setting of antenna pattern uses the 3GPP's model [20] as shown in (1).

$$A(q) = -\min\left[\ 12\left(\frac{q}{q_{3dB}}\right)^2,\ \ A_m\right]\ \ where -180^{\circ} \le q \le 180^{\circ} \qquad (1)$$

, where q_{3dB} is $70°$ and A_m is $20dB$.

The cell's frequency reuse factor in our simulation platform has two select, one (1x1) or three (1x3). The total bandwidth of frequency reuse factor 1 is 6 MHz in our simulation. Frequency reuse factor 3 need triple bandwidth, 18 MHz. The cells with frequency reuse factor 3 has longer distance between two cell used the same bandwidth than frequency reuse factor 1. Therefore, the interference in frequency reuse factor 3 scenario is lower due to stronger interference pathloss caused by longer distance. But the cost is the triple bandwidth need be used. Figure 3-4 shows the deployment of frequency reuse factor. In frequency reuse factor 1 scenario, the overall cells use the same bandwidth. In frequency reuse factor 3 scenario, only partial cells use the same bandwidth. For instance, #0 (# means number), #7, #9, #11, #13, #15, #17 cells use the identical cell bandwidth in Figure 3-4.

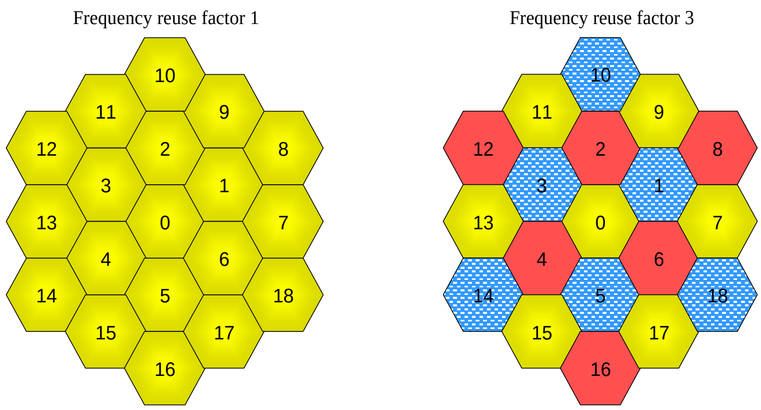

Figure 3-4 Example of the Deployment of Frequency Reuse Factor

3.2 The Architecture of Frame Transmission

In this thesis, we focus on the downlink transmission and use the OFDMA technique with TDD mode. The IEEE 802.16 standard can support an asymmetric downlink and uplink transmission of TDD mode, which adjusts the ratio according to the traffic loading of downlink and uplink transmission. In our simulation, we use a simple assumption. We assume the downlink and uplink transmission have a ratio of equality and they use a half frame, respectively. In 2.3.1, we introduce the time duration of a frame. In our simulation, the frame length we used is 10 ms. The frame structure we used is 2048-FFT OFDMA downlink carrier allocations –PUSC mode defined in the standard. The carrier distribution is shown in Table 3-1. In the 2048 subcarriers, only 1680 subcarriers can carry data information and other subcarriers are used for guard band and dc subcarrier.

Table 3-1 2048-FFT OFDMA downlink carrier allocations with PUSC

Subcarrier types	**number**
Total subcarriers	2048
DC subcarriers	1
Guard subcarriers	184 (Left), 183 (Right)
Sub-channels	60
Data sub-carriers within each sub-channel	24

The length of an OFDMA symbol is an important parameter which involves how many symbols a frame has. The IEEE 802.16 standard provides the method to calculate the OFDMA symbol time, just like (2).

$$\begin{aligned}
&\text{Sampling frequency} &&: F_S = floor(n \times BW/\ 8000\ ) \times 8000 \\
&\text{Subcarrier spacing} &&: \Delta f = F_S / N_{FFT} \\
&\text{Useful symbol time} &&: T_b = 1/\Delta\Delta \\
&\text{CP Time} &&: T_g = G \times T_b \\
&\text{OFDMA symbol time} &&: T_S = T_b + T_g \\
&\text{Sampling time} &&: T_b / N_{FFT}
\end{aligned} \tag{2}$$

BW is the nominal channel bandwidth. N_{FFT} is the smallest power of two greater than Nused which is the number of used subcarriers includes the DC subcarrier. n is a sampling factor. This parameter, in conjunction with BW and Nused determines the subcarrier spacing, and the useful symbol time. This value is set to 8/7. G is the ratio of CP time to "useful" time. The following values shall be supported: 1/32, 1/16, 1/8, and 1/4. From these equations, we can get the OFDMA symbol time with our settings. We use 6 MHz bandwidth and set G value equal to 1/32. The OFDMA symbol time will be 308 μs. Using this value, one subchannel has sixteen OFDMA symbols roughly per frame. Under the equal downlink and uplink transmitted ratio, a subcannel of downlink or uplink subframes owns eight OFDMA symbols, respectively. In the downlink PUSC mode, a transmission slot occupies two OFDMA symbols, so one subchannel has four slots in the downlink subframe. The total slots of a downlink subframe of PUSC mode are 240 (4*60).

3.3 Link Budget

The link budget settings of downlink transmission in our simulation are as far as possible to match the IEEE 802.16e real environment. In IEEE S802.16e-03/23 document [19], it makes deployment scenario assumptions for 802.16e, like Table 3-2. In our simulation, we adopt the outdoor vehicular scenario, which the BS transmitted power is 46 dBm, the BS antenna gain is 17 dBi, the SS (MS) antenna gain is 3dBi on the downlink transmission. The BS back off which is used to avoid the RF circuit working in the non-linear region due to the peak-to-average power ratio (PAPR) of OFDM system is 5 dB. The common usage value of thermal noise density is -173.93 dB / Hz. The receiver noise figure of MSs is 9dB [21].

Table 3-2 Link Budget Parameter of 802.16e system

Scenario / Parameter	Indoor	Outdoor to indoor	Outdoor vehicular
BS Tx power	27 dBm (0.5 W)	36 dBm (4 W)	46 dBm (40 W)
MS Tx power	17 dBm	17 dBm	27 dBm
BS ant gain	6 dBi	17 dBi	17 dBi
MS ant gain	0 dBi	0 dBi	3 dBi
BS ant height		15 m	30 m

In wireless channel, the transmitted signals will suffer the fading effect, which can change the original signals. The fading can be divided into three type: pathloss, shadow fading, and fast fading (multipath and doppler effect). In our simulation, we only consider the pathloss and shadow fading. The fast fading will be used in the future work. The pathloss model is used to present the signal strength decreases with increasing distance between transmitter and receiver. In Winner D5.4 document [22], it provides several pathloss models, such as Table 3-3. Because the cell radius is 1 km in our simulation and the signal transmission in 2~11 GHz is non-line of sight (NLOS), the C2 scenario is more suitable and we use it in our simulation.

Table 3-3 Pathloss Model Scenarios

Scenario		Path-loss [dB]	Shadow fading standard dev.	Applicability range
A1	LOS	18.7 $\log_{10}(d[\mathrm{m}])$+46.8	σ = 3.1 dB	3m < d < 100m
A1	NLOS	PL(d) = 36.8 $\log_{10}(d[\mathrm{m}])$+38.8	σ = 3.5 dB	3m < d < 100m
B1	LOS	22.7 $\log_{10}(d[\mathrm{m}])$+41.0	σ = 2.3 dB	10m < d < 650m
B1	NLOS	0.096 d_1[m]+65+ (2.8-0.024 d_1[m]) $\log_{10}(d_2[\mathrm{m}])$	σ = 3.1 dB	10m < d_1 < 550m w/2 < d_2 < 450m
C2	NLOS	35.0 $\log_{10}(d[\mathrm{m}])$+31.5	σ = 8 dB	50m < d < 5km
D1	LOS	PL(d) = 21.5 $\log_{10}(d[\mathrm{m}])$+44.6 = 40.0 $\log_{10}(d/d_{BP})$+44.6+ 21.5 $\log_{10}(d_{BP})$	σ = 3.5 dB σ = 6.0 dB	30m < d < d_{BP} d_{BP} < d < 10km
D1	NLOS	PL(d) = 25.1 $\log_{10}(d[\mathrm{m}])$+55.8	σ = 8.0 dB	30m < d < 10km

The main reason forms shadow fading is from the shelters, like buildings, or mountain, on the signal transmitted path. According to the test result of the real wireless environment, we can know the variant of shadow fading is a log-normal distribution statistically. So, we can use the log-normal distribution to produce the shadow fading effect. The standard deviation of this distribution is based on the simulation environment. In our simulation, we use 8 dB for the standard deviation [22]. When the user is fixed, the shadow fading effect will not alter. On the other hand, the shadow fading changes with different locations at the mobile user. In the different simulation points, we can use the log-normal distribution to produce a value for the shadow fading, respectively. But this method has a problem, the time between two neighbor simulation points is very small so that the mobile user location will not change very obvious even at high mobile speed. It means the variance the shadow fading will not be large and have a correlated relationship between two neighbor time points. Hence, we use the concept of a correlation model, called Gudmundson's correlation model [23], in (3).

$$r(\Delta x) = e^{-\frac{|\Delta x|}{d_{cor}}\ln 2} \qquad (3)$$

,where ρ is the auto-correlation constant between two simulation sample points, Δx is the distance of two sample points and is a function of sampling times between them, sampling duration, and user speed. The d_{cor} is de-correlation distance and the values in the suburban macro, urban macro, and urban micro environments are 200m, 50m, and 5m, respectively. In our simulation, we use 5m as our parameter. Using log-normal distribution and correlation model, the simulation can get more actual shadow fading.

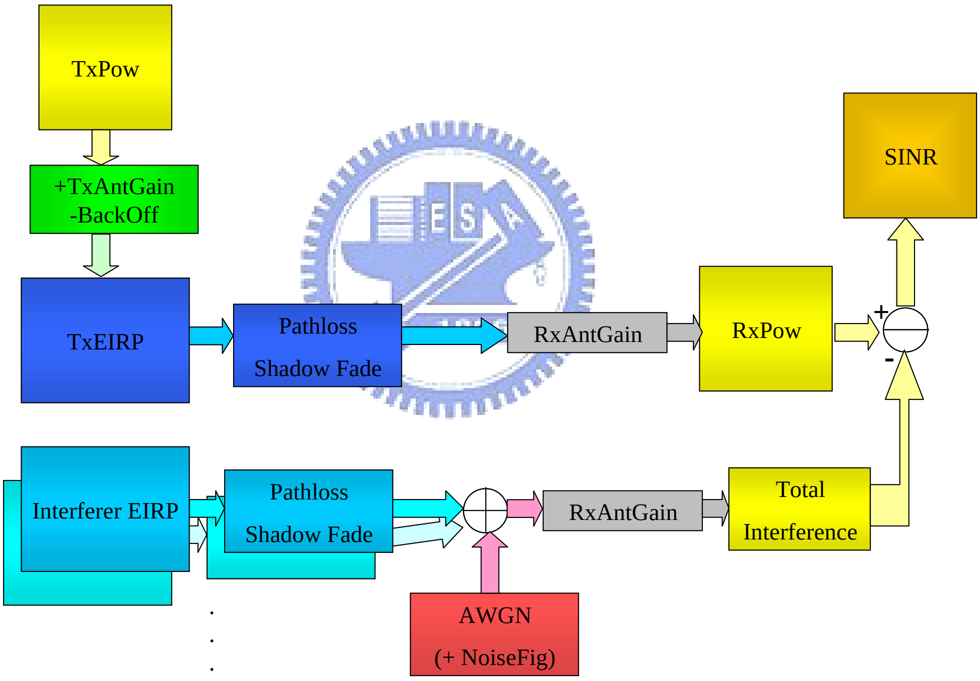

Figure 3-5 Example of SINR computation

In Figure 3-5, we present the flow of signal-to-interference and noise ratio (SINR) computation. The OFDMA technique uses the multiple carrier to transmit signal and we should compute carrier-to-interference and noise ratio (CINR), not SINR. But the MSs of 802.16 system with PUSC or FUSC mode only compute and report the sum of received

CINR per carrier to BS, not individual CINR. Hence, the SINR and CINR are the same under these conditions. Finally, the mobility model we use is like below. The MS speed is 30 km/hr. Probability to change direction is 0.2 when position update. Max. angle for direction update is 45^o.

3.4 Basic Radio Resource Management

The purpose of radio resource managements is to raise the efficiency and reliability of wireless transmission. In our performance analysis, we will use the basic radio resource management method as follow.

- **Power Control:** In 802.16e system, the power control is not a major method to maintain the transmitted quality and is more required for MSs. So, we assume the BS use maximum and fixed power to transmit signals. The power of per subcarrier is the same.
- **Rate Control (AMC):** The adaptive modulation and coding scheme is a major method to keep the quality of wireless transmission. The IEEE 802.16e standard supports a variety of modulation and coding scheme, which we introduce in 2.2.3. In our simulation, we only get the link performances of BPSK, QPSK, 16-QAM, and 64-QAM modulation schemes, so we use these modulation schemes. In formal, the BPSK is not available in 802.16e OFDMA version, but we can use QPSK which repeats the data of BPSK once to transmit. The coding scheme is used to correct errors in the receiver and we use convolution code (CC) with 1/2 code rate. From 2.2.2 and 3.2, we can get one slot using BPSK, QPSK, 16-QAM, and 64-QAM with CC 1/2 can carry 24, 48, 96, 144 bits, respectively.
- **Channel assignment:** The OFDMA frame structure has two dimensions, the slot with two OFDMA symbols and the subchannel, for channel assignment. In our simulation, we obey the definition of 802.16e standard that we introduce in 2.2.2. The basic principles

are to segment the data after the modulation block into blocks sized to fit into one OFDMA slot, and map the slots in the subchannels with higher priority than that in the OFDMA symbols. In other words, the data mapping method is frequency first.

- **Subcarrier permutation:** In our platform, we use the distributed subcarrier permutation. If we use the permutation formula and series to implement the permutation, it is too complex and low efficient in simulation computation. So, we use the statistic method to simulate the permutation effect, like Figure 2-5. The interference of one slot produced by the other cells will be dispersed to all subframe.
- **Scheduling method:** The scheduling control is an important radio resource management. Many research has made about all kinds of scheduling methods. In our simulation, we choose some famous and suitable scheduling method for OFDMA techniques, such as round robin (RR), proportional fair (PF), max CINR (MC), fair throughput (FT), and early deadline first (EDF) [24][25][26][27]. The detail will be introduced as follow:

- **Round Robin (RR):** With respect to Round Robin scheduling, where users are cyclically scheduled irrespective of the channel condition.
- **Proportional Fair (PF):** The PF scheduler allocates the user *m** who maximizes the ratio of achievable instantaneous data-rate over average received data-rate. The PF provide a good tradeoff between allocation fairness and system throughput by utilizing the multiuser diversity. The user in the OFDMA system is scheduled at frame n using following function (4):

$$m^* = \arg_m \max\{\frac{\sum_{s=1}^{S} DRC_m^s(n)}{R_m(n)}\} \qquad (4)$$

, where $DRC_m^s(n)$ denoting the achievable instantaneous data-rate for user m at time n on subcarrier s, $R_m(n)$ denotes the moving average of data-rate at user m has received up to time n according to the following equation (5):

$$R_m(n) = (1 - \frac{1}{N_T})R_m(n-1) + \frac{1}{N_T} DRC'_m(n-1) \qquad (5)$$

, where N_T denoting the length of moving average, which has been set to 750.

ü **Max CINR (MC):** The MC scheduler allocates the user *m** who has maximum received CINR. The MC provides a good multiuser diversity, but less fairness. The user in the OFDMA system is scheduled at frame *n* using following function (6):

$$m^* = \arg_m \max\{\sum_{s=1}^{S} CINR_m^s(n)\} \qquad (6)$$

, where $CINR_m^s(n)$ denoting the carrier-to-interference and noise ratio for user *m* at time *n* on subcarrier *s*.

ü **Fair Throughput (FT):** The FT scheduler allocates the user *m** who has minimum average received data-rate and it is a special case of PF. The FT provides a good fairness, but less multiuser diversity. The user is scheduled at frame *n* using following function (7):

$$m^* = \arg_m \min\{R_m(n)\} \qquad (7)$$

, where $R_m(n)$ denotes the moving average of data-rate at user *m* has received up to time *n* and is equal to equation (5).

ü **Early Deadline First (EDF):** The EDF scheduler allocates the user *m** who has minimum remainder time needs be transmitted. The EDF provides a guarantee of QoS for real time services. The user is scheduled at frame *n* using following function (8):

$$m^* = \arg_m \min\{DB - Age - T_t\} \qquad (8)$$

, where *DB* is delay bound, *Age* is the time that the user's packet has stayed in the MAC layer, and the T_t is the required time to finish the transmission of the packet.

n **Handoff method:** In this thesis, handoff is not a weight-bearing point. So, we use the simplest method: hard handoff. This method is "Break-Before-Make".

- **ARQ retransmission:** In our simulation, we only implement the ARQ retransmission and don't use the HARQ. When the PDU is error, the ARQ retransmission will work. We don't limit the retransmission times. The PDU can be retransmitted at all times until it is correct.

3.5 Traffic Models

In IEEE 802.16e standard, the data traffics are divided into five QoS classes, such as UGS, rtPS, ertPS, nrtPS, and BE. The details are described in 2.3.4. In general, UGS, rtPS, and ertPS are designed for real time services and the major differences among these classes are bandwidth request-and-grant on the uplink. The nrtPS and BE are devised for non-real time services. In order to reduce the simulation time and computed loading, we reasonably use the HTTP services to stand for non-real time services and the VoIP services to represent real time services on the downlink based on the previous discuss. The HTTP traffic model adopts 3GPP model [28] as shown in Table 3-4. The VoIP traffic model uses G729-1 codec [29] as shown in Table 3-5. The HTTP services use TCP/IP protocol to transmit, so the HTTP

Table 3-4 HTTP Traffic Model

events	**distribution**	**number**
Number of packet call requests per session	Geometrically	5
Reading time between two consecutive packet call	Geometrically	412
Number of packets in a packet call	Geometrically	25
Time interval between two packets inside a packet call	Geometrically	4/rate
Packet size	Pareto	A=1.1 K=81.5

packet needs to add 20 bytes TCP header and 20 bytes IP header. The VoIP services use RTP/UDP/IP protocol to transmit. The VoIP packet must add 12 bytes RTP header, 8 bytes UDP header, and 20 bytes IP header.

Table 3-5 VoIP Traffic Model

codec	Framesize(byte)	samples	Interval(ms)	Rate(bps)
G729-1	10.0	2.0	20.0	8k

Finally, we use Table 3-6 to summarize this chapter and present the arrangement of the parameter setting in our simulation platform.

Table 3-6 The Parameter Setting in Simulation Platform

Parameters	**Value/Comment**
Cell layout	Hexagonal grid, 19 cells (wrap around)
Sectors per cell	3
Frequency reuse factor	1x1 and 1x3
Available bandwidth	6 MHz in 1x1 reuse 18 MHz in 1x3 reuse
Antenna pattern	70° with 20 dB front-to-back ratio, according to [20]
Cell radius	1 km
Transmitter/Receiver	Downlink (from BS to MSs)
Duplex	TDD mode
DL/UL subframe ratio	1:1
Frame length	10ms, according to [1]

Frame structure	2048-FFT OFDMA downlink carrier allocations with PUSC, according to [1]
OFDMA symbol length	308 μ s, according to [1]
OFDMA symbols per slot	2 symbols
BS Tx power	46dBm (40 Watt), according to [19]
BS Antenna gain	17 dBi, according to [19]
BS back off	5 dB, according to [21]
Thermal Noise Density	-173.93 dB/Hz, according to [21]
MS Noise Figure	9dB, according to [21]
MS Antenna gain	3 dBi, according to [19]
Pathloss model	35.0log(d[m])+31.5, 50m<d<5km, according to [22]
Shadow fading model	Log-normal distribution with STD=8dB and Gudmundson's correlation model, according to [23]
Mobility model	MS speed : 30 km/hr Probability to change direction : 0.2 Max. angle for direction update : 45^{o}
BS Power contol	Max power
AMC	BPSK+CC 1/2, QPSK+CC 1/2, 16-QAM+CC 1/2, 64-QAM+CC 1/2, according to [2]
Channel assignment	Frequency first, according to [2]
Scheduling control	Round Robin (RR), Proportional Fair (PF) Max CINR (MC), Fair Throughput (FT), Early Deadline First (EDF)
Handoff	Hard handoff
Traffic model	HTTP and VoIP, according to [28][29]

Chapter 4

Simulation Results

In this chapter, we will show the simulation results with different scheduling controls and frequency reuse factors. First, the performances of non-real time services are presented include slot usage, PHY layer throughput, MAC layer throughput, the CDF of user throughput, admission control criterion, scheduling delay and jitter, AMC usage, and spectrum efficiency. Second, the performances of real time services are demonstrated include packet loss rate and admission control criterion. Finally, the affect of mixed traffic will be addressed and discussed.

4.1 Non-Real Time Service

In this section, we only use the users of non-real time services to simulate all kinds of performance targets.

4.1.1 Slot Usage per Frame

A slot is the minimum unit for data transmission. We simulate the slot usage per frame with non-real time services at different active users as shown in Figure 4-1 and 4-2. Figure 4-1 and 4-2 use frequency reuse factor one and three, respectively. From these figures, we can find that frequency reuse factor 3 has lower slot usage at the same active users. The reason is the users at frequency reuse factor 3 own higher transmission rate and can use less slots to transmit the same data. But the slots will be full usage over 30 active users, no matter what frequency reuse factor is.

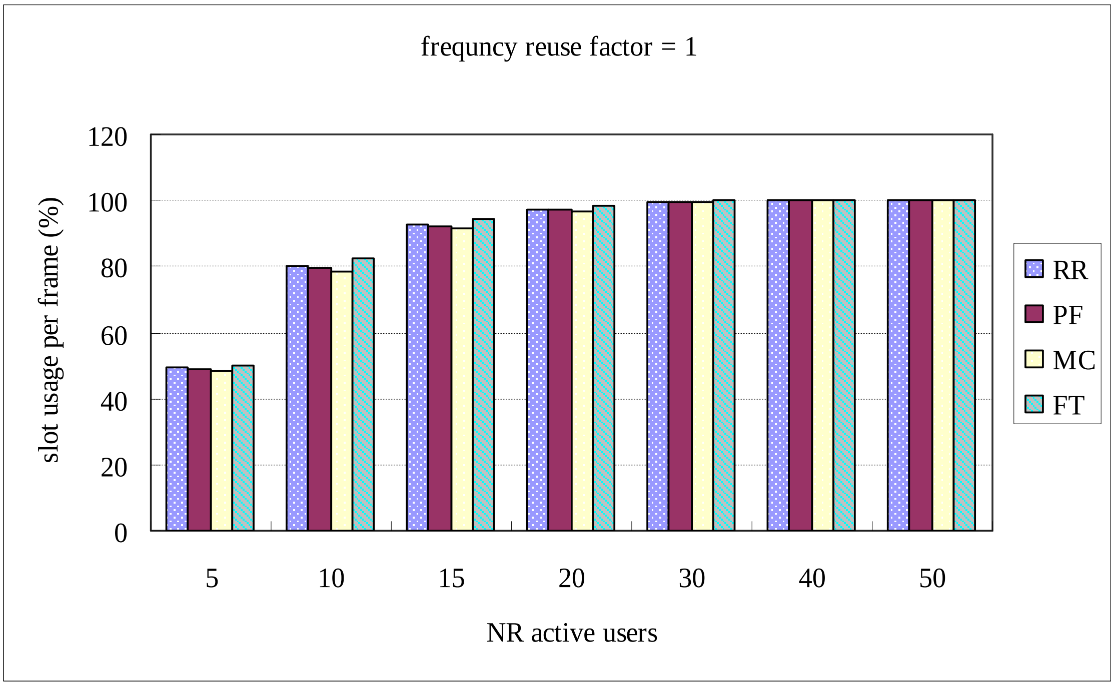

Figure 4-1 Slot usage per frame in frequency reuse factor 1

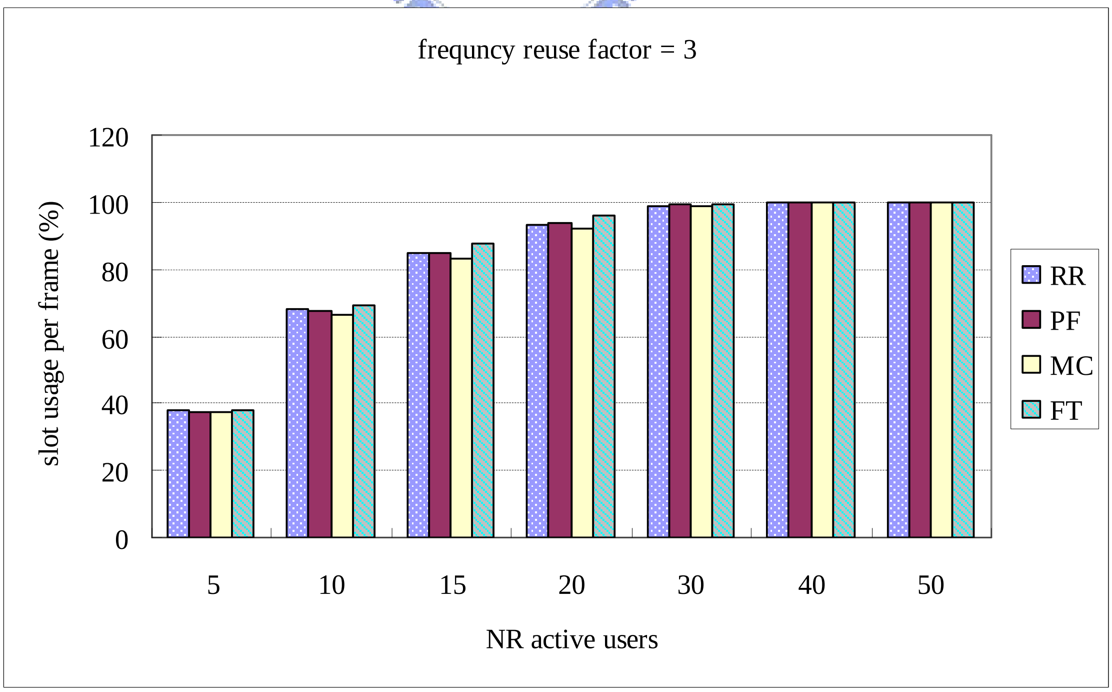

Figure 4-2 Slot usage per frame in frequency reuse factor 3

4.1.2 PHY Throughput per BS

Figure 4-3 and 4-4 display the PHY throughput per BS with different scheduling controls and frequency reuse factors. The PHY throughput per BS means the data throughput transmits between physical layers include the header and ARQ retransmission of MAC PDUs. From these figure we can discover the order of the dimension of throughput is maximum CINR (MC), proportional fair (PF), round robin (RR), and then fair throughput (FT), no matter what frequency reuse factor is. Frequency reuse factor three has higher throughput because of lower interference. At frequency reuse factor one the PF has the range from 50% to 70% more than RR. The PF just has the range from 30% to 40% at frequency reuse factor three. So the scheduling controls which consider channel condition are more important at frequency reuse factor one. The throughput of frequency reuse factor three is 60% more than that of frequency reuse factor one with RR. But the throughput of frequency reuse factor three is only 35% more than that of frequency reuse factor one with PF. The PF with frequency reuse factor one and RR with reuse factor three can get the same throughput.

4.1.3 MAC Throughput per BS

The MAC throughput per BS means the data throughput transmits between MAC layers except the header and ARQ retransmission of MAC PDUs. The PHY throughput per BS which subtracts the header and ARQ retransmission of MAC PDUs, called overheads, will become the MAC throughput per BS. Figure 4-5 and 4-6 show the MAC throughput per BS at frequency reuse factor one and three. From Figure 4-3 and 4-5, we can observe the overheads of PF and MC are 10% and those of RR and FT are 20% at frequency reuse factor one, respectively. Because PF and MC let the users with better channel conditions to transmit data,

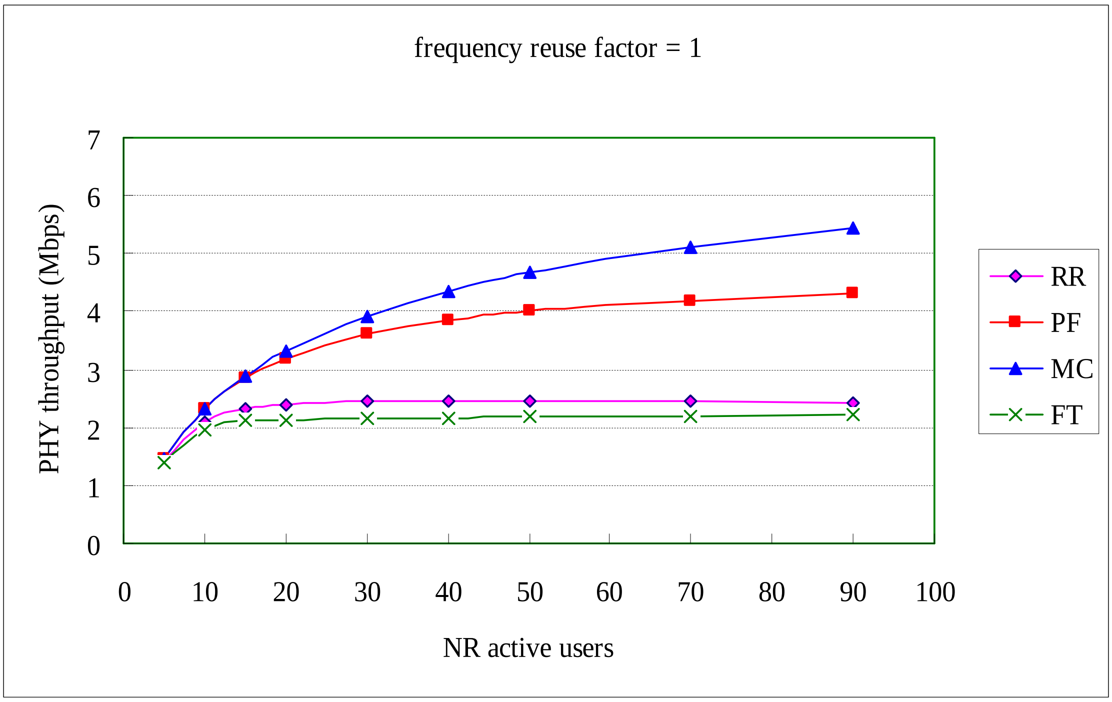

Figure 4-3 PHY throughput of frequency reuse factor 1 per BS

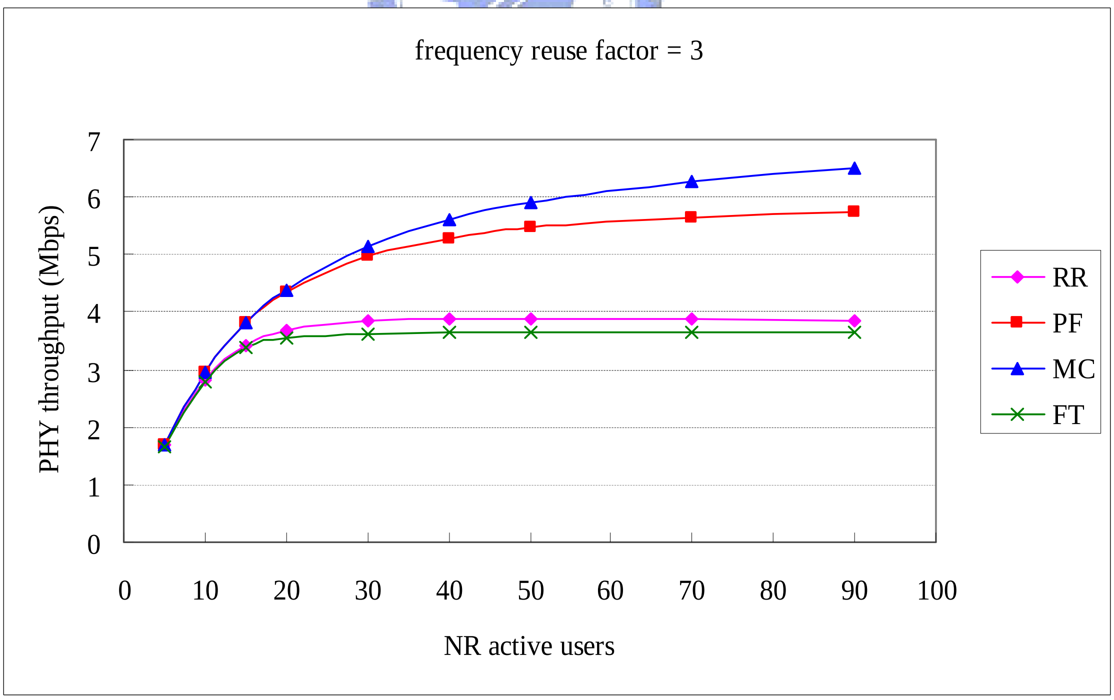

Figure 4-4 PHY throughput of frequency reuse factor 3 per BS

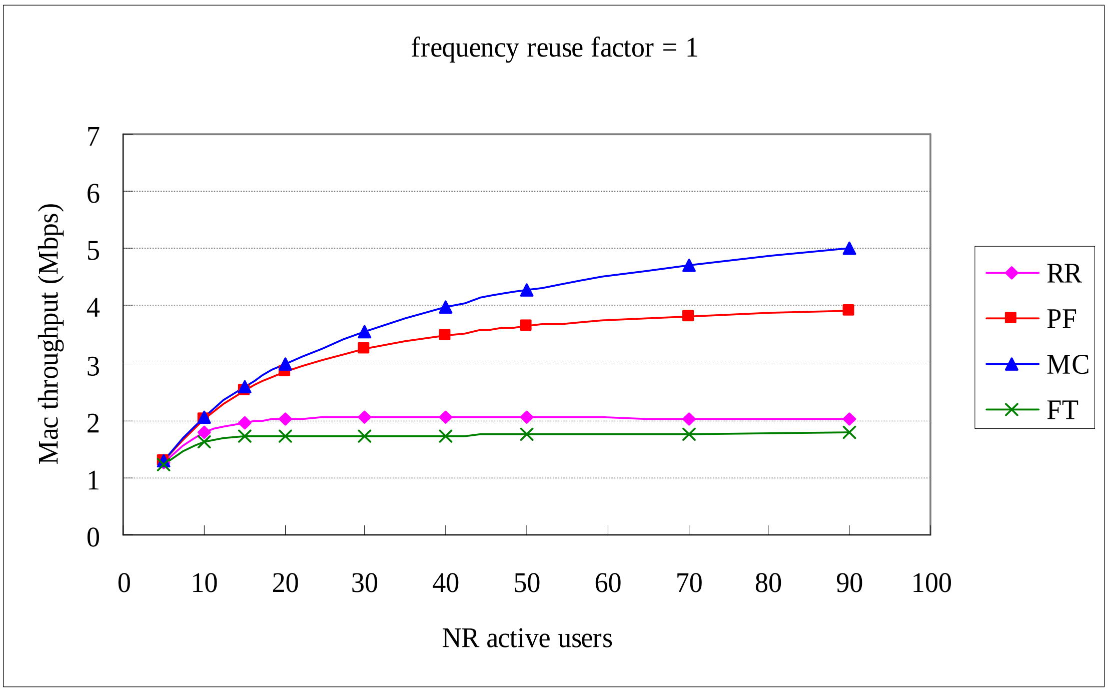

Figure 4-5 MAC throughput of frequency reuse factor 1 per BS

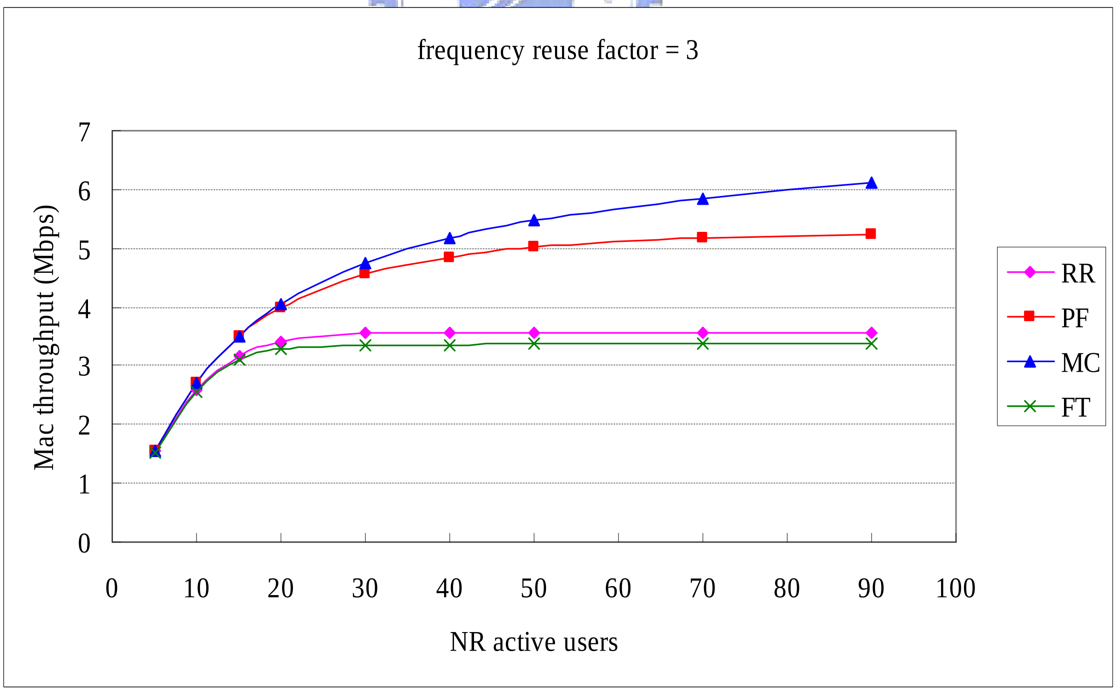

Figure 4-6 MAC throughput of frequency reuse factor 3 per BS

the packet error rate is lower caused lower overheads for retransmissions. From Figure 4-4 and 4-6, the overheads of PF and MC are 7% and those of RR and FT are 9% at frequency reuse factor three, respectively. When frequency reuse factor three is adopted, the overheads can be roughly reduced half.

4.1.4 CDF of User Throughput

Figure 4-7 and 4-8 address the CDF of user's PHY throughput with frequency reuse factor one and three at 30 active users. Regardless of frequency reuse factor one or three the trends of curves are identical. Although MC has maximum system throughput, the throughput per user has a large variation and the fairness of throughput for different users is disappointing. The fairness of throughput with FT is the best, but it has lowest system throughput. The throughput of users with PF has an identical trend of the curve and is always higher than that with RR. The channel condition and fairness can be looked after both sides with PF.

4.1.5 Admission Control Criterion for Non-Real Time Service

In the 802.16d/e standard, the non-real time services have a QoS parameter which is the minimum data rate. Figure 4-9 and 4-10 present the minimum rate of users with frequency reuse factor one and three at different active users. If we assume the minimum data rate is 50 kbps, the minimum data rate at frequency reuse factor one will below 50kbps over 30, 60, 25, and 40 active users with RR, PF, MC, and FT, respectively and we should do admission control over these active users. The same situation at frequency reuse factor three is over 50, 90, 35, and 70 active users with RR, PF, MC, and FT. Therefore, the needs for admission control obey the order: MC, RR, FT, and then PF.

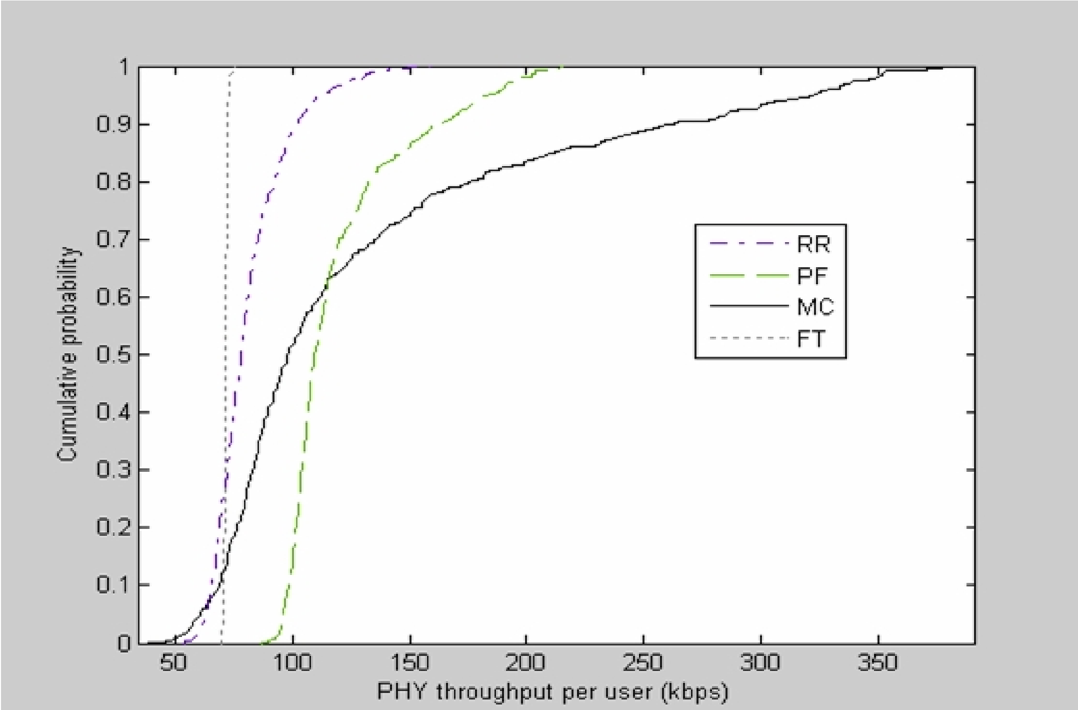

Figure 4-7 CDF of user’s PHY throughput with frequency reuse factor 1 at 30 users

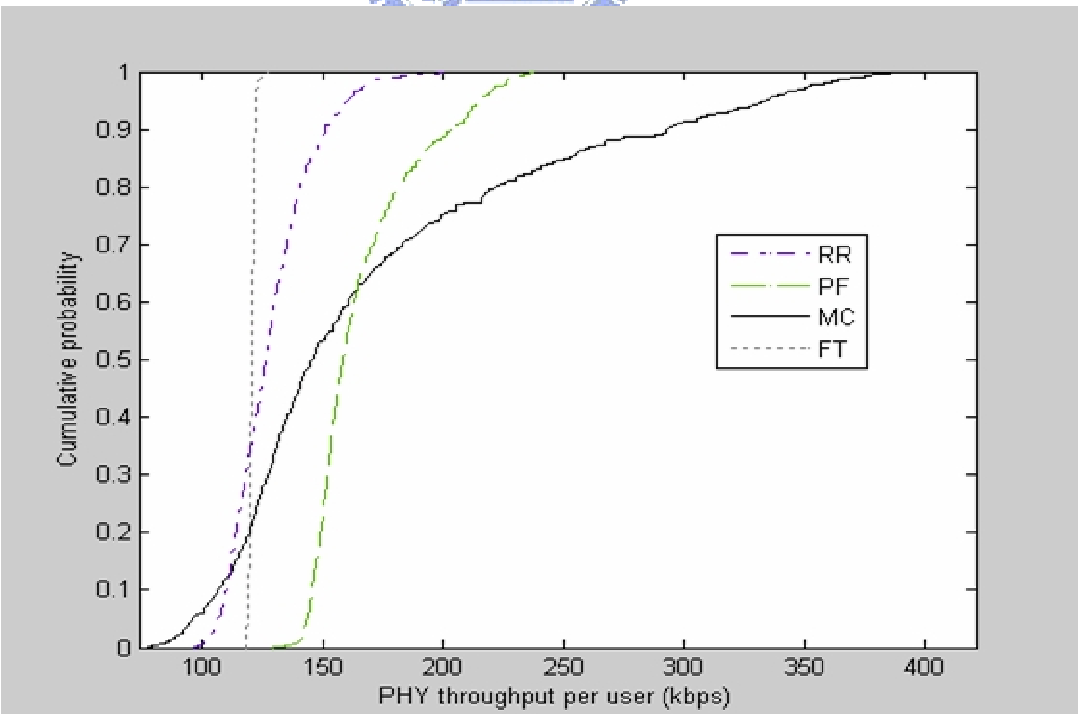

Figure 4-8 CDF of user’s PHY throughput with frequency reuse factor 3 at 30 users

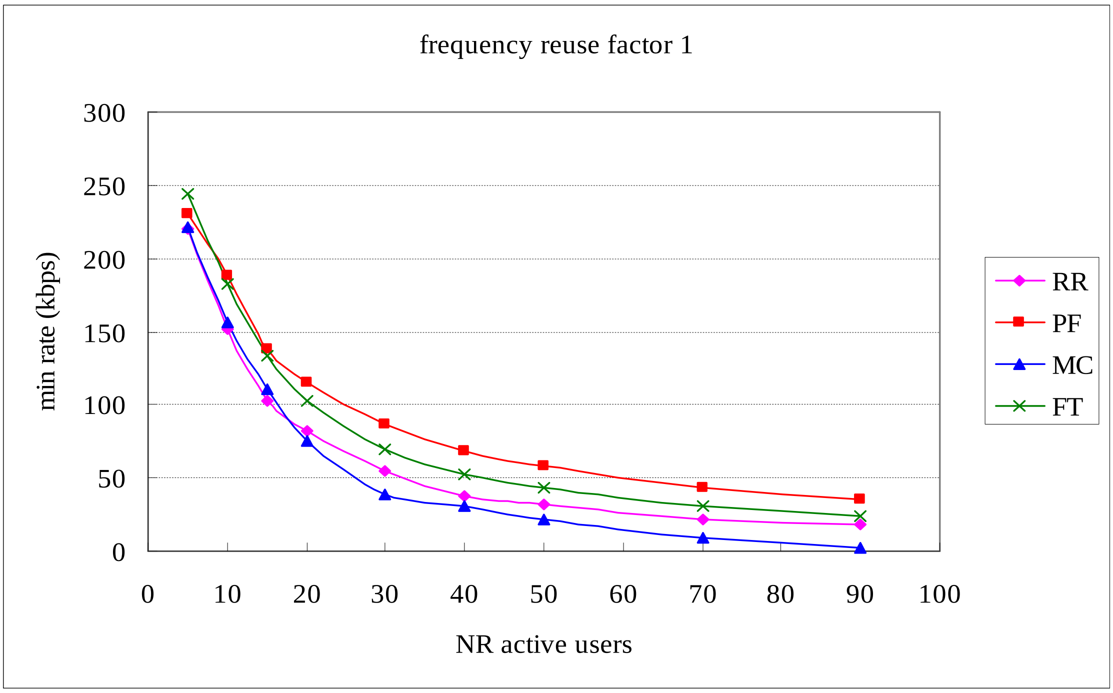

Figure 4-9 Minimum data rate of non-real time services at frequency reuse factor 1

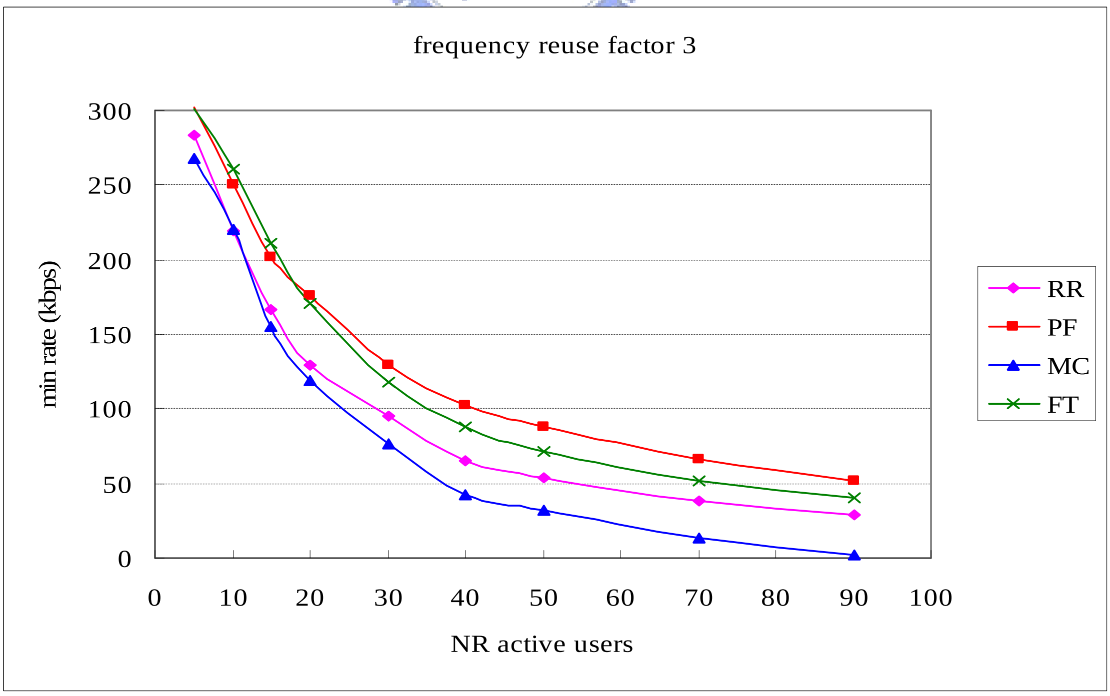

Figure 4-10 Minimum data rate of non-real time services at frequency reuse factor 3

4.1.6 Scheduling Delay

Scheduling delay is the mean value of the interval time of single active user from this schedule to next schedule. Figure 4-11 and 4-12 show the scheduling delays of different scheduling controls with frequency reuse factor one and three. From these figure we can find the delay time increases when the active users rise and the trends of curves are linear relationship, no matter what frequency reuse factor is. Frequency reuse factor three has lower scheduling delay than one. The order of scheduling delay is FT, RR, PF, and MC at the same active users.

4.1.7 Scheduling Jitter

Scheduling jitter is the standard deviation of the scheduling delay. It stands for the variation of the interval time of single user. Figure 4-13 and 4-14 display the scheduling jitters of different scheduling controls with frequency reuse factor one and three, respectively. From these figure we can observe the scheduling jitter rise when the active users add. No matter what frequency reuse factor is, the scheduling jitters of RR, FT, and PF are linear relationship with active users. But scheduling jitter of MC is the exponential relationship with active users. The order of scheduling jitter is MC, PF, FT, and RR at the same active users.

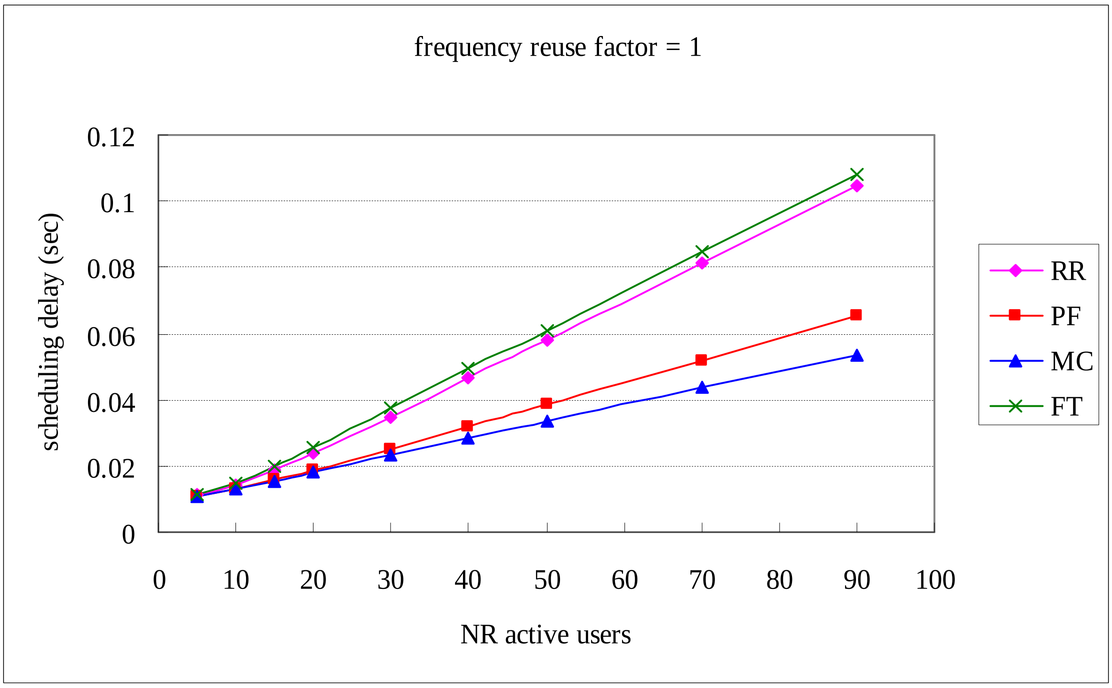

Figure 4-11 Scheduling delay at frequency reuse factor 1

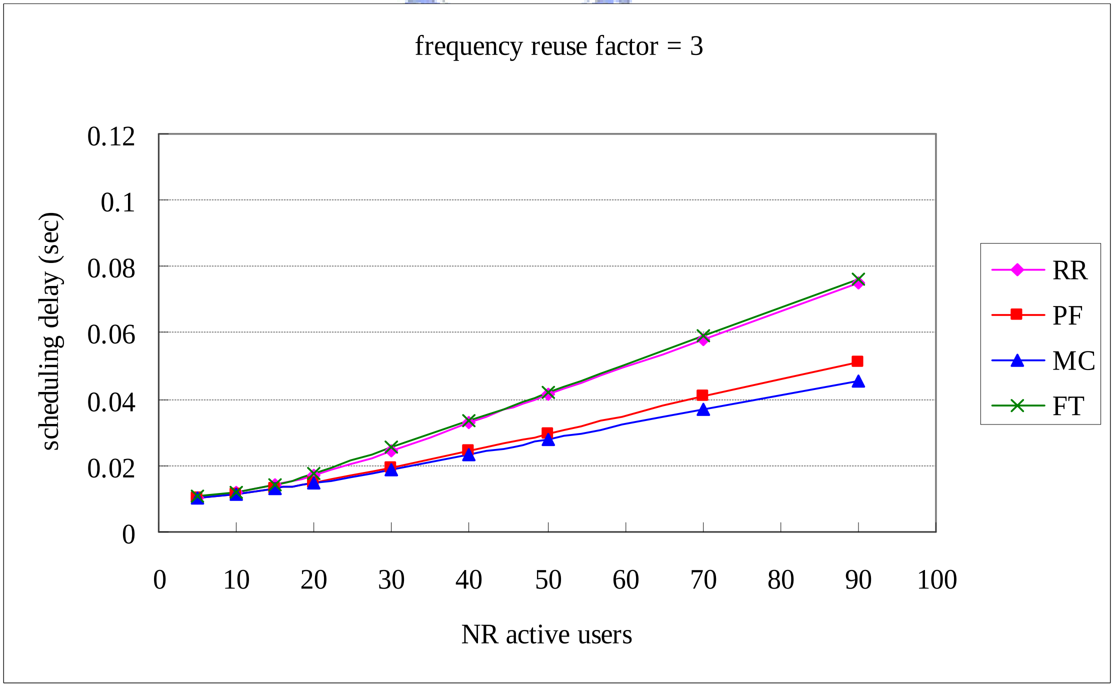

Figure 4-12 Scheduling delay at frequency reuse factor 3

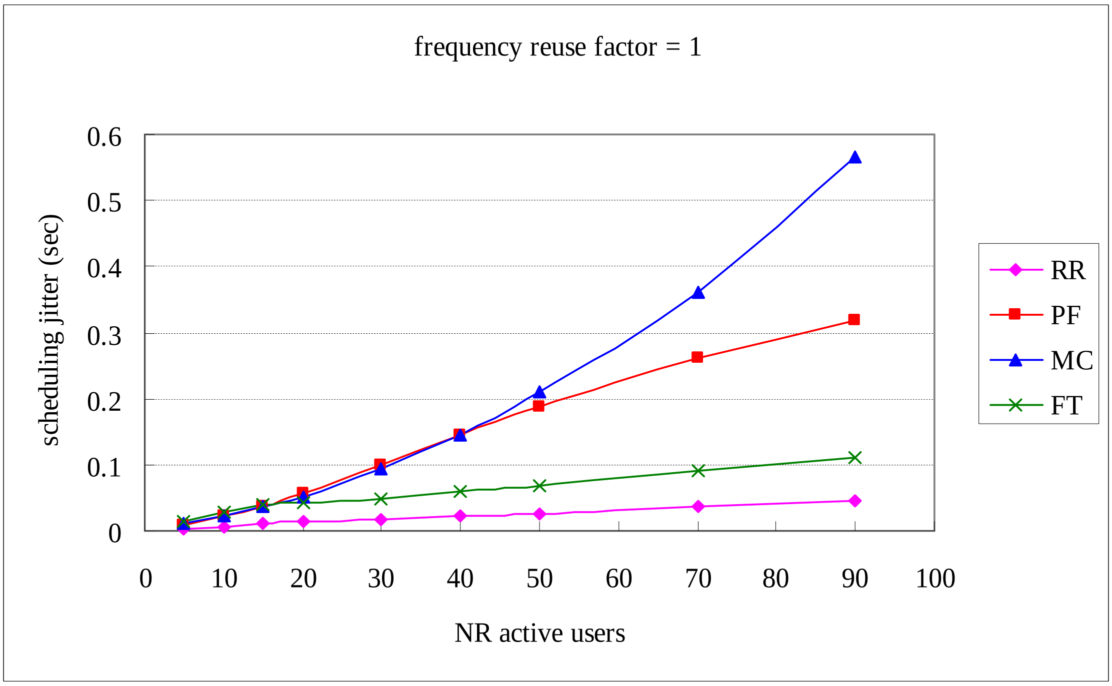

Figure 4-13 Scheduling jitter at frequency reuse factor 1

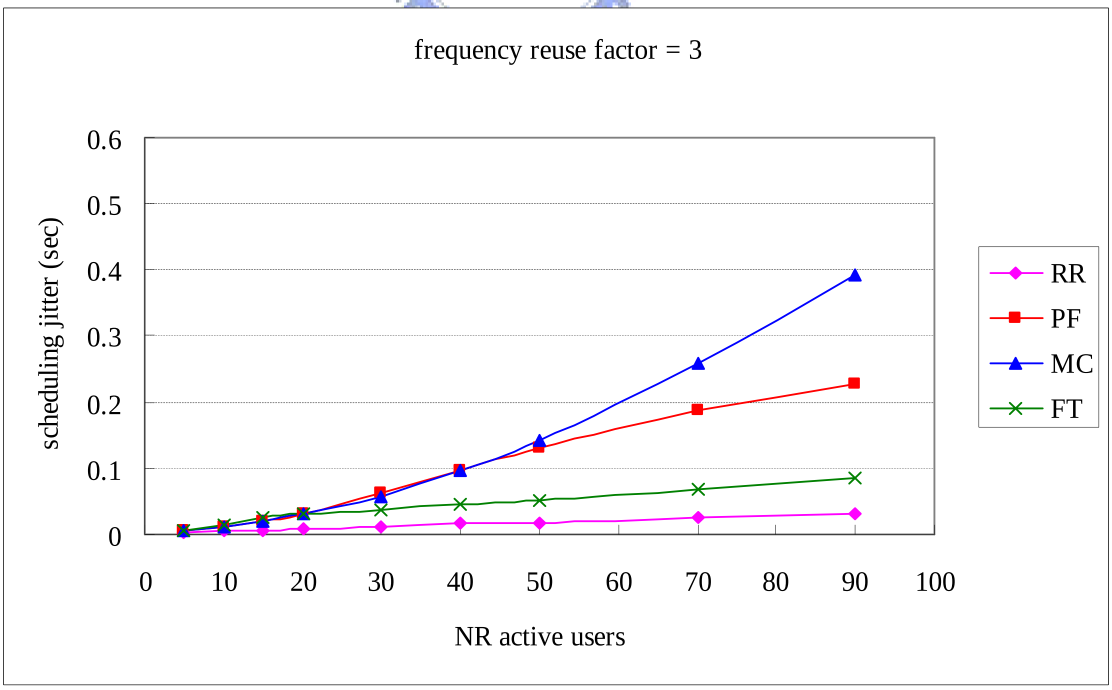

Figure 4-14 Scheduling jitter at frequency reuse factor 3

4.1.8 AMC Usage

In our simulation, we use BPSK, QPSK, 16-QAM, and 64-QAM modulations and convolution code with half code rate which executes the forward error correction. Figure 4-15 and 4-16 show the usage of modulation and coding scheme with RR at frequency reuse factor one and three. At frequency reuse factor one 66.7% of scheduled users occupy low-level AMCs, BPSK and QPSK, with high loading. At frequency reuse factor three 67.3% of scheduled users use high-level AMC, 16-QAM and 64-QAM with high loading and almost no users use BPSK to transmit data. Figure 4-17 and 4-18 address the usage of AMCs with FT at frequency reuse factor one and three. We can find the trends of curves with FT are similar to that with RR. With high loading low-level AMCs, BPSK and QPSK, dominate the 74.5% of AMCs at frequency reuse factor one and high-level AMCs, 16-QAM and 64-QAM, hold the 61.9% of AMCs at frequency reuse factor three. Figure 4-19 and 4-20 display the AMC usage of PF at frequency reuse factor one and three. The user diversity increases with the rise of active users, so more scheduled users use high-level AMC. High-level AMCs, 16-QAM and 64-QAM, occupy 45% to 75% usage with high loading at frequency reuse factor one. 78%~97% scheduled users hold high-level AMCs with high loading at frequency reuse factor three. Figure 4-21 and 4-22 show the AMC usage of MC at frequency reuse factor one and three. The trend of user diversity with MC is alike to that with PF. At frequency reuse factor one the scheduled users use high-level AMCs form 43% to 93% with high loading. At frequency reuse factor three high-level AMCs from 78% to 99% are occupied by the scheduled users in high loading.

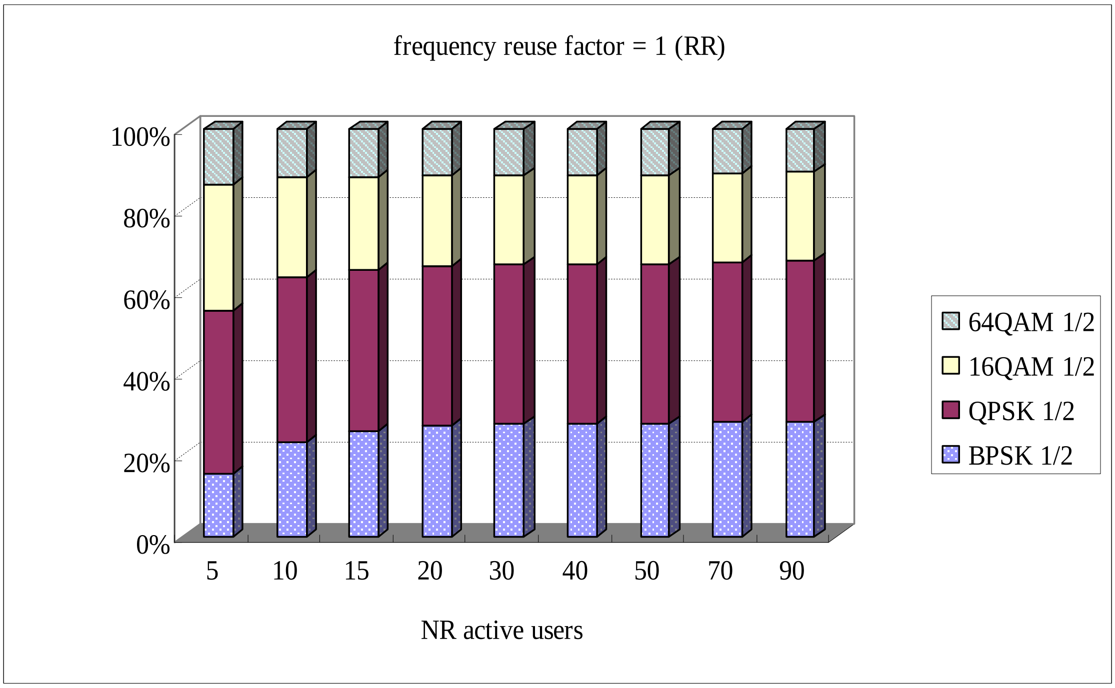

Figure 4-15 AMC usage of RR at frequency reuse factor 1

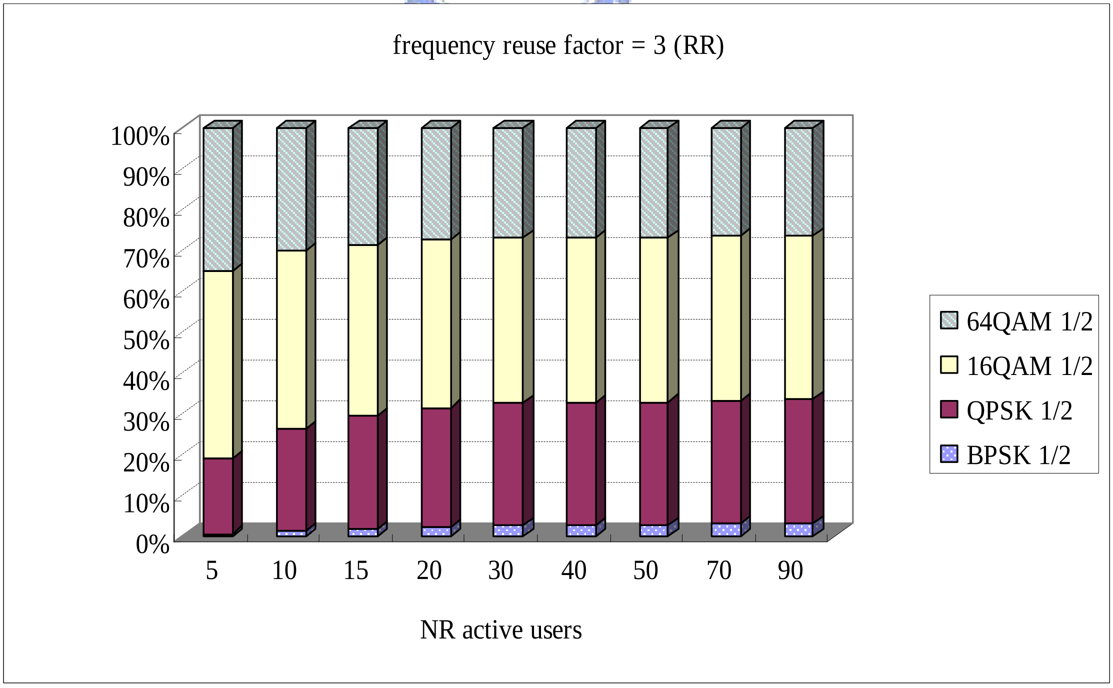

Figure 4-16 AMC usage of RR at frequency reuse factor 3

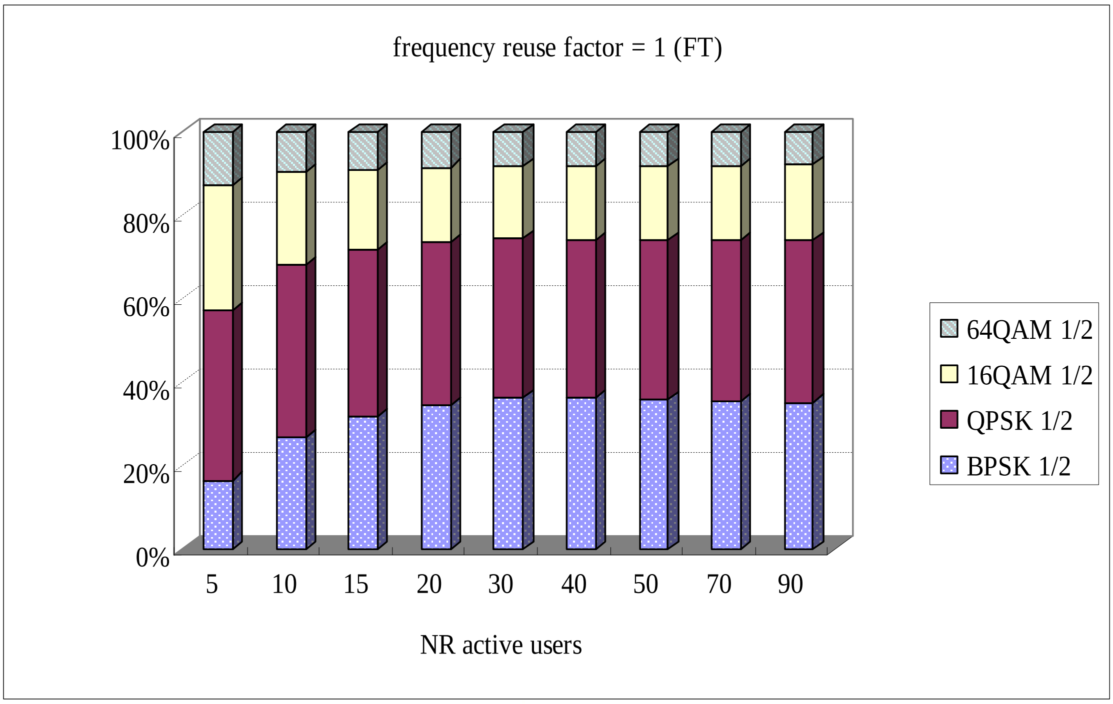

Figure 4-17 AMC usage of FT at frequency reuse factor 1

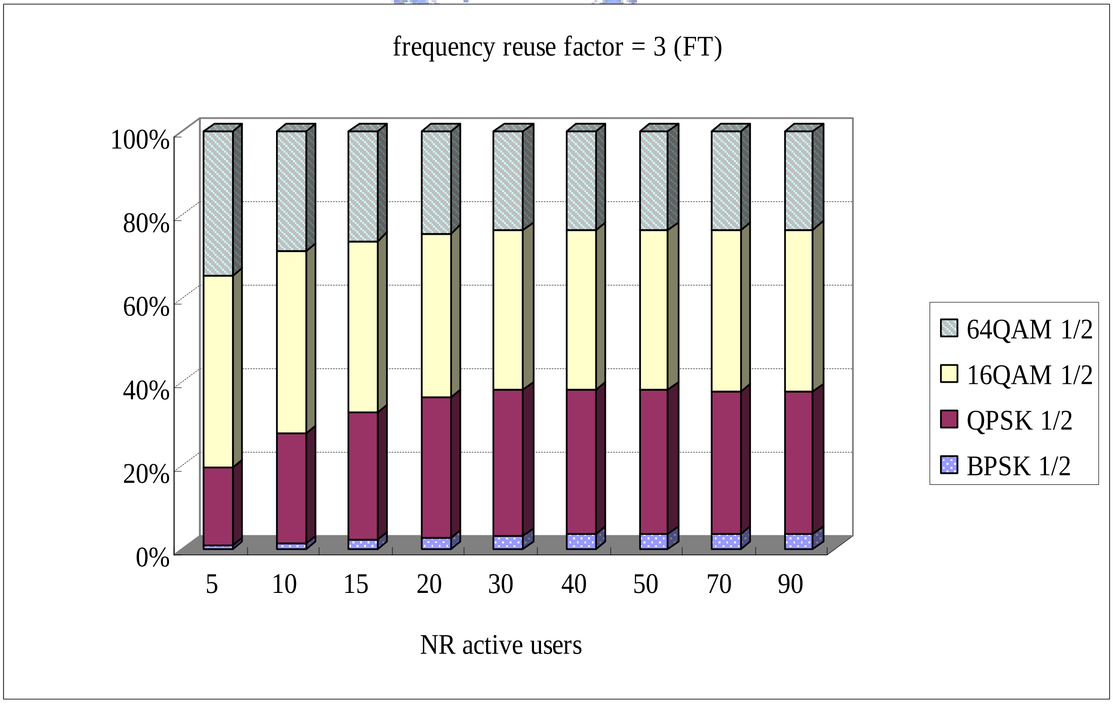

Figure 4-18 AMC usage of FT at frequency reuse factor 3

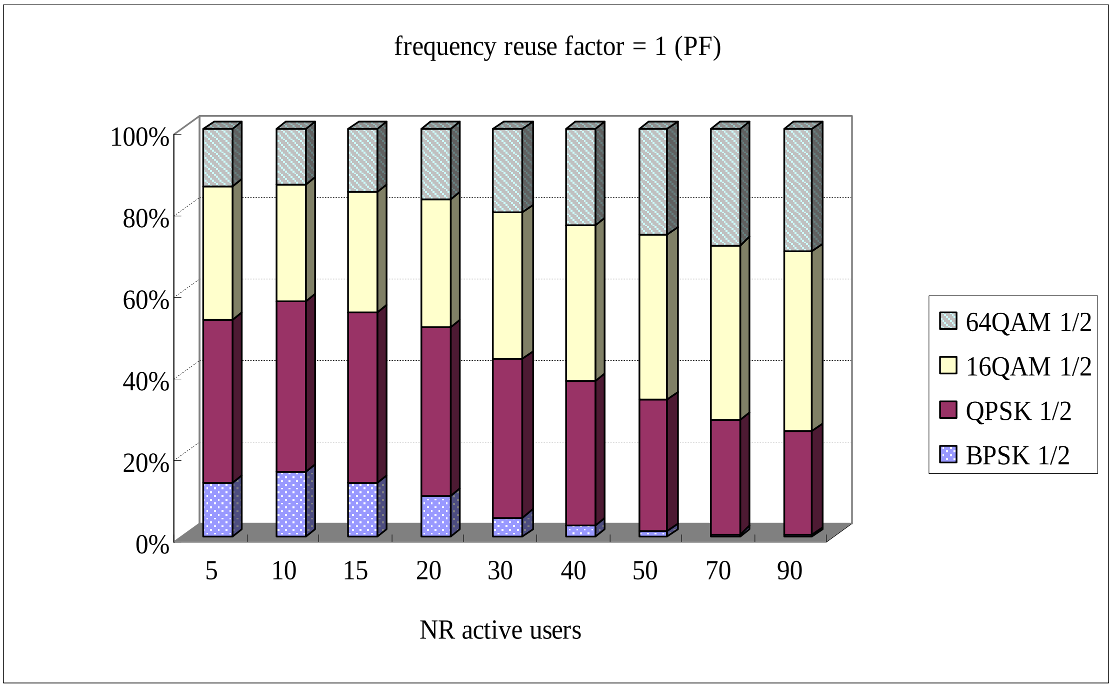

Figure 4-19 AMC usage of PF at frequency reuse factor 1

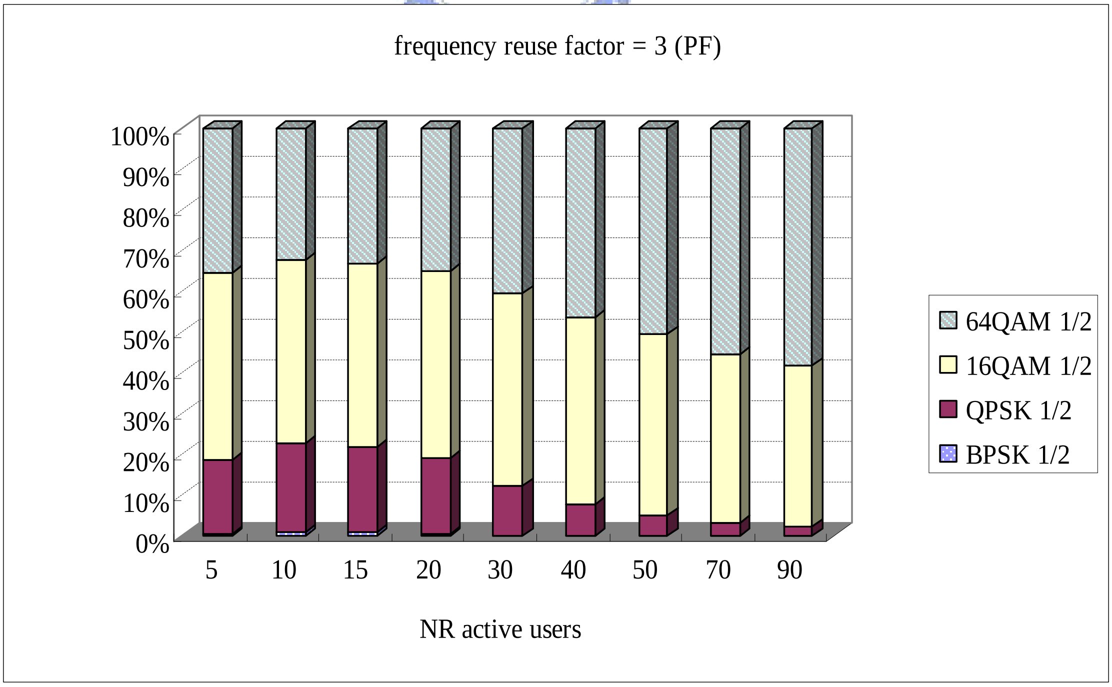

Figure 4-20 AMC usage of PF at frequency reuse factor 3

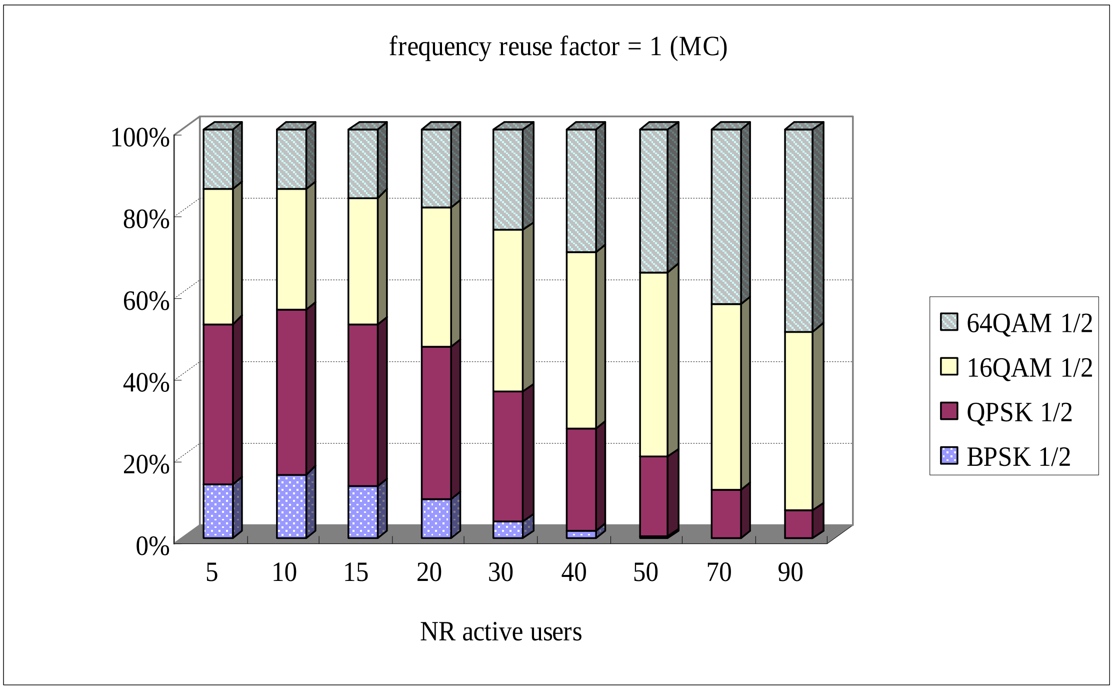

Figure 4-21 AMC usage of MC at frequency reuse factor 1

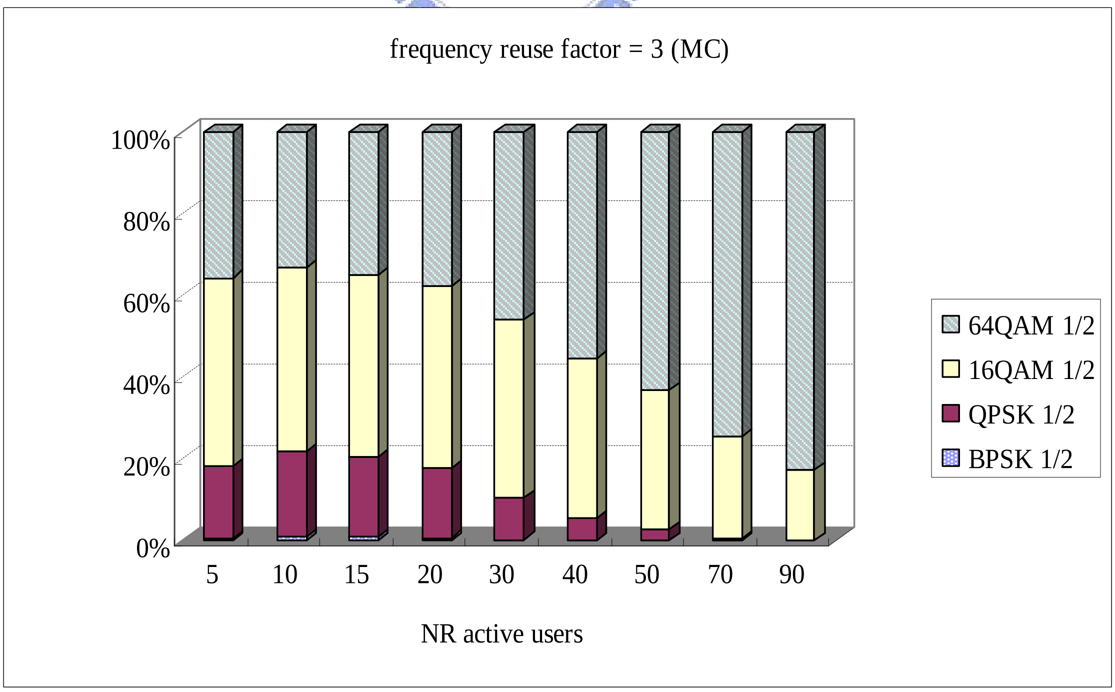

Figure 4-22 AMC usage of MC at frequency reuse factor 3

4.1.9 Spectrum Efficiency

The spectrum efficiency is the ratio of total throughput per BS and overall system bandwidth. Figure 4-23 presents the spectrum efficiency at 30 active users. Spectrum efficiency of FT and RR at frequency reuse factor one is 78% and 90% more than at frequency reuse factor three, respectively. Spectrum efficiency of PF and MC at frequency reuse factor one is 116% and 128% more than at frequency reuse factor three, respectively. We can obvious find the frequency reuse factor one has better spectrum efficiency than frequency reuse factor three. Notice that frequency reuse factor three need three times available bandwidth than frequency reuse factor one. Although it has higher system throughput and lower interference, spectrum efficiency must divide into three times bandwidth.

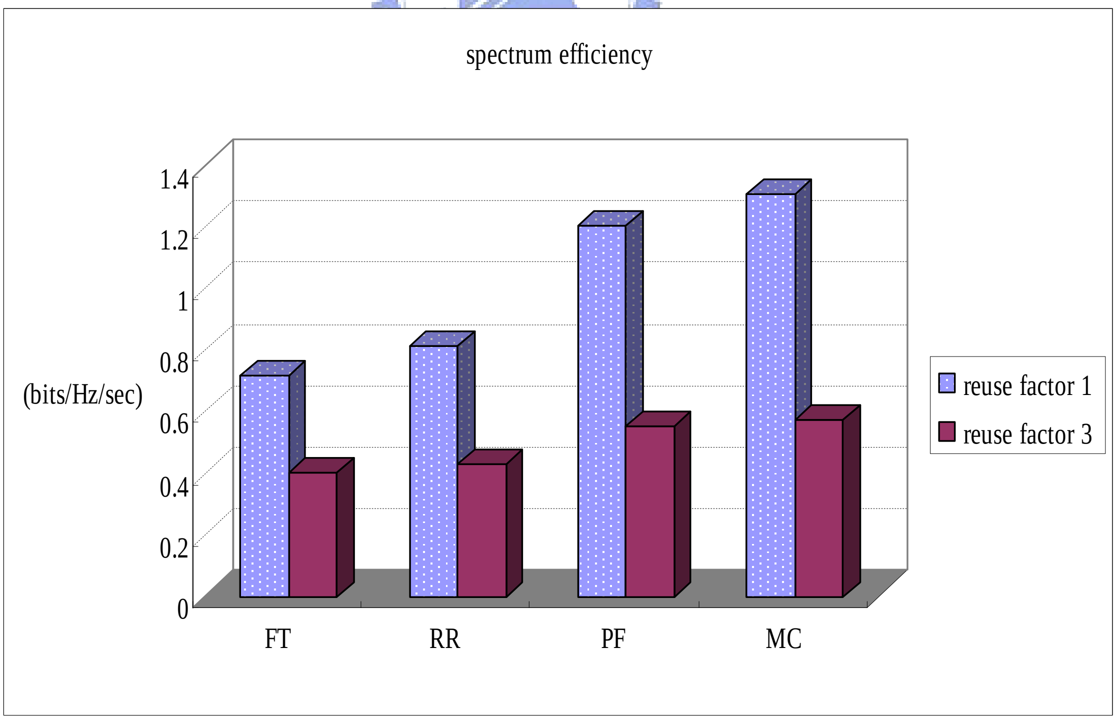

Figure 4-23 Spectrum efficiency at 30 active users

4.2 Real Time Service

In this section, we only use the users of real time services to simulate the performance targets.

4.2.1 Admission Control Criterion for Real Time Service

For real time services, packet loss rate is an important performance target. If the packet loss rate is too high, the user will feel the QoS of services is bad. The acceptable packet loss rate for VoIP services is below 1%. Figure 4-24 and 4-25 address the packet loss rates with different scheduling controls at frequency reuse factor one and three. When active users increase, the packet loss rates also rise, especially over 60 and 105 active users at frequency reuse factor one and three, respectively. The rough order of the need of admission control is FT, RR, PF, MC, and EDF at 1% of packet loss rate. The need of admission control for PF, MC, and EDF is very close. The packet loss rates of PF and MC are lower than that of EDF over 75 and 125 active users at frequency reuse factor one and three, respectively. The reason is the system has exceeded its loading over those users too much. So, the PF and MC considered channel condition can accommodate more packets and has lower packet loss rate. Although PF and MC have lower packet loss rate in high loading than EDF, the packet loss rates of a large part of users have exceeded 1%. On the contrary, the EDF has lower packet loss rate per user in high loading as shown in Figure 4-26, 4-27, 4-28, 4-29, 4-30, 4-31, 4-32, and 4-33. From Figure 4-26 to 4-33, we present the percentage ratio of the user number of different packet loss rate ranges and total active users. The different ranges of packet loss rate are 0~0.1%, 0.1~0.5%, and 0.5~1%. When the active users rise, the transmitted quality will decrease.

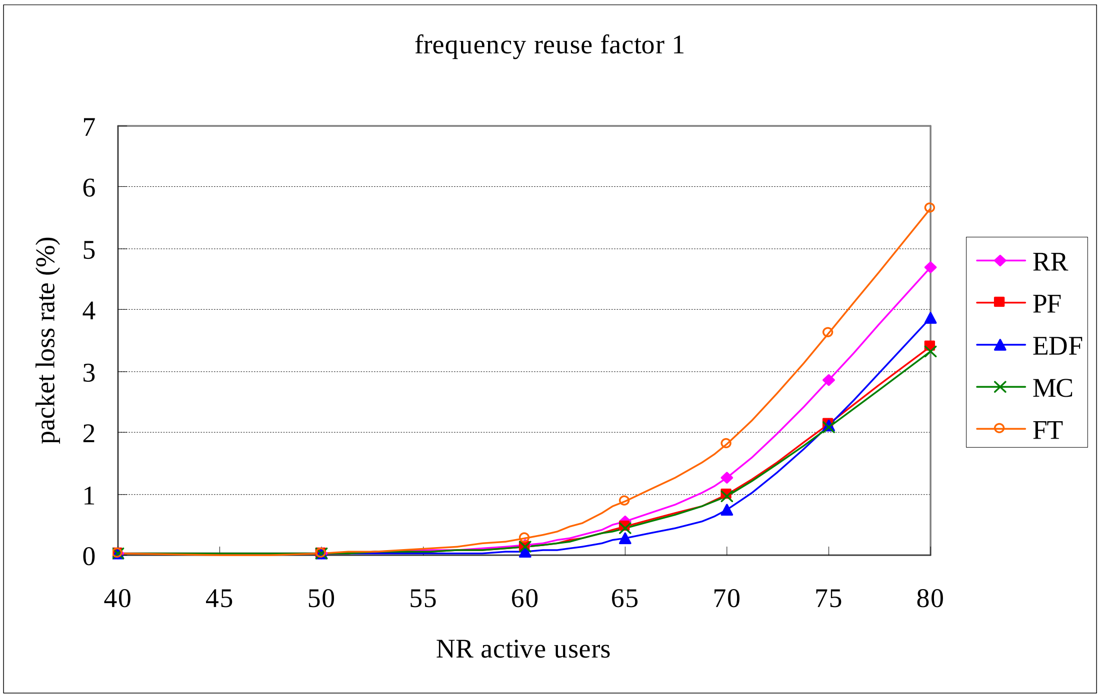

Figure 4-24 Mean value of packet loss rate at frequency reuse factor 1

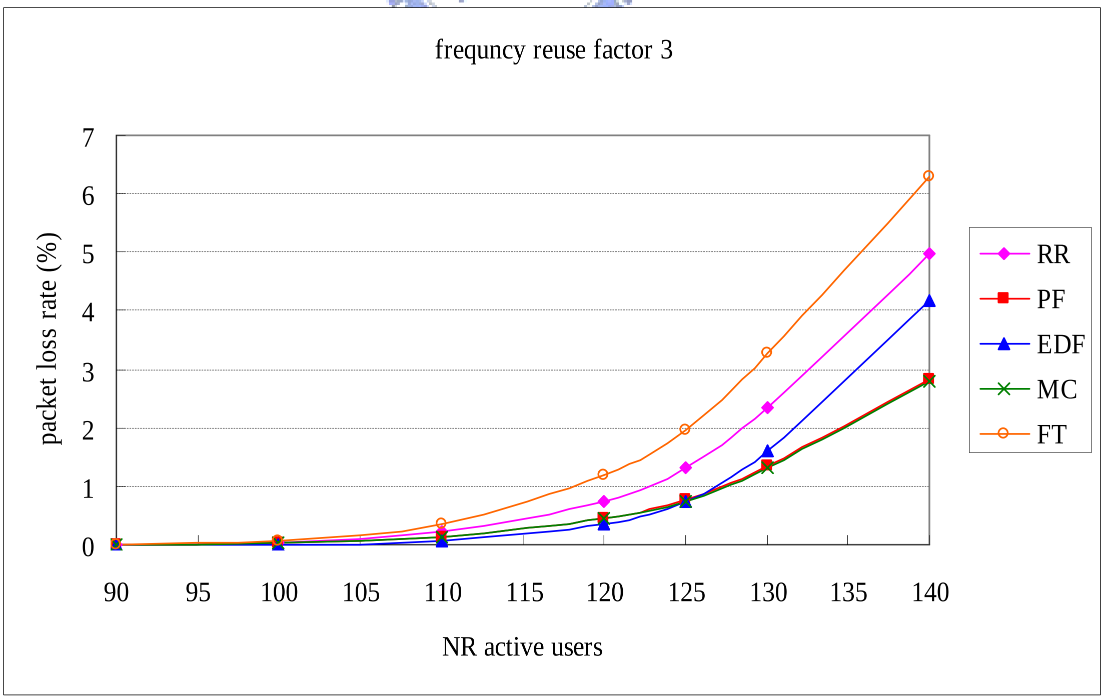

Figure 4-25 Mean value of packet loss rate at frequency reuse factor 3

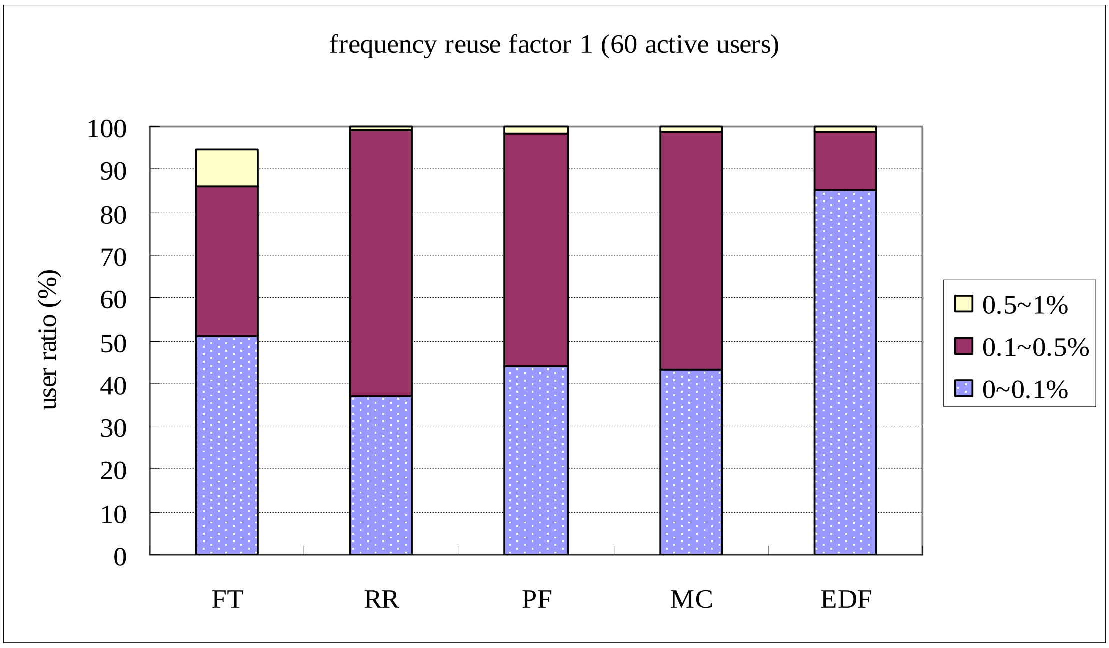

Figure 4-26 Ratio of user number of different packet loss rate ranges and 60 active users at frequency reuse factor 1

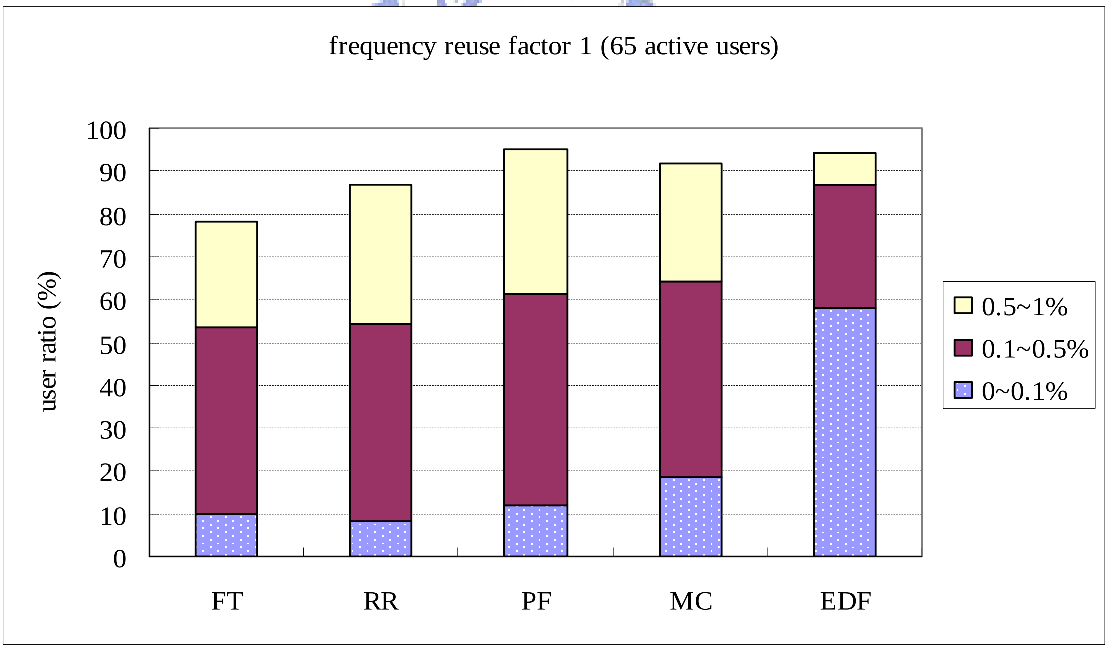

Figure 4-27 Ratio of user number of different packet loss rate ranges and 65 active users at frequency reuse factor 1

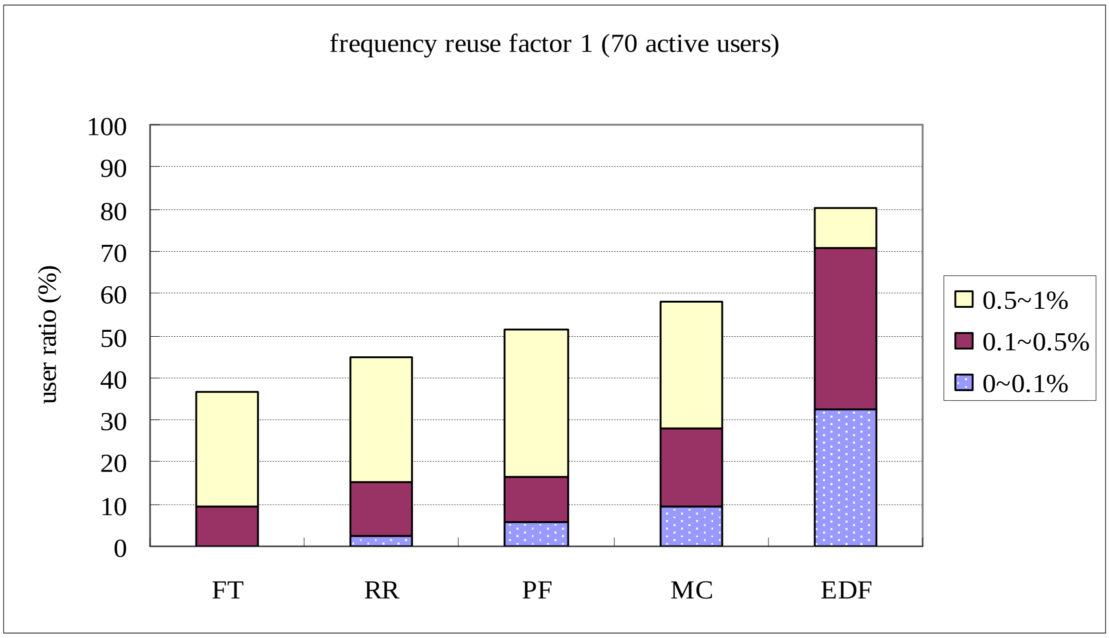

Figure 4-28 Ratio of user number of different packet loss rate ranges and 70 active users at frequency reuse factor 1

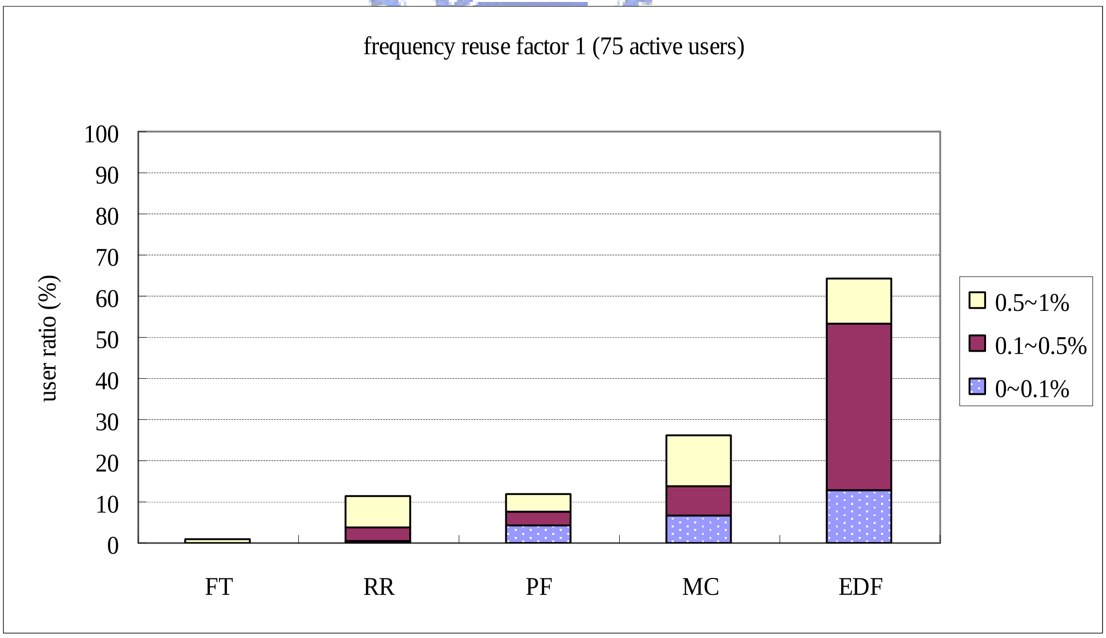

Figure 4-29 Ratio of user number of different packet loss rate ranges and 75 active users at frequency reuse factor 1

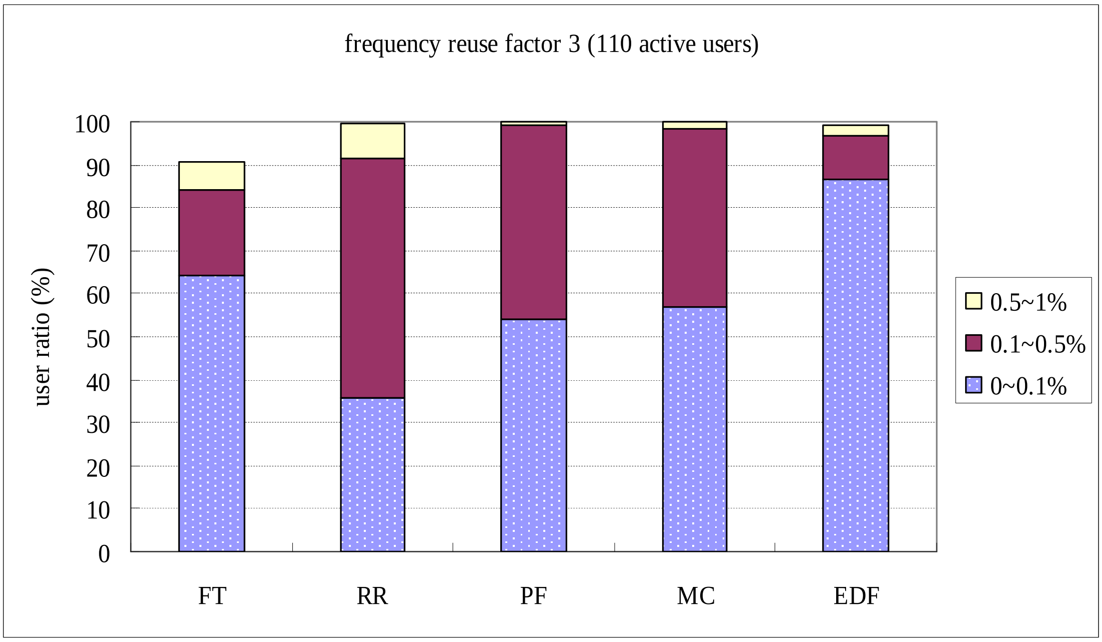

Figure 4-30 Ratio of user number of different packet loss rate ranges and 110 active users at frequency reuse factor 3

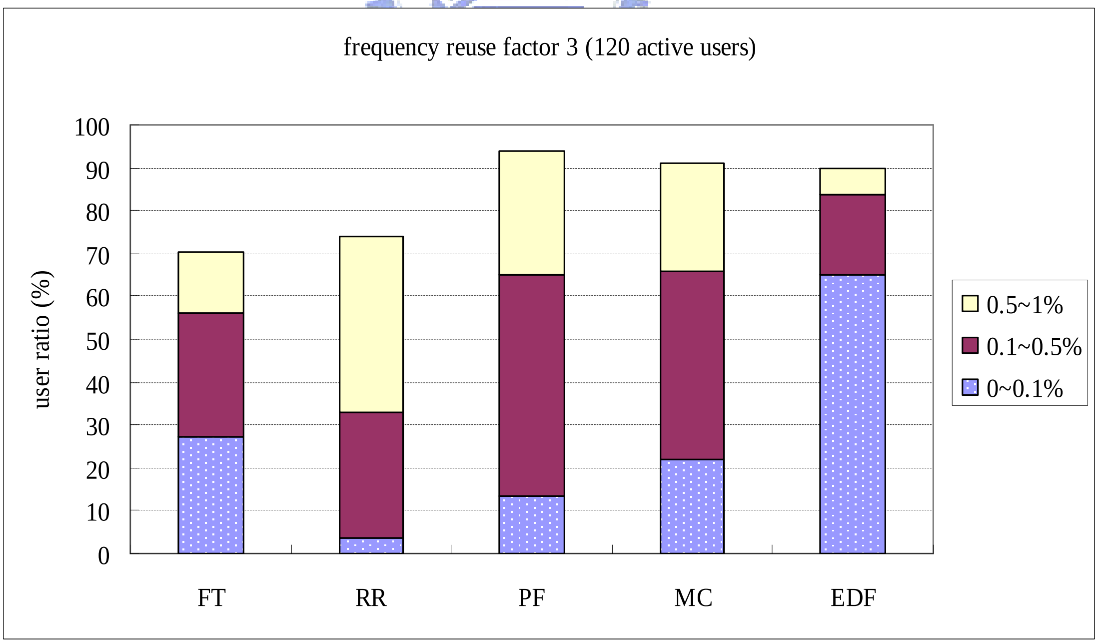

Figure 4-31 Ratio of user number of different packet loss rate ranges and 120 active users at frequency reuse factor 3

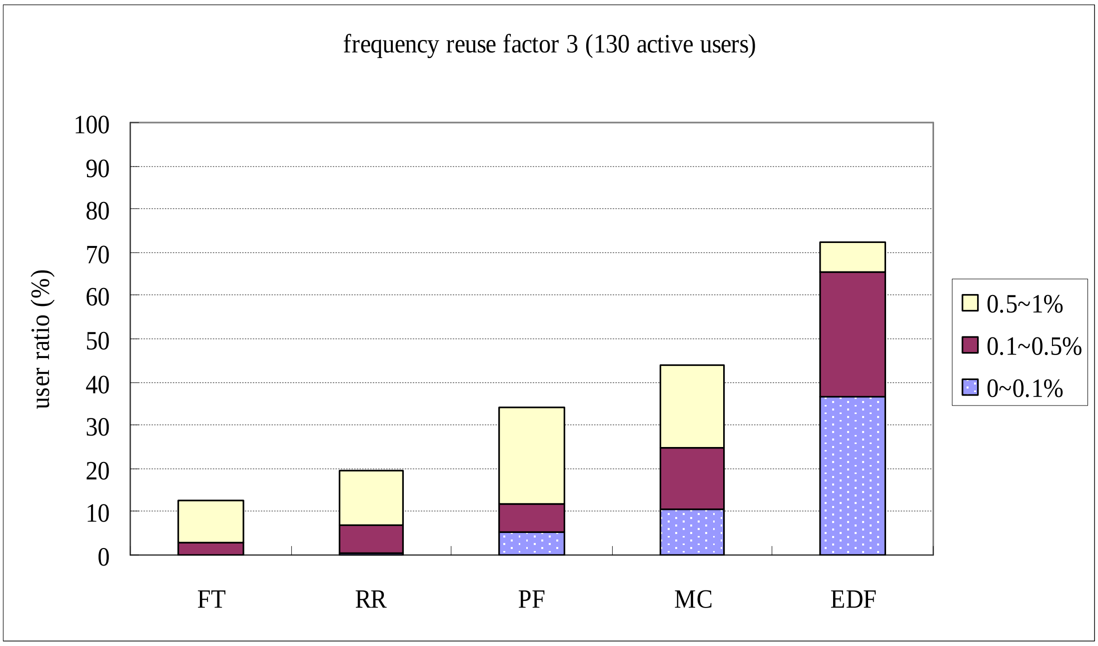

Figure 4-32 Ratio of user number of different packet loss rate ranges and 130 active users at frequency reuse factor 3

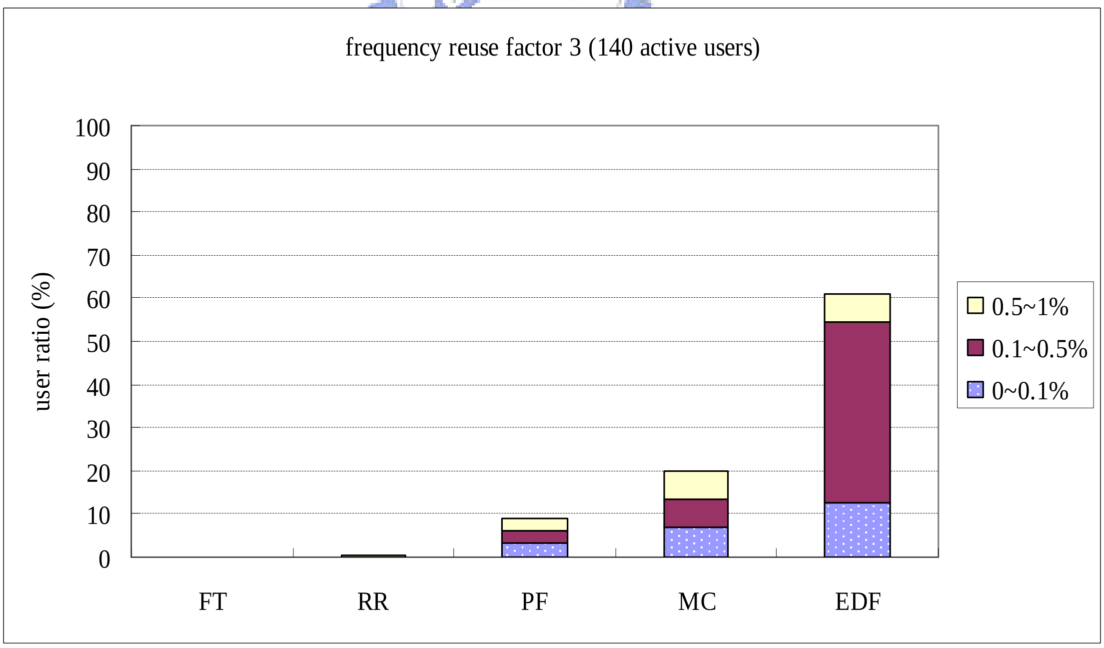

Figure 4-33 Ratio of user number of different packet loss rate ranges and 140 active users at frequency reuse factor 3

4.3 Mixed Traffic

In this section, we both use the users of non-real time services and real time services to simulate the performance targets and to observe the effects of mixed traffic. The simulation method for different service types is to fix the users of non-real time services and change the users of real-time services cyclically. In the 802.16 standard, the data packets per connection can be classified in convergence layer and sent to priority queue. It means we can distinguish different traffic type, such as real time and non-real time services. Hence, we can use hierarchy scheduling controls to maintain QoS requirements and increase system performance. Without hierarchy scheduling controls, all connections are scheduled together whatever the traffic types are and we use RR and PF to simulate. With hierarchy scheduling controls, different traffic types can have different scheduling methods and the RT users have higher priority to transmit data than NR users. We use EDF for the users of real time services and RR and PF for the users of non-real time services with hierarchy scheduling controls.

4.3.1 Throughput of Non-Real Time Service

From Figure 4-34 to 4-37, we can see the system throughputs of non-real time services per BS with and without hierarchy scheduling controls at frequency reuse factor one. The throughputs always reduce with the increase of total active users no matter if the hierarchy scheduling controls are adopted. For the non-real time services, the PF has higher throughput than RR with and without hierarchy scheduling controls. This result closely meets the discussion of 4.1.2. From these figure, we can obvious find the throughput of non-real time services with hierarchy scheduling control is lower than without in high loading. This is a tradeoff between the priorities for real time services and system throughputs for non-real time

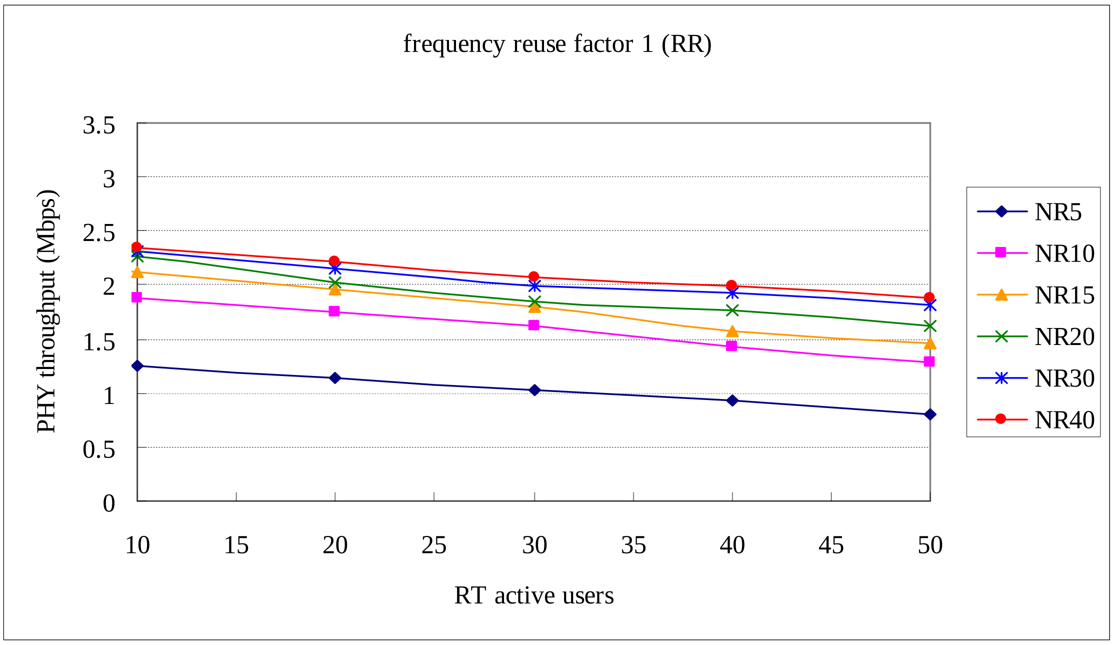

Figure 4-34 PHY throughputs of non-real time services of mixed traffic per BS with RR at frequency reuse factor 1

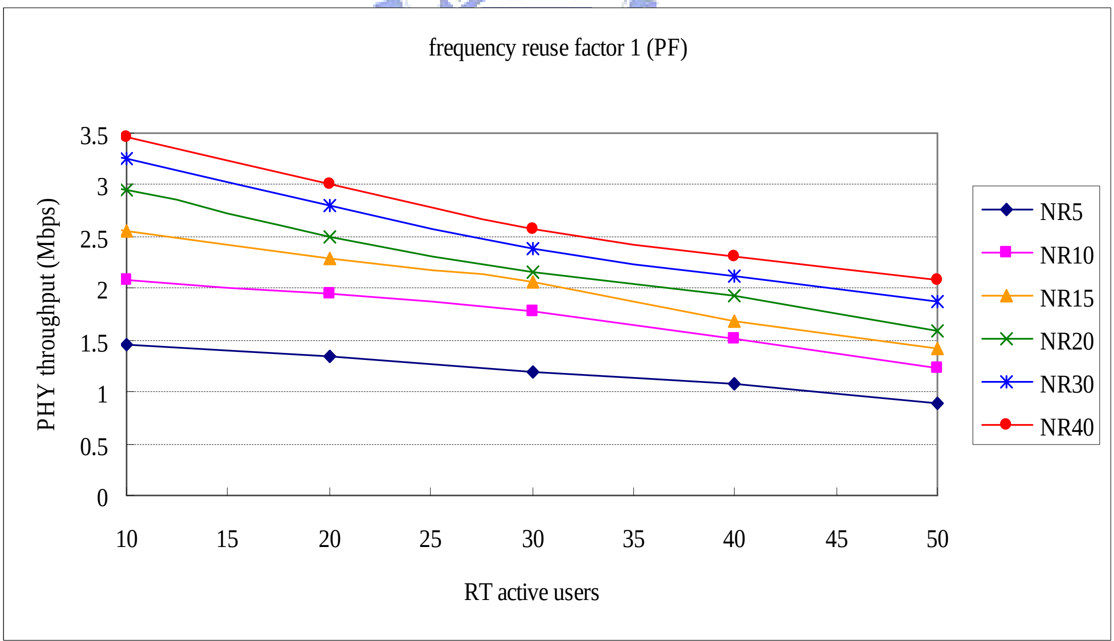

Figure 4-35 PHY throughputs of non-real time services of mixed traffic per BS with PF at frequency reuse factor 1

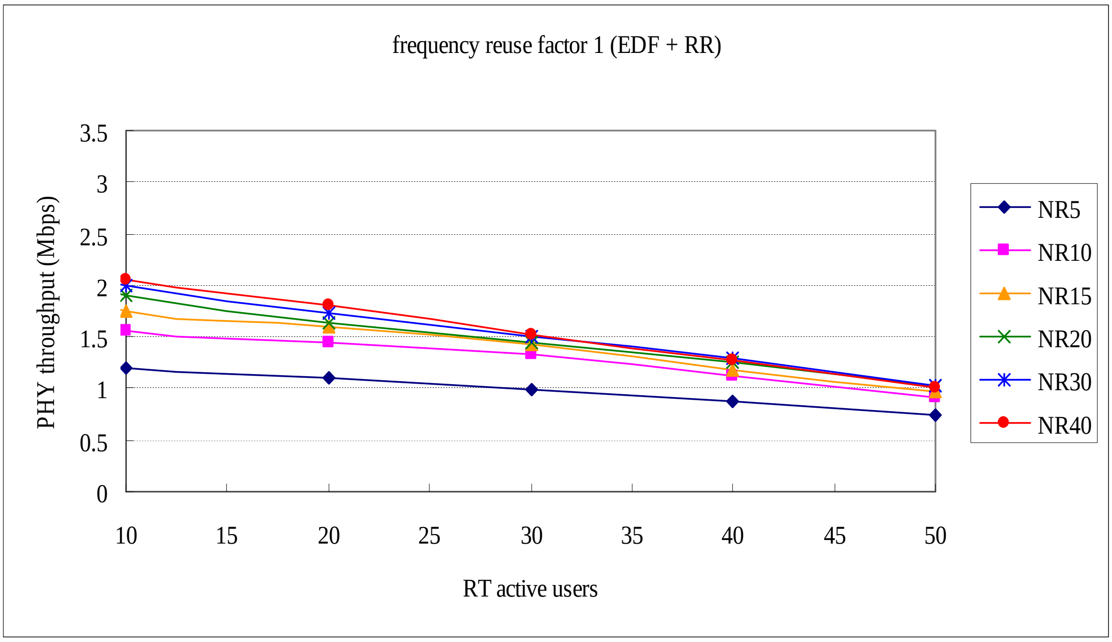

Figure 4-36 PHY throughputs of non-real time services of mixed traffic per BS with EDF+RR at frequency reuse factor 1

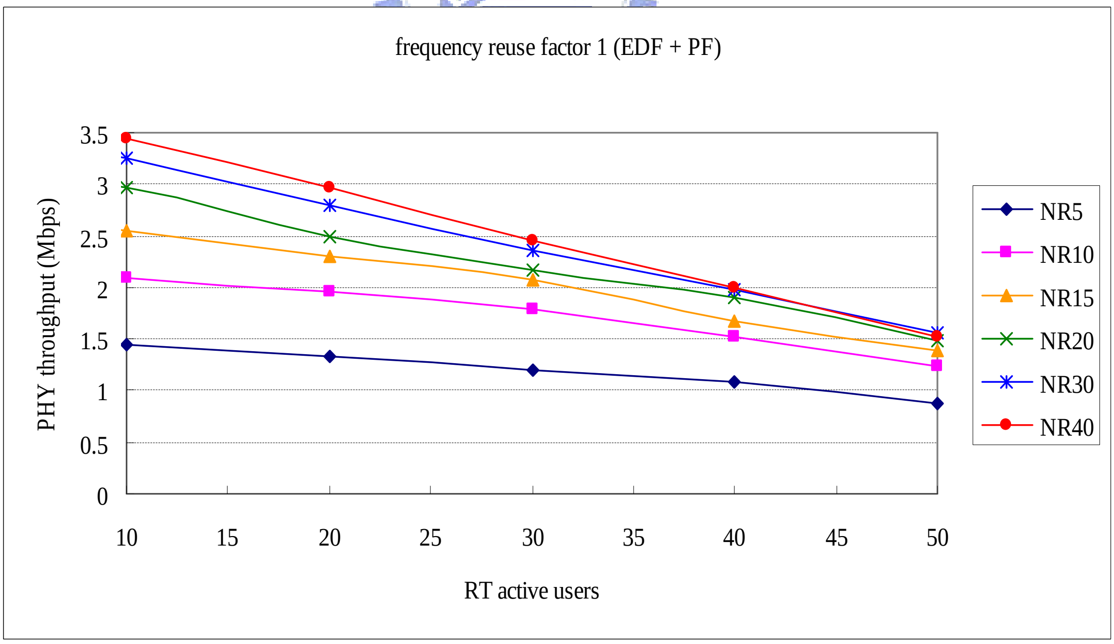

Figure 4-37 PHY throughputs of non-real time services of mixed traffic per BS with EDF+PF at frequency reuse factor 1

services. In order to have priorities to guarantee the QoS of real time services, the victims are the throughputs of non-real time services with hierarchy scheduling control.

4.3.2 Minimum Data Rate of Non-Real Time Service

Figure 4-38 and 4-39 show the minimum data rate of non-real time services without hierarchy scheduling control and Figure 4-40 and 4-41 present hierarchy scheduling controls at frequency reuse factor. From these figure, we can see the minimum data rate will decrease when the total users rise no matter with or without hierarchy scheduling controls. In Figure 4-38, if there are 40 non-real time connections (NR40) of RR without hierarchy scheduling control, the minimum data rate will below 50 kbps no matter what real time service's users (RT) are. It means no NR users can achieve 50 kbps. The minimum data rate of NR30 is below 50 kbps over 30 RT active users. The minimum data rates of NR5, NR10, NR15, and NR20 are below 50 kbps over 60 RT active users. In Figure 4-39, NR40 of PF without hierarchy scheduling control has 50 kbps data rate at 30 RT active users. NR30 and NR 20 have 50 kbps data rate at 45 and 60 RT active users, respectively. The minimum data rates of NR5, NR10, and NR15 are below 50 kbps over 60 RT active users. In Figure 4-40, NR30 and NR40 of EDF+RR with hierarchy scheduling control can't maintain the 50 kbps data rate at any RT active users. NR10, NR15, and NR20 own 50 kbps data rate at 50, 33, and 15 RT active users. In Figure 4-41, NR15, NR20, NR30, and NR40 of EDF+PF with hierarchy scheduling control achieve 50 kbps data rate at 55, 50, 37, and 23 RT active users, respectively. The minimum data rates of NR5 and NR10 will be below 50 kbps over 60 RT active users.

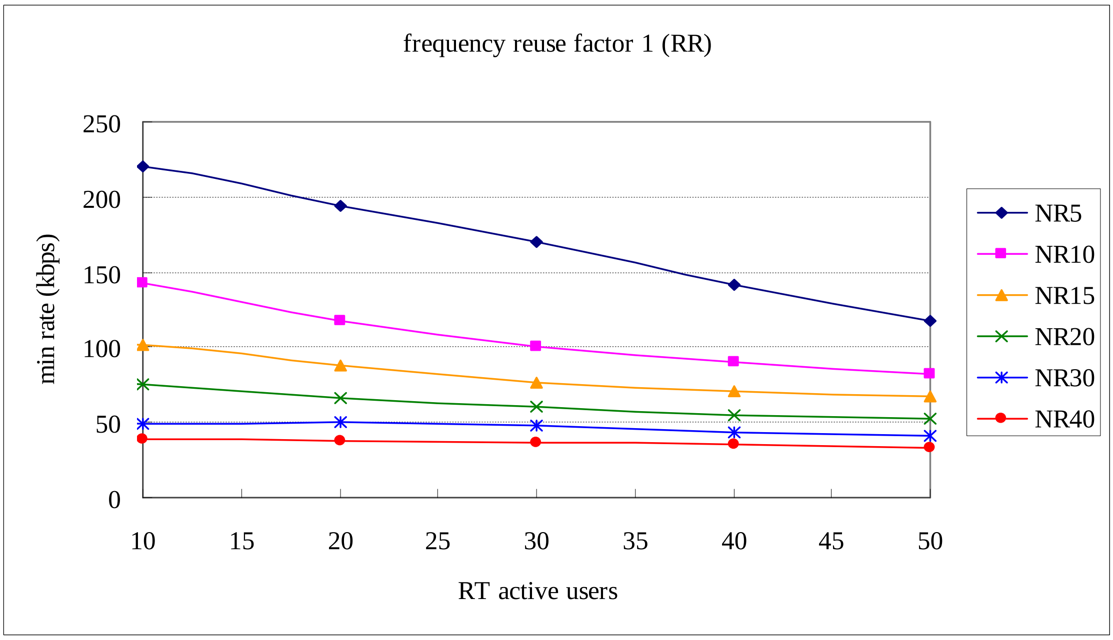

Figure 4-38 Minimum data rate of non-real time service of mixed traffic with RR at frequency reuse factor 1

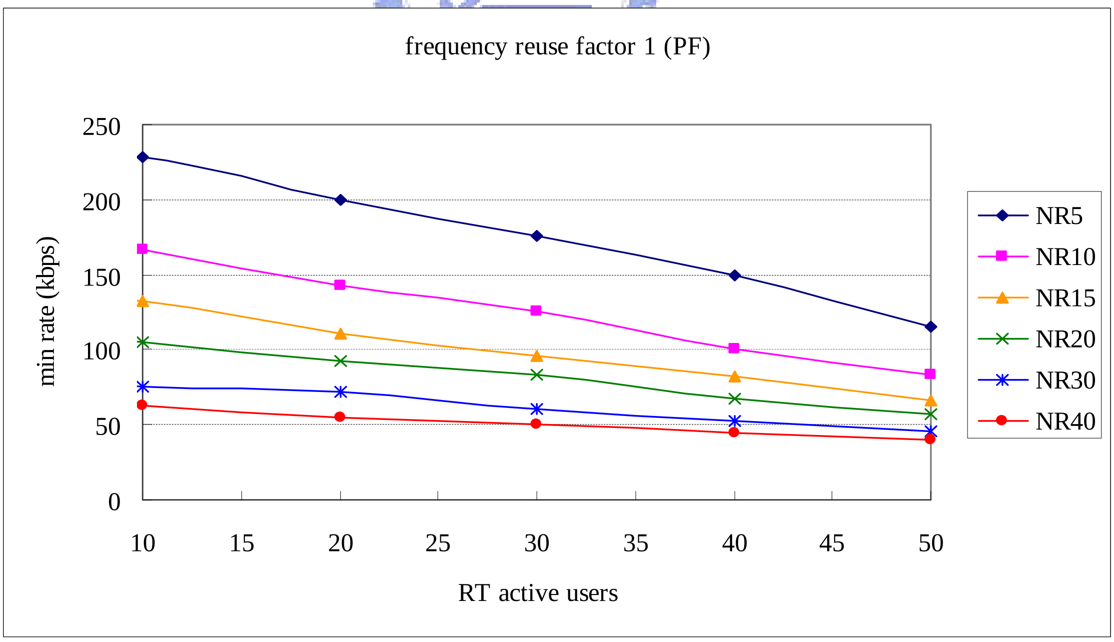

Figure 4-39 Minimum data rate of non-real time service of mixed traffic with PF at frequency reuse factor 1

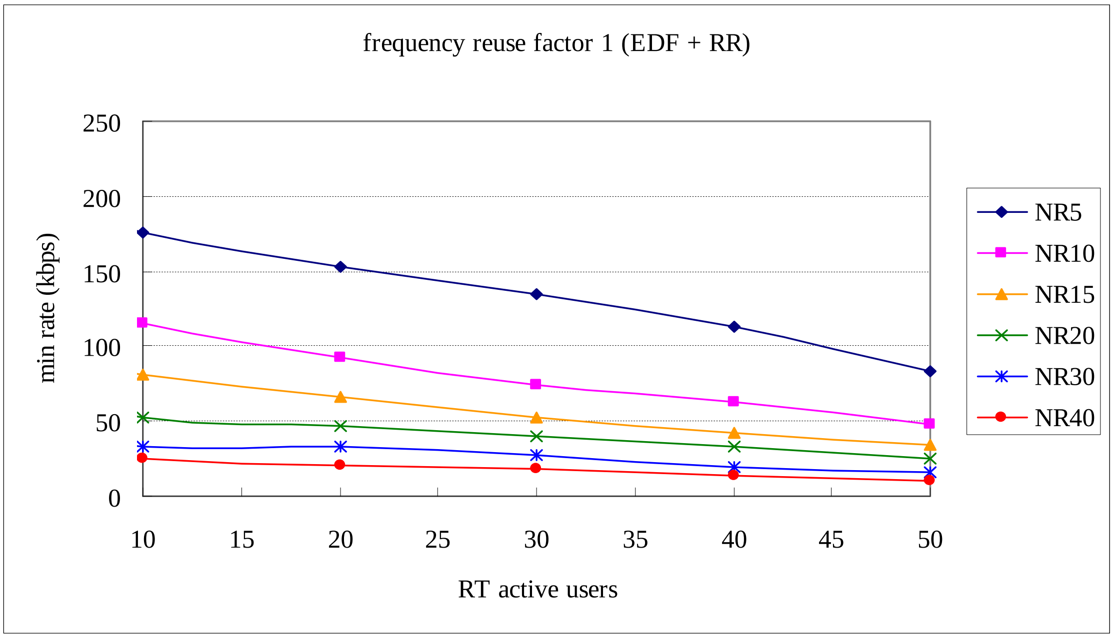

Figure 4-40 Minimum data rate of non-real time service of mixed traffic with EDF+RR at frequency reuse factor 1

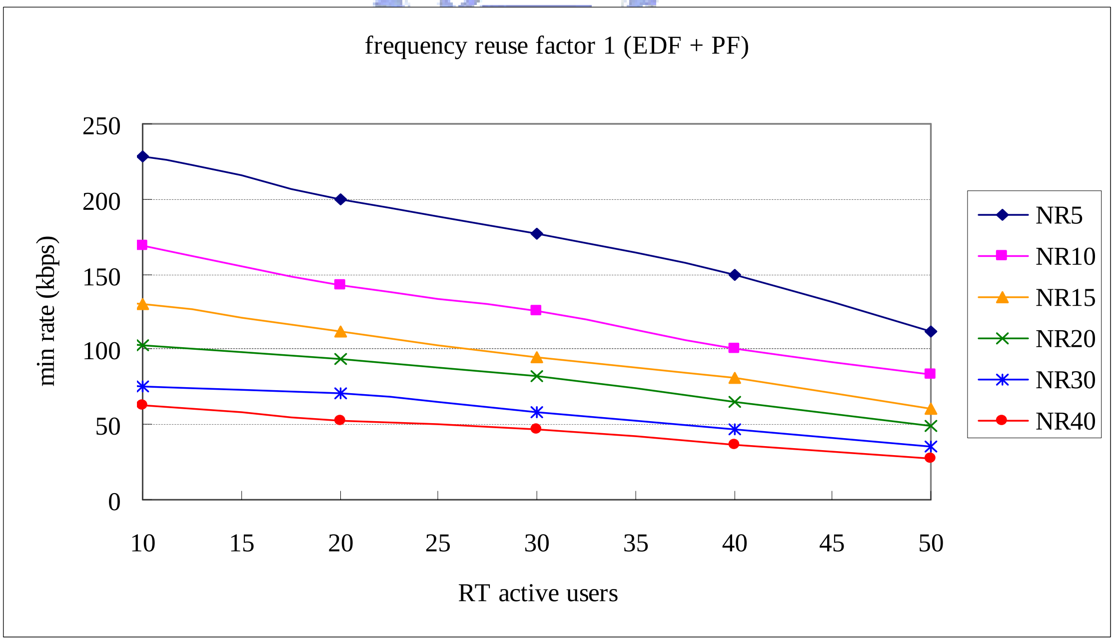

Figure 4-41 Minimum data rate of non-real time service of mixed traffic with EDF+PF at frequency reuse factor 1

4.3.3 Packet Loss Rate of Real Time Service

Figure 4-42, 4-43, 4-44, and 4-45 address the packet loss rate of real time services with and without hierarchy scheduling controls at frequency reuse factor one. In Figure 4-42, we can obvious find the packet loss rates of RR without hierarchy scheduling control are large than 1% no matter what total active users are. It means no RT active users can maintain the QoS requirement. In Figure 4-43, NR10, NR15, and NR20 of PF without hierarchy scheduling control will make packet loss rate exceed 1% at 58, 51, and 44 RT active users. The packet loss rates of NR30 and NR40 are over 1% at 28 and 11 RT active users, respectively. In Figure 4-44 and 4-45, we can see the packet loss rates with hierarchy scheduling control are very low and exceed 1% over 60 RT active users.

4.3.4 Capacity of Mixed Traffic

In 4.3.2 and 4.3.3, we can find without hierarchy scheduling control can serve more users of non-real time services which the minimum data rate is set to 50 kbps than with hierarchy scheduling control. But with hierarchy scheduling control can serve more users of real time services which the maximum packet loss rate is set to 1% than without hierarchy scheduling control. So, the capacity of mixed traffic will be defined by maintaining these two QoS requirements simultaneously. It means the 50 kbps data rate and 1% packet loss rate must be keep simultaneously no matter the ratio of users of non-real time and real time services. Figure 4-46 present the capacity of mixed traffic. From these figure, we can observe EDF+RR has higher capacity than RR. EDF+PF also has higher capacity than PF. So, hierarchy scheduling controls indeed can improve capacity. The relationship of PF and EDF+RR deserves to be mentioned. The capacity of EDF+RR is only higher than PF at lower

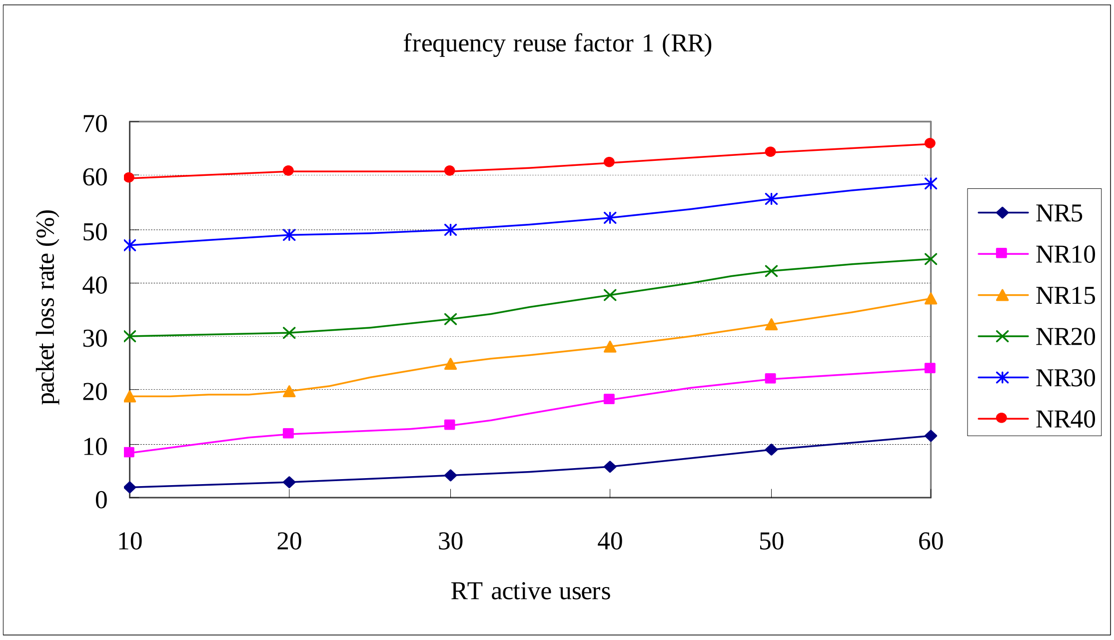

Figure 4-42 Packet loss rate of real time service of mixed traffic with RR at frequency reuse factor 1

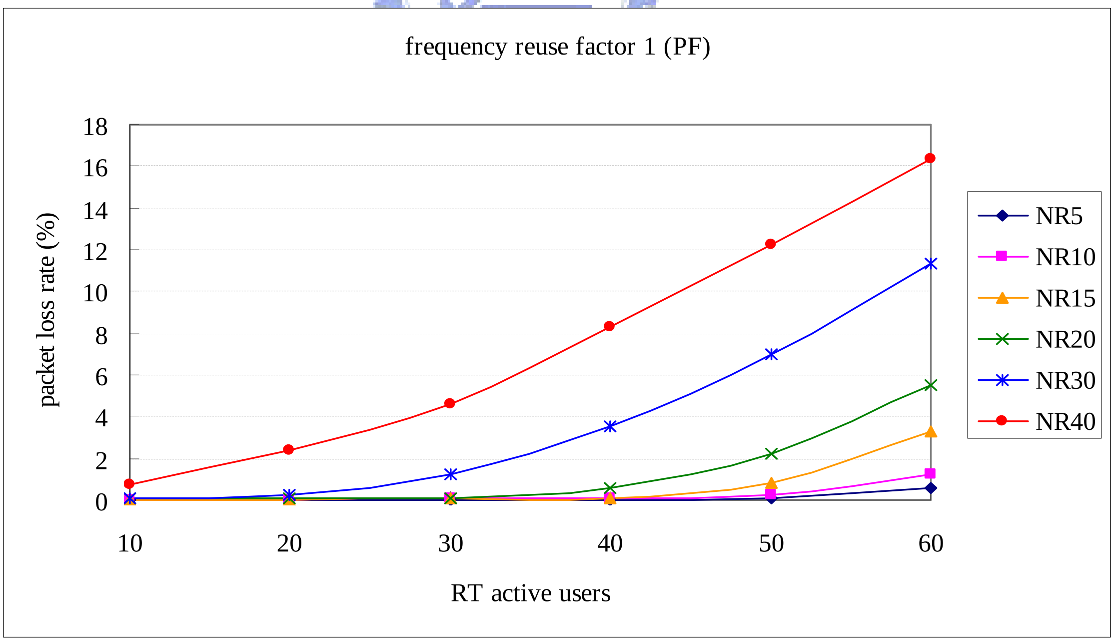

Figure 4-43 Packet loss rate of real time service of mixed traffic with PF at frequency reuse factor 1

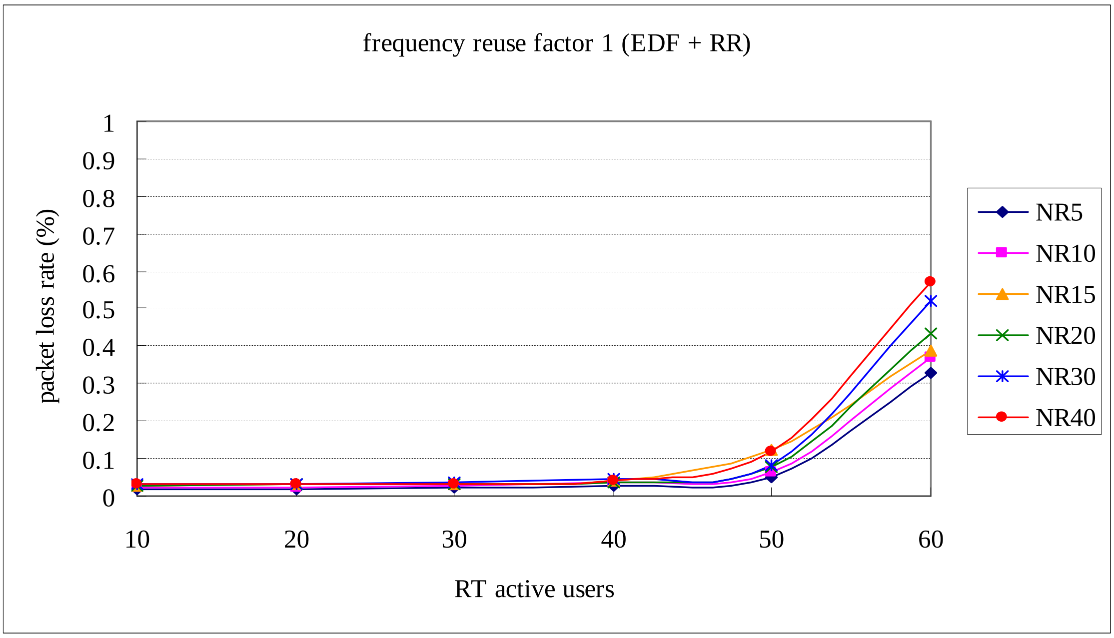

Figure 4-44 Packet loss rate of real time service of mixed traffic with EDF+RR at frequency reuse factor 1

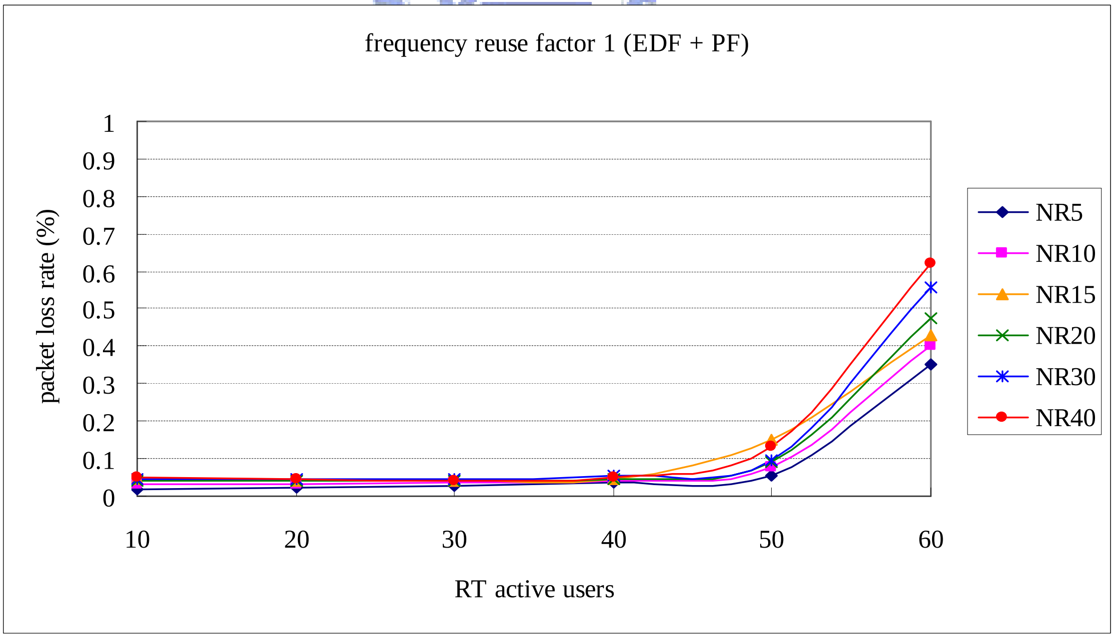

Figure 4-45 Packet loss rate of real time service of mixed traffic with EDF+PF at frequency reuse factor 1

NR active users and takes a sudden turn and then develops rapidly. The reason is the RR of EDF+RR causes low capacity of the NR active users. Even the RT active users can keep 1% packet loss rate, a large part of NR active users can't maintain 50 kbps data rate. Hence, the capacity of EDF+RR is encumbered with RR and is lower than PF.

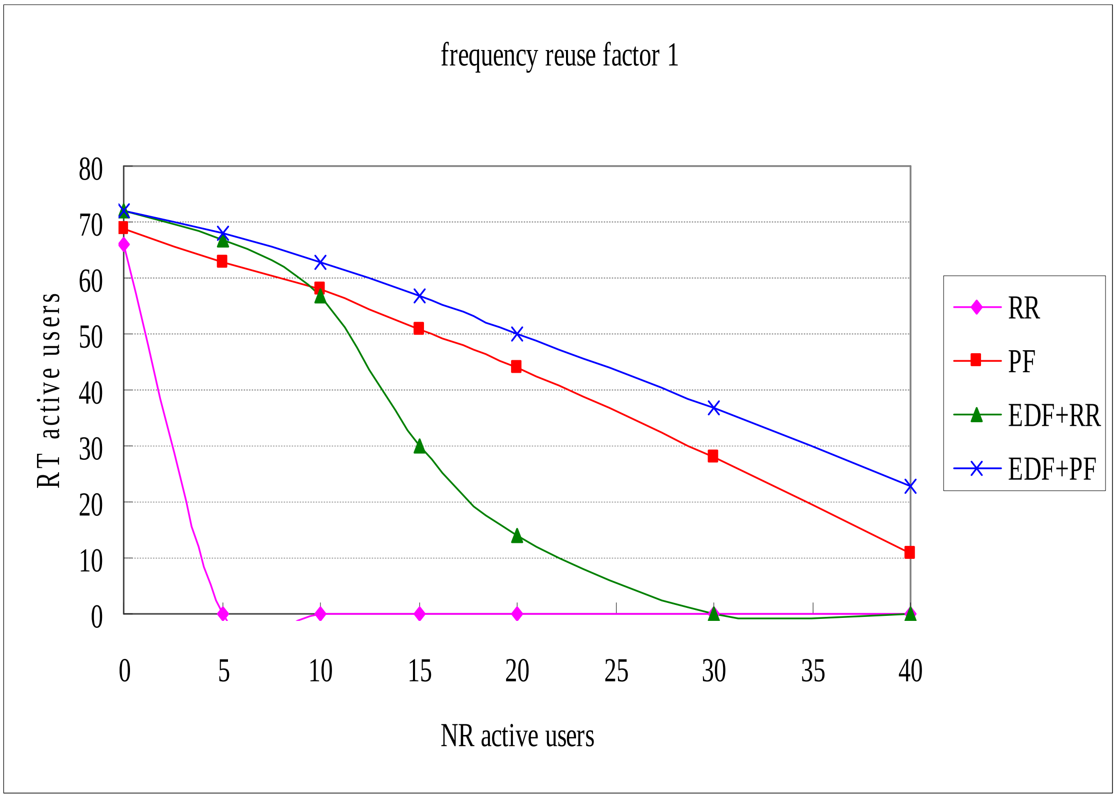

Figure 4-46 Capacity of mixed traffic at frequency reuse factor 1

Chapter 5

Conclusions and Future Works

In this thesis, our contributions are (1) the system level platform establishment of MAC layer with basic resource management in IEEE 802.16e standard and (2) the performance study of IEEE 802.16e system. First, the MAC layer in IEEE 802.16e system is implemented in our study. This platform is used for preliminary understanding of IEEE 802.16e system. Secondly, non-real time service's performance with different scheduling controls is studied. After that, real time service's performance with different scheduling controls is investigated. We also address the admission control criterion for non-real and real time service. The impact from different frequency reuse factor is presented, too. Finally, the impact of hierarchy scheduling control with priority queues is shown.

From these performance simulations, we can get the conclusions as below. For non-real time service, Maximum CINR has maximum throughput and minimum scheduling delay than other scheduling controls, but it has maximum scheduling jitter and need to do admission control the most. On the contrary, proportional fair can handle these performance tradeoff more balance and is more suitable for non-real time service. For real time service, although the needs of admission controls for early deadline first, maximum CINR, and proportional fair are very closely, the early deadline first has lower packet loss rate than other methods and is more agreeable to real time services. The scheduling control, considered the channel condition (with CQICH) and OFDMA permutation, can still offer a good performance in single frequency cell coverage of mobile WiMAX system. Finally, the simulations of hierarchy scheduling controls tell us that using hierarchy and appropriate scheduling controls can improve the system capacity with QoS guaranteed in a mixed traffic of IEEE 802.16e system. Therefore, the complete performance of IEEE 802.16e system is investigated in this thesis.

In the future works, we will add the fast fading which is caused by Doppler and multipath effects in our simulation platform. And, we will focus on the methods for capacity boost, such as advanced scheduling controls, multi-input and multi-output (MIMIO) provided by space time coding (STC) and Adaptive Antenna Systems (AAS), and hybrid-ARQ (HARQ) provided by turbo coding.

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