國 立 交 通 大 學

電 信 工 程 學 系 碩 士 班

碩 士 論 文

無線區域網路資料速率估計排程演算法

Data Rate Estimation Algorithm for the Scheduler of IEEE 802.11e Wireless LANs

研究生　：謝景融

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中文摘要

隨著今日無線網路上即時性應用的增加,我們需要某些方案來提供更適合的服務，而混和式協調功能(HCF)即是為了提供網路協定(IP)品質的服務保證給IEEE 802.11e 架構下的無線區域網路而設計。比起傳統的盡力式傳輸模式,IEEE 802.11e 對於不同的應用提出了針對優先性分級的架構，以達到服務品質的區別。在本論文中,我們將提出適用於 IEEE 802.11e 混合式協調功能控制之通道存取機制(HCCA)排程器的資料速率估計演算法。

從 IEEE 802.11e 所提供的參考排程方案的評估,我們得知排程中負責的資料流若非嚴格的固定位元率(CBR)時，這個方案的效能並不是很好。因此我們需要設計一個更有彈性的方式，對於擁有不同特性應用的服務品質保證站台(QSTA)，動態地調整分配之傳送機會(TXOP)的估計。藉由提出的排程演算法，服務品質增進擷取點(QAP)能夠為這些對服務品質敏感的資料流，提供一些可以保證的服務品質參數，如延遲、封包漏失率與頻寬。提出的演算法的效能已藉由在網路模擬器第二版(NS-2)的電腦模擬來做評量,並將與 IEEE 802.11e 草案中提出的參考排程器作一個比較。

Data Rate Estimation Algorithm for the Scheduler of IEEE 802.11e Wireless LANs

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Abstract

As the real-time applications used in today's wireless network grow, we need some schemes to provide more suitable service for them. HCF was designed to provide IP quality of service guarantees in IEEE 802.11e WLANs. Compared with the traditional best-effort transmission scheme, IEEE 802.11e presents architecture to do traffic differentiation according to different QoS requirements. This thesis presents a data rate estimation algorithm for the scheduler of the IEEE 802.11e Hybrid Coordination Function (HCF) Controlled Channel Access (HCCA) mechanism.

From the evaluation of referenced scheduling scheme provided in IEEE 802.11e, we know that it does not perform well on traffic which is not strictly CBR. Therefore, we need to design a more flexible scheme to dynamically adjust the estimation of TXOP allocated to the QSTA with different characteristics of applications. With the proposed scheduling algorithm, the QAP can provide guaranteed quality of service parameters such as delay, packet loss rate, and throughput for the QoS-sensitive traffic. The performance of the algorithm is evaluated through computer simulation on network simulator 2 (ns-2) and compared with the referenced scheduler proposed in the draft of IEEE 802.11e task group.

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謹將此論文獻給所有我愛與愛我的人。

2005 年 6 月 於風城交大

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Acronym

AC	access category
ADDTS	add traffic stream
AIFS	arbitration interframe space
BSS	basic service set
CA	collision avoidance
CAP	controlled access phase (period)
CCK	complementary code keying
CDF	cumulative distribution function
CF-Poll	contention-free polling frame
CFP	contention-free period
CSMA	carrier sense multiple access
CW	contention window
DCF	distributed coordination function
DIFS	DCF interframe space
DSSS	direct sequence spread spectrum
EDCA	enhanced distributed channel access
HC	hybrid coordinator
HCCA	HCF controlled channel access
HCF	hybrid coordination function
IEEE	Institute of Electrical and Electronics Engineers
MAC	medium access control
NAV	network allocation vector
OFDM	orthogonal frequency division multiplexing
PIFS	PCF interframe space
PC	Point Coordinator
PCF	Point Coordination Function
QAP	QoS enhanced access point
QoS	quality of service
QSTA	QoS enhanced station
SI	service interval
SIFS	short interframe space
TS	traffic stream
TSPEC	traffic specification
TXOP	transmission opportunity

Chapter 1

Introduction

In recent years, wireless networks such as the IEEE 802.11 WLANs are deployed widely and rapidly in many environments around us. We can enjoy the freedom and convenience of connecting to the internet with portable computing devices on the campus, at home, or in coffee shops. Today, 802.11 WLAN (referred to as legacy 802.11 in this article) can be interpreted as a wireless version of Ethernet that supports best effort service.

However, as the demand of new application such as real-time audio/video traffic keeps increasing, the interest in wireless network that supports quality of service (QoS) has grown. There are already available mechanisms in the legacy 802.11 which are designed to support QoS, but because of their limitations they have not been implemented in real hardware. Therefore, the 802.11 working group initiated a new group "E" to define new MAC protocols in order to enhance the ability of supporting the applications that requires QoS.

The 802.11e introduces the hybrid coordination function (HCF) and defines two channel access mechanisms. The first one is a contention-based channel access referred to as enhanced distributed channel access (EDCA). The other is a controlled channel access referred to as HCF controlled channel access (HCCA). The controlled channel access is a polling-based scheme enhanced from point coordination function (PCF) of legacy 802.11. The HCCA mechanism uses a QoS-aware centralized coordinator, called hybrid coordinator (HC), and operates under rules that are different from the point coordinator (PC) of the PCF.

Since real time traffic has stricter delay constrain than non-real-time traffic, it can only wait for a very short time before it is transmitted. Therefore, it needs higher priority and enough time to access the medium. In the draft of IEEE 802.11e, the HC can negotiate with the QSTAs that have real time traffic to send using the TSPEC field in ADDTS frame. The parameters HC obtains in its scheduler are mean values of the traffic specifications. But the inter-arrival time, data rate, and packet size may be variable for some application such as video conference. Therefore, if HC always estimates the possible traffics that need to be cleared off in service period by the TSPEC parameters, it may cause the delay and loss rate of VBR traffic to increase.

In this thesis, the challenge we face is that HC wants to know how much traffic will need to be cleared at the beginning of next service interval (SI). If the HC can predict the possible amount of traffic well, it can allocate suitable and enough time to the QSTA and achieve the goal of providing QoS. To forecast the queue level at the QSTA, the scheduler needs a mechanism to do rate estimation in order to track the possibly fluctuating data rate. In the following paragraphs, we will discuss some mechanisms of related works for the scheduling and propose a new method. The performance of our scheme will be evaluated with the simulation results from network simulator 2 (NS-2).

The remainder of the thesis is organized as follows. In chapter 2, we describe the MAC mechanisms of legacy IEEE 802.11 WLANs and the enhanced mechanisms in the upcoming IEEE 802.11e specification. In chapter 3, we give a survey on related scheduling researches. In chapter 4, we will introduce our architecture of data rate estimation for the scheduler. In chapter 5, the delay of the traffic in the polling-scheme will be analyzed. Simulation results are shown in chapter 6. Finally, in chapter 7, our conclusions are presented.

Chapter2

Backgrounds

In 1999, the standard about IEEE 802.11 Wireless LANs which specified the specification of Medium Access Control (MAC) layer and Physical layer was defined. There are different versions of WLANs in the market nowadays, which apply different modulation scheme and operate in different frequency bands. The 802.11b version provides data rate up to 11 Mb/s and operates in ISM at 2.4 GHz. It applies complementary code keying (CCK) and direct sequence spread spectrum (DSSS) as transmission scheme. Another scheme, 802.11a, operates in the unlicensed 5 GHz band, and is able to achieve data rate up to 54 Mb/s, applying the multi-carrier technique orthogonal frequency-division multiplexing (OFDM) as the transmission scheme. The 802.11g version is still another scheme which applies OFDM as 802.11a, but operates in the 2.4 GHz ISM like 802.11b.

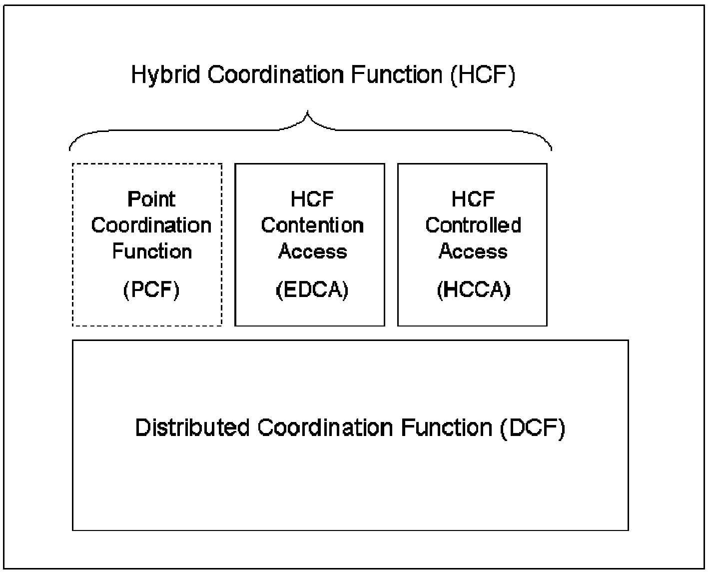

Figure 2.1: MAC architecture [2]

The MAC layer of legacy 802.11 has two MAC protocols, Distributed Coordination Function (DCF) and Point Coordination Function (PCF), while the upcoming 802.11e standard also defines two MAC function, Enhanced Distributed Channel Access (EDCA) and HCF controlled channel access (HCCA), which enhance the ability to provide more QoS qualities. Coordination function is the function to determine when a station operating within a BSS is permitted to transmit or receive MPDUs via WM.

2.1 Distributed Coordination Function [1]

DCF is the basic and fundamental medium access mechanism in IEEE 802.11 wireless LANs. The stations in infrastructure mode or Ad hoc mode should provide this operation. Using the technique of CSMA/CA, stations located closely can share the same wireless medium, and the collisions can be solved by it. IEEE 802.11 standard defines four different classes for Frame Space. The frame with different classes should wait for separately specific period before it can be transmitted. The period is described as Inter-Frame Spaces (IFS). Short IFS (SIFS) is used for immediate response, like Clear to Send (CTS) frame and Acknowledge (Ack) frame. PCF IFS (PIFS) is used when AP wants to access medium in PCF mode. DCF IFS (DIFS) is the period that station should wait before transmission in DCF mode. Finally, Extend IFS (EIFS) is used when stations retransmit frames. The length order of above IFS should be SIFS < PIFS < DIFS < EIFS. The higher priority frame like the Management type frame or Control type frame should use shorter IFS and will have more chance to access the wireless medium.

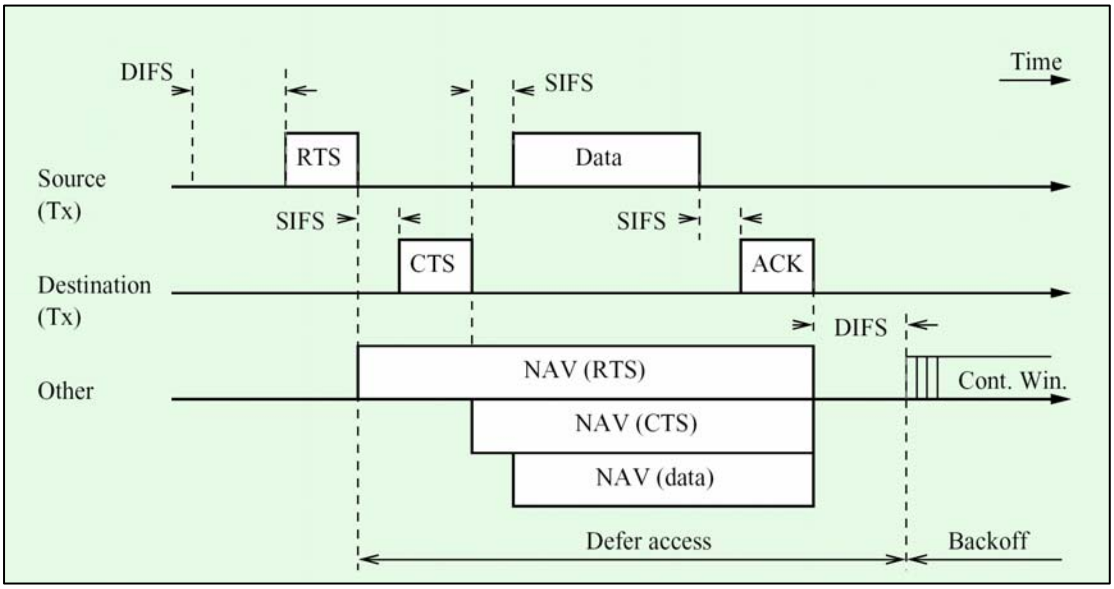

Figure 2.2: Basic operation of DCF (Including RTS/CTS mechanism) [14]

Even with these definitions, the Data type frames which should wait for DIFS will collide with each other if there are more stations that need to transmit frames at the same time. Therefore, it should add a mechanism to decrease the probability of collision. When the stations which have some packets to send sense that medium is busy and wait after the medium is idle for DIFS, they should perform a random backoff before transmitting packets. The backoff interval randomly generated by the station is uniformly distributed between zero and a maximum value called Contention Window (CW). The station should hold a backoff timer that counts down when the medium becomes idle and freeze when the medium becomes busy. By this mechanism, the probability of collisions decreases. Furthermore, after an unsuccessful transmission attempt, the CW is doubled until it reaches a specific maximum value, CWmax. To solve the hidden node problem that may occur on nodes locating in the same BSS but cannot detect the existence of each other, the standard presents a RTS/CTS mechanism.

2.2 Point Coordination Function [1]

PCF is designed to support the time-bounded data transmission such as voice or video. While in DCF the control of medium is distributed to each station, PCF is a central coordinating mode controlled by the Point Coordinator (PC) on AP. PC will initiate a contention-free period to perform polling to the stations. RTS and CTS frames are not used in PCF mode, and PC will not provide Backoff mechanism. If an infrastructure WLAN supports PCF, the periods of contention-free service alternate with the standard DCF-based service and these two periods compose a super frame. When a contention-free period starts, stations in this basic service set (BSS) should keep silent and set their NAV to the end of the contention-free period. After the PC has gained control of the wireless medium, it polls the stations on a polling list for data transmissions. During the contention-free period, the PC will transmit a Contention-free polling frame (often abbreviated as CF-Poll) to a station, which will then have an opportunity to transmit one frame. Multiple frames can be transmitted only if the PC sends multiple poll requests.

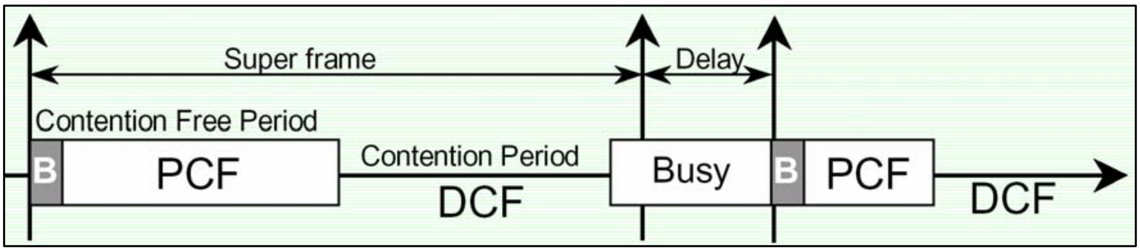

Figure 2.3: The structure of Super frame and the possible delay of TBTT [14]

In the standard of legacy 802.11, the mode of PCF is optional and it still has some limitations. PCF defines a polling-based architecture but does not provide clear methods to exchange the information of the traffic from stations. Therefore it can not design a suitable scheduling to serve the stations with real-time traffic. Besides, the transmission time for one packet of the station is out of the control of PC after the PC gives the access of medium to the station. Hence the remaining time of the contention-free period may not be enough for stations locating around the end of the polling list. The transmission of beacon may also be delayed and this unpredictable factor will influence the start of contention-free period. (This phenomenon is referred to as unpredictable delay of TBTT.) Because of the above-mentioned problems, PCF is not often implemented in the products of current wireless networks.

2.3 Enhanced Distributed Channel Access [2]

The EDCA mechanism provides differentiated, distributed access to the wireless medium for QSTAs using eight different priorities. These priorities will be mapped into four access categories (ACs) and will be treated as different backoff entities. MSDUs which are delivered by parallel backoff entities are prioritized using AC-specific contention parameters, referred to as EDCA parameter set from the beacon announced by the AP. The four ACs in the 802.11e station are labeled according to their target application, i.e., AC_VO (voice), AC_VI (video), AC_BE (best effort), and AC_BK (background), while legacy 802.11 station has only one backoff entity.

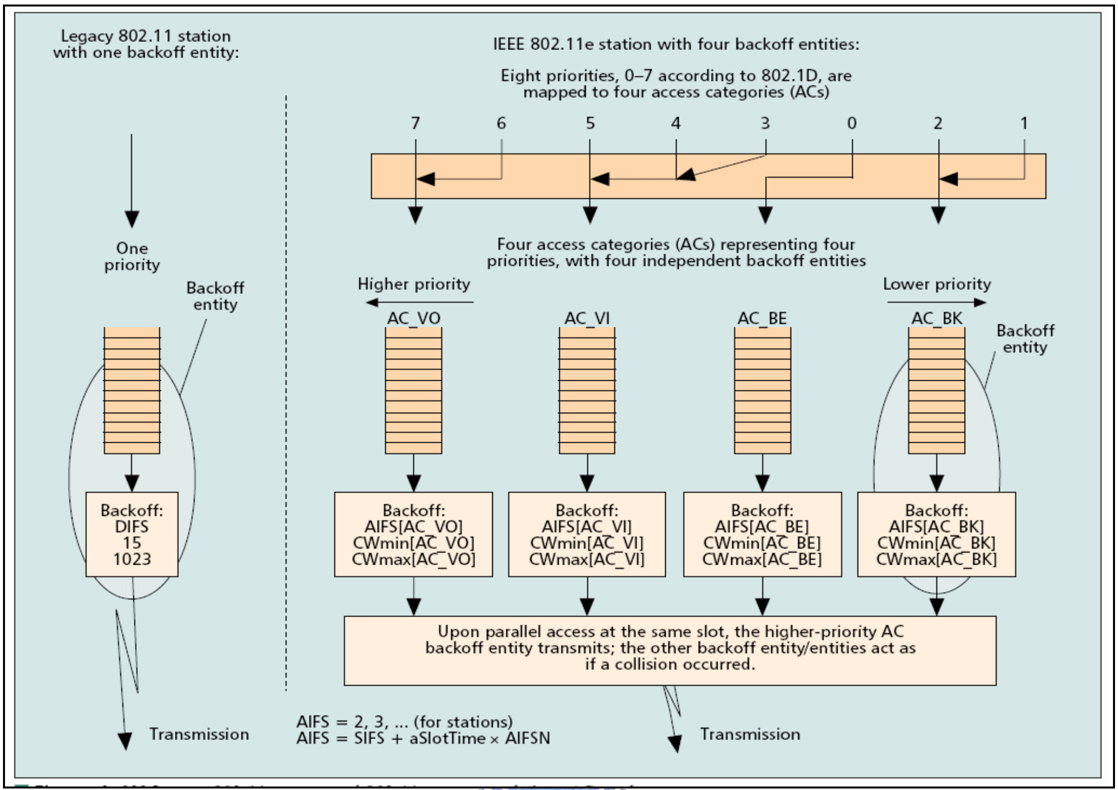

Figure 2.4: Comparison of backoff entities between 802.11 and 802.11e [3]

The EDCA parameter set defines the priorities in medium access by setting individual IFS, CW, and some other parameters for each AC. The same EDCA parameter set should be used by the backoff entities of the same AC in different stations. Each backoff entity within a station independently contends for a TXOP. The IFS used for backoff entity of IEEE 802.11e QSTA is arbitration inter-frame space (AIFS[AC]) instead of DIFS, which is used by legacy stations. The AIFS[AC] is at least DIFS, and can be enlarged for the lower priority ACs by the arbitration interframe space number (AIFSN[AC]). The AIFSN[AC] defines the duration of AIFS[AC] according to:

$$AIFS[AC] = SIFS + AIFSN[AC] \cdot aSlotTime, \quad AIFSN[AC] \geq 2.$$

The minimum size of the contention window, CWmin[AC], is another parameter controlling the access probability of the specified AC. The following figure will describe the correlation between the priority order and the parameters. The smaller the CWmin[AC] and AIFS[AC], the higher the priority in medium access.

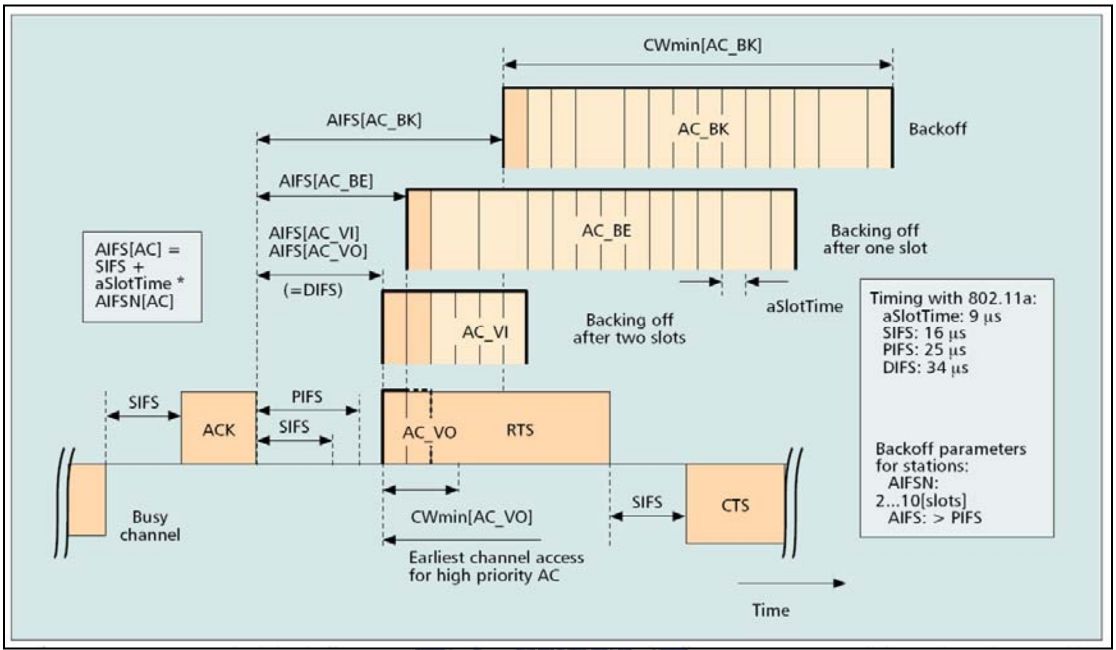

Figure 2.5: Correlations between AC and EDCA parameters [3]

In addition to the backoff parameters, the TXOPlimit[AC] is defined per AC as part of the EDCA parameter set. The larger TXOPlimit[AC] is, the larger the share of capacity for this AC will be. During a TXOP obtained from contention, a backoff entity can continue to deliver multiple MSDUs. By the introducing of this new characteristic, the upper bound of the TXOP belonging to some AC is under control and other ACs still can have reasonable opportunity to use the medium. Continually sending the frame with SIFS long inter-frame space can also reduce the overheads in transmission to improve the efficiency.

2.4 HCF Controlled Channel Access [2]

Besides EDCA, IEEE 802.11e specifies another MAC function extension referred to as Hybrid Coordination Function (HCF) Controlled Channel Access. The HCCA mechanism uses a Hybrid Coordinator (HC) which is collocated with the QoS access point (QAP) of the QBSS and has higher priority to access the medium. The HC can initiate frame exchange sequences and allocate TXOPs to itself or other QSTAs at any time when it senses the wireless medium (WM) has been determined to be idle for one PIFS period. In other words, it can provide limited-duration controlled access phase (CAP) at the contention period and initiate CFP after beacon frame for contention-free transfer of QoS data with higher priority than other non-AP QSTAs. The interval between frames during the CFP/CAP is one SIFS period, and therefore improves efficiency of the channel utilization.

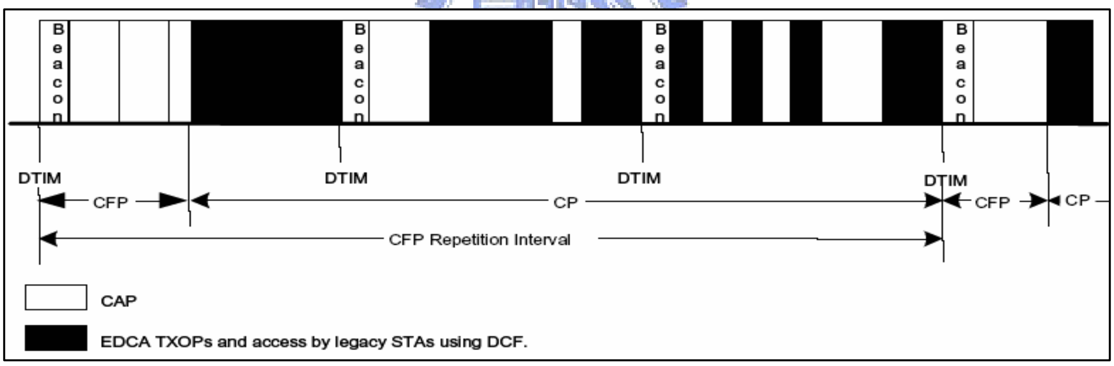

Figure 2.6: CFP, CAP, and CP periods in IEEE 802.11e [2]

The HC traffic delivery and TXOP allocation may be scheduled during the CP and any locally-generated CFP (generated optionally by the HC) to meet the QoS requirements of a particular traffic category or traffic stream. TXOP allocations can be based on the information obtained from negotiation of traffic specification (TSPEC). Through the TSPEC, the HC can have a QBSS-wide knowledge of the amounts of pending traffic belonging to different TS and/or TCs and schedule the traffic. When it is the time to poll the QSTA, HC should allocate suitable TXOP whose limit is notified at the QoS Control field of the QoS (+) CF-Poll frame. Within the polled TXOP a QSTA may initiate multiple frame exchange sequences when the remaining time is sufficient. Besides, the HCF protects the transmission during each CAP using the virtual carrier sense mechanism. The use of this mechanism provides improved protection of the CFP, in addition to the protection provided by having all STAs in the BSS setting their NAVs to a value of CFP max duration at TBTT.

Element ID (13)	Length (55)	TS Info	Nominal MSDU Size	Maximum MSDU Size	Minimum Service Interval	Maximum Service Interval	Inactivity Interval	Suspension Interval
Service Start Time	Minimum Data Rate	Mean Data Rate	Peak Data Rate	Maximum Burst Size	Delay Bound	Minimum PHY Rate	Surplus Bandwidth Allowance	Medium Time

Table 2.1: TSPEC element fields [2]

In the following chapter, we will give a survey of the scheduling algorithm related with the HCCA or HCF mechanism. Since the HC is collocated in the QAP, we will use QAP to replace the term HC when making mention of the central coordinator of HCCA.

Chapter 3

Related Works

There are many researches on the topic of Quality of Service (QoS) at present, but most of them are based on Differentiated Services (DiffServ). For the Integrated Service (IntServ) architecture, it needs to implement many functions on the side of QAP and may not be very scalable compared with DiffServ. However, if the service schedule is designed well for each kind of traffic, it can provide more guaranteed and stable QoS for the admitted traffic. In the following paragraphs of this chapter, we will present a survey of related scheduling algorithms of nowadays.

3.1 Referenced Scheduler of IEEE 802.11e [2]

The IEEE 802.11e draft provides an example scheduler in the annex as a reference design to meet the minimum performance requirements of different types of traffic. Each QoS station (QSTA) requiring strict and guaranteed QoS support can send an Add-Traffic-Stream (ADDTS) frame to do QoS request with the HC. The QoS request frame includes a Traffic Specification (TSPEC) element that brings the information to notify the requirements of the traffic stream (TS). This simple scheduler uses the mandatory set of TSPEC parameters to generate a schedule; these parameters are Mean Data Rate, Nominal MSDU Size and Maximum Service Interval (MSI) or Delay Bound. If both MSI and Delay Bound are specified by the non-AP QSTA in the TSPEC, the scheduler uses the MSI to do calculation for the schedule.

After gathering the requests, the QAP first determines the minimum value of all the MSI required by the admitted TSs. Then it will compute the highest sum-multiple of the beacon interval that is lower than the determined minimum of the MSI. This value will become the Scheduled Service Interval for all non-AP QSTAs with admitted streams. Therefore, the beacon interval is divided into multiple Sis and the admitted TS will be polled in a round-robin sequences during the CFP/CAP of each SI.

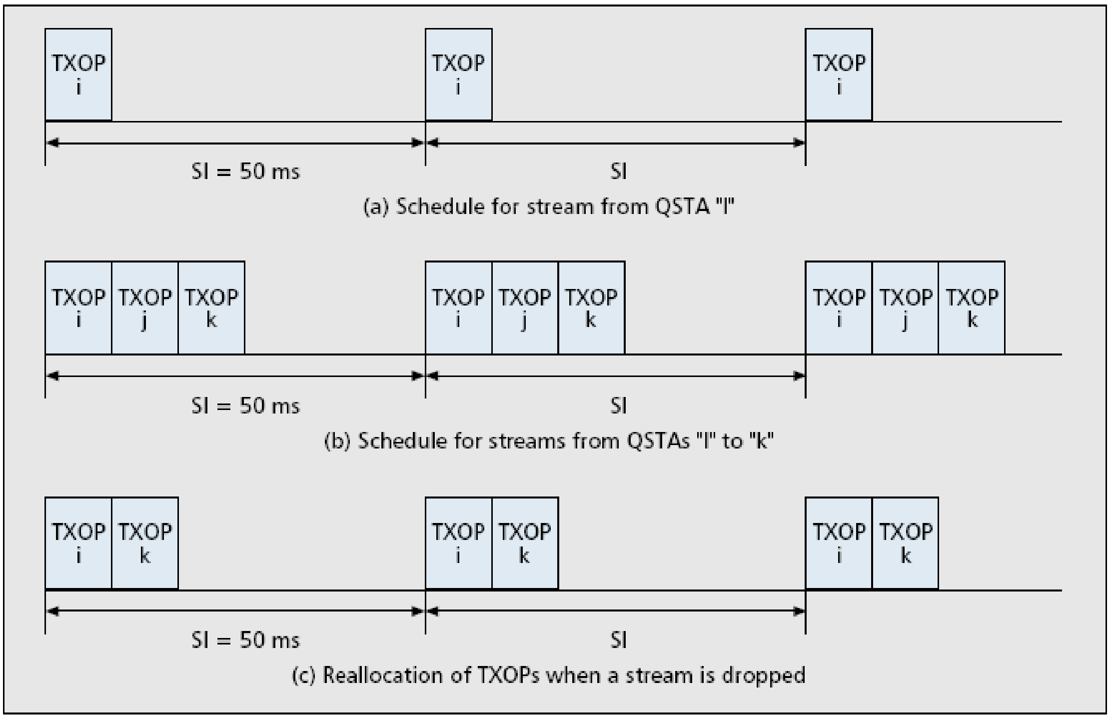

Figure 3.1: Structure of service interval in referenced scheduler [16]

To calculate the allocated TXOP of specified TS, the QAP uses the following parameters: Mean Data Rate (ρ), Nominal MSDU Size (L), the Scheduled Service Interval (SI) derived above, Physical Transmission Rate (R), Maximum allowable Size of MSDU (M), and Overheads in time units (O). The Overheads in time is composed of IFS, ACKs and CF-Polls duration. The TXOP is calculated as follows. First, the scheduler need to calculate the number of MSDUs reached during a SI:

$$N_i = \left\lceil \frac{SI \times \rho_i}{L_i} \right\rceil \tag{3.1}$$

Then the scheduler calculates the TXOP duration to clear the generated MSDUs:

$$TXOP_i = \max\left(\frac{N_i \times L_i}{R_i} + O, \frac{M}{R_i} + O\right),\ O = \left(\frac{L_i}{R} + 2 \cdot SIFS + ACK\right) \tag{3.2}$$

If the application is strictly constant bit rate (CBR), then the data rate will always follows the exchanged value of the mean data rate. Thus the TXOP duration derived can fulfill the requirement of traffic streams of this kind. However, when the type of application becomes variable bit rate (VBR), the data rate and packet size may fluctuate with time. When the rate is much higher then the mean value, using this scheme may possibly increase the packet delay or drops and then it cannot provide guaranteed QoS for the admitted traffic streams.

3.2 The SETT-EDD Scheme (Scheduling Based on Estimated Transmission Time: Earliest Due Date) [8]

The SETT-EDD Scheme uses a much different way to decide the allocated TXOP and the calculation of service interval. While enforcing the allocated TXOP equal to the average value in the referenced scheduler, the SETT-EDD scheme suggest that the QAP should maintain a TXOP timer. The TXOP timer is a token bucket of time units whose token adding rate is equal to $\frac{TD_j}{mSI_j}$ which is the maximum fraction of time the STA j can spend in polled TXOPs (mSI_j is the minimum interval between two successive polls and TD_j is the average TXOP duration). The depth of the TXOP timer is set to MTD_j which is bounded by the transmission time of the aggregate maximum burst size of traffic. The time token will be deducted form the TXOP timer after the station use the polled TXOP. Only when the time token in the TXOP timer is above the minimum TXOP duration can the station be polled.

Regarding the service interval, the SETT-EDD does not use a fixed duration for all stations in the polling sequences. Instead, each station in this scheme will have an independent service interval equal to its mSI. If the current due time to poll a station is t, the next poll to be issued should fall on t' that satisfies the following:

$$t + mSI \le t' \le t + MSI \tag{3.3}$$

The lower bound of t' is the instant after which the station can be polled, and equivalent the release time in the real-time scheduling theory. The upper bound reveals the maximum time by which the next poll should be done, or deadline of the traffic. The relations between MSI and delay bound D can be written as:

$$MSI_j = \beta \times (D - MTD_j) \text{ with } 0 < \beta \le 1 \text{ and } D = \min D_i, \text{for } i = 1 \sim n \tag{3.4}$$

After having these bounds, the QAP has to decide which station to poll first at a given moment. Because it uses deadline as the reference to perform polling, the scheduling algorithm is referred to as delay bound earliest due date (Delay-EDD). Real-time scheduling theory has already settled that EDD is optimal in a wide set of real-rime scheduling problems. The admission control calculation for SETT-EDD remains similar as referenced scheduler of TGe, but the SI is now replaced with mSI for station j.

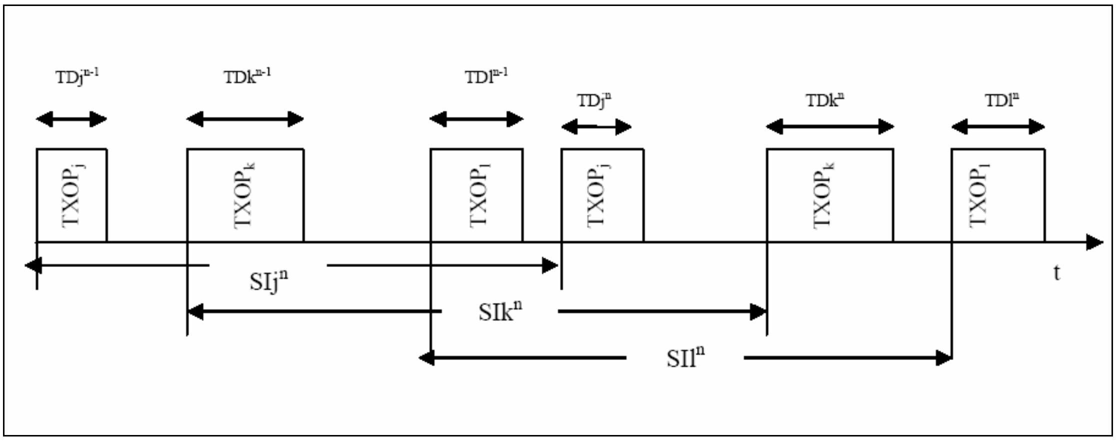

Figure 3.2: The diagram of SETT-EDD scheme [10]

For the SETT-EDD algorithm, TXOP durations are still calculated based on TSPEC parameters and the interval between two consecutive transmissions of the same station as that in the referenced scheduler. One possible extension is to try to collect the exact required duration of the TXOP for a QSTA. From the MAC header of the Data frame, the QAP can read the Queue Size or TXOP duration requested information that is filled in QSTAs to reveal the instant information of queue. The mechanism is called Traffic Scheduling Based Actual Requirements [10]. Having these requests about current traffic load, the scheduler can assign a TXOP duration satisfying the QSTA requirements in the next polling sequence. The definition of service interval will be changed as following equation in this mechanism.

$$MSI_j = \frac{1}{2} \times (D - MTD_j) \qquad (3.6)$$

The method of SETT-EDD should be able to improve the performance of the referenced scheduler of TGe by reducing the packet loss ratio and delay of video streams. However, finding the traffic stream with earliest deadline to achieve the optimality at each moment will cause a scalability problem in real implementation. When the number of QSTAs and the traffic stream that needs to be scheduled increases, this scheduling algorithm become inefficient because that the QAP will need to calculate each traffic stream in every QSTA.

3.3 The FHCF Scheme [6]

FHCF is composed of two schedulers: a QAP scheduler and a node scheduler. Before next SI starts, the QAP scheduler estimates the queue length of each admitted traffic stream for every QSTA. Since the traffic may be VBR, the estimation based on mean data rate is not always correct. Therefore, each time the estimation is performed, the QAP scheduler needs to add some correct terms to minimize the estimation error derived from comparison with the real situation.

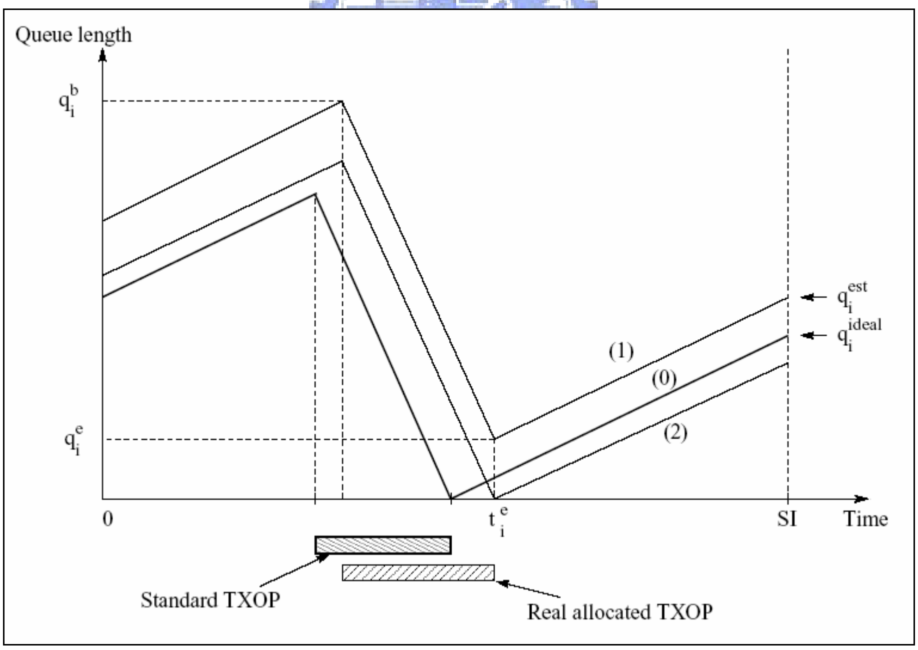

Figure 3.3: The diagram of queue length evolution for a TS [6]

For the correct terms, first the QAP can use the queue length information q_i^e

and the time t_i^e derived at the end of TXOP. The queue length estimation at the beginning of next SI is:

$$q_i^{est} = \frac{\rho_i (SI - t_i^e)}{M_i} + q_i^e \tag{3.7}$$

Besides, the FHCF proposes a window method to get the absolute value $|\Delta_i^n|$ of average estimation error derived from difference between the real queue length and the estimation at the beginning of TXOP of i-th TS: $\Delta_i^n = q_i^{b,real} - q_i^{b,est}$. Thus, in order to allow the QSTA to transmit more packets that arise from possible fluctuating data rate that is above the mean value, the QAP scheduler records a window of w previous $|\Delta_i|$, whose average value is supposed to be close to the expectation of $|\Delta_i|$.

$$q_{i,new}^{b,est}(n) = q_i^{b,est}(n) + \frac{\sum_{j=n-w}^{n-1} |\Delta_i^j|}{w} \tag{3.8}$$

After deriving the above new term of estimation, the QAP needs to calculate the additional number of packets, DN_i^{est}, which is the difference between the estimated queue length and the one calculated based on TSPEC values (as the way in referenced scheduler). Since the data rate term used in estimation is the same as the TSPEC value, the queue length will both evolve linearly during time, and then, we can obtain:

$$DN_i^{est} = q_{i,new}^{b,est}(n) - q_i^{b,ideal}(n) = q_{i,new}^{est}(n) - q_i^{ideal}(n) = q_i^{est}(n) - q_i^{ideal}(n) + \frac{\sum_{j=n-w}^{n-1} |\Delta_i^j|}{w} \tag{3.9}$$

Then, the additional required time will be:

$$t_i^{est} = DN_i^{est} \cdot \left(\frac{M_i}{R} + 2SIFS + ACK\right) \tag{3.10}$$

Collecting the positive and the negative time for additional packets, the QAP will examine that if the available remaining time is sufficient to do the compensation. If the additional required time is above the available time, then the QAP will redistribute the requiring time for all the streams.

$$t_i^{add} = \begin{cases} (1+\beta)t_i^{est} & \text{, if } t_i^{est} \geq 0 \\ (1-\beta)t_i^{est} & \text{, if } t_i^{est} < 0 \end{cases}, \text{ where } \beta = -\frac{(T_P - T_N) - T^r}{T_P + T_N} \text{ (if } T_P - T_N > T^r\text{)} \quad (3.11)$$

The node scheduler will also perform a similar mechanism to redistribute the polled TXOP which received from QAP.

Since the corrective term is supposed to be close to expectation, this scheme may be much more suitable for the traffics that do not fluctuate too much and exhibits the same behavior for the different flows. In the long term average performance, it can achieve a pretty good performance but sometimes still does not absorb the estimation error that is bigger than the average. Therefore, if the traffic varies a lot around its nominal requirements, the FHCF scheme may not provide enough time to absorb the variation.

3.4 The P-HCCA Scheme [9]

To adapt to the fluctuating flows, the P-HCCA scheme provide a way to predict the possible queue length at the beginning at next SI. The QAP maintains a polling list of the admitted TSs. After receiving a QoS request or QoS-NULL message from a QSTA, it will update the terms in the corresponding polling list.

Item	Description
TC-ID	Traffic Stream's ID
Mean Rate	Mean application data rate of the TS

Max SI	Maximum Service Interval
MSDU	MSDU Size
S-Queue-Len	Queue length at the beginning of TXOP
F-Queue-Len	Queue length at the end of TXOP
Packet-Num	Packets transmitted by TS during TXOP
TXOP	TXOP of the next polling round

Table 3.1: Polling list of P-HCCA scheme

The principle of P-HCCA scheduling scheme is to evaluate the mean application data rate and to predict the TXOP during next polling sequences for the corresponding TS. Since it is very difficult to evaluate the instant data rate of VBR traffic at a specified time point, the QAP can only calculate the mean data rate for a given interval. Therefore, the P-HCCA scheme proposes an evaluation of application data rate over a TXOP duration allocated to a TS:

$$DataSize = \int_0^{TXOP} d\rho \times dt = M \times (F + N - S) \quad \Rightarrow \quad \overline{Rate} = \frac{DataSize}{TXOP} \qquad (3.12)$$

On the above equation, $DataSize$ is the size of data that the application generates during a TXOP, $d\rho$ is the instantaneous application data rate at a specified transmission time point, M is MSDU size, S is the queue length at the beginning of TXOP, F is the queue length at the end of TXOP, and N is the amount of packets sent during the TXOP. Therefore, the QAP has to update the polling list items when it gets the corresponding information. The QAP also records the previous data rate for prediction of the mean application data rate of next polling round. Although the data rate of VBR traffic fluctuates with time, the curve of rate is near smooth during a very short period of time. So, the QAP can roughly evaluate the data rate of next polling round with the following equation:

$$\overline{NextRate} = \overline{Rate} + (\overline{Rate} - \overline{PreRate}) \times \beta \tag{3.13}$$

where $\overline{NextRate}$ is the mean application data rate in next TXOP, $\overline{Rate}$ is the mean data rate of current TXOP, and $\overline{PreRate}$ is the rate of previous TXOP. Besides, β is the adapting factor evaluated by the ratio of the amount of packet that TS has sent during the current TXOP and previous TXOP. So, by using the equation:

$$Next\_TXOP = N\left(\frac{M}{\overline{NextRate}} + 2 \times SIFS + ACK\right) \tag{3.14}$$

the QAP can allocate TXOP for the TS during next polling round.

The P-HCCA provides a different way from FHCF to estimate the queue length at the beginning of next polling round. However, because it only uses the information during a TXOP (it is a very short time) to do rate estimation, the derived mean data rate may not be representative for the entire service interval. After examining the above equations in P-HCCA, we can know that there is a mistake in equation (3.14): the duration spent on a MSDU should be evaluated over "Physical Transmission Rate", not the mean data rate which is the generation rate of MSDU from upper layer of computer. Using N (amount of packets sent during current TXOP) as the number of packets to be sent during next TXOP may also cause some error. Finally, this prediction mechanism does not provide any compensating function to remedy prediction errors.

Chapter 4

Data Rate Estimation Algorithm for the Scheduler

In this chapter, we propose our enhanced algorithm for estimating of data rate of specified traffic stream to be used for IEEE 802.11e HCCA scheduler for providing good delay and throughput guarantee. Our thought is motivated by the scheme of P-HCCA and presents some improvements to it.

From the data revealed in [12], we get the typical QoS requirements of various kind of traffic for several service classes: non-real-time variable bit rate (nrt-VBR), available bit rate (ABR), unspecified bit rate (UBR), constant bit rate (CBR) and real-time VBR (rt-VBR). Notice that the delay is measured between the QAP and QSTAs:

Class	Application	Bandwidth (b/s)	Delay bound (ms)	Loss rate
CBR	Voice	32 k-2 M	30-60	10^{-2}
nrt-VBR	Digital video	1 M-10 M	Large	10^{-6}
rt-VBR	Videoconference	128 k-6 M	40-90	10^{-3}
UBR	File transfer	1 M-10 M	Large	10^{-8}
ABR	Web browsing	1 M-10 M	Large	10^{-8}

Table 4.1: QoS requirements of different kind of traffic [12]

In our scheme, we focus on the challenge that in IEEE 802.11e the QAP cannot instantaneously know the exact requirements of VBR traffic, like videoconference streams, during transmission time. If we estimate the possible TXOP duration only based on TSPEC parameters, as provided in referenced scheduler, QoS performance can be bad when the real rate is above mean value. Hence, under the inspiration from P-HCCA, we need a dynamical data rate estimating scheme with time to adjust the TXOP allocated to traffic streams of QSTAs.

4.1 Data Rate Estimation and Prediction

Different from P-HCCA, we estimate the mean data rate of a specified traffic over duration of service interval since the allocated TXOP aims to clear the traffic load that comes in a service interval. First, we determine the service interval for all traffic streams of each QSTA in the same way described in referenced scheduler. The QAP picks up the value, which is smaller than all of the *MSI* and is a sub-multiple of beacon interval, as service interval. Note here that the relation between *MSI* and Delay Bound is not specified in the upcoming standard of IEEE 802.11e, so we follow the guideline of SETT-EDD [8] to set the *MSI* as $MSI_i \le D_i - MTD_i$ for TS i when generating the TSPEC in the upper layer of QSTAs. Therefore, from the QoS requirements shown in table 4.1, we suggest that the MSI of audio and that of real-time video be set to 25*ms* and 50*ms*, respectively. Then QAP will periodically initiate a CAP/CFP after duration of service interval, and all QSTAs should set their NAVs to the end of CAP/CFP duration each time it starts.

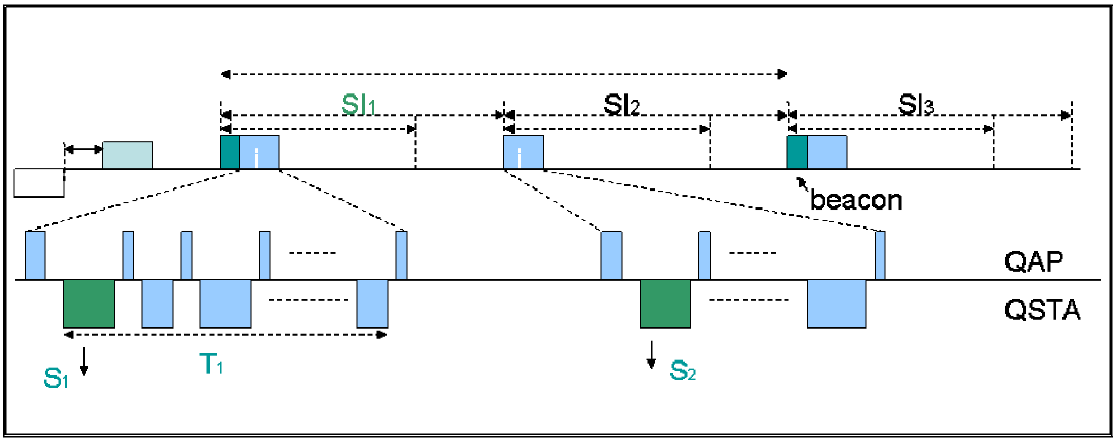

Figure 4.1: Structure of the beacon interval

Then, we have to gather the queue length information at the beginning of current and the previous TXOP (in the previous service interval) for the TS and the amount of MSDU size belonging to the TS that are sent during previous TXOP. Before these terms are fully collected, the QAP temporarily uses the mean value from negotiated TSPEC parameters. Notice that only VBR traffic, whose values of Mean Data Rate and Peak Data Rate items of TSPEC are not the same, needs to do rate estimation. From the following equation, we can have the total amount of traffic that comes in previous service interval:

$$Traffic_i = \int_0^{SI} M_i(t) \times dt = L_i \times (S_2 - S_1)_i + T_i \tag{4.1}$$

where the $M_i(t)$ means the MSDU size coming on time t, L is the mean MSDU size, S_2 is the queue length at the beginning of TXOP of current SI, S_1 is the queue length at the beginning of TXOP of previous SI, and T is the total amount of data size sent in TXOP. Then the mean application data rate of previous service interval can be written as:

$$\overline{Rate_i} = \frac{Traffic_i}{SI} \tag{4.2}$$

For the prediction of data rate of next *SI*, we use an AR-model equation. The equation is as following:

$$\widehat{NextRate_i}(n) = (1-\alpha)\overline{Rate_i}(n) + \alpha\,\overline{PreRate_i}(n) \quad (4.3)$$

In the above equation, n is the index of service interval and α is an adjustable parameter. In order to reduce the complexity of calculations in hardware, we can choose α as 2 to the power of $-k$, where k is a positive integer. In this way, $\widehat{NextRate_i}(n)$ can be derived using bit-shift instead of multiplication.

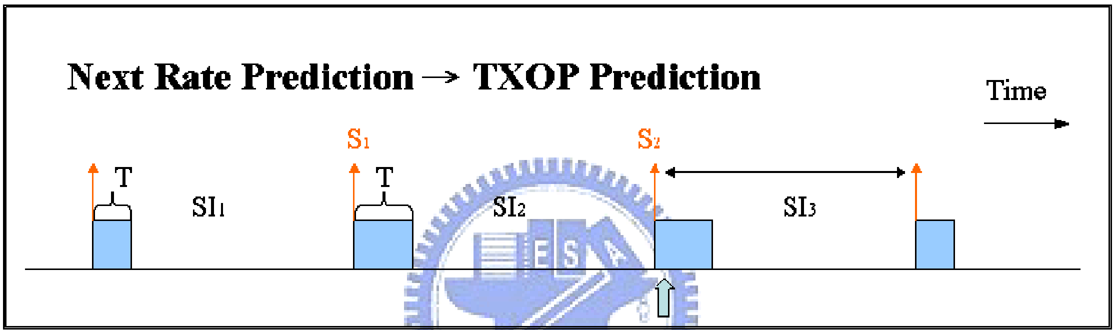

Figure 4.2: The diagram of data rate prediction

4.1 TXOP Calculation

Having the new rate information, we can utilize it to predict the TXOP duration needed to be allocated. First we should estimate the corresponding number of packets belonging to traffic stream i:

$$N_{next,i} = \left\lceil \frac{\widehat{NextRate_i} \times SI}{M_i} \right\rceil \quad (4.4)$$

Now, we derive the number of packets that will come in current service interval, and the traffic load should be cleared during next round of polling. The QAP then needs to calculate the required time corresponding to the number of traffic.

$$Next\_TXOP = \sum_{j}\left[N_{j,next}\cdot\left(\frac{M}{R}+2\times SIFS+ACK\right)+O\right] \tag{4.5}$$

Here we also utilize the method of aggregation. The term "Aggregation" means that if a QSTA has more than one admitted traffic stream in the schedule, each time when polling is performed the QAP will allocate TXOP considering all traffic streams belonging to the QSTA. In this way, it can reduce the number of polling packets and overheads and then make more efficient utilization of wireless medium.

Since the estimation may not always be very precise, we need a compensation mechanism to remedy the prediction error. The things we can utilize are the remaining time T^r after allocating all the polled TXOP to QSTAs in the CAP/CFP duration and the queue length information q_i^e at the end of TXOP for traffic stream *i*. After collecting the queue length information, we do remaining time redistribution according to the weight which is derived from the proportion of q_i^e to the sum of all queue information:

$$T_i^c = T^r \times \frac{q_i^e}{q_{sum}},\ \text{if}\ \left(q_{sum}=\sum_{\forall j} q_j^e\right) > 0 \tag{4.6}$$

This compensating time will also be combined with the TXOP allocated to QSTA to decrease the possible additional overheads of polling. In order to limit the long term average number of traffic that goes into the network, we add a token bucket mechanism to police the traffic. The depth of the token is set to the time to absorb the maximum burst size and the token adding rate is set to the mean data rate of the traffic stream. Hence, before allocating the polled TXOP, the QAP must examine the available token (will be transformed into available time) and the required time. Only when there is enough token in the bucket belonging to corresponding TS of the QTSA can the required time be fulfilled:

$$Next\_TXOP = \sum_j \min\left\{\left[N_{j,next} \cdot \left(\frac{M}{R} + 2 \times SIFS + ACK\right) + O\right] + T_j^c, token_j\right\} \quad (4.7)$$

To implement the proposed scheme, the QAP should also maintain a polling list for all the admitted traffic streams of each QSTA and update the corresponding items when it receives the information:

Item	Description
TID	Traffic Stream's ID
AID	Association ID
Pre Mean Rate	Mean data rate of the TS of the one before previous SI
Next Mean Rate	Mean data rate of the TS of next SI
Max SI	Maximum Service Interval
MSDU	MSDU Size
S_1 -Queue-Len	Queue length at the beginning of TXOP of previous SI
S_2 -Queue-Len	Queue length at the beginning of TXOP of current SI
End-Queue-Len	Queue length at the end of TXOP of previous SI
Traffic	Amount of traffic generated for TS during TXOP
Token	Available token in the token bucket
Depth	Depth of the token bucket

Table 4.2: Polling list maintained by the QAP

After receiving the polled TXOP, the QSTA should simply seek for the queues of admitted stream from high priority to low priority to examine if there is a packet that needs to be sent. Only when the remaining time of polled TXOP is sufficient to send the next packet, can the QSTA utilize the time. If there is no packet that needs to be sent or the time is not sufficient to send the next packet, the QSTA should return a QoS Null packet to the QAP and give the control of medium back to it. The QAP can then poll the next QSTA in the round-robin polling sequence order. The CAP/CFP will end after the finish of polling sequence by an announcement of CF-End packet with zero duration to reset the NAV of all QSTAs. Then the medium is given back for contention.

There should also be a reserved time during service interval for contention-based medium access as it is needed for important management tasks (e.g., association of stations with the QAP during handover or initial connection).

The following pages are the simulation flow charts. We will give the delay analysis for the traffic between the QAP and QSTA in the round-robin polling scheme after this chapter. Then, the performance of our proposed scheme will be evaluated by computer simulations in chapter 6.

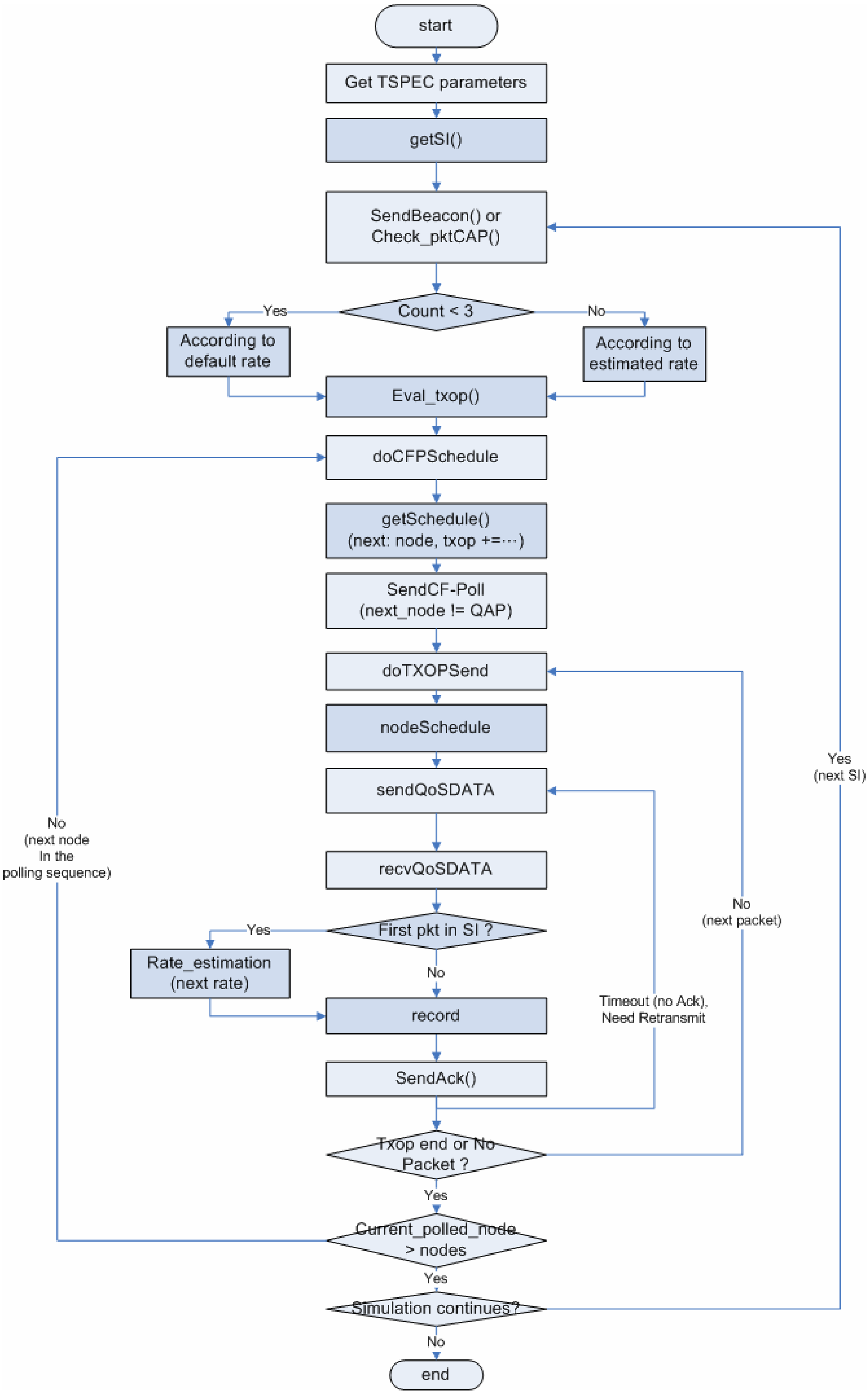

Figure 4.3: Flow chart of simulation of scheduling

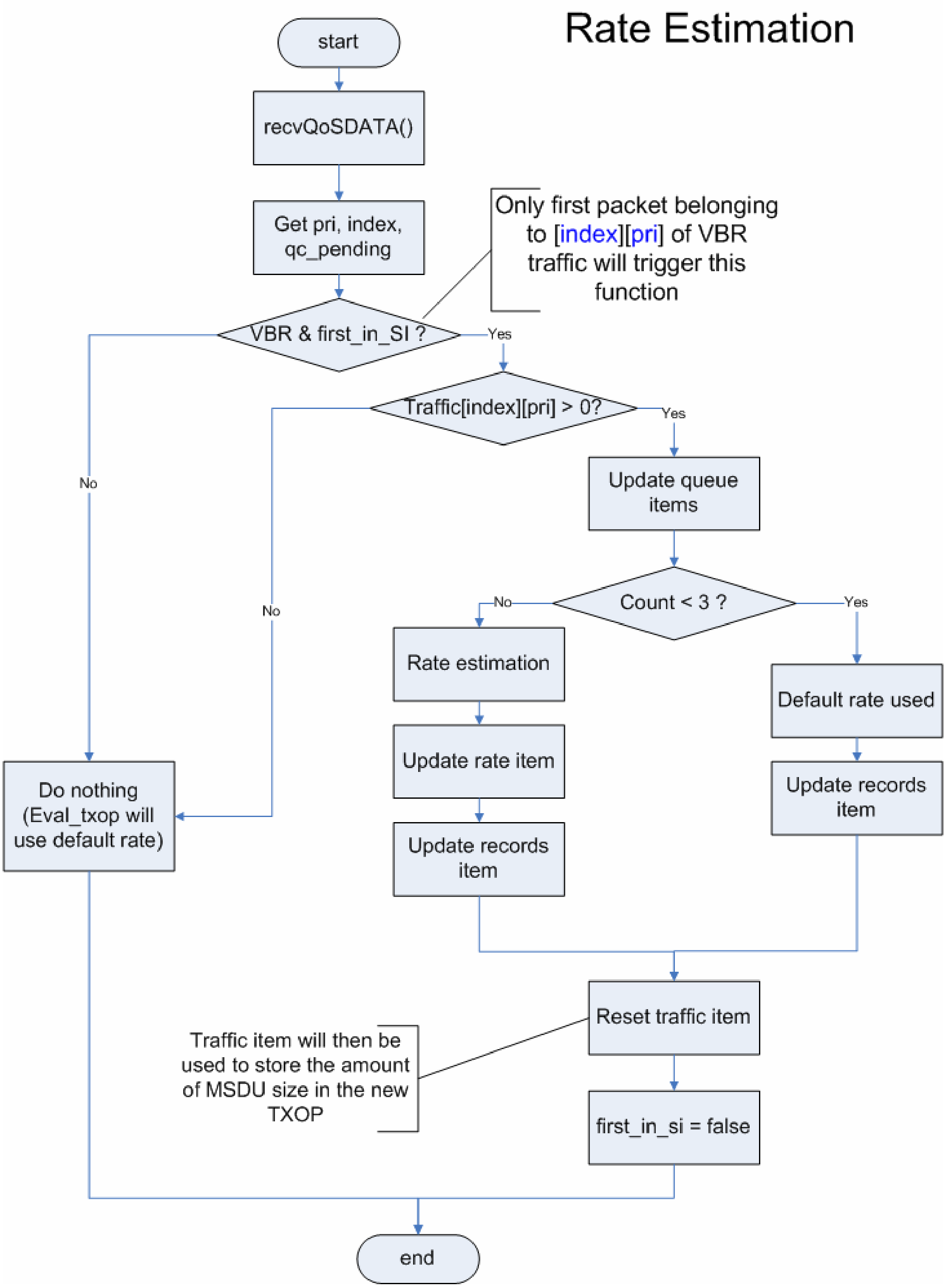

Figure 4.4: Flow chart of rate estimation function

Chapter 5

Delay Analysis [15]

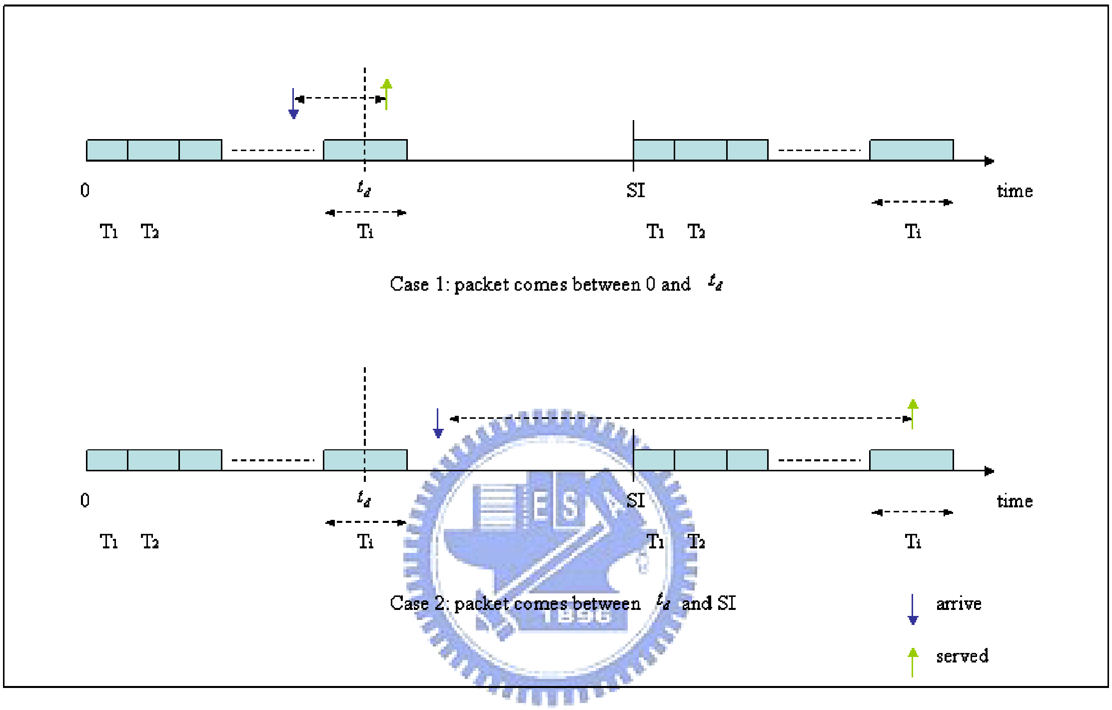

Figure 5.1: Delay analysis

Regarding the round-robin polling scheme, suppose that the queue length is q_i^e at the end of TXOP for traffic stream *i*. Let t_d denotes the time when the packets that come after this point cannot be transmitted during the TXOP in current SI, T_j denotes the TXOP duration of traffic stream with index *j*, and t is the arrival time of the packet that we want to analyze. t_d can be written as the following equation:

$$t_d = \sum_{j=1}^{i} T_j - \frac{q_i^e L_i}{\rho_i} \tag{5.1}$$

For case 1, packet comes between 0 (referred to as the beginning of current SI) and t_d, and the delay of packet belonging to traffic stream i can be written as:

$$d_i(t) = \left((\sum_{j=1}^{i-1} T_j - t) + \frac{q_i^0 L_i}{R} + \frac{\rho_i t}{R} \right) \tag{5.2}$$

In the above equation, q_i^0 is the queue length of traffic stream i at the beginning of current service interval. For case 2, packet comes between t_d and SI (referred to as the end current service interval), and the delay of packet belonging to traffic stream i can be written as:

$$d_i(t) = \left((SI - t + \sum_{j=1}^{i-1} T_j) + \frac{\rho_i (t - t_d)}{R} \right) \tag{5.3}$$

Thus the maximum delay denoted by D_i is obtained at the packet arrival time of t_d and is equal to:

$$D_i = \max_t d_i(t) = d_i(t_d) = SI - T_i + \frac{q_i^e L_i}{\rho_i} \tag{5.4}$$

If the packet comes uniformly during the SI, we can estimate the mean latency as:

$$\frac{1}{2}(SI - T_i + 2\frac{q_i^e L_i}{\rho_i}) \tag{5.5}$$

For the possible case that the QAP allocates enough time for the traffic stream and the queue will be empty at the end of TXOP ($q_i^e = 0$), since T_i is much smaller than SI, we can roughly know that the mean delay of simulation will be about half of the SI. In other cases, the result may vary a little for their traffic specification such as the arrival distribution. (Note that the delay including the waiting time till the next TXOP, queuing delay, and the transmission time of the packet.) Since TXOP duration may be variable in different service interval for each TS of separate QSTA and the interval between successive service periods may not be exactly equal to SI, the full analysis should take these into consideration.

Chapter 6

Simulation Results

6.1 Simulation Parameters

The traffic we use in our simulation is composed of three types of applications: burst on/off audio stream of priority 6, VBR video trace derived from the VIC videoconferencing tool using H.261 coding of priority 5 [7], and CBR video of the lowest priority 4. Their specification parameters are as follows:

application	Inter-arrival time (ms)	Nominal MSDU Size (bytes)	Mean Data Rate (kb/s)	Maximum Service Interval (ms)	Peak Data Rate (kb/s)	Traffic Stream/ User Priority	Time for packet in 54Mbps (μ s)
audio	4.7	160	64	25	64	6	71
VBR video	~26	~660	~200	50	500	5	145
CBR video	2	800	3200	50	3200	4	166

Table 6.1: Parameters of traffic in our simulation

Note that the *MSI* is set following the rule: $MSI \leq D - MTD$. Since the delay bound of audio traffic is around 30 ms and the delay bound of real-time video is around 50 ms, we set the *MSI* of the traffic as above table. The parameter α that we use in simulation is set to 2^{-3} . Hence the equation of data rate prediction can be written as

$\widehat{NextRate}_i(n) = (1-2^{-3}) \times \overline{Rate}_i(n) + 2^{-3} \times \overline{PreRate}_i(n)$. The PHY (we use OFDM scheme parameters) and MAC parameters are as follows:

PHY parameters	
SIFS	16 μs
DIFS	34 μs
ACK size	14 *bytes*
PHY Rate	54 *Mb/s*
Minimum Bandwidth	6 *Mb/s*
Slot Time	9 μs
CCA Time	4 μs
MAC header length	38 *bytes*
PLCP header length	4 *bits*
Preamble length	20 *bits*

MAC parameters	
CWmin (audio)	7
CWmax (audio)	15
CWmin (video)	15
CWmax (video)	31
Beacon interval	100 *ms*
CAPLimit	90 *ms*

Table 6.2: Environments parameters of our simulation

VBR trace	1	2	3	4	5	6
Duration (ms)	48294	38027	30995	31995	32327	33696
Mean data rate (b/s)	242189	199600	206817	197960	171355	236563
Mean packet size (bytes)	671	698	680	645	597	687

Table 6.3: Analysis of trace files [7]

6.2 Simulation Scenarios

6.2.1 Scenario: Three streams per QSTA (aggregation)

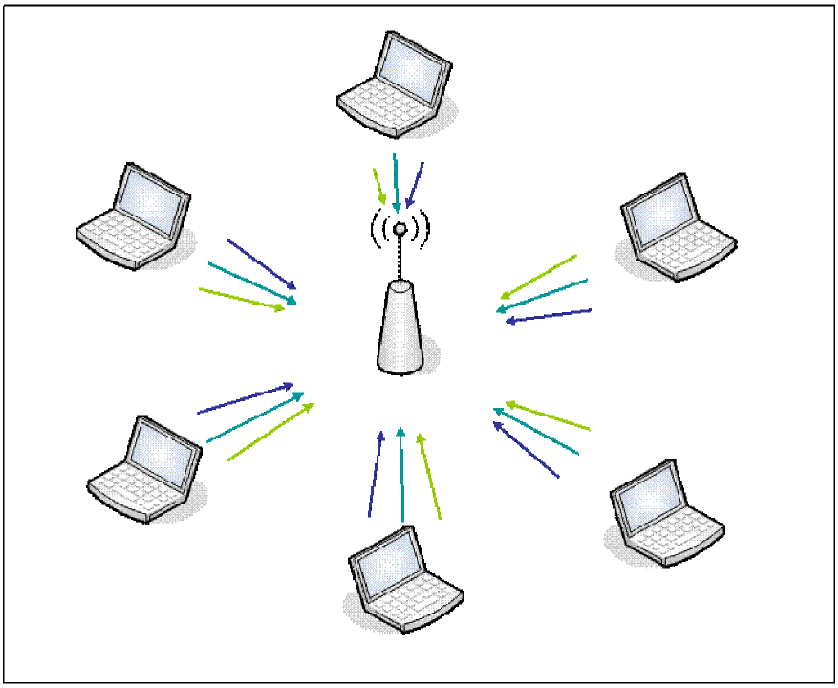

Figure 6.1: The diagram of simulation scenario

This scenario is composed of 6 QSTAs and a QAP. Each QSTA generates 3 streams to uplink to the QAP. Note that the implementation of referenced scheduler (HCCA/HCF) is from [7]. Our implementation is also adapted from it. The implementation constructs most of the architecture of the IEEE 802.11e standard in the NS-2 code. We change the scheduling function to do queue and TXOP estimation, as described in chapter 5 and chapter 6.

Besides, we will evaluate the performance for various load condition using EDCA, HCCA, and our proposed scheme, and make some comparisons in section 6.2.2. In this scenario, the CBR MPEG flow plays the most important role in changing the load condition since it has the largest packet size. We will change the CFP load (the ratio of total required time to CFP duration in a service interval) by increasing or decreasing the MSDU size of CBR traffic.

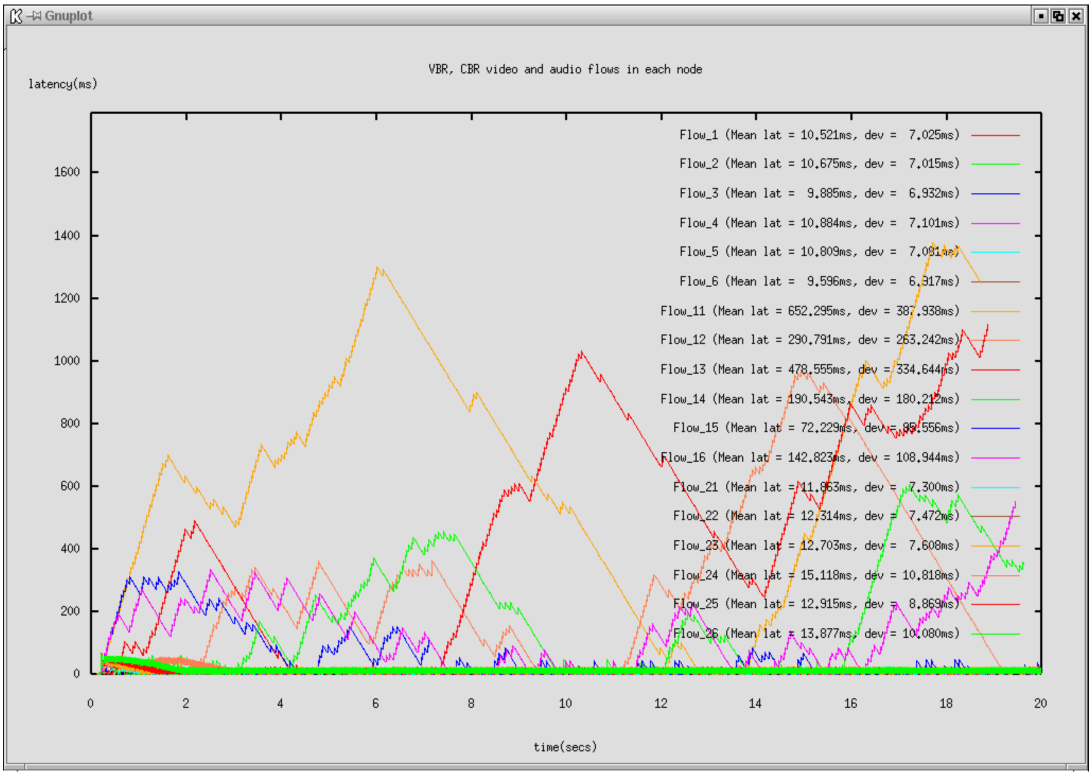

Figure 6.2: Latency versus Time based on referenced scheduler

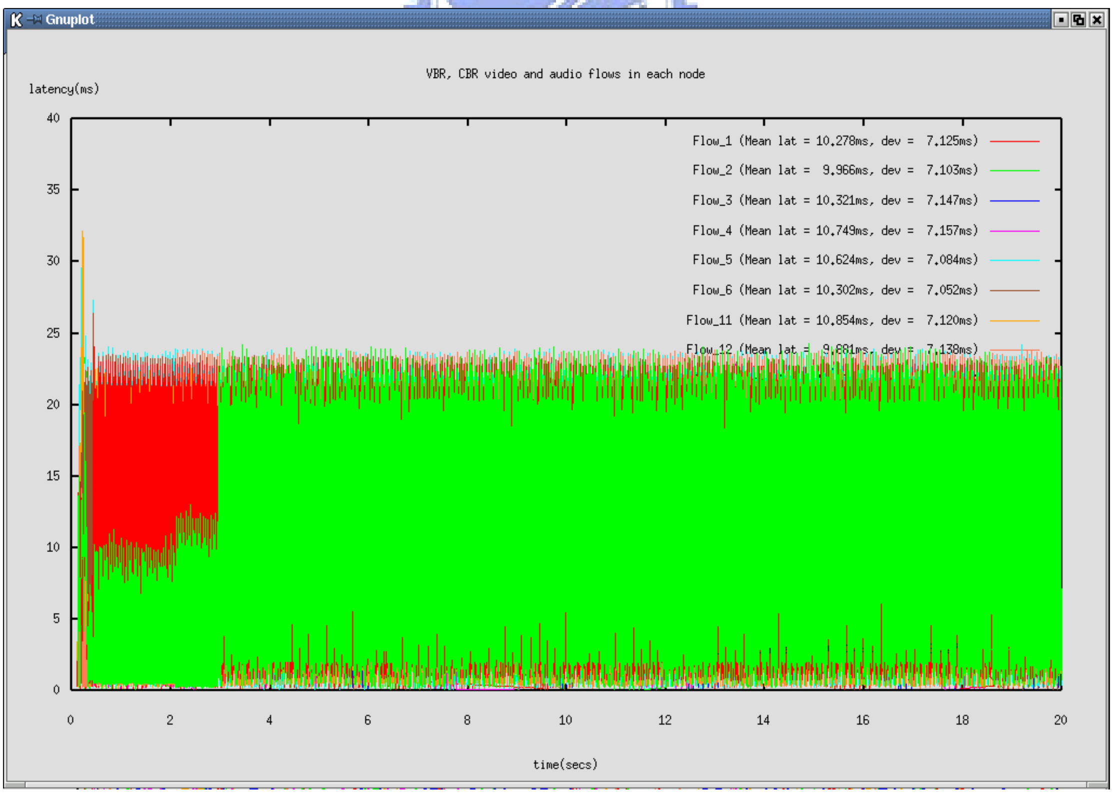

Figure 6.3: Latency versus Time based on proposed scheme

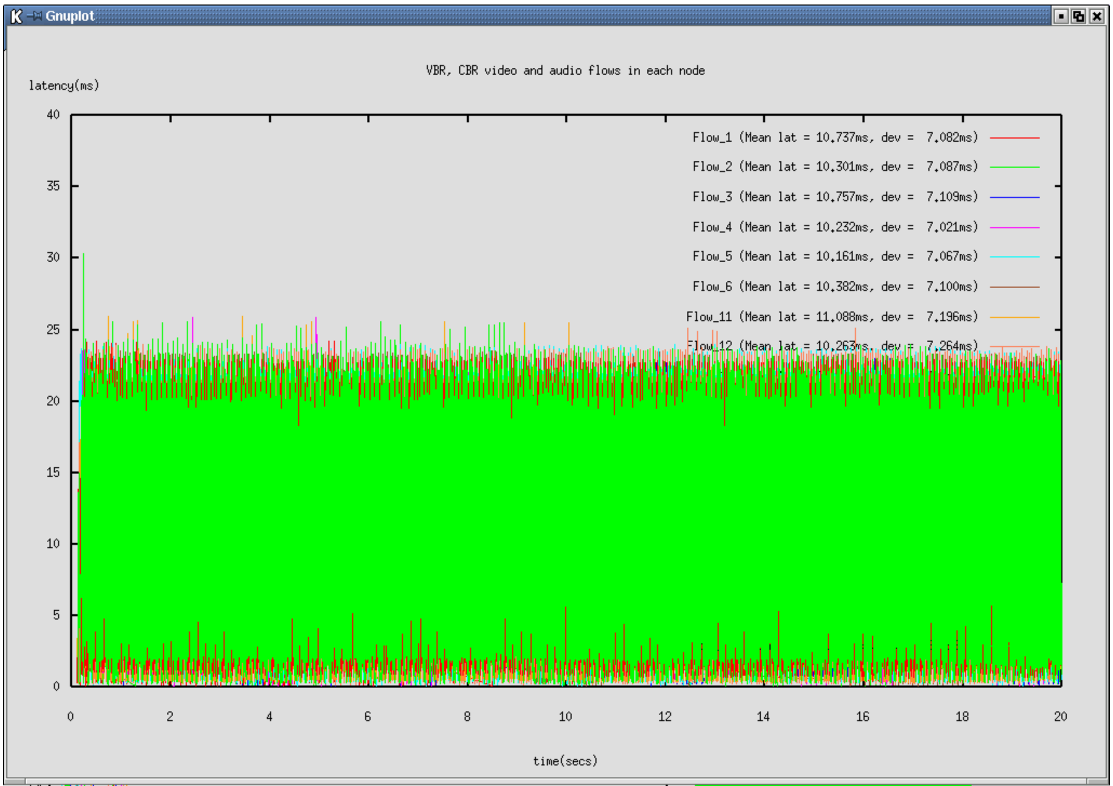

Figure 6.4: Latency versus Time based on FHCF scheme

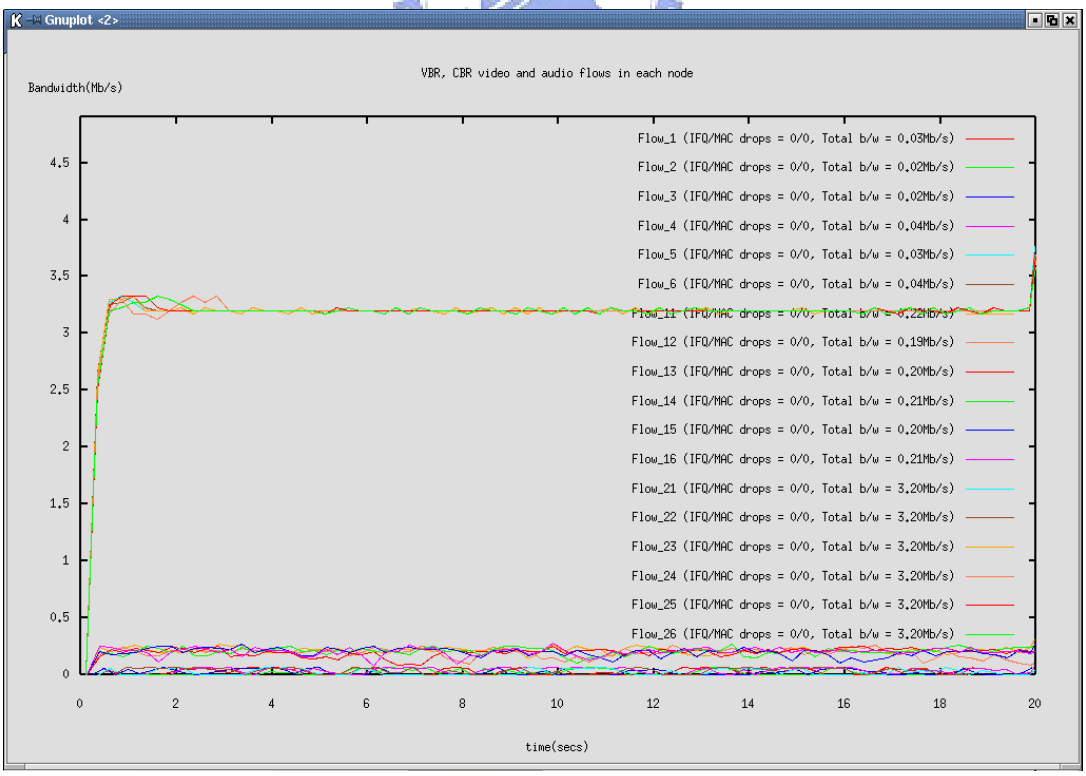

Figure 6.5: Bandwidth versus Time based on referenced scheduler

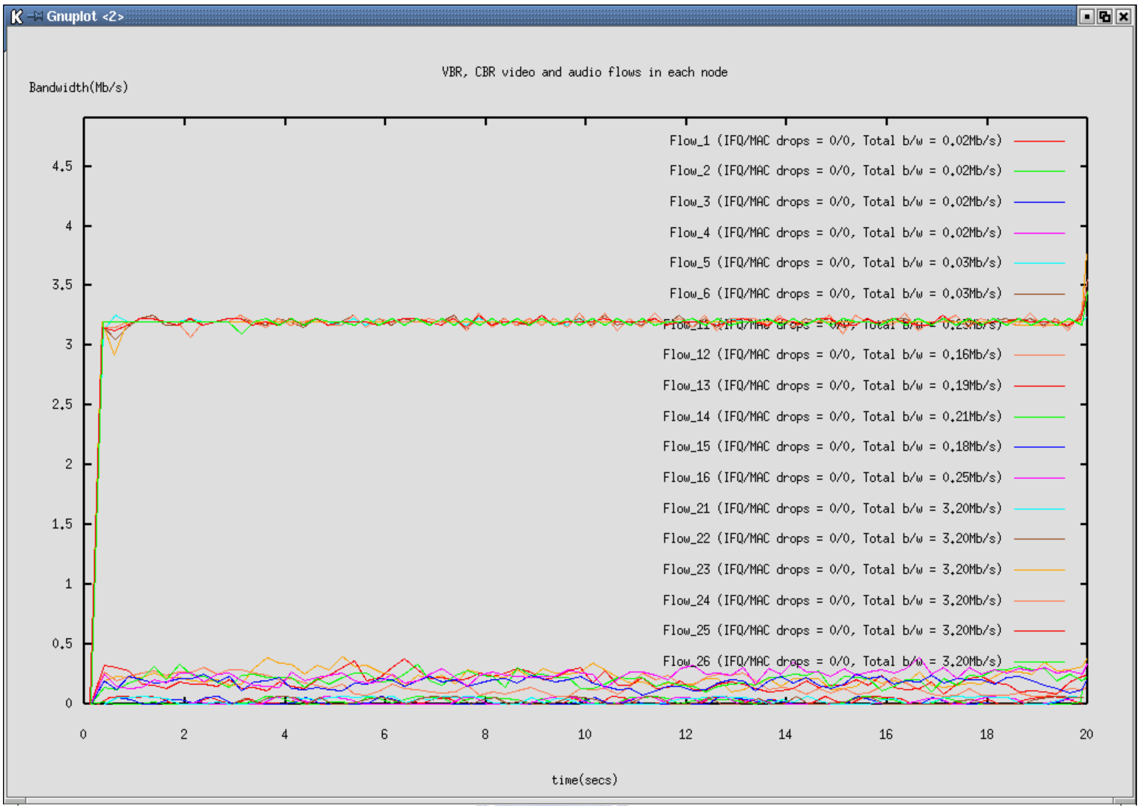

Figure 6.6: Bandwidth versus. Time based on proposed scheme

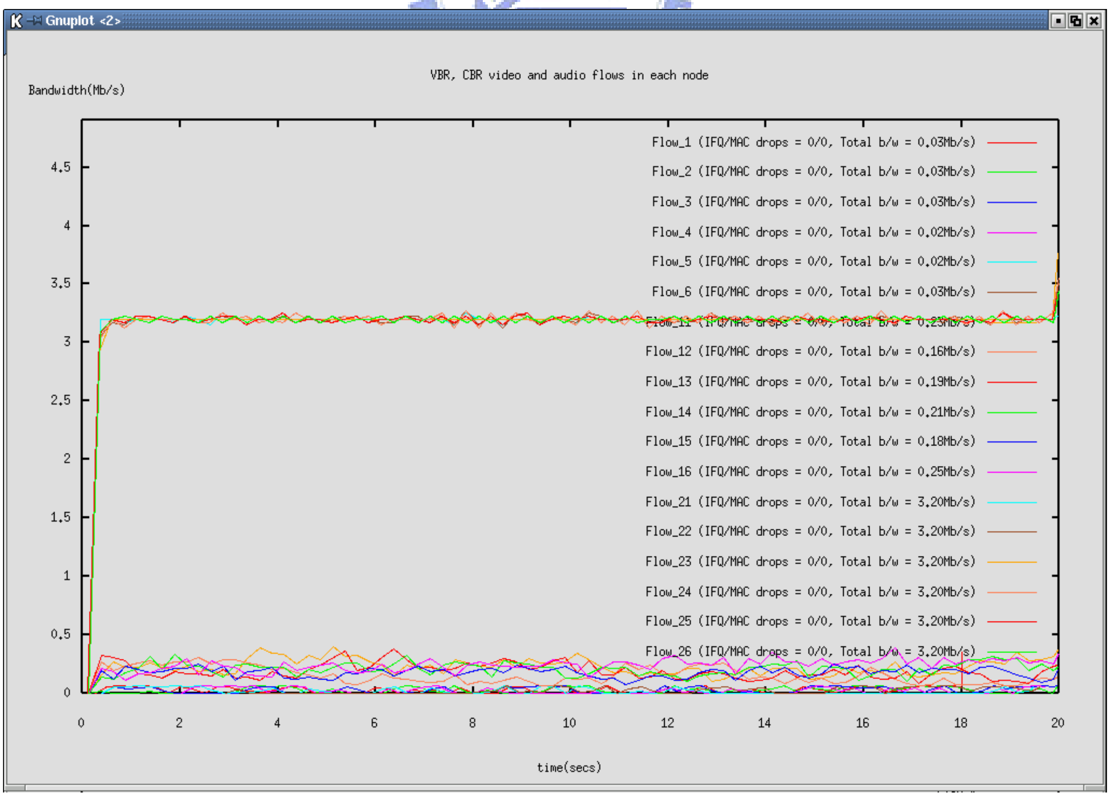

Figure 6.7: Bandwidth versus. Time based on FHCF scheme

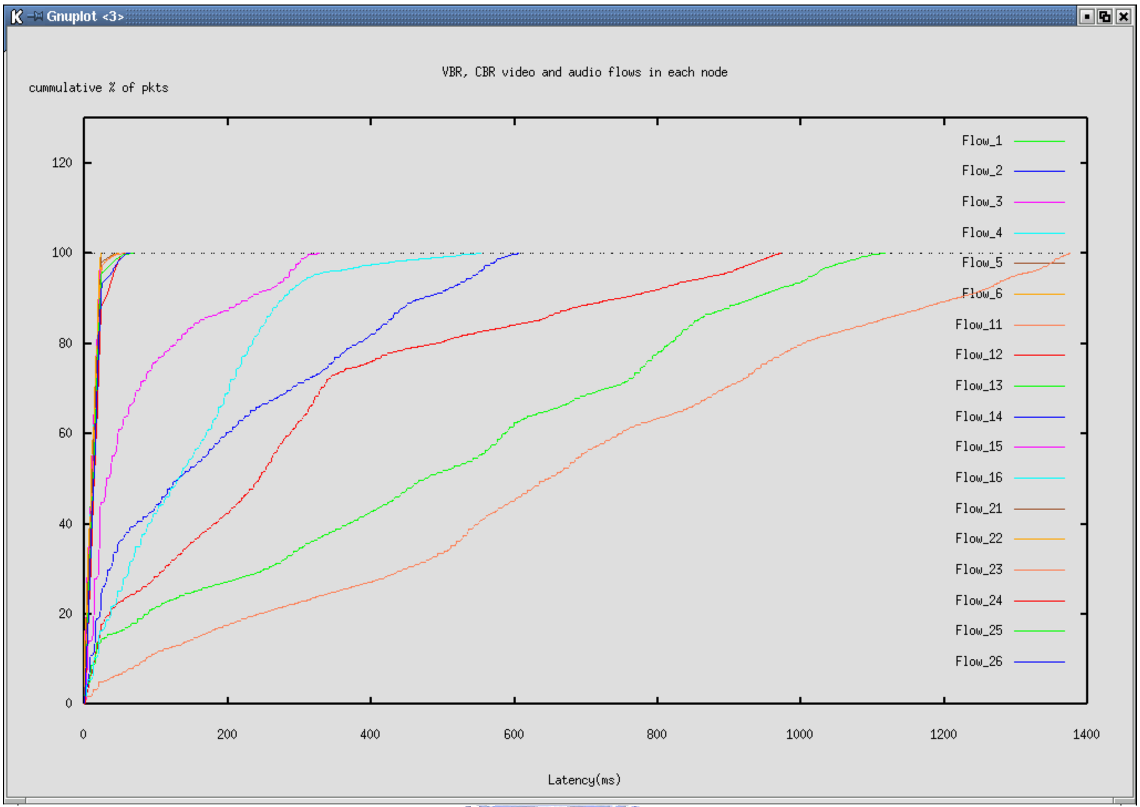

Figure 6.8: C.D.F of packets versus Latency based on referenced scheduler

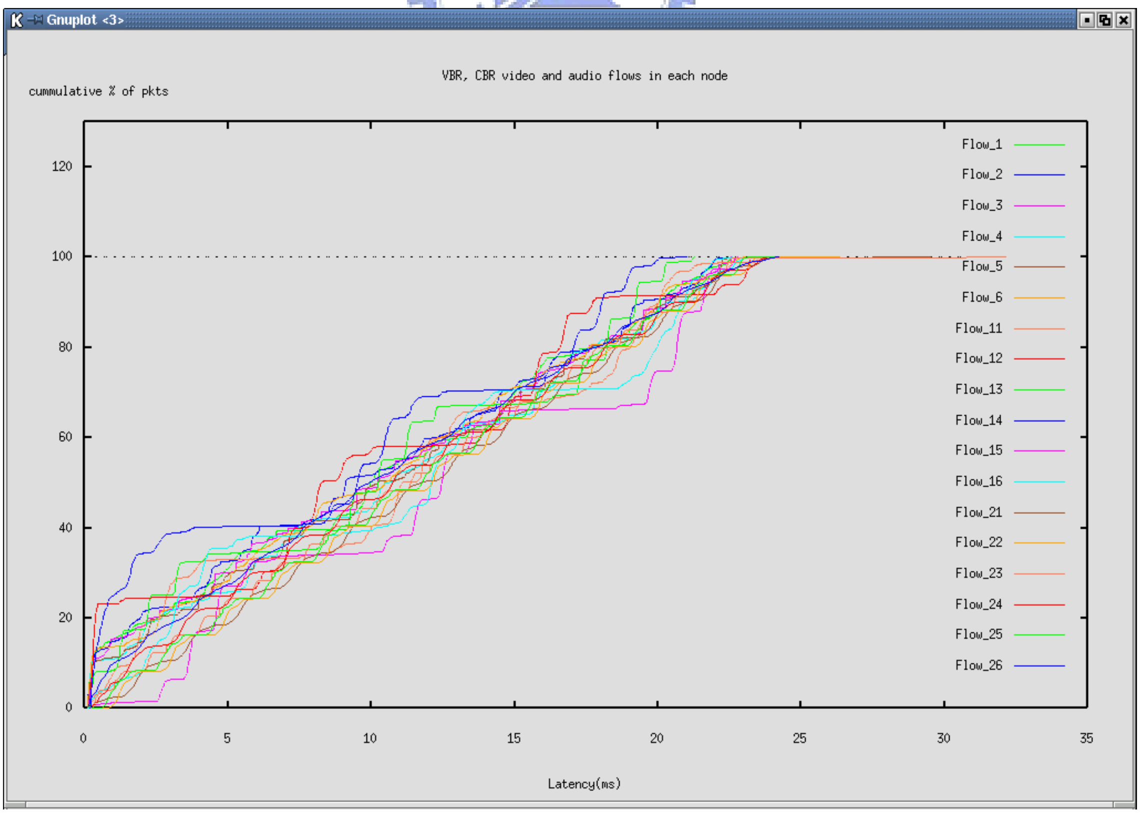

Figure 6.9: C.D.F of packets versus Latency based on proposed scheme

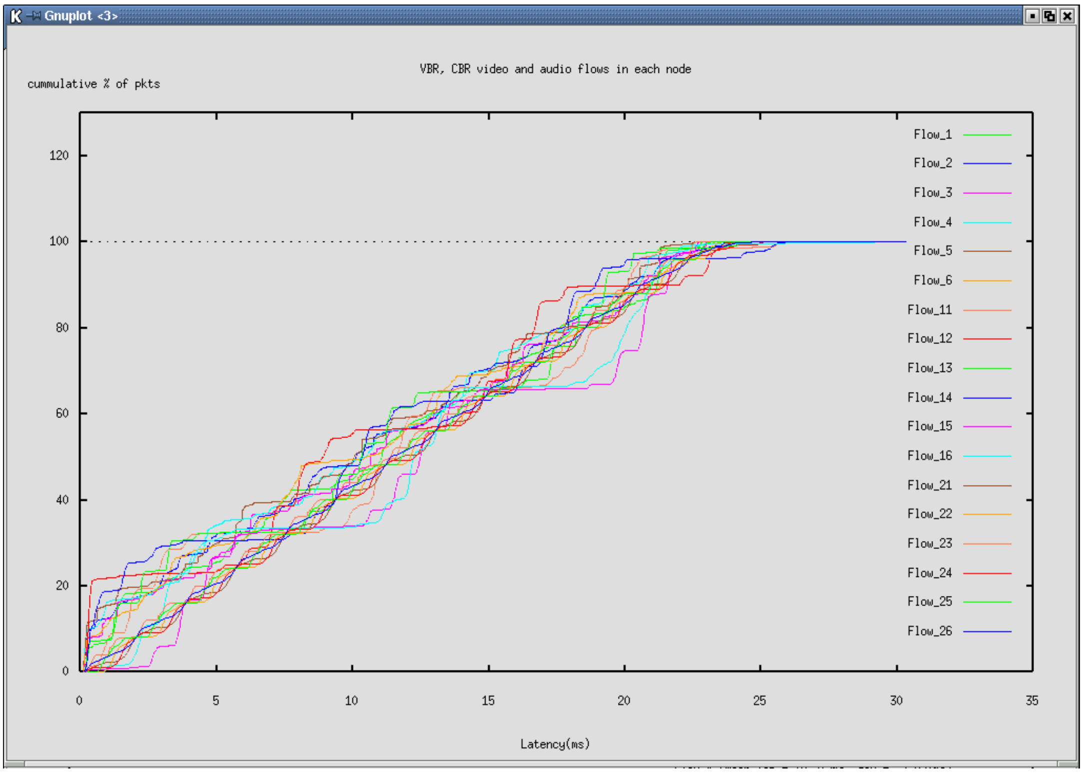

Figure 6.10: C.D.F of packets versus Latency based on FHCF scheme

The CFP load is about 70 % in this scenario. We can discover that all the schemes can provide fairly good performance for CBR traffic. However, the referenced scheme can not control the delay for the VBR traffic trivially. On the contrary, both the proposed scheme and the FHCF scheme can provide a controlled average delay for VBR and CBR traffic that is fit for the requirement of latency. The results also obey the analysis in chapter 5 in the stable state of simulation. Because the inter-arrival time is fixed and packets arrive regularly, the mean delay of VBR traffic is a little bit different from the analysis. For the comparison of proposed scheme and FHCF scheme, even in the long term average the performance is almost the same, sometimes the FHCF scheme cannot absorb the excess requiring time since the real estimation error may be above the average and hence it has larger maximum and mean latency in VBR traffic.

	Referenced scheduler			Proposed Scheme			FHCF Scheme		
Flow ID	Peak Latency (ms)	Mean Latency (ms)	Total Bandwidth (KB/s)	Peak Latency (ms)	Mean Latency (ms)	Total Bandwidth (KB/s)	Peak Latency (ms)	Mean Latency (ms)	Total Bandwidth (KB/s)
1	22.559	10.521	3.687	24.433	10.278	2.742	22.889	10.737	3.502
2	22.466	10.675	2.531	24.578	9.966	2.878	23.667	10.301	3.446
3	22.414	9.885	2.816	22.561	10.321	1.950	22.971	10.757	3.554
4	22.593	10.884	5.093	23.111	10.749	2.897	23.130	10.232	2.941
5	23.396	10.809	3.065	22.586	10.624	4.268	22.570	10.161	2.375
6	22.728	9.596	4.381	23.088	10.302	3.429	23.274	10.382	3.130
Mean	**22.693**	**10.395**	**3.5955**	**23.393**	**10.373**	**3.027**	**23.034**	**10.428**	**3.158**
11	1377.141	652.295	26.705	24.156	10.854	28.363	27.213	11.088	28.363
12	973.627	290.791	23.722	23.375	9.881	19.259	25.102	10.263	19.259
13	1117.849	478.555	24.575	21.272	10.020	23.791	24.232	10.452	23.791
14	606.745	190.543	25.464	21.035	8.707	26.076	26.346	10.314	26.076
15	328.296	72.229	24.382	22.833	12.440	22.158	22.738	12.525	22.158
16	553.956	142.823	25.136	22.595	11.128	29.917	29.182	12.044	29.917
Mean	**826.629**	**304.539**	**24.997**	**22.544**	**10.505**	**24.927**	**25.802**	**11.101**	**24.927**
21	45.321	11.863	390.642	29.553	11.898	390.658	29.380	11.945	390.655
22	48.869	12.314	390.639	26.383	11.851	390.633	27.819	11.845	390.630
23	57.710	12.703	390.630	32.150	11.295	390.641	24.548	11.324	390.639
24	55.949	15.118	390.632	23.865	11.099	390.628	28.512	11.924	390.625
25	68.691	12.915	390.658	24.107	11.543	390.637	25.183	11.606	390.635
26	64.366	13.877	390.628	24.262	10.504	390.650	30.348	11.687	390.647
Mean	**56.818**	**13.132**	**390.683**	**26.720**	**11.365**	**390.641**	**27.632**	**11.722**	**390.639**

Table 6.4: Simulation Static of referenced scheduler, proposed scheme and FHCF scheme

6.2.2 Performance versus CFP Load Condition

application	Inter-arrival time (ms)	Nominal MSDU Size (bytes)	Mean Data Rate (kb/s)	Maximum Service Interval (ms)	Peak Data Rate (kb/s)	Traffic Stream/ User Priority
audio	4.7	160	64	25	64	6
VBR video	~26	~660	~200	50	500	5
CBR video	2	600 →1250	2400 →5000	50	2400 →5000	4

Table 6.5: Parameters of traffic in our simulation

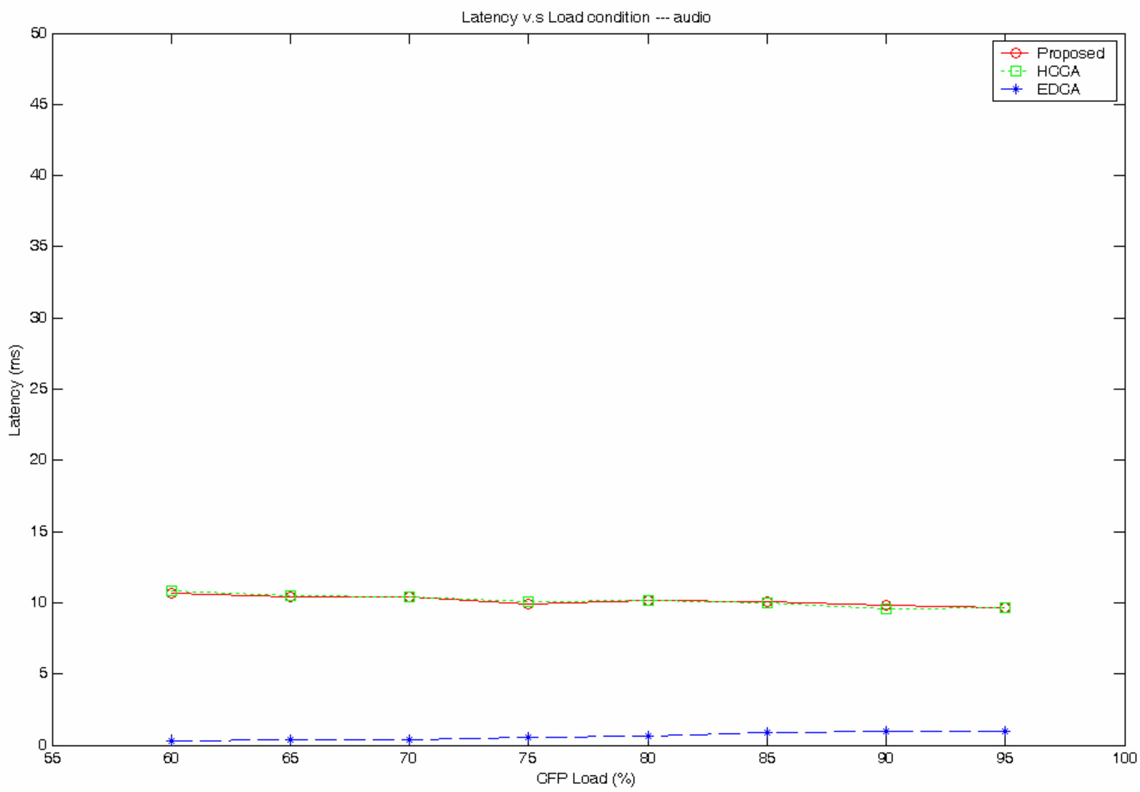

Figure 6.11: Mean delay of the audio flows versus CFP load

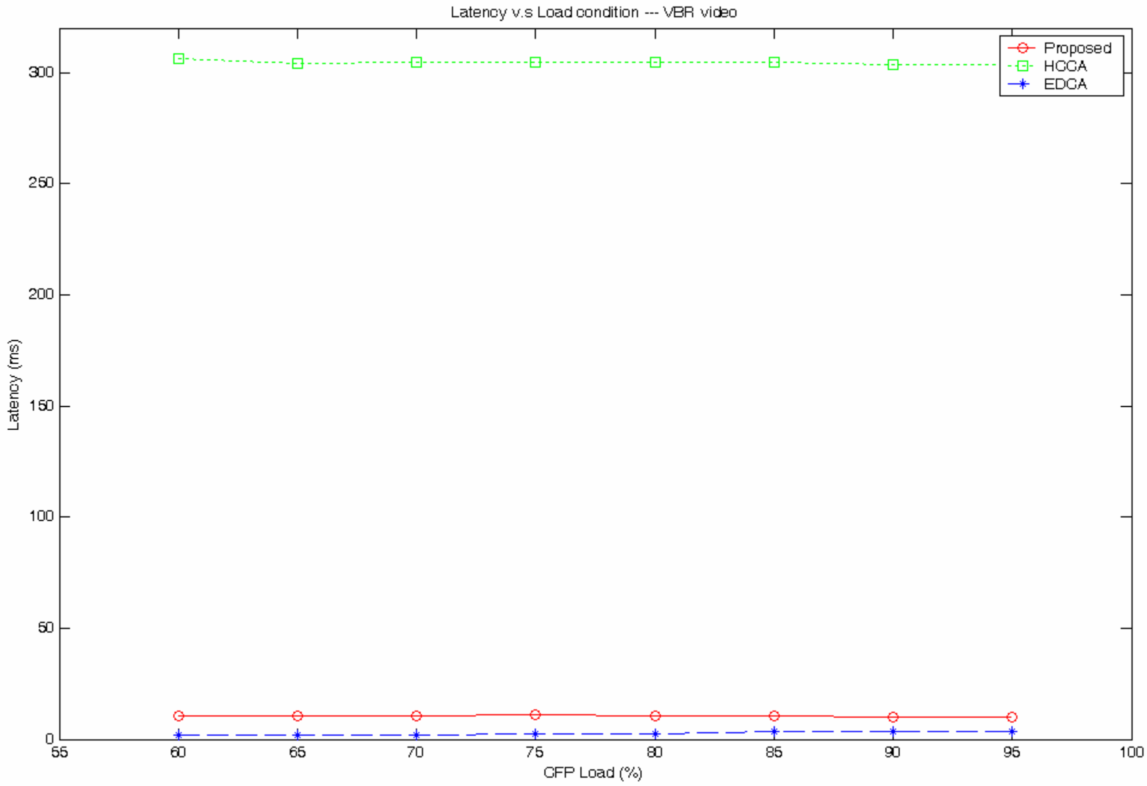

Figure 6.12: Mean delay of the VBR video flows versus CFP load

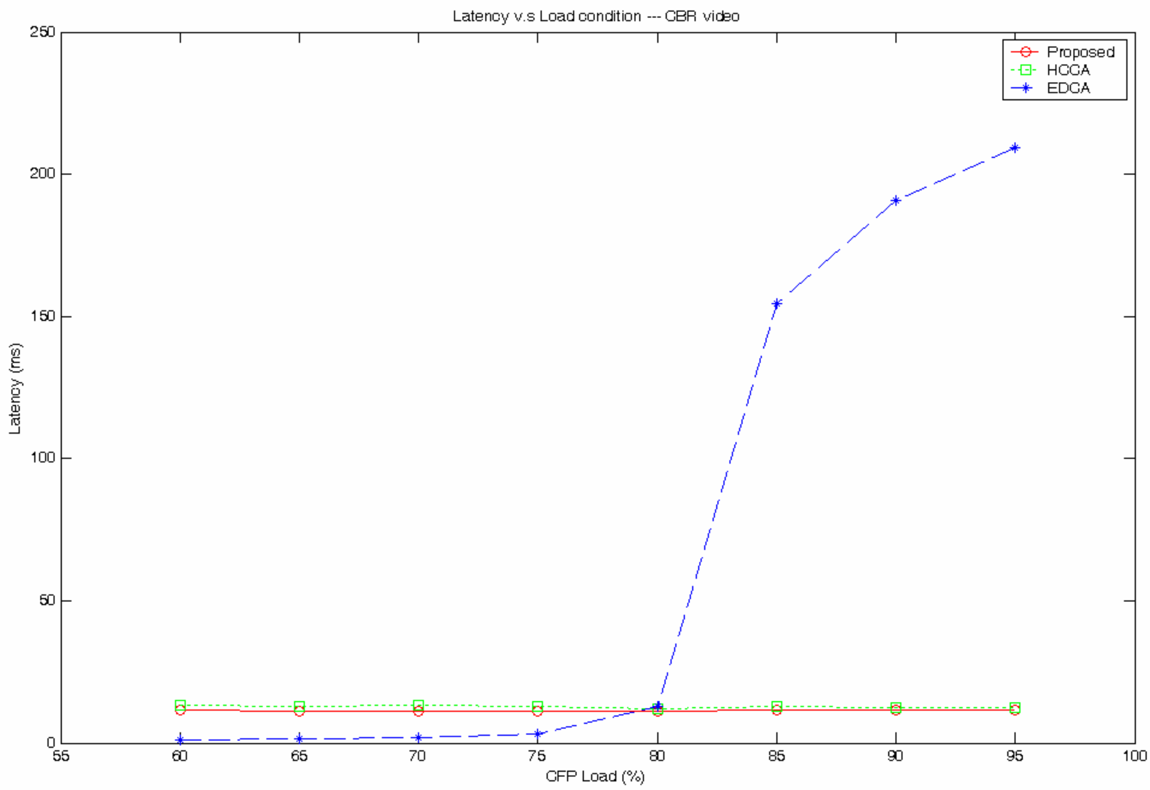

Figure 6.13: Mean delay of the CBR video flows versus CFP load

		EDCA			HCCA			Proposed Scheme		
Load (%)	Latency (ms)	audio	VBR	CBR	audio	VBR	CBR	audio	VBR	CBR
60	mean	0.317	1.690	1.174	10.845	306.219	13.305	10.665	10.255	11.473
	max	6.398	11.162	13.544	23.506	1377.284	66.851	25.590	24.790	27.484
	drop	0/0	0/0	0/0	0/0	0/0	0/0	0/0	0/0	0/0
65	mean	0.356	1.764	1.324	10.476	304.124	12.974	10.435	10.582	11.379
	max	4.103	10.643	23.113	23.809	1377.141	66.688	24.013	24.197	26.788
	drop	0/0	0/0	0/0	0/0	0/0	0/0	0/0	0/0	0/0
70	mean	0.381	2.025	1.764	10.395	304.539	13.132	10.373	10.505	11.365
	max	4.897	15.543	45.599	23.396	1377.141	68.691	24.578	24.156	32.150
	drop	0/0	0/0	0/0	0/0	0/0	0/0	0/0	0/0	0/0
75	mean	0.544	2.214	3.180	10.045	304.813	12.894	9.937	10.889	11.266
	max	7.746	12.292	98.068	23.458	1377.141	68.543	24.595	23.98	37.750
	drop	0/0	0/0	0/0	0/0	0/0	0/0	0/0	0/0	0/0
80	mean	0.665	2.533	12.941	10.148	304.718	12.156	10.162	10.683	11.356
	max	7.591	23.876	218.148	23.225	1377.141	70.199	22.940	23.420	54.387
	drop	0/0	0/0	2/1	0/0	0/0	0/0	0/0	0/0	0/0
85	mean	0.855	3.257	154.641	9.966	304.491	12.649	10.091	10.394	11.436
	max	11.669	20.633	294.782	22.297	1377.141	67.475	35.047	23.903	62.461
	drop	0/0	0/0	362/1	0/0	0/0	0/0	0/0	0/0	0/0
90	mean	0.986	3.328	190.644	9.603	303.331	12.402	9.781	9.939	11.529
	max	12.049	19.981	298.159	23.104	1377.141	59.559	34.101	23.392	89.142
	drop	0/0	0/0	802/1	0/0	0/0	0/0	0/0	0/0	0/0
95	mean	0.945	3.421	209.108	9.684	303.53	12.643	9.692	10.118	11.393
	max	9.489	24.255	331.972	38.332	1377.141	58.986	35.110	10.458	95.122
	drop	0/0	0/0	1096/1	0/0	0/0	0/0	0/0	0/0	0/0

Table 6.6: Simulation Static of EDCA, referenced scheduler, and proposed scheme

When testing the EDCA scheme, we let the entire simulation time be reserved for contention-based medium access. We can discover that the central-coordinated scheme can provide much more stable delay control for all load conditions compared with distributed-coordinated scheme. When the medium is not very congested, the EDCA scheme achieves the best performance and has very short delay for all the traffic. However, as the load is getting heavier, the delay increases rapidly and the lowest priority traffic suffers more packet drops. We also simulate the proposed scheme under different values of α, like 2^{-2}, and get similar results. Therefore, we can know that the proposed scheme can provide controlled delay and low packet drops under the upper bound of CFP load.

Chapter 7

Conclusions

Aiming to provide bounded delay for both VBR and CBR traffic, we present a data rate estimation algorithm and a simple queue-length-based weighted compensating time allocation. From the performance evaluation, our scheme is good for both VBR and CBR traffic and is stable for various load conditions comparing with the referenced scheduling scheme and the contention-based access method.

Comparing with FHCF scheme, we achieve equivalent performance in the same simulation scenario using, however, a scheme with lower complexity. Besides, our scheme has lower mean delay for both VBR and CBR traffics revealed in data and figures of simulation. Regarding the FHCF scheme, since it uses an average of estimation error to do the adjustment of queue length estimation, when the VBR traffic fluctuates a lot around the mean value , like the traffic with bigger variance in data rate, it may not absorb the change that is much higher than average. As for our proposed scheme, we can successfully track the variation of data rate if the short time data rate variation is smooth. Besides, with the compensating method, we can easily remedy the estimation error to keep the good performance.

Therefore, we provide a simple but efficient way to estimate fluctuating data rate and provide delay, packet loss rate, and throughput guarantee in error-free wireless circumstance. If the channel condition is varying with time, our scheme needs to combine other techniques to lower the BER, like link adaptation. Finally, the polling-based scheme is much more suitable to operate under interference-free circumstance, while contention-based scheme can still operate normally when there are other WLANs within the range.

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