國立交通大學

網路工程研究所

碩士論文

一個支援 IEEE 802.16e 輪詢的適應性頻寬需求機制

An adaptive bandwidth request scheme for IEEE 802.16e polling services

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摘　要

隨著無線網路的逐漸普及，為了能夠有大範圍、高頻寬傳輸效能且達到“最後一哩”目標的 IEEE 802.16 協定因而產生。IEEE 802.16(WiMAX)還是個開發中的技術，所以尚有許多空間值得改進及討論，我們論文的重點是針對服務品質來加以研究。

在 IEEE 802.16 的機制中，在傳輸各種資料前，都需要用戶端傳送頻寬需求給基地台，當頻寬不足以讓各個用戶端都用單一輪詢的方式做頻寬需求時，基地台會讓群組用戶端使用多點輪詢和廣播輪詢的方式競爭傳輸頻寬需求的時槽。802.16e 提出增強的即時輪詢服務(ertPS)是為了服務靜音壓縮的 IP 網路語音傳輸技術(VoIP)，這種傳輸技術有時間上的要求，延遲要低才能達到較好的效能。

為了降低碰撞產生的延遲，我們提出一種方法讓增強的即時輪詢服務的資料流透過不用競爭的區段去取得頻寬傳輸的時槽，且同時也保有原本競爭的方法以確保能夠在一定時間內傳輸頻寬需求，達到減少延遲、增加流通量的目標。最後我們模擬出來的結果，在用戶端逐漸增加的情況下，增強的即時輪詢服務的需求頻寬延遲可以保持較佳的數值，且平均流通量也有較好的輸出值。

An adaptive bandwidth request scheme

for 802.16e polling services

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Abstract

Due to the population of wireless networks, IEEE 802.16 protocol was developed in order to achieve large coverage, high bandwidth and last-mile access. IEEE 802.16, also called WiMAX, is still a technology under developing, and there are various issues that are worthy of investigation. This thesis focused on the study of QoS over WiMAX. Before transmitting various kinds of packet based on the scheme of 802.16, subscriber stations (SS) must transmit a bandwidth request message to the base station (BS). When bandwidth is not sufficient to allow each SS using unicast polling to get bandwidth, BS will form a group of stations that utilize multicast polling and broadcast polling to contend the slots of transmitting bandwidth request. Extended real-time polling service which serves the VoIP flows with silence suppression is proposed in 802.16e. Since this technology has certain delay time requirement, and the lower delay will achieve higher QoS performance.

In order to decrease the delay which was caused by collisions, we proposed a scheme that could utilize contention-free period to get slots for bandwidth request, they also save the original contention period to achieve our goal that decreases delay and increases average throughput. The simulation results show that our method features lower delay time and better performance of ertPS flows due to the increasing number of stations.

Keywords: 802.16, WiMAX, QoS, bandwidth request, contention period

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Chapter 1 Introduction

During recent years, the increasing demands for high-speed Internet access and multimedia services in the residential and small office sectors have expected the development of last-mile access technologies, especially through wireless.

In the mid-1990's, various groups began to promote "last-mile" fixed wireless access solutions, they tried to achieve some goals. First, to provide the capacity and reliability of wireline but with the flexibility and ease of deployment of wireless. Second, to provide a hierarchical system for corporate or institutional backhaul / distribution networks. Third, to break the monopolies of incumbent carriers.

In 2001, the IEEE 802.16 standard for BWA (Broadband Wireless Access) systems which operate in the 10-66 GHz range was released. Although the single carrier modulation during 10-66 GHz features 30 miles transmission radius and 70Mbps transmission rate, but the Line-of-Sight (LOS) limitation is still a problem. Until January 2003, 802.16a was published. It does not only include new PHY layer specification and enhanced MAC functionality, but also add 2-11 GHz in spectrum. So the signal can be used in Non-Line-of-Sight (NLOS) environment and the single antenna can also serve multiple stations. Several years later, IEEE 802.16a / b / c and various updates were incorporated into IEEE 802.16-2004 [1].

The Worldwide Interoperability for Microwave Access (WiMAX) technology based on the IEEE 802.16-2004 Air Interface Standard is rapidly proving itself as a technology that will play a key role in fixed broadband wireless metropolitan area networks. The first certification lab, established at Cetecom Labs in Malaga, Spain is fully operational and lots of trials are underway in Europe, Asia, Africa and North and South America [21]. Unquestionably, Fixed WiMAX, based on the IEEE 802.16-2004 Air Interface Standard, has proven to be a cost-effective fixed wireless alternative to cable and DSL services. In December, 2005 the IEEE ratified the 802.16e amendment to the 802.16 standard [2]. This amendment adds the features and attributes to the standard that is necessary to support mobility. The WiMAX Forum is now defining system performance and certification profiles based on the IEEE 802.16e Mobile Amendment. Beyond the air interface, the WiMAX Forum is defining the network architecture necessary for implementing an end-to-end Mobile WiMAX network. [3]

A network that utilizes a shared medium shall provide an efficient sharing mechanism. The WiMAX architecture depends on point to multipoint (PMP) and Mesh networks to achieve medium sharing. The medium means the space which can propagate the radio waves. The PMP mode is illustrated in Figure 1.1.

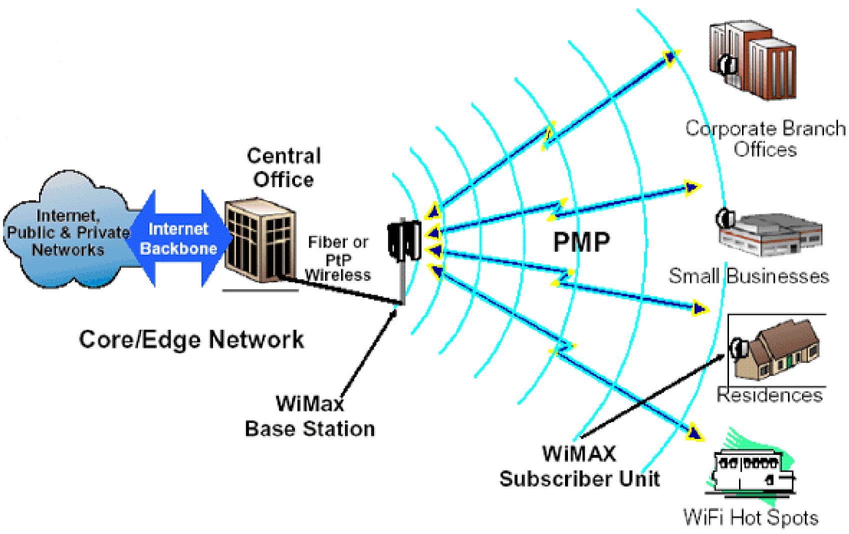

Figure 1.1 IEEE 802.16 PMP mode [22].

WiMAX provides fixed, nomadic, portable, and mobile wireless broadband connectivity without the need for line-of-sight with a base station. In a typical cell radius deployment of three to ten kilometers, WiMAX Forum Certified systems can be expected to deliver capacity of up to 40 Mbps per channel, for fixed and portable access applications. This bandwidth is sufficient to simultaneously support hundreds of T-1 speed connectivity and thousands of residences with DSL speed connectivity. Mobile network deployments are expected to provide up to 15 Mbps of capacity within a typical cell radius deployment of up to three kilometers. It is expected that WiMAX technology will be incorporated in notebook computers and PDAs by 2007, allowing for urban areas and cities to connect with each other for portable outdoor broadband wireless access.

We know that the WiMAX has been analyzed in lots of countries and organizations, such as Nokia and Intel will cooperate to develop the WiMAX technologies. South Korea's electronics and telecommunication industry spearheaded by Samsung Electronics and ETRI has developed its own standard, WiBro. In late 2004, Intel and LG Electronics have agreed on interoperability between WiBro and WiMAX. The telephone and telegraph office in Taiwan also work hard for opening the spectrum up.

We may ask whether one size of WiMAX transmission scope really fit all conditions. Different applications have different requirements and constraints for spectrum and performance. So to allow options within a consistent framework is a suitable idea, because factories can choose serviceable standard profiles from limited set of standard profiles.

IEEE 802.16 seeks to provide the required features to serve users. One main element of WiMAX technology is the interoperability of WiMAX equipment that can integrate much wireless stations or products and let them work together. A common platform could achieve lower cost. Fixed wireless equipments will be able to use the same modem chipset used in personal computers (PCs) and PDAs. For short distance, the equipments will be similar to a cable or DSL and the base stations will be able to use the same chipsets developed for low-cost WiMAX access points. So they also provide lower cost equipments which support WiMAX and original functionality.

The technology behind WiMAX has been optimized to provide Non-Line-of-Sight (NLOS) coverage. Non-Line-Of-Sight advantages are coverage of wider area and better predictability of coverage. A key advantage of WiMAX is to use OFDM over single carrier modulation schemes with the ability to deliver higher bandwidth efficiency and therefore higher data throughput, with more than 1 Mbps downstream and even much higher data rates, even in NLOS with multipath conditions. Adaptive Modulation also increases link reliability for carrier-class operation and the possibility to keep 64 QAM modulation at wider distance extend full capacity over longer distances. Optimized handover also ensures the time being less than 50ms.

QoS (Quality-of-Service) is an important issue in wireless network especially for those service flows belonging to voice and video. IEEE 802.16 has already specified complete construction, including how they achieve QoS requirement. In 802.16, there are five types of scheduling services specified for different traffic models, i.e. Unsolicited Grant Service (UGS), real-time Polling Services (rtPS), non-real-time Polling Service (nrtPS), Best Effort (BE), and Extended real-time Polling Service (ertPS) which was published in IEEE 802.16e. Among the five service flows, the QoS support in UGS, rtPS, and ertPS is necessary. Here we briefly introduce what problem may occur in 802.16 and how can we solve it.

Before the uplink data transmission in IEEE 802.16 PMP mode, the subscriber station must send their bandwidth request first to base station and let it know how much bandwidth is required. When the whole bandwidth isn't enough for all stations to send individual bandwidth request, the service flows belonging to ertPS, nrtPS, and BE have to count on contention to send their requests. There is certain collision probability during contention period and that would cause the request delay for the ertPS flows. When the request was delayed, the station couldn't get proper bandwidth and as a request its QoS will degrade. So how to solve the collision problem will be discussed in this thesis.

The rest of this thesis is organized as follows. Chapter 2 provides an overview of the 802.16 system. Chapter 3 describes the detailed problem in 802.16 and our proposed scheme in detail. The numerical analysis and simulation evaluation are presented in Chapter 4. Conclusions are stated in Chapter 5.

Chapter 2 Background

In this chapter, we first introduce the 802.16 MAC, including the scheduling services, bandwidth allocation, request, and polling mechanisms. Regarding the scheduling services, QoS issues would be discussed in more detail. As we have addressed in Chapter 1, bandwidth allocation will be based on bandwidth request from each subscriber station (SS). So the timing of sending request and the probability of successful polling would affect the transmission efficiency.

Next we briefly review the WiMAX frame architecture and MAC support of PHY. Several duplexing techniques are supported by MAC protocol. The choice of duplexing technique may affect certain PHY parameters as well as impact the features that can be supported. So we will introduce this architecture first, then we will address related works that have been proposed.

2.1 Review of IEEE 802.16 MAC

2.1.1 Scheduling services

Scheduling services represent the data handling mechanisms supported by the MAC scheduler for data transport in a connection. Each connection is associated with a single scheduling service. A scheduling service is determined by a set of QoS parameters about its behavior. These parameters are managed by using the Dynamic service addition (DSA) and Dynamic service change (DSC) message dialogs. Five services: Unsolicited Grant Service

(UGS), Real-time Polling Service (rtPS), Non-real-time Polling Service (nrtPS), Best Effort (BE), and extended real-time Polling service (ertPS) are supported in 802.16e.

Uplink request / grant scheduling is performed by the BS with the intent of providing each SS with bandwidth for uplink transmissions or opportunities to request bandwidth. By specifying a scheduling service and its associated QoS parameters, the BS scheduler can anticipate the throughput and provide polls and/or grants at the appropriate times. Table 2.1 summarizes the scheduling services and the poll / grant options. The following sections define flow scheduling services for uplink operations.

Scheduling type	**PiggyBack request**	**Bandwidth stealing**	**Polling**
UGS	Not allowed	Not allowed	PM bit is used to request a unicast poll for bandwidth needs of non-UGS connections.
rtPS	Allowed	Allowed	Scheduling only allows unicast polling.
nrtPS	Allowed	Allowed	Scheduling may restrict a service flow to unicast polling via the transmission / request policy; otherwise all forms of polling are allowed.
BE	Allowed	Allowed	All forms of polling allowed.
ertPS	Allowed	Allowed	All forms of polling allowed.

Table 2.1 Scheduling services and usage rules

Unsolicited Grant Service (UGS)

The UGS is designed to support real-time service flows that generate fixed-size data packets on a periodic basis, such as T1/E1 and Voice over IP without silence suppression.

The services privide fixed-size grants on a real-time periodic interval which eliminate the overhead and latency of SS requests and assure that grants are available to confirm the flow’s real-time requirement.

The BS should provide Data Grant Burst information elements (IEs) to the SS at periodic intervals based on the Maximum Sustained Traffic Rate of the service flow. The size of these grants should be sufficient to hold the fixed-length data associated with the service flow. In order for this service to work correctly, the Request / Transmission Policy must be set so that the SS is prohibited from using any contention request opportunities for this connection.

Real-time Polling Service (rtPS)

The rtPS is designed to support real-time service flows that generate variable size data packets on a periodic basis, such as moving pictures experts group (MPEG) video.

This service provides real-time, periodic, and unicast request opportunities which conform to the flow's real-time requirement and allow the SS to specify the size of the desired grant. This service requires more request overhead than UGS, but supports variable grant sizes for optimum data transport efficiency. The BS should provide periodic unicast request opportunities, and it may issue unicast request opportunities even if prior requests are currently unfulfilled. SSs which are belonging to this category should use only unicast request opportunities in order to obtain uplink transmission opportunities.

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, for example, FTP may use nrtPS.

The service provides unicast polls on a regular interval, which assures that the service flow receives request opportunities even during network congestion. The BS polls nrtPS flows on an interval. The BS should provide timely unicast request opportunities.

Best Effort Service(BE)

The BE service is designed to support data streams without minimum service level required, therefore it may be handled on a channel-free interval, such as web browsing and E-mail. The purpose of the BE service is to provide efficient service for best effort traffic and use contention period to get transmission opportunities.

Extended real-time Polling service (ertPS)

The Extended rtPS is designed to support real-time service flows that generate variable size data packets on a periodic basis, such as Voice over IP services with silence suppression.

It is a scheduling mechanism which builds up the efficiency of both UGS and rtPS. The BS should provide unicast grants in an unsolicited service flow like in UGS to save the latency of a bandwidth request. However, the UGS allocations are fixed size and ertPS allocations are dynamic. The BS may provide periodic UL allocations that may be used for requesting the bandwidth as well as for data transfer. By default, size of allocations corresponds to the current value of Maximum Sustained Traffic Rate of the connection.

The BS should not change the size of UL allocations until receiving another bandwidth change request from the MS. When the bandwidth request size is set to zero, the BS may only provide allocations for bandwidth request header or no allocations at all. In case that no unicast bandwidth request opportunities are available, the MS may use contention request opportunities for that connection to inform the BS regarding that it has data to send.

2.1.2 Bandwidth allocation and request mechanisms

Increasing (or decreasing) bandwidth requirements is necessary for all services except incompressible constant bit rate UGS connections. The bandwidth allocations of incompressible UGS connections do not change between connection periods. The requirements of compressible UGS connections, such as channelized T1, may increase or decrease depending on the traffic condition. When an SS needs to ask for bandwidth on a connection with its scheduling service, it sends a message to the BS containing the immediate requirements. QoS for the connection was established at connection establishment and is looked up by the BS.

Requests

Requests refer to the mechanism used by SSs to inform to the BS regarding their need of uplink bandwidth allocation. A Request may come as a stand-alone bandwidth request header or it may come as a PiggyBack Request.

The Bandwidth Request message may be transmitted during uplink bandwidth allocation, except during initial ranging interval. Bandwidth Requests may be incremental or aggregate. When the BS receives an incremental Bandwidth Request, it adds the quantity of bandwidth requested to its current bandwidth in the connection. Otherwise, if the BS receives an aggregate Bandwidth Request, it should replace its assigned bandwidth of the connection with the quantity of bandwidth requested. The Type field in the bandwidth request header indicates whether the request is incremental or aggregate. Since Piggybacked Bandwidth Requests do not have a type field, Piggybacked Bandwidth Requests should be always incremental.

Finally, when the BS receives a Bandwidth Request from any SS, it should make use of admission control scheme to check whether the request is permitted or not.

Polling

Polling is the process in which the BS allocates bandwidth to SS by sending request. These allocations may be to an SS or to a group of SSs. The polling should be done on SS basis. Note that bandwidth is always requested by a CID and it is allocated to an SS. According to the bandwidth congestion situation, there are three polling types, unicast, multicast, and broadcast.

First, we introduce the procedure of unicast polling. When an SS is polled, there is no explicit message that can be sent to poll SS. Rather, the BS actively allocate SS slots in UL-MAP (introduced in section 2.1.3) for sending bandwidth request. UGS connections do not need to be polled individually, because they need a much efficient way to get bandwidth unless they set the poll-me (PM) bit in the packet header. Unicast polling could only be used if the bandwidth is sufficient for polling whole individual SSs.

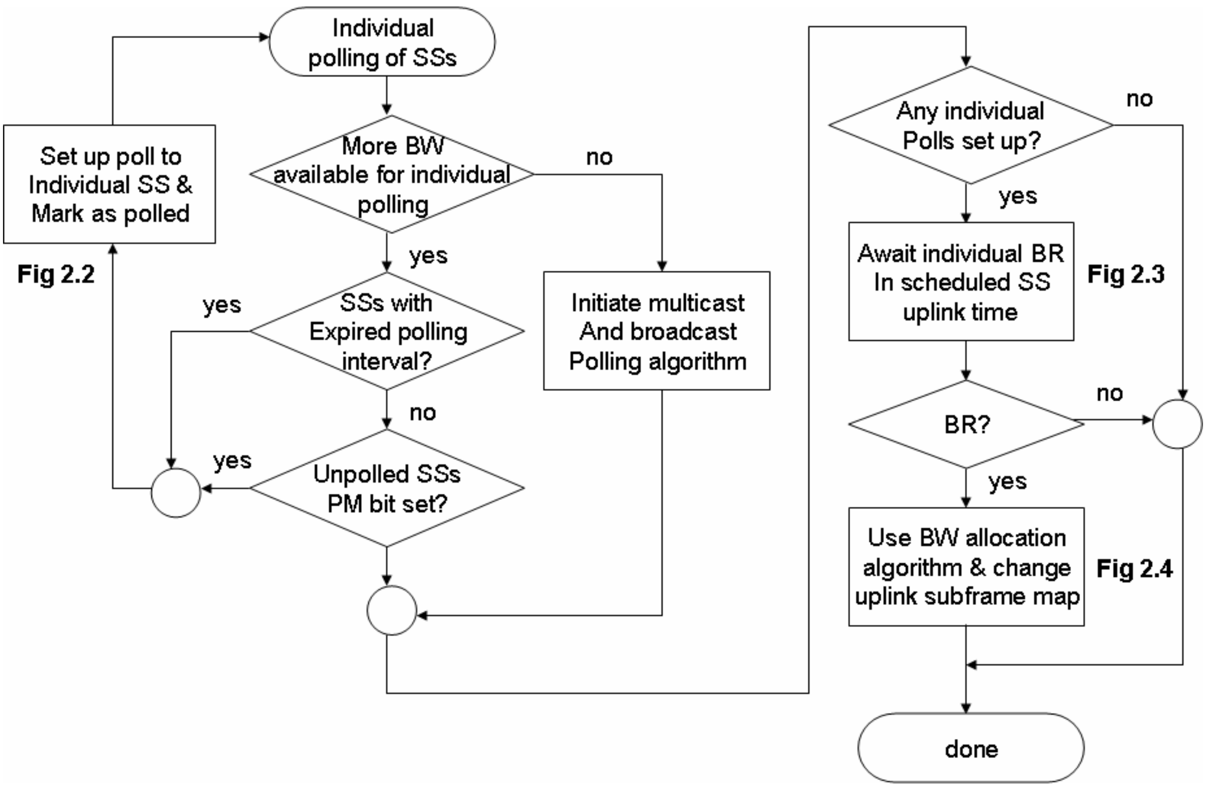

Figure 2.1 Unicast Polling

As illustrated in Figure 2.1, first the unicast polling procedure should check whether more BW is available for individual polling or not. If it is not enough, the multicast and broadcast polling algorithm should be initiated. Then we continue to make some check, for example, were SSs with expired polling interval? Did unpolled SSs PM bit set? After the necessary checking procedure, the SS would be polled.

In Figure 2.2, it shows that SS's polling interval was already expired and BS allocated the bandwidth to SS. The allocated slots were defined in UL-MAP.

Then, we can realize that SS would transmit bandwidth request in allocated slots, just like Figure 2.3. It also provides contention period to a group SSs.

Finally, BW allocation algorithm would be used and uplink subframe map would be changed, as shown in Figure 2.4.

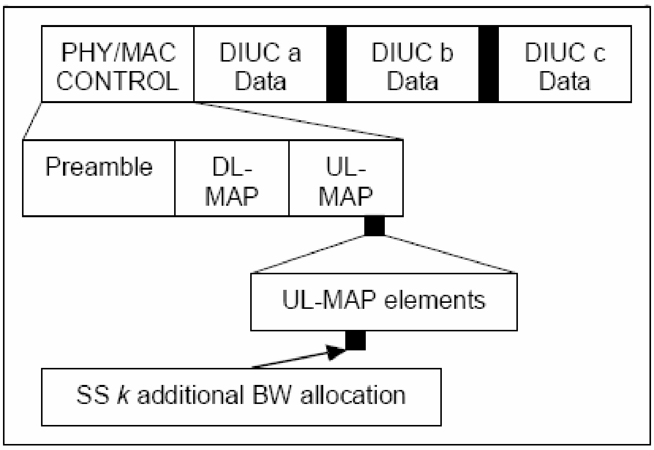

Figure 2.2 Additional BW allocation in unicast polling

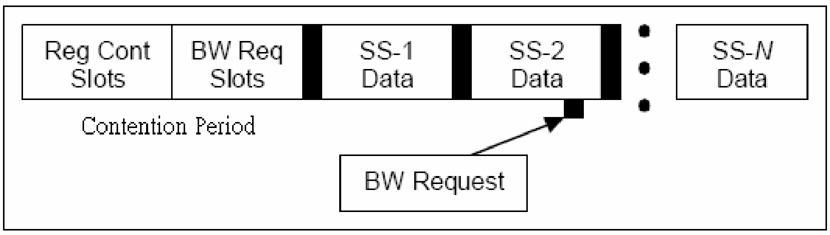

Figure 2.3 Unicast Polling BW Request

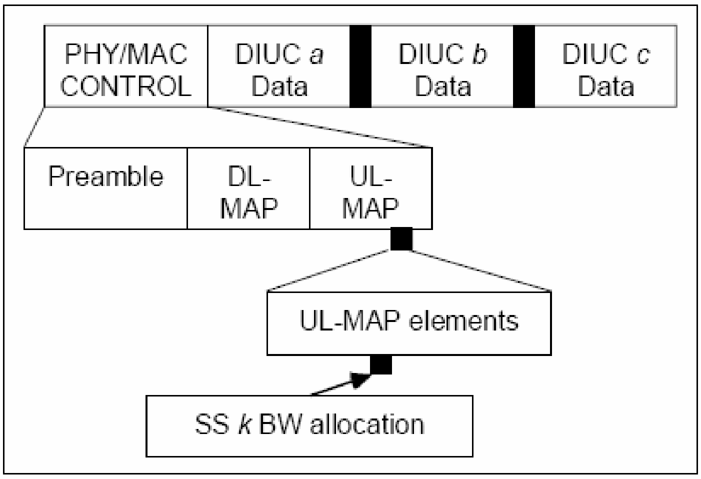

Figure 2.4 Unicast Polling BW allocation for data

If available bandwidth is insufficient for polling whole individual SS, some SSs may be polled by multicast polling or broadcast polling. Certain CIDs are reserved for the two polling types, when some SSs share one multicast or broadcast polling, they should be assigned the same CID. An example is provided in Table 2.2.

Interval description (UIUC = uplink interval usage code)	UL-MAP IE fields		
	CID (16 bits)	UIUC (4 bits)	Offset (12 bits)
Initial Ranging	0000	2	0
Multicast group 0xFFC5 Bandwidth Request	0xFFC5	1	405
Multicast group 0xFFDA Bandwidth Request	0xFFDA	1	605
Broadcast Bandwidth Request	0xFFFF	1	805
SS 5 Uplink Grant	0x007B	4	961
SS 21 Uplink Grant	0x01C9	7	1136

Table 2.2 Sample UL-MAP with multicast and broadcast IE

The information exchange procedure for multicast and broadcast polling is shown in

Figure 2.5. Different from individual polling that depends on contention period to select the slot through which to transmit its bandwidth request. In order to avoid collision, they use contention resolution algorithm to decrease the collision probability. If no grant has been received during the specified timeout duration, we could assume that the transmission was unsuccessful.

Based on Figure 2.5, we could discuss the procedure in more detail. Just like unicast polling, SS must be allocated the slots first when SS's polling interval was expired, as shown in Figure 2.6.

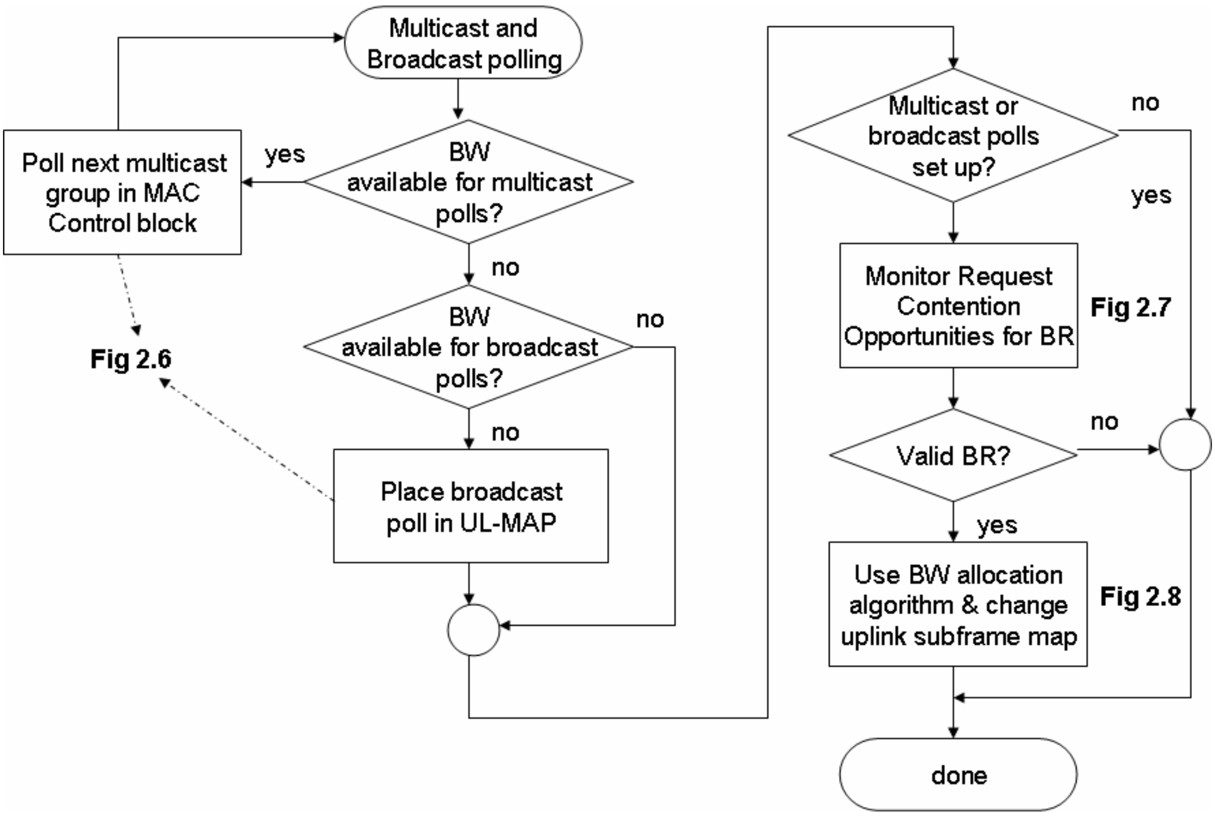

Figure 2.5 Multicast and Broadcast Polling

After the SSs have been allocated a uplink duration, they could participate the contention period to get the transmission opportunity. If a certain SS has already obtained the slot and transmitted the request successfully, the BS would check the request in accordance with admission control scheme. Otherwise, SSs must wait for backoff time until it gets TXOP. The BW request message must include the SS ID, connection ID, and the requested bandwidth as shown in Figure 2.7. Last, the data transmission could be transmitted as illustrated in Figure 2.8.

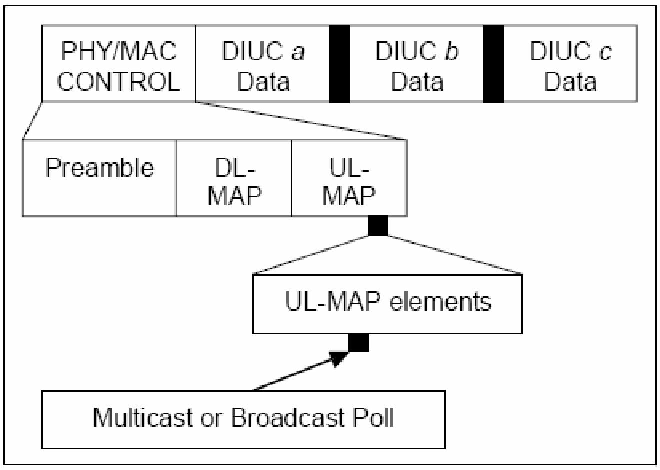

Figure 2.6 Multicast or Broadcast Polling BW allocation for BR

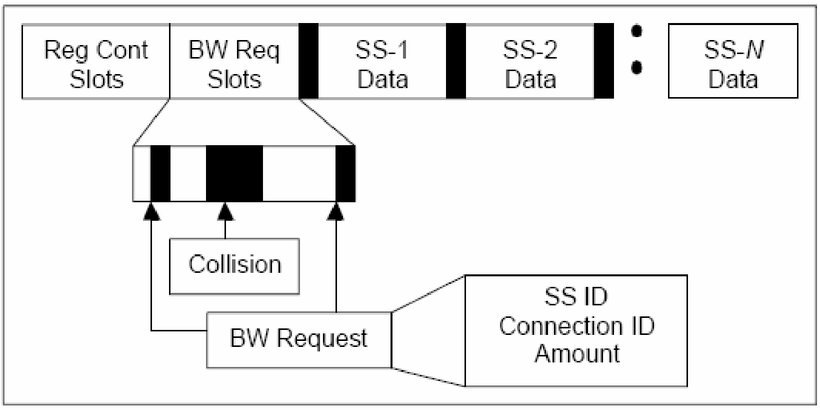

Figure 2.7 Multicast or Broadcast Polling BW Request

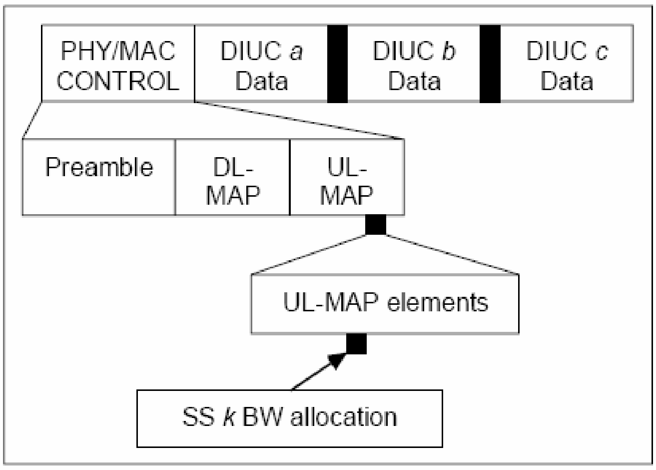

Figure 2.8 Multicast or Broadcast Polling BW allocation for data

Then, we briefly introduce how the contention resolution algorithm be implemented. The BS has more flexibility in controlling the contention resolution. The mandatory method of contention resolution that should be supported is based on a truncated binary exponential backoff, with the initial backoff window and the maximum backoff window controlled by the BS. The values are specified as part of the uplink channel descriptor (UCD) message which represents a power-of-two value. For example, a value of 4 indicates a window between 0 and 15; a value of 10 indicates a window between 0 and 1023.

2.2 MAC support of PHY

In this section, we introduce two major duplexing techniques and frame structure. One of the duplexing techniques is Frequency Division Duplex (FDD), the other one is Time Division Duplex (TDD).

2.2.1 Frequency Division Duplex (FDD)

FDD is a duplex scheme in which uplink and downlink transmissions use different frequencies but are typically simultaneous. In an FDD system, the uplink and downlink channels are separated in different channels, so they can transmit two-way data at the same time.

Both uplink and downlink transmissions could be used in a fixed duration. It also allows simultaneous use of full-duplex SSs which can transmit and receive simultaneously and half-duplex SSs which cannot. If half-duplex scheme is used, an SS can not transmit and receive data at the same time. Otherwise, both of them could operate simultaneously. Figure 2.8 is an example which describes the process of an FDD mode.

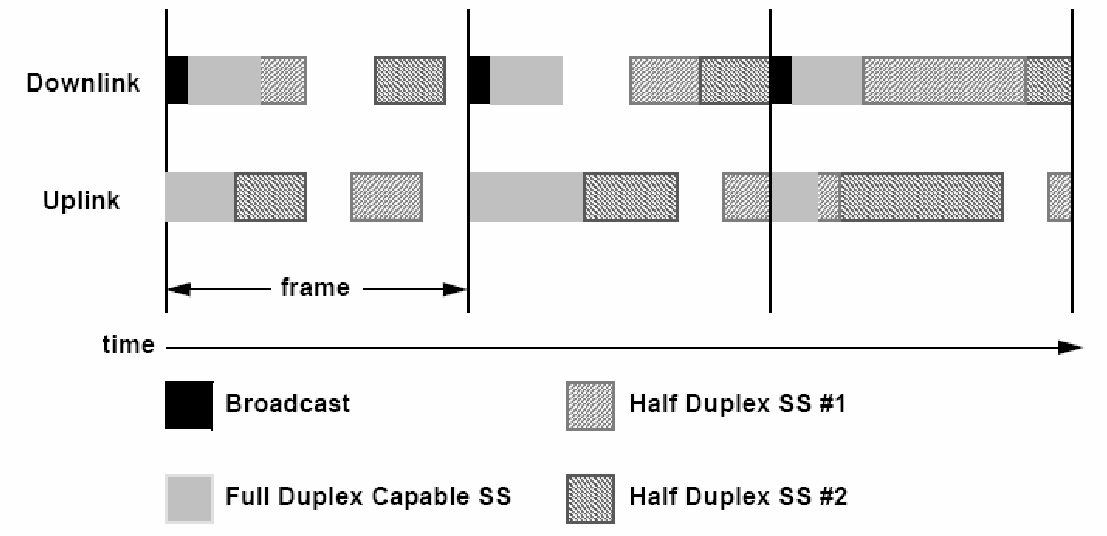

Figure 2.9 Example of FDD bandwidth allocation

2.2.1 Time Division Duplex (TDD)

In the TDD system, the downlink and uplink transmissions operate at different times but could share the same frequency. A TDD frame has a fixed duration and contains one downlink and uplink subframe. The frame is divided into an integer number of slots which make the bandwidth allocation easier. The splitting of downlink and uplink can be controlled by higher layers using a system parameter.

The 802.16e PHY supports TDD, full-duplex FDD, and half-duplex FDD operation, however the initial release of Mobile WiMAX certification profiles will only include TDD. With recently releases, FDD profiles will be considered to address specific market. TDD is the preferred duplexing mode for the following reasons:

- TDD allows adjustment of the downlink and uplink ratio to efficiently support asymmetric two-way traffic. With FDD, downlink and uplink always have fixed and similar bandwidth.
- Unlike FDD, which requires a pair of channels, TDD only requires a single channel for both downlink and uplink. It also can provide greater flexibility for adaptation to varied global spectrum allocations.
- The TDD system designed by factories are less complex and therefore less expensive.

Figure 2.9 illustrates the simple frame structure for Time Division Duplex (TDD).

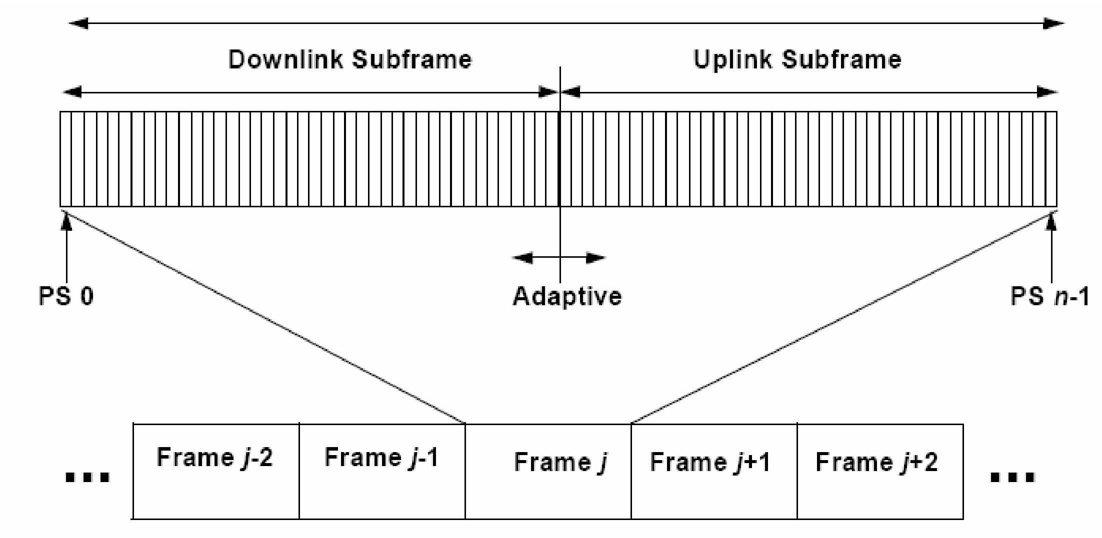

Figure 2.10 TDD frame structure.

Each TDD frame is divided into downlink and uplink subframe separated by Transmit /

Receive transition Gaps (TRG) or Receive / Transmit transition Gaps (RTG) to prevent downlink and uplink transmission collisions.

The following control information is used to control the system operation in a frame:

- Preamble: The preamble is used for synchronization and is also the first OFDM symbol of a frame.
- Frame Control Head (FCH): The FCH follows the preamble and is used for providing the frame information such as MAP message length, two-way channel descriptor (UCD and DCD), and usable sub-channels.
- DL-MAP and UL-MAP: The two MAPs provide some control information including DL bursts, UL bursts, and ranging period. The burst configurations, composed of slots, are always built in the form of rectangles.
- Ranging and Bandwidth Request subchannel: Ranging and Bandwidth Request are always combined to form a signal channel which can simplify the design complexity. Via the Initiate Ranging IE, the BS specifies an interval in which new stations may join the network. Ranging period is an interval, equivalent to the maximum round trip propagation delay plus the transmission time of the RNG-REQ message, shall be provided in some UL-MAPs to allow new stations to perform initial ranging. Packets transmitted in this interval shall use the RNG-REQ MAC Management message. And the transmission process of Bandwidth Request is similar to Ranging Request.

Figure 2.10 illustrates an example of OFDMA frame structure. A slot in the OFDMA PHY requires both a time and subchannel dimension and is the minimum possible data allocation unit. The definition of an OFDMA slot depends on the OFDMA structure, for example, if uplink and downlink using the adjacent subcarrier permutation, one slot is one subchannel by one OFDMA symbol.

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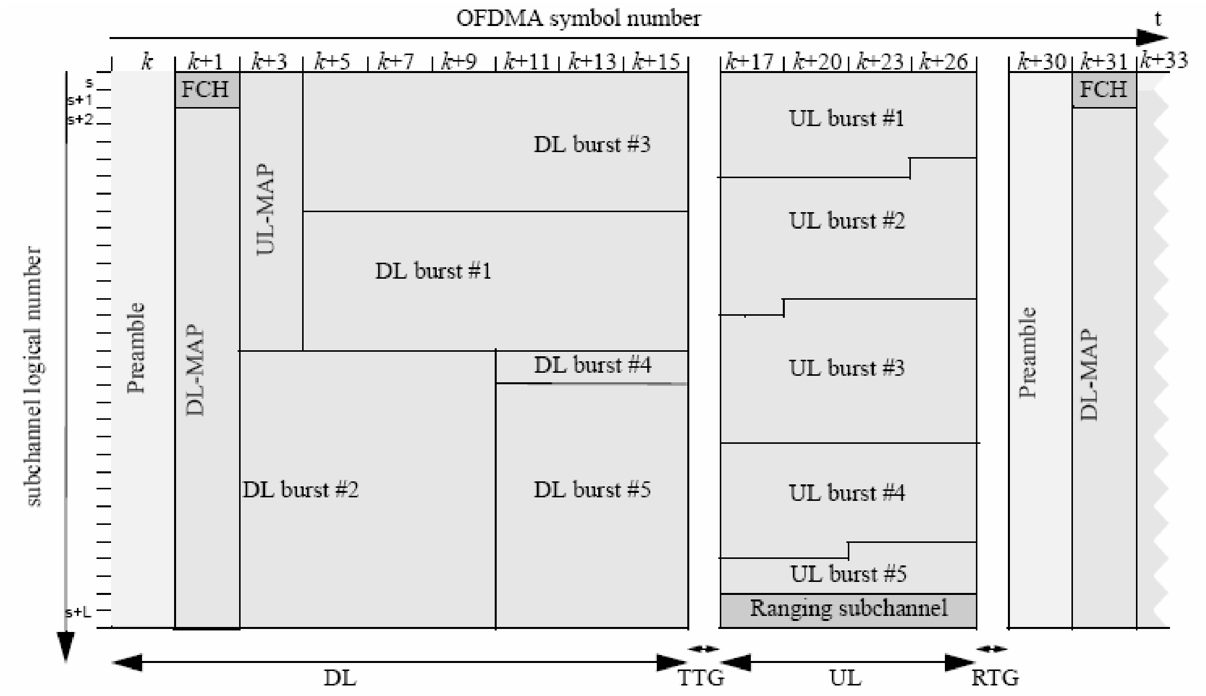


Figure 2.11 OFDMA frame structure

2.3 Related Works

In our survey, most of those scheduling algorithm were focusing on QoS issues especially for UGS, rtPS, and ertPS. In [4], the author proposed an simple and efficient algorithm which calculates each connection allocated slots and controls order of slots to decrease the jitter. In [5], the proposed scheme adaptively allocates bandwidth in order to control queue length at a target level so that the requirements for delay and PDU dropping probability can be met. It also presented an analytical queueing framework to analyze the proposed scheme. On [6], authors suggest a system model that could consider the voice state transitions. They use voice activity detector and silence detector to check whether the state is on or off, then rely on system to calculate the steady-state probability.

Several adaptive schemes were proposed to dynamically allocate resource allocations. The scheduling problem is to allocate time slots on a subset of subcarriers to confirm clients demands and maximize system throughput. In [7], it presented linear programming relaxations for resource allocation problem and provide optimal allocations for all users. The method would be based formulations to achieve maximum throughput.

IEEE 802.16e provides ertPS, a new scheduling algorithm for supporting silence suppression, it has some advantages that not only reduce MAC overhead and access delay of the rtPS algorithm, but also prevent the uplink resources waste of the UGS algorithm. On [8], the authors make some analyses to verify the ertPS performance. Based on the simulation results, the ertPS algorithm indeed has better performance than both UGS and rtPS algorithms.

Although lots of related scheduling algorithm has been proposed, different environment should have different requirements. How to achieve the distinct purposes in the whole systems is still an issue that we should investigate.

Chapter 3 Proposed Approaches

Firstly, we explain the problem that could be found in WiMAX and address our motivation. In 802.16e, the ertPS was proposed to support real-time service flows that generate variable size data packets on a periodic basis, such as Voice over IP services with silence suppression. VoIP uses conventional IP network to provide voice communication service between end users. It is much cost effective and easier to construct IP networks than traditional telecommunication networks, and it is also convenient for the users of voice service, so VoIP market grows up quickly.

Therefore, lots of users try to use handset devices in place of traditional telephones to make VoIP calls through networks. This means that providing QoS is necessary for VoIP connections, especially in WLAN. But in WiMAX system, when bandwidth isn't enough for unicast polling, ertPS must use multicast polling or broadcast polling to participate contention with nrtPS or BE. This usually causes collisions during contention period and the bandwidth request delay could not be guaranteed, and the packets could not be transmitted immediately. Except the delay, SSs could not get suitable bandwidth (larger or less) and may cause the bandwidth wasting.

Based on the above observation, we proposed a scheme for providing QoS in WiMAX environment. The scheme adjusts their transmitting sequence to allow ertPS connections to have better priority. We will use some mathematic equations to calculate the delay.

3.1 Main Scheme

The bandwidth request must be implemented in uplink subframe, so we will focus our researches on the uplink period. In WiMAX standard, the multicast polling and broadcast polling period will not be restricted in specific channel, but based on the frame burst allocations, they are always built in the form of rectangles, as illustrated in Figure 3.1. So the multicast polling and broadcast polling should be defined in one or multiple specific channels to reduce the difficulty of UL-MAP definition. Some factories combine the ranging and bandwidth request to a specific channel called signal channel. Our simulation will use one channel for a contention period of multicast polling or broadcast polling.

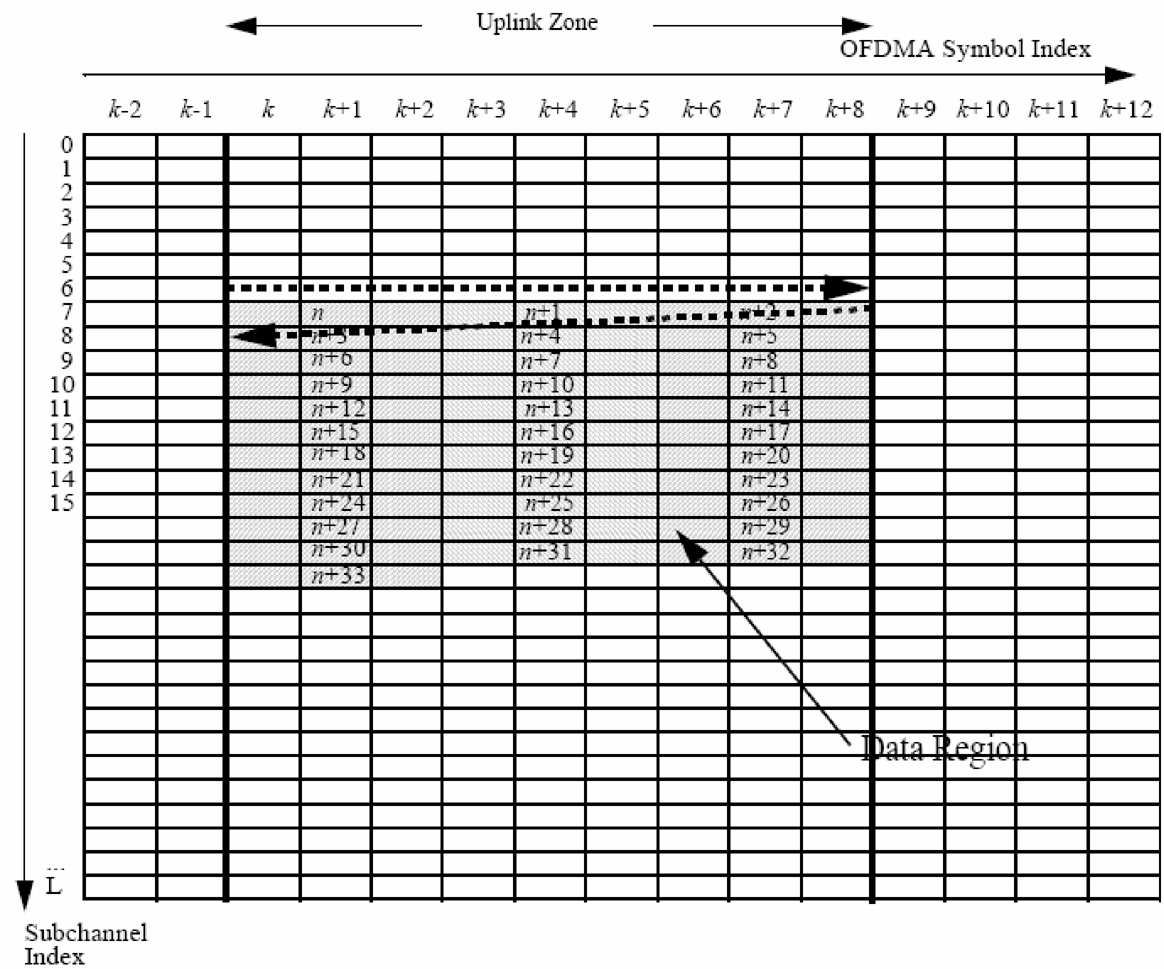


Figure 3.1 mapping OFDMA slots to subchannels and symbols in the uplink.

For uplink, there are two kinds of permutation to choose from. One is adjacent subcarrier permutation, in which each slot is one subchannel by one OFDMA symbol; the other one is distributed subcarrier permutations, in which each slot is one subchannel by three OFDMA symbol. We adopt the adjacent subcarrier permutation to implement our scheme because it is easier to implement (The permutation in Figures 3.1 is distributed subcarrier permutation).

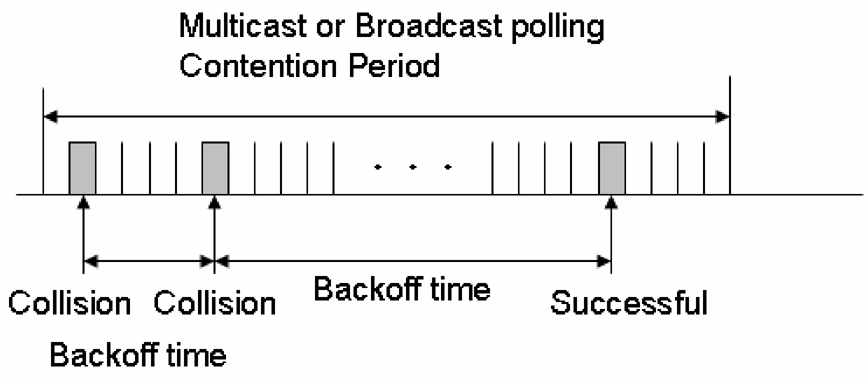


Figure 3.2 Slots allocation of original scheme

In Figure 3.2, we provide a diagram to illustrate the original multicast or broadcast polling situation. Each slot is a transmission opportunity, service flows have to contend the slots if more than one flows want to transmit request at the same slot. If collision occurs, backoff will be performed to request again.

Our goal is to reduce the collision of ertPS flows. The key idea of our main scheme is to sacrifice the request contention time of nrtPS and BE connections, because these two service flows does not have immediate QoS demand. If there are VoIP packets to be sent by ertPS flows, we assign the first several slots to ertPS connections and the slots could not be occupied by nrtPS or BE flows.

First the BS must define one period called contention-free (CF) period to ertPS bandwidth requests so as to avoid contention possibility and increase the success probability. The CF period does not occupy other bandwidth but divide the original polling contention period into two periods. The heading slots are integrated into CF period which could be allocated to ertPS flows only, and the remainding slots keep the original functionality. Both two periods could be defined in UL-MAP which combines the information element (IE) of unicast polling and multicast polling (or broadcast polling). In Figure 3.3, the diagram illustrates the framework and slots allocation.

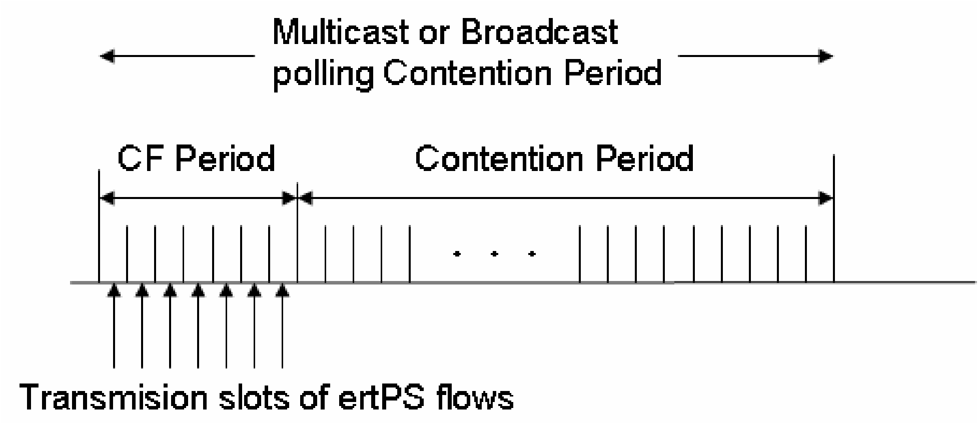


Figure 3.3 Slots allocation of proposed scheme

In our scheme, the ertPS flows are guaranteed to have two kinds of transmission opportunities. If an ertPS flow was allocated a CF period slot, it can transmit the bandwidth request packet immediately; otherwise, it can participate the contention period with nrtPS flows and BE flows.

3.2 Contention-free period definition

In order to calculate how many slots in contention period could be allocated to CF period, first we must figure out the ratio of ertPS flows to all service flows. For example, if there are 20 slots in multicast or broadcast polling contention period, 5 ertPS flows and 20 nrtPS or BE flows, the ratio is 1 to 4. According to the ratio to allocate the CF slots, the number is 4 (20*5/(5+20)=4).

Each CF slot could only be used by the assigned service flow. For example, if the first slot was allocated to 5-th ertPS flow, the first slot can only be used by 5-th flow, even if there is no request to send. This is the disadvantage of our scheme that has possibility to waste bandwidth, because the CF slots are allocated in advance. Beside wasting bandwidth, shorten nrtPS and BE contention period would cause the increasing of their request delay. As illustrated in Figure 3.3, the situation is clear at a glance. Although our scheme brings these two disadvantages, it is still worthy to improve the performance of ertPS flows.

Since the transmission range of WiMAX is much larger than that in WiFi, hundreds of users may utilize the same WiMAX base station at the same time. In general, it is not sufficient for allocating the CF slots to service flows one by one. The BS is not aware when the SSs have bandwidth request to send, so we allocate each CF slot to ertPS flow with random selection for fairness. Although we divide some slots to let ertPS flows transmit bandwidth request first, the number of CF slots still has its limitation in providing essential bandwidth for ertPS and BE flows. So we set threshold which is half of the contention period. The number of CF slots can be obtained as follows:

$$CF = \begin{cases} C \times (\frac{N\_ert}{N}) & if\,(C \times (\frac{N\_ert}{N}) < \frac{N}{2}) \\ \frac{N}{2} & if\,(C \times (\frac{N\_ert}{N}) >= \frac{N}{2}) \end{cases} \tag{1}$$

Where, CF is the number of allocated slots.

C is the total slots in contention period.

N is the number of whole service flows.

N_ert denotes the number of ertPS flows.

According to Equation (1), if C multiplied by the ratio of N_ert to N is no larger than half of N, we do the allocation directly; otherwise, it could only get half of N.

Figure 3.4 shows a slots allocation sample of our scheme. We assume that the randomly selected number is four, so the CF slots are allocated to the ertPS flows whose serial numbers are from 4 to 8. The slots of 4 and 7 indicate that the stations are transmitting bandwidth requests; other slots (5, 6, and 8) are for nothing because the station may not be ready for transmission or have no buffered packets to be handled. Other ertPS flows which are not allocated CF slots could only participate contention period with nrtPS and BE flows.

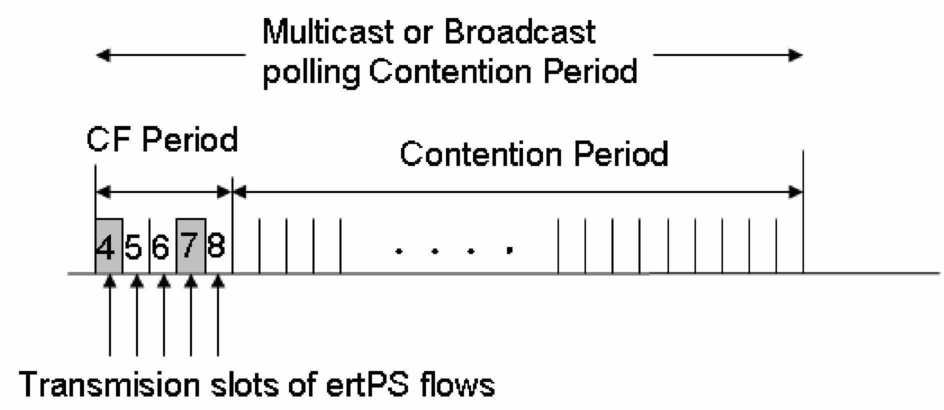


Figure 3.4 Slots allocation example of proposed scheme

3.3 Policy decision procedure

As illustrated in Figure 3.5, the policy decision of SSs would obey the procedure. One SS may have more than one service flow and each of them is uncertain. So during the SS decision procedure, SS will check whether sending bandwidth request is necessary or not, then determine its service flow type. If the type is ertPS flow and has been allocated CF slot, it could transmit bandwidth request immediately. Otherwise, they participate in contention, no matter what kind of service flow it is. When collision occurs, the service flow must wait for backoff timer to finish, then retry the transmission procedure.

When many service flows were ready to transmit bandwidth request before transmitting a frame, they always try to occupy the first slot in contention period, this may cause higher collision probability.

Since each frame or each uplink subframe could have only one signal channel which we introduced in Chapter 2, the procedure would operate only once in each uplink subframe.

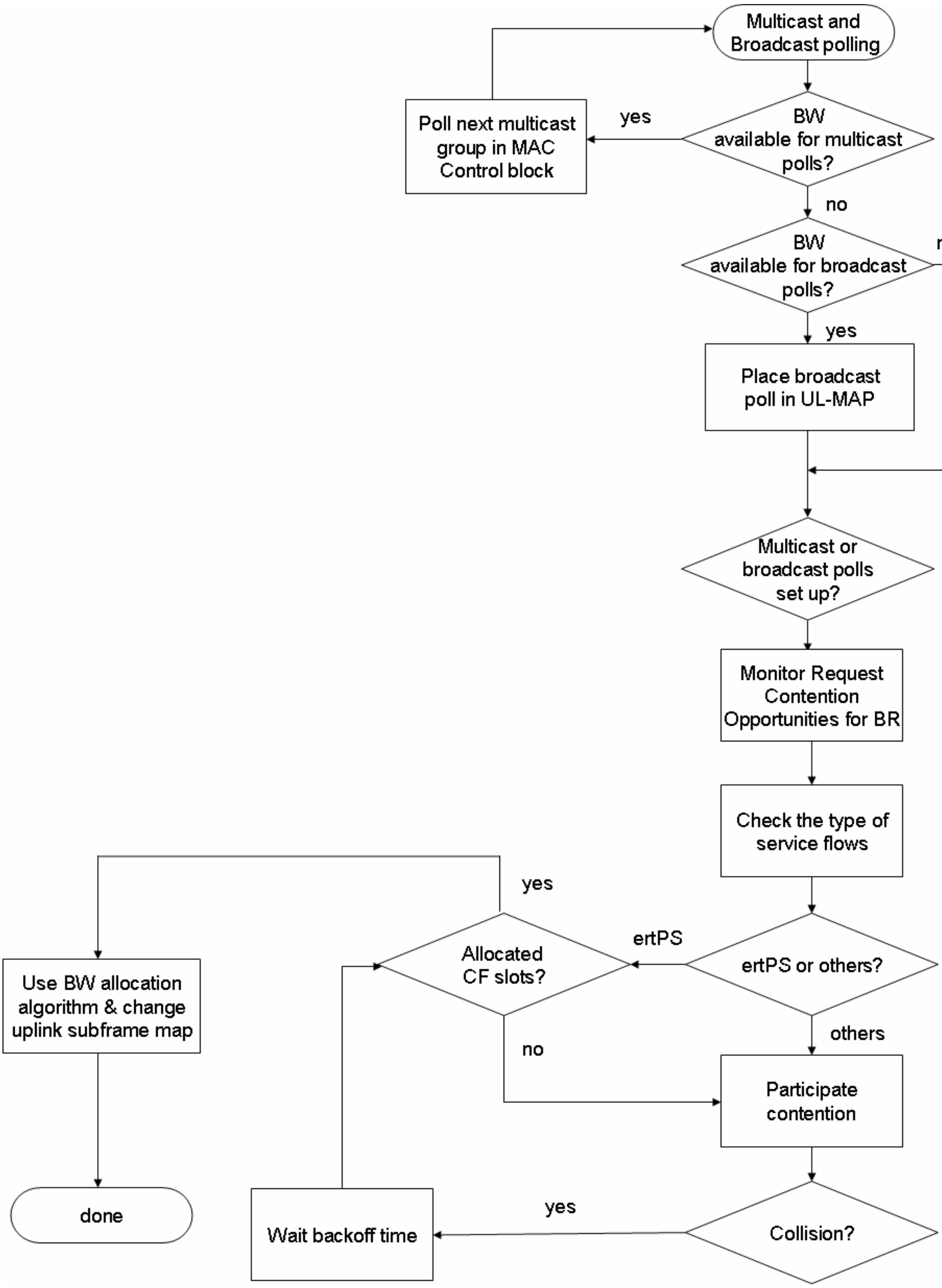


Figure 3.5 SS decision flow chart.

After introducing the SSs procedure, we discuss the BS policy decision. In our scheme, multicast and broadcast polling will be implemented in signal channel, so the first BS must define which channel to be used. Then BS checks the situation of each SS, because an initial service flows or a leaving one will cause the allocated CF ratio to redistribute. Finally, BS relies on the information which was collected in prior steps, and changes uplink subframe map, as illustrated in Figure 3.6.

Like SS decision flow charts, BS decision flow charts will also operate once in each uplink subframe.

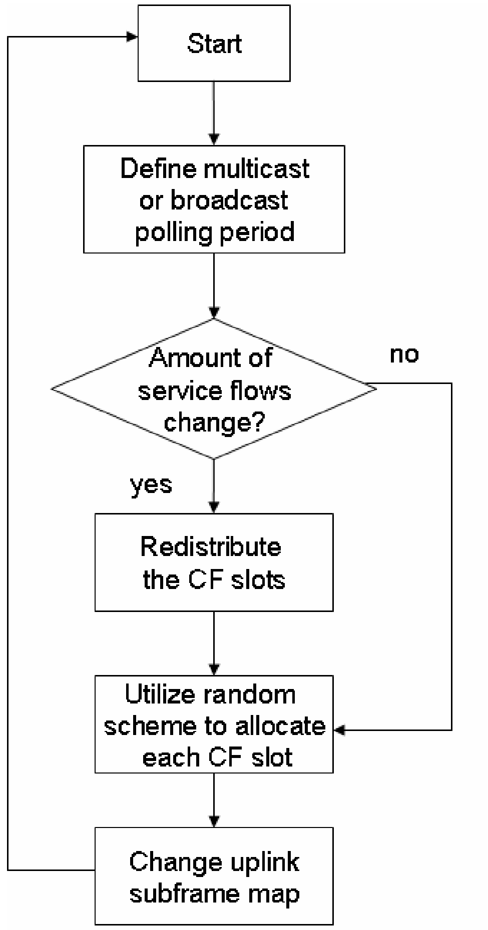


Figure 3.6 BS decision flow chart.

3.4 Mathematical analysis

The simulation results and performance evaluation are presented in Chapter 4. Before it, we introduce some mathematical analysis to compare the performance with the original scheme in the standard. We define that the delay is the time spent until a bandwidth request message is successfully transmitted, and this information is the most important observations in our analysis. In order to obtain the delay information, we need to find the count of retransmissions first, then add the backoff time to get our theoretical delay data.

3.4.1 Numerical analysis of the original scheme

The probability that one bandwidth request message is located in a specific slot is 1/C, where C is the total number of slots in contention period (according to Equation (1)). This is based on the assumption that the bandwidth request is uniformly distributed during a contention period.

We analyze the statistics from the point of view of a specific service flow. The probability ($P_{not\_the\_same}$) that no other bandwidth request messages are transmitted in the same slot is given by

$$P_{not\_the\_same} = \left(1 - \frac{1}{C}\right)^{N-1} \tag{2}$$

where, N is the number of total service flows.

According to Equation (2), we could derive the probability (P_{suc_slot}) that successfully transmit a bandwidth request message in a slot.

$$P_{suc\_slot} = \left(1 - \frac{1}{C}\right)^{N-1} \times \frac{1}{C} \tag{3}$$

Since the contention period size is C slots, the probability (P_{suc}) of successfully transmitting a bandwidth request message during a contention period can be derived as follows:

$$P_{suc} = \frac{1}{C}\left(1-\frac{1}{C}\right)^{N-1} \times C = \left(1-\frac{1}{C}\right)^{N-1} \tag{4}$$

Since we need to find out the retransmission time, utilizing collision probability is necessary for calculation. We derive the probability (P_{col}) of unsuccessfully transmitting a bandwidth request:

$$P_{col} = 1 - P_{suc} = 1 - \left(1-\frac{1}{C}\right)^{N-1} \tag{5}$$

In our scheme, we have a specific channel called signal channel to implement contention period. A frame could only have one signal channel, so the collision times in a signal channel is the same as the collision times in a frame. Using geometric distribution, we can derive the probability ($P_{suc}(k)$) for transmitting a successful bandwidth request in the k-th frame:

$$P_{suc}(k) = P_{col}^{k-1}(1-P_{col}) \tag{6}$$

In Equation (6), it means that ransmissions during the first (k-1) frames collided with other service flows and won't be successful until the k-th frame.

We use an expected value to represent the average number of retransmission for a bandwidth request message:

$$E = \sum_{k=1}^{\infty} kP_{suc}(k) = \frac{1}{\left(1-\frac{1}{C}\right)^{N-1}} \tag{7}$$

where, E is the expected value.

Because we do not know when the bandwidth request will be transmitted successfully, we set the number of frames from one to infinity and count it accumulately. Finally, we convert the expected value to an integer. For example, if E equals to 4.2, it means that the successful transmission is in 5-th frame. Beside the expected collision count, we also need to consider backoff delay. So we derive backoff slots first:

$$Backoff\_slot = P_{suc\_slot} + \frac{\sum_{n=1}^{10} P_{col\_slot}{}^{n} \times P_{suc\_slot} \times 2^{n}}{2} \times \min\_backoff\_window \qquad (8)$$

where, Backoff_slot means the number of total backoff slots.

n is the power-of-two value.

P_{suc_slot} is from Equation (3) and P_{col}=1-P_{sul_slot}.

min_backoff_window is the minimal backoff window size

The mandatory method of contention resolution that should be supported is based on a truncated binary exponential backoff, and the initial backoff window and the maximum backoff window controlled by BS. The values are specified as part of the UCD message and represent a power-of-two number. We set the initial backoff window from 0 to 15 and the maximum backoff window from 0 to 1023. So in Equation (8), n is the power-of-two number that is no larger than 10. Finally, we divide it by 2 to get the average number of backoff slots. In Equation (8), it accumulates the backoff slots whose n is from 0 to 10. When n is larger than 10, relative to less than or equal to 10, the values of P_{suc_slot} and P_{col_slot} are very small. So we can ignore it to make the equation simpler.

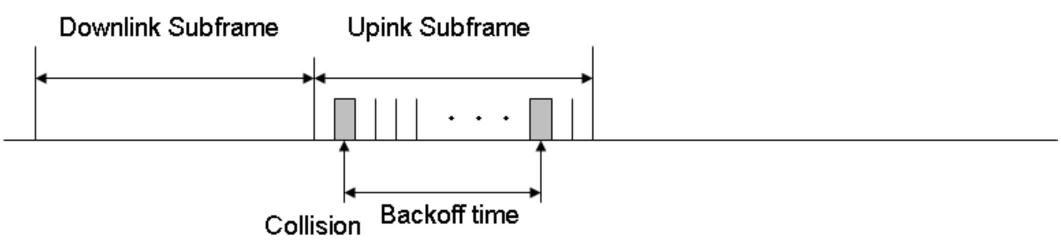
We can utilize the Backoff_slot to calculate Backoff_delay:

$$Backoff\_delay = \frac{Backoff\_slot}{slots\_perframe} * frame\_size \qquad (9)$$

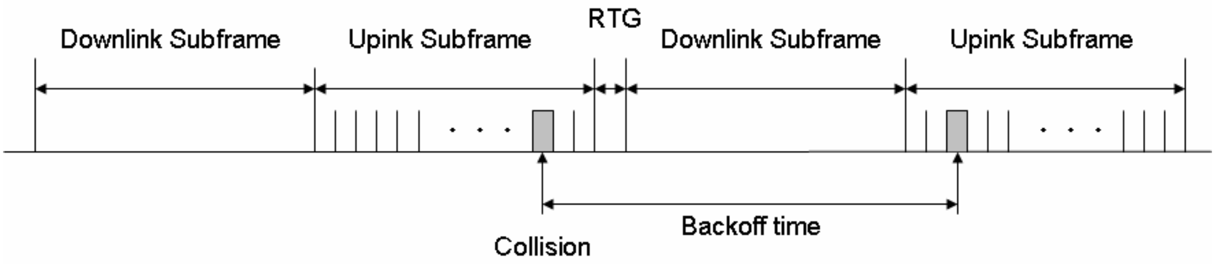
where, we exchange Backoff_slot for Backoff_delay; in other words, they are different units.

So the unit obtained from Backoff_slot divided by slots_perframe is frame, then we multiply it with frame_size (5ms in general) to get Backoff_delay with mini-second (ms) as the unit.

This exchange will cause some inaccuracy, because we do not know when the backoff starts. Figure 3.7 illustrates the different backoff situations:



(a)



(b)

Figure 3.7 Different backoff situations

In Figure 3.7 (a), if the collision happened in heading slots of a frame, it shows that the backoff transmission might be still in the same frame. But in (b), the collision happened in tailing slots of a frame, it shows that the backoff transmission might wait till the next uplink subframe. We are unable to predict which one will happen, and it must experience larger backoff delay in Figure 3.7 (b). Although Equation (9) features some inaccuracy, the value is not too far from what we think and is acceptable.

Finally, we combine the expected value and backoff delay to get real collision delay:

$$Delay = Frame\_size \times E + Backoff\_delay = \frac{F}{\left(1 - \frac{1}{C}\right)^{N-1}} + Backoff\_delay \qquad (10)$$

where, F means the frame size; in general, the value is 5ms.

3.4.2 Numerical analysis of proposed scheme

Because the contention situations in ertPS flows are different from others, we introduce them in two subsections.

3.4.2.1 Analysis of nrtPS and BE flows

In this section, we introduce the mathematical analysis for nrtPS and BE flows. The basic idea is the same, because contention period in our proposed scheme reduces some slots.

The probability of successfully transmitting a bandwidth request message during a frame (or a contention period) is derived as follows:

$$P_{suc} = \frac{\sum_{i=1}^{CF} \left(1 - \frac{1}{C-i}\right)^{N-i}}{CF} \tag{11}$$

where, C is the number of total slots,

CF is the number of slots in contention free period,

N is the number of total service flows (all service flows can participate contention including ertPS flows),

i means a variable number of service flows that do not participate the contention.

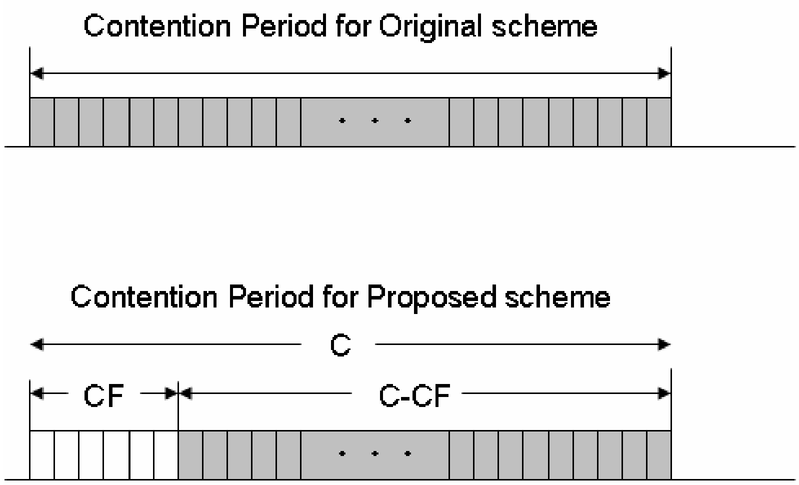


Figure 3.8 Comparisons with two schemes.

In Figure 3.8, the shaded slots represent real contention periods. It clearly shows us that our proposed scheme has a smaller successful transmission probability comparing to the original scheme (Equation (4)). In Equation (11), we do not know how many ertPS flows that already successfully transmit its bandwidth request, so we utilize a variable value i to control the situation. We accumulate each probability and divide it by CF to get an average value.

$$P_{col} = 1 - P_{suc} = 1 - \frac{\sum_{i=1}^{CF}\left(1 - \frac{1}{C-i}\right)^{N-i}}{CF} \tag{12}$$

$$P_{suc}(k) = P_{col}^{k-1}\left(1 - P_{col}\right) \tag{13}$$

Like that in Section 3.4.1, Equation (12) represents the probability of unsuccessful transmission of a bandwidth request. Equation (13) means that bandwidth request message would be transmitted successfully in k-th frame, so the order of P_{col} must be k-1.

$$E = \sum_{k=1}^{\infty} kP_{suc}(k) \tag{14}$$

$$Delay = frame\_size \times E + Backoff\_delay = F \times E + Backoff\_delay \tag{15}$$

The process of deriving Backoff_delay is same as that in Equation (8) and Equation (9), so we do not give details here again. Finally, we also get an expected value and utilize it to derive the collision delay.

3.4.2.2 Analysis of ertPS flows

In this kind of service flows, two transmission opportunities must be considered; first is CF transmission opportunity, and the other is contention transmission opportunity.

We list the equations first and explain the definition of parameters, then we discuss the meaning of the equations.

If (CF<N_ert)

$$P_{suc} = \frac{CF}{C} \times \frac{CF}{N\_ert} + \left(1 - \frac{CF}{C} \times \frac{CF}{N\_ert}\right) \times \frac{\sum_{i=1}^{CF} \left(1 - \frac{1}{C-i}\right)^{N-i}}{CF} \qquad (16)$$

else

$$P_{suc} = \frac{CF}{C} + \left(1 - \frac{CF}{C}\right) \times \frac{\sum_{i=1}^{CF} \left(1 - \frac{1}{C-i}\right)^{N-i}}{CF} \qquad (17)$$

Where, C is the number of total slots, the value is constant,

CF is the number of contention free period slots, the value will reset when the ratio of ertPS flows to all service flows is changed.

N is the number of total service flows, the value will be different, if there are joining or leaving new service flows.

N_ert is the number of ertPS flows.

According to Equation (16) and Equation (17), we could clearly know that we have two different situations. When CF is less than N_ert, it means that if all ertPS flows have to transmit bandwidth request messages in the same frame, the number of CF slots is not sufficient to be allocated for each ertPS flow. When one specific ertPS flow wants to transmit bandwidth request message during CF period (the probability is CF/C), the transmission does not collide with other service flows if it has been allocated one CF slot, so the probability must be multiplied by CF/N_ert. Otherwise, if the service flow is not prepared in CF period or is not allocated CF slot, it must participate contention and the probability is 1-(CF/C)*(CF/N_ert). According to Equation (11), we could derive the successful probability in contention period.

If CF is larger than or equal to N_ert, it means that when ertPS flows want to transmit a bandwidth request message in CF periods, it will be allocated one CF slot and do not occupy others'. In Equation (17), it illustrates this situation.

3.4.2.3 Performance comparison

In this section, we make some comparison of our theoretical analyses. We will list the expected value of frame, delay, and goodput. E is the expected value of delivered frame, and we could compare improvement different performance to derive the goodput value, utilizing Equation (18).

$$Improvement = \frac{Delay\_original - Delay\_proposed}{Delay\_original} \tag{18}$$

Theoretical Service flows			Original scheme	Proposed scheme		
				ertPS(QoS)		Others(non-QoS)
Total number of flows	ertPS	Others	Delay (ms)	Delay (ms)	Improvement	Delay (ms)
10	5	5	20.03	13.04	34.90%	14.12
15	5	10	30.24	15.25	49.57%	16.39
20	5	15	37.65	18.48	50.92%	20.15
25	5	20	40.17	27.09	32.56%	32.45
30	5	25	42.99	37.38	13.05%	42.17

Table 2.3 Theoretical performance comparison.

As illustrated in Table 2.3, the number of stations is from 10 to 30 and ratio of ertPS flows to others (including nrtPS and BE flows). When the number of ertPS flows is kept constant and the number of others is increasing, we could see that proposed ertPS has better delay result. But by the number of station increases, the delay between original scheme and proposed scheme will be close because our CF slots allocation is based on the ratio of ertPS flows. This improvement is still acceptable.

We analyzed the statistics and illustrate the delay variation in Figure 3.9, it shows that the proposed scheme has better delay performance for ertPS flows

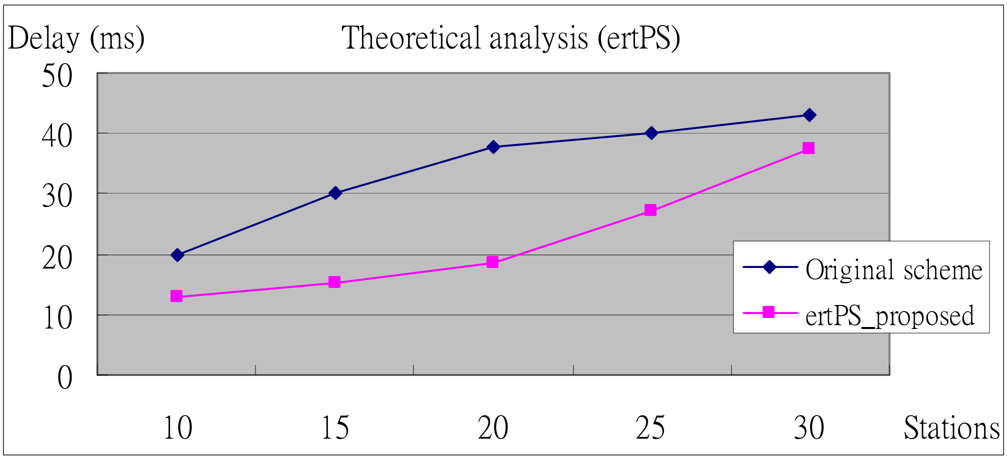


Figure 3.9 Theoretical analysis of delay comparison 1.

The other services flows would experience higher delay than the original scheme, because our method is based on reducing the transmission opportunity of QoS-free service flows, as illustrated in Figure 3.10.

The number of service flows in our theoretical analyses is from 10 to 30, and the real simulation could not achieve the assumption. It only support no larger than 30 service flows. This problem would be discussed in Chapter 4, we skip it here.

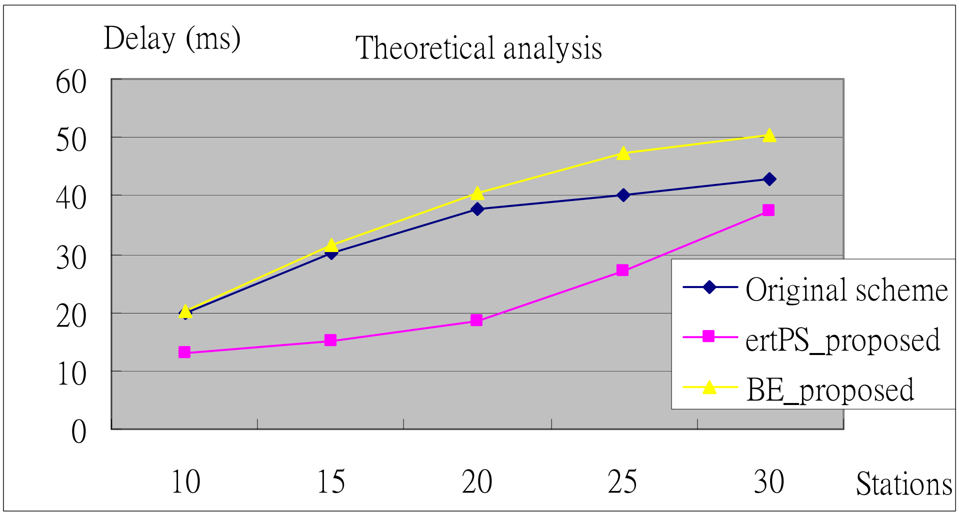


Figure 3.10 Theoretical analysis of delay comparison 2.

3.5 Summary

In this chapter, we proposed a scheme that decreases the delay of ertPS flows. Because ertPS flows are designed to support VoIP with silence suppression, it is necessary to guarantee their QoS requirement.

The delay of ertPS flows is caused by collisions in contention period. In our scheme, we divide the period into two periods; one is contention-free (CF) period, the other is shorten contention period. CF period could only be allocated to ertPS slots, and the shorten contention period could be participated by all kinds of service flows. So this scheme could achieve our goal that allows the bandwidth request messages of ertPS flows to avoid collisions. Then we proposed mathematical equations to compare the performance between the original scheme and the proposed scheme. Although our scheme may increase the delay in nrtPS and BE flows, we think that is still acceptable due to their QoS-free characteristics.

Chapter 4 Simulation and Numerical Results

4.1 Simulation Environment

In addition to demonstrating the performance of our proposed scheme, we use NS-2 (version 2.29) tool [19] with WiMAX PMP module [20] which is originally designed and developed by Networks & Distributed Systems Laboratory (NDSL) members in Chang Gung University, Taiwan.

To simplify the simulation, we neglect the sevice flows (UGS and rtPS) that do not use multicast polling or broadcast polling to participate contention, and one station corresponds to one service flow. Our network topology of the simulation is shown in Figure 4.1 and we focus our simulation in the area which can be controlled by one BS.

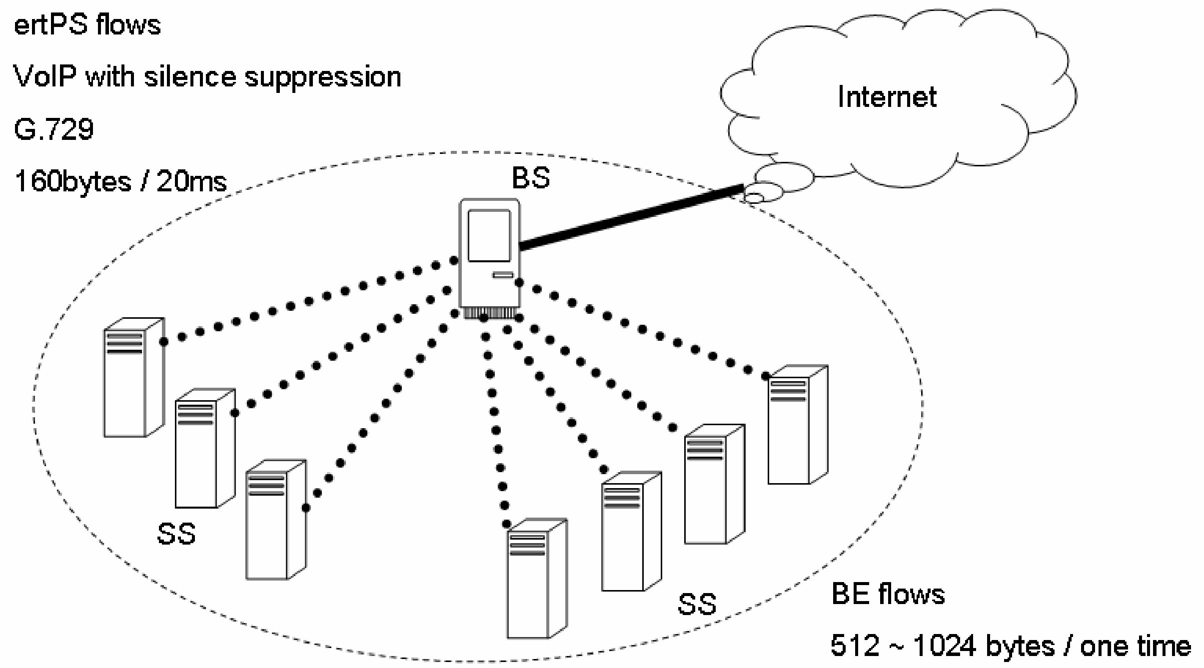


Figure 4.1 Simulation Configuration.

Each wireless station either runs bidirectional VoIP traffic with silence suppression or BE traffic which may be an application program such as E-mail or web browsing. VoIP traffic supporting silence suppression, with format of G.729 codec, 160 bytes payload and 20ms intervals are used as real-time traffic. G.729 codec could compress 64kbit/s data into 6.4kbit/s to 11.8kbit/s. Other traffics with 512 to 1024 bytes payload are used to simulate the best effort traffic (web browsing or E-mail).

4.2 Simulation Results

Before the simulations, we explain a problem which we made in simulation environments. WiMAX is a new technology in wireless transmission, so only few modules can be obtained and utilized to simulate the real environments. Our modules just support the WiMAX MAC layer and do not include physical layer, so some simulations do not fit in with actual situation, such as no sufficient bandwidth to deal with large data packets. Even though the PHY module does not satisfy our demands, the simulations still show the results of performance improvement.

4.2.1 Bandwidth request delay

As we described in Chapter 2 and Chapter 3, bandwidth request represents the requirement of SSs. Bandwidth request delay will cause data packets delay because of insufficient bandwidth for transmission. Of course, it will also lose some throughput. The simulation about throughput is introduced in Section 4.2.2.

In our bandwidth request delay simulation, the number of stations is from 10 to 30 and the number of ertPS flows is fixed in 5 to observe delay variation conveniently. We are unable to simulate larger than 30 stations due to the lack of a suitable WiMAX PHY.

STAs			Average delay (ms)	
all	ertPS	others	Original	Proposed
10	5	5	25.67	13.09
15	5	10	33.52	17.76
20	5	15	39.34	23.31
25	5	20	41.84	27.21
30	5	25	44.18	35.3

Table 4.1 Simulation result of bandwidth request delay 1.

Table 4.1 illustrates the bandwidth request delay comparison between the original scheme and the proposed scheme. The delay represents the time for which bandwidth requests of ertPS flows were buffered in SS until BS has received the messages.

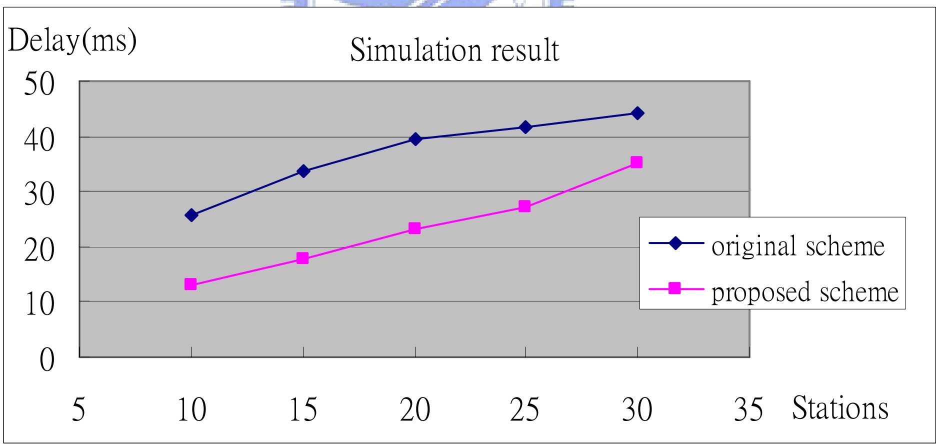


Figure 4.2 Simulation result of bandwidth request delay 1.

According to the delay data illustrated in Figure 4.2, our proposed scheme has over 10ms improvement. Delay data is close in 30 stations because the allocated CF slots are based on ratio of ertPS flows to total number of flows (5:25). The fewer the number of CF slots is, the higher the delay will be.

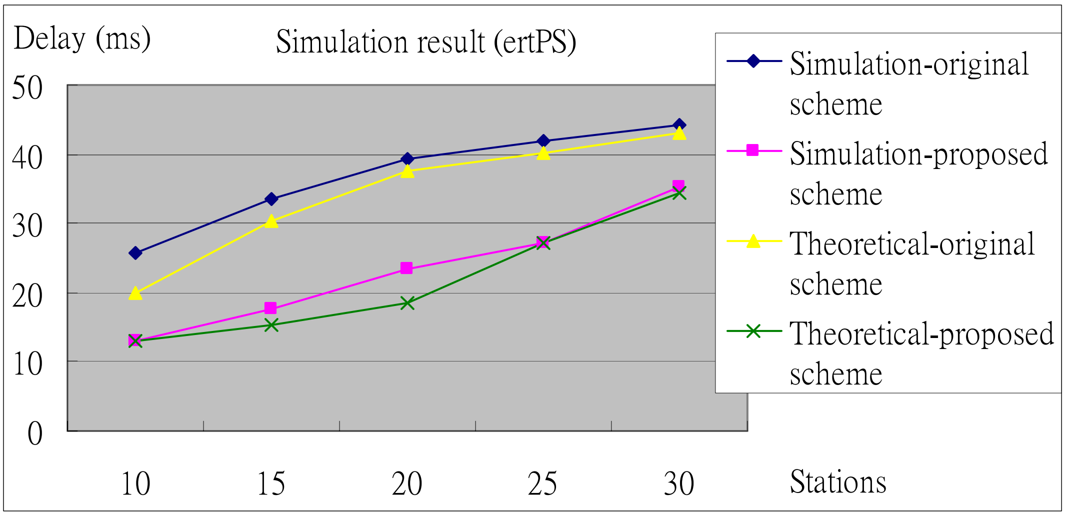


Figure 4.3 Simulation result of Bandwidth request delay 2.

The ertPS delay comparison of four situations which are simulations of original scheme, the proposed scheme, the theoretical original scheme, and the theoretical proposed scheme, respectively are illustrated in Figure 4.3. The deviations between two original schemes (theoretical and simulated) are from 2.69% to 21.97% and those between two proposed schemes (theoretical and simulated) are from 0.4% to 22.67%.

In our theoretical equations, we do not know whether collisions will happen and supposed that our backoff equations are calculated from the first slot. So simulation results will be a little less than theoretical statistics in Figure 4.3. Simulation value is larger than theoretical value when the number of stations is 20, and we think that the number of stations reaches a bottleneck because 20 is the same as the number of total contention period slots. If there were collisions, the delay will increase and backoff time is always larger than a frame (backoff window is from 0 to 31 or larger), but our theoretical scheme could not express this situation. When the number of stations is larger than 20, theoretical value and simulation value are closed because the backoff window is large enough to handle the situation.

The amount of ertPS flows in the first delay simulation is fixed, so the CF ratio will decrease by increasing the amount of total service flows. Then we try another simulation in which the ratio of ertPS flows is kept at one-third.

STAs			Average delay (ms)	
all	ertPS	others	Original	Proposed
5	1	4	21.54	17.08
10	3	7	25.67	21.61
15	5	10	33.52	17.76
20	6	14	39.34	20.02
25	8	17	41.84	15.81
30	10	20	44.18	17.94

Table 4.2 Simulation result of bandwidth request delay 2.

We could see that the ratio of ertPS flows is close to one-third and the allocated slots are almost 6 or 7. The number of stations is the same as before and is not larger than 30.

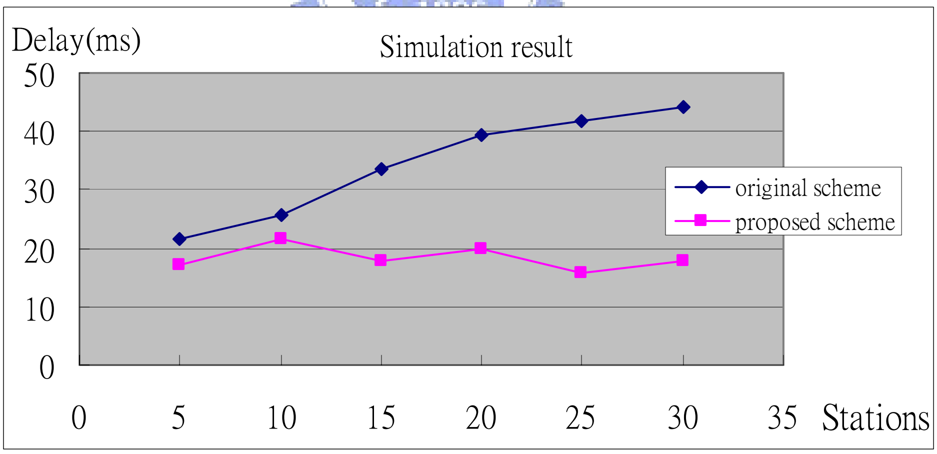


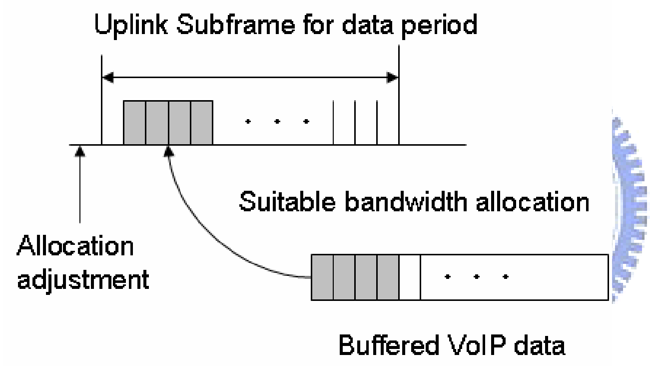
Figure 4.4 Simulation result of bandwidth request delay 3.

According to Figure 4.4, the bandwidth request delay of our proposed scheme is almost close to 20ms. This is because we keep the ratio of ertPS flows to a fixed number and the allocated CF slots could handle the demands of those service flows.

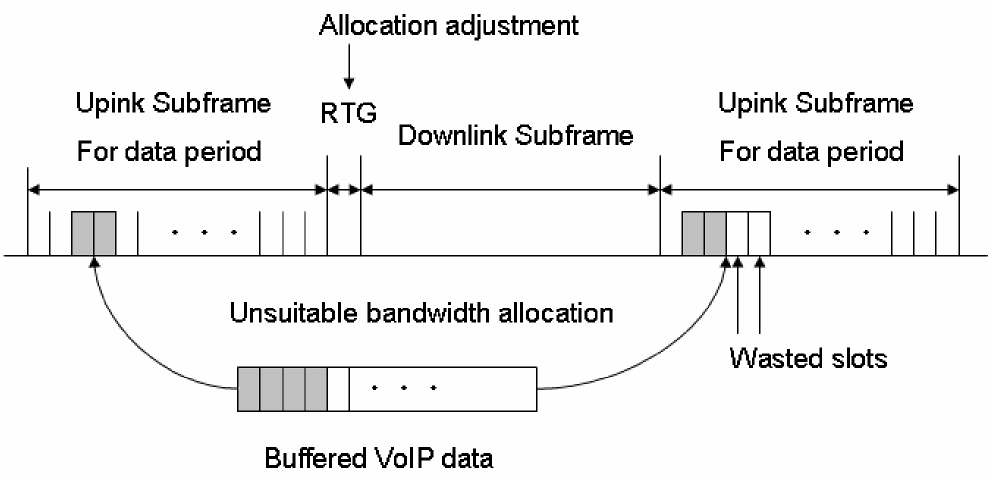
4.2.2 Throughput

VoIP traffic with format of G.729 codec, 160 bytes payload and 20ms intervals are used as real time traffic, it is compressed from 64kbit/s data into 6.4kbit/s to 11.8kbit/s. Bandwidth request delay relates to throughput, so smaller delay means that the bandwidth allocation is more sensitive. This situation affects the transmission time of VoIP data and we devoted our attention to transmit VoIP data as soon as possible, so decreasing bandwidth request delay allows VoIP service flows to get proper bandwidth faster.

The difference of data bandwidth allocation between original scheme and the proposed scheme is illustrated in Figure 4.5.



(a) Suitable bandwidth allocation.



(b) Unsuitable bandwidth allocation.

Figure 4.5 Comparison of two different bandwidth allocations.

The allocation adjustment means that BS already received bandwidth request message and changed the bandwidth allocation in UL-MAP. In Figure 4.4 (a), do allocation adjustment if buffered VoIP data has suitable bandwidth allocation, data could be transmitted immediately and does not waste slots. But in Figure 4.4 (b), heading frame does not have enough bandwidth until the allocation adjustment has been dealt with, so the next frame would be allocated more bandwidth, however, additional slots would be wasted.

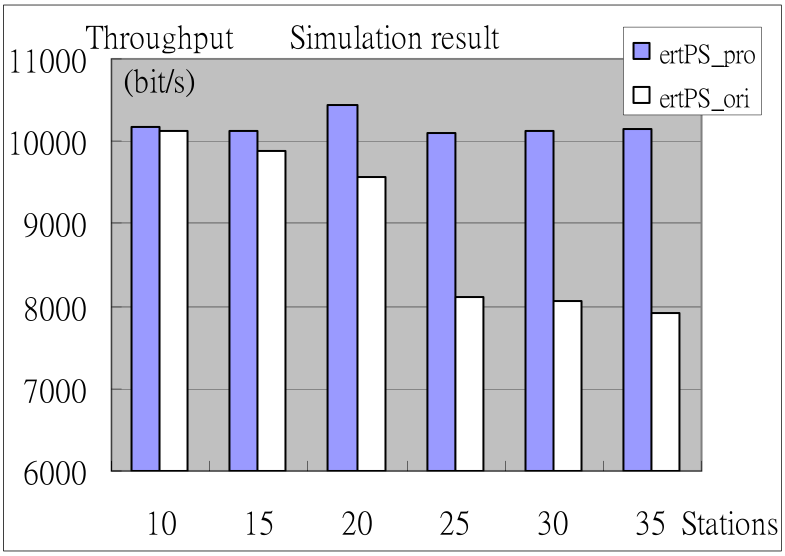
Table 4.3 illustrates the throughput in different number of stations and our ertPS stations are fixed in 5, so the number of other stations increases. To observe the values, throughput of two different ertPS scheme, which are almost close to 1k (bit/s) through compressing using the G.729 codec.

Average throughput (bit/s)				
STAs	Proposed scheme		Original scheme	
	ertPS	BE	ertPS	BE
10	10133	46424	10169	44764
15	9872	44208	10138	41092
20	9563	33160	10439	26833
25	8104	26410	10102	16330
30	8072	19477	10133	15464
35	7907	16413	10147	9919

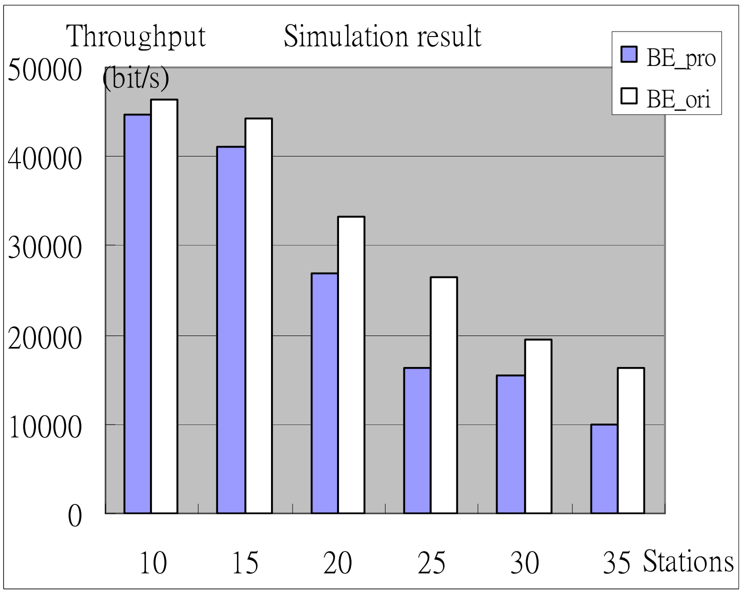
Table 4.3 Simulation result of throughput.

We could understand the throughput simulation results more clear in Figure 4.5. The two curves of ertPS looks similar, but the values are just calculated in one second. If the time goes on like this, throughput will experience bigger difference. The throughput of two BE schemes decreases as the number of stations increases, because stations shared the same bandwidth and got fewer transmission opportunities.

BE throughput of our proposed scheme is less than the original one because we neglect their priorities of bandwidth requests. It is still acceptable by its non-QoS characteristic.



(a)



(b)

Figure 4.6 ErtPS and BE throughputs in two different schemes.

Throughputs of two ertPS schemes in Figure 4.6(a) are averaged values in one second, and the other is the comparison of two BE schemes. Now we observe the throughput during first 10 seconds when the number of station is 25 and ertPS stations is 5. As illustrated in Figure 4.7 and 4.8, ertPS throughputs in proposed scheme are a little larger than the original one and BE throughputs in our proposed scheme are always less than the original one.

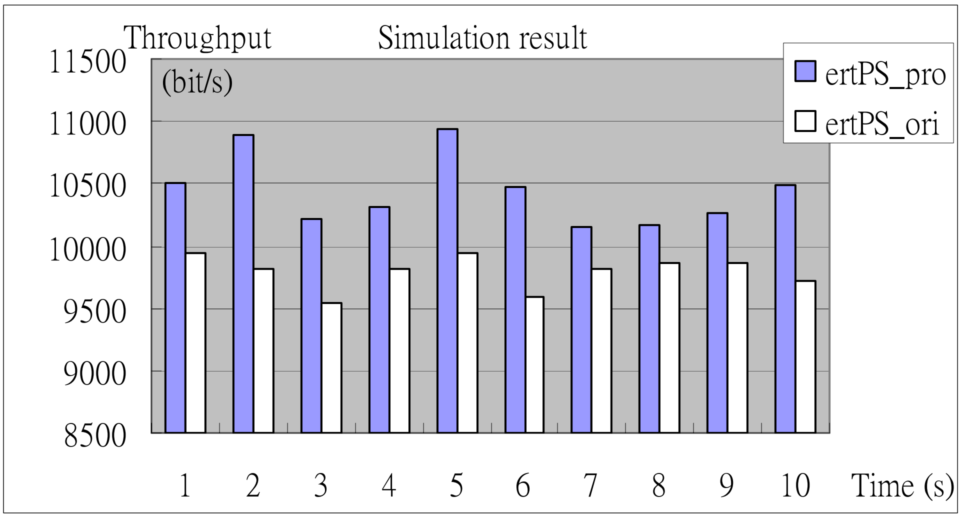


Figure 4.7 ErtPS throughputs in two different schemes (25 stations).

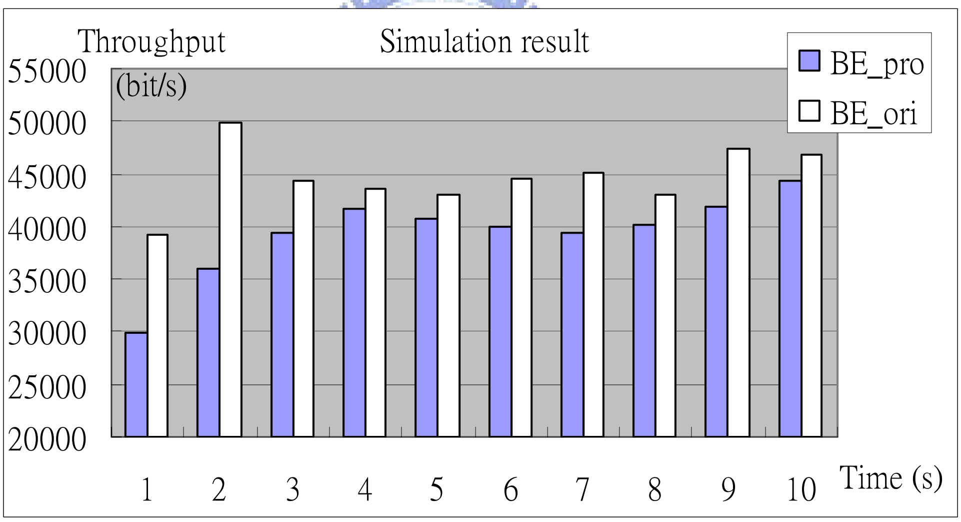


Figure 4.8 BE throughputs in two different schemes (25 stations).

Chapter 5 Conclusion and Future Works

In this thesis, first we spend some time to analyze entire scheme of bandwidth allocation and understand that before transmitting data in one SS, which must be allocated certain bandwidth. The process which acquires for bandwidth is called polling, the BS's allocation of bandwidth to SS by sending request has been defined in 802.16. If the request is successfully received by BS and it passes the admission control, BS would allocate suitable bandwidth to SS that has sent the request message; otherwise, data must be buffered and wait for BS to allocate bandwidth. So we know bandwidth request plays an important role for bandwidth allocation in IEEE 802.16.

When bandwidth is not sufficient to allocate unicast polling slots for each service flow, multicast polling and broadcast polling should be used. The two polling schemes utilize contention period to let a group of stations contend their bandwidth request transmission opportunity. UGS and rtPS have QoS issue, so they can not participate in contention. Other three type service flows (ertPS, nrtPS, and BE) must utilize contention period to get transmission slots. There are some collision probabilities when a group of stations participate in contention, so we aimed at the problem that may be caused by contention.

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 with silence suppression. When ertPS flows participate in contention period with other service flows, the VoIP packets delay may be unsettled by the failure of bandwidth request.

We proposed a scheme which let ertPS to have higher priority to transmit its request messages. Our method is to utilize the characteristic in which nrtPS and BE is QoS-free, and we divide original contention period into two different periods; first is contention-free (CF) period, the other is contention period. Contention-free period can only be occupied by ertPS flows, and each CF slot is allocated by BS. Bandwidth requests of ertPS flows will have twice opportunities to be transmitted. If one service flow is not allocated any CF slot, it also can participate the divided contention period. But in our scheme, we do not design a precise algorithm to calculate the suitable CF period, and just set the threshold which is half of the contention period. We think that a precise algorithm can be designed to utilize slots and avoid wasting of other slots. So we will improve this incomplete part in future and let our scheme feature better performance.

We use NS-2 (version 2.29) tool with WiMAX PMP module which is originally designed and developed by Networks & Distributed Systems Laboratory (NDSL) members in Chang Gung University. Although this module still has some problems to be improved, we modified the necessary functions to accommodate our task. Based on the simulation results, our scheme decreases the bandwidth request delay and increases a certain amount of throughput. Our scheme still has some defective situations and we can improve it in future, such as that we could not simulate large number of stations and the environments with wider range. Because 802.16 is a new protocol and we could not find suitable PHY layer module, our simulation just use original wireless physical layer.

In the future, we will further improve the scheme by investigating the scenario about bandwidth request. The observation will focus on the QoS degradation induced by collision in contention period. Then we will correct our method to avoid sacrificing the BE service flow transmission opportunities and achieve more improvement in throughput. Also we try to investigate the characteristic of WiMAX PHY layer and integrate it to our scheme.

In our simulation scenario, it is just a small scope that can be handled by a BS. We will try to construct an integrated network environment and simulate more accurate access delay and throughput.

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