國 立 交 通 大 學

電子工程學系電子研究所碩士班

碩 士 論 文

在 IEEE 802.11e EDCA無線區域網路下以服務品質導向的服務允許控制演算法

QoS Based Call Admission Control in 802.11e Enhanced Distributed Coordinate Access Communication System

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摘要

最近數年之間無線網路(IEEE 802.11) 已迅速在世界各地布建.但是無線網路的服務品質並未在最初的規範中被提及.雖然 IEEE 以在著手訂定 IEEE 802.11e 來嘗試提升無線網路在服務品質控制上的能力.不過如何滿足不同的服務需求仍然是一個問題.在眾多常見的服務當中. 即時通訊服務例如 VoIP 對於服務品質的要求更是嚴苛.本論文主要研究目的在於從服務允許控制演算法的角度來確保在無線網路的即時通訊服務能夠保持在令人滿意的程度. 在本論文中提出一個根據 IEEE 802.11e EDCA 無線網路架構的服務允許控制演算法. 這個演算法的目的有二.一為確定系統仍然可以滿足一個新即時通訊服務的服務要求.二為保證已存在這個系統中的即時通訊服務其服務品質仍然維持在令人滿意的範圍.除了對於即時通訊服務作服務允許控制外,一個針對非即時通訊系統的頻寬維持控制也在這篇論文中被提出來控制非即時通訊服務對於即時通訊服務和服務允許控制演算法的影響.

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Abstract

How to fulfill the QoS of different kinds of services is getting more and more important with the increasing demands of WLAN system. Although different applications have different priority settings under the definition of IEEE 802.11e spec, the contention process still disturbs hard QoS guarantee in EDCA. In this thesis, Virtual traffic source estimation (VTSE) algorithm is proposed to judge the access request of a new real-time service by estimating the impacts of the new real-time service and the system loading. By using VTSE, WLAN system can protect real-time services' QoS effectively. A bandwidth reservation algorithm is also proposed to implement with VTSE to prevent the impacts of non-real-time services.

誌謝

碩士班的時光匆匆，似乎又到了離別的時刻。在人生最精華的這段歲月裡，感謝上天讓我有這個緣分遇見你們，陪我一起度過這段難忘的時光。

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Chapter 1
Introduction

IEEE 802.11[1] is a specification that defines the medium access control (MAC) layer and the physical (PHY) layer. IEEE 802.11 standard defines two channel access mechanisms. One is the mandatory contention based distributed coordinate function (DCF) and another is the optional polling based point coordinate function (PCF). Based on the spread spectrum technique, the transmission rate can achieve up to 2 Mbps.

After the release of IEEE 802.11 standard, there are many related specifications are proposed to increase the system performance. In order to increase the data rate in the network, IEEE has released IEEE 802.11a/b/g specification to modify the mechanism in the PHY layer. The utilization of orthogonal frequency division multiplexing (OFDM) in the 802.11a/g increases the packet transmission rate up to 54 Mbps.

Besides the high transmission rates, quality of service (QoS) is also an important topic in the WLAN system. Different services have different requirements on the QoS. Services such as VoIP or video streaming require steady bandwidth and small delay and jitter. On the other hand, Non-real-time services such as FTP service are not sensitive to the delay but require low packet error rate. To resolve QoS problems, IEEE constructs the 802.11e standard [2] to differentiate service priorities and to utilize network resources more effectively.

An enhanced distributed coordinate Access (EDCA) is proposed in [2] to improve the traditional DCF, There are many enhanced features to support QoS control under the original contention process. The basic concept of EDCA is to provide service differentiation in WLAN system by changing different transmission parameters to prioritize applications.

Although [2] is proposed to enhance the ability of QoS control, it still has not mentioned how to fulfill requirements of different services. In general, there are three different approaches to analyze QoS control mechanism: (1) Service differentiation (2) Call admission control (3) Bandwidth reservation.

The concept of service differentiation is to prioritize different kinds of applications in the network and preserve resources for high priority services. EDCA is a typical service priority algorithm. In EDCA, different transmission parameters are assigned to services of different priorities. In [6], transmission parameters changes adaptively with the packet collision rates. [7] differentiates the service priority by adjusting backoff process. In [8], the authors propose a Distributed fair scheduling (DFS) algorithm to achieve fairness between service priority and packet length. In [9], size of contention window of is adjusted according to their actual throughput and expected throughput. [10] calculates an idle period of a service which it has to wait before the contention process and the decision rule of idle period is based on their achieved throughput.

The function of a call admission control is to decide whether a new service can get the access of channel or not. WLAN system needs the call admission control because the system performance such as the packet collision rate and packet delay would be degraded seriously when the system loading is high. There are two types of call admission control. A measurement-based call admission control makes decision based on the measurements of existing condition such as the collision rate of packet transmission or system throughput [11-12]. Calculation-based schemes construct performance criteria for evaluating the status of the network and predict the QoS of services. For example, the calculation of the available system throughput is an indication to trigger call admission controls in [13-14].

Bandwidth reservation is to reserve resources to fulfill requirements of different applications. In [15], an algorithm for the AP and mobile host to negotiate the minimum required bandwidth is proposed. [16] reserves the minimum requesting service bandwidth by adjusting service priority. In [17], it defines a set of reservation-based algorithm to solve a hidden point problem in a multi-channel system.

In this thesis, a new call admission control based on the IEEE 802.11e EDCA is proposed. When a new real-time service requests to access the system, a virtual traffic source estimation (VTSE) algorithm will be used to estimate the impacts from admitting the new service. Besides the virtual traffic source estimation algorithm, a bandwidth reservation is also applied to reserve resources for real-time services. The goal of the proposed bandwidth reservation is to prevent the QoS of real-time services from being destroyed by unstable traffic loading.

The rest of the thesis is organized as follows: Chapter 2 introduces the existing IEEE 802.11 and 802.11e systems. Chapter 3 describes the QoS requirements of different services and related prior works on WLAN QoS controls. Chapter 4 discusses the proposed virtual traffic source estimation (VTSE) algorithm and bandwidth reservation algorithms. Chapter 5 provides the simulation platform and the simulation results. Finally, a conclusion is drawn in chapter 6.

Chapter 2

Overview of IEEE 802.11 system

This chapter describes the MAC layer mechanisms accepted in the IEEE 802.11 and IEEE 802.11e. Section 2.1 is the introduction of the mandatory channel access mechanism in [1], which is called distributed coordinate function (DCF). The Enhanced DCF, EDCA are described in Section 2.2. Section 2.3 describes the parameters and settings used in PHY layer.

2.1 802.11 MAC Layer

Medium access control (MAC) layer is the kernel to process packet transmission. It handles the packet contention, acknowledgement, retransmission, and etc. IEEE 802.11 accepts two different medium access control mechanisms: the mandatory contention-based distributed coordinate function (DCF) and the optional polling-based point coordinate function (PCF).

2.1.1 Distributed Coordinate Function (DCF)

DCF works based on carrier sense multiple access with collision avoidance (CSMA/CA) mechanism. In CSMA mode, stations which want to access the channel have to listen to the channel condition. Stations stay in idle mode when the channel is occupied. Otherwise, they will start to contend for the channel. But collisions will happen when there are more than one station transmit their packets at the same time. There are two different packet transmission mechanisms standardized in [1]. They are basic assess mechanism and RTS/CTS mechanism.

2.1.1.1 Basic Access Mechanism

First, there are different interframe spaces (IFS) used in the basic access mechanism. When the channel becomes idle, stations have to wait for at least an IFS period before it starts the contention procedure. There are four types of IFS for stations to operate. They are the short IFS (SIFS), DCF IFS (DIFS), PCF IFS (PIFS), and the extended IFS (EIFS). The shortest SIFS is selected when the transmitted packet is a control frame. The PCF IFS is used for AP in the network to start point coordinate function in the network. DIFS is used for stations that need to transmit data packets in DCF. The extended IFS is defined for a station which receives an error ACK frame.

Fig. 2.1 shows scenarios that different stations contend for the channel in the DCF mode. In the basic access mechanism, every station with data packets has to monitor the channel. If stations detect that the air medium keeps idle longer than a period of time called distributed interframe space (DIFS), MAC layer starts backoff procedure by decreasing a counter called a backoff timer. The backoff timer is decreased when channel stays idle. Backoff timer is frozen when a packet transmission of other station is detected by the station. Backoff timer will reactivate when channel is idle again for more than a DIFS. Stations can begin to transmit a packet only if channel keeps idle and backoff timer of the station decreases to zero. In Fig. 2.1, stations halt their backoff timer when user *i* is transmitting packet. Backoff timer of stations will be reactivated again when user *i* finished its transmission and channel is in idle state longer than a DIFS. Station which has the shortest backoff timer will get the access to transmit packet.

DCF adopts an exponential backoff procedure to avoid collisions. A backoff timer is selected randomly between (0, ω-1), ω is a value called contention window. For example, the value of backoff timer T_b will be

$$T_b = SIFS + C_a * slottime \ \ (us) \tag{2-1}$$

when a value C_a is randomly selected from (0, ω-1). Slottime is a parameter defined in [1].

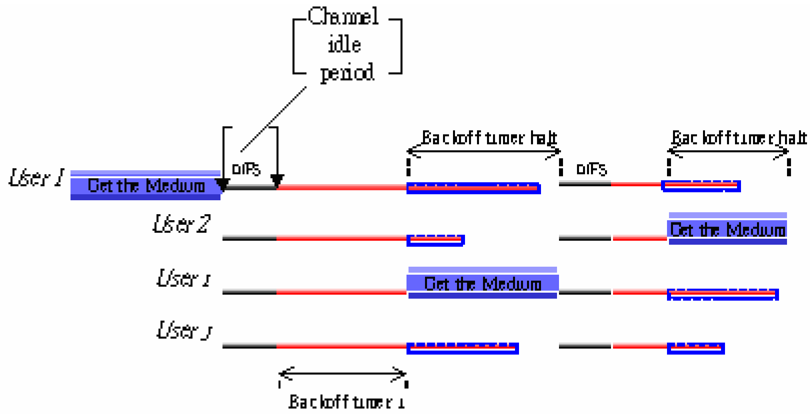

Fig. 2. 1 Station contention process in IEEE 802.11 standard. User i's backoff timer decreases to zero first and begins to transmit packets

The Value of contention window depends on the number of fail transmission attempts. The initial value of ω is set to a value CW_{min} called the minimum contention window. When a collision happens, the collided station doubles the value of contention window before reaching CWmax. CWmax is called the maximum contention window. The Contention window will be reset to CWmin when the station transmits a packet successfully. So, function of the contention window is equal to

$$\omega(n) = \min\left(2^{n} * (CW_{\min} + 1) - 1,\ CW_{\max}\right) \qquad (2\text{-}2)$$

n is number of successive fail attempts of the transmitter station that has tried to transmit the waiting packet.

Fig. 2.2 depicts the timing schedule of packet transmission in basic access mechanism. A transmission period under basic access mechanism is composed by the following time intervals: DIFS deferral time, backoff time if necessary, data transmission, SIFS deferral time, and ACK transmission. The schedule of a successful packet transmission is illustrated in Fig. 2.2(a). When the destination station receives a packet successfully, it will transmit an acknowledgement frame (ACK frame) to the transmitter station after a short interframe space (SIFS). Fig. 2.2(b) is the timing schedule if no ACK frame is received after an SIFS interval. It is possibly due to collision or an erroneous reception of the data packet, i.e. received with an incorrect frame check sequence (FCS). At this time, transmitter will contend again for the medium by retransmitting the packet after an ACK_Timeout period. Sometimes the transmitter station receives an error ACK frame. An error ACK frame means the received ACK frame is decoded in error. The transmitter station will retransmit the packet after an EIFS period when it receives an error ACK frame, as shown in Fig. 2.2(c).

The packet will be dropped eventually if the number of attempts for a station to retransmit the packet has reached an retry limit. The retry limit for a station will be set by the system designer.

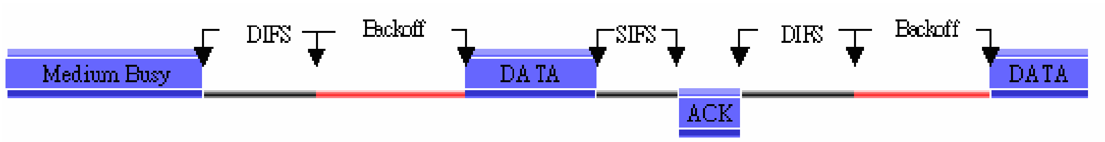

Fig. 2.2 (a)

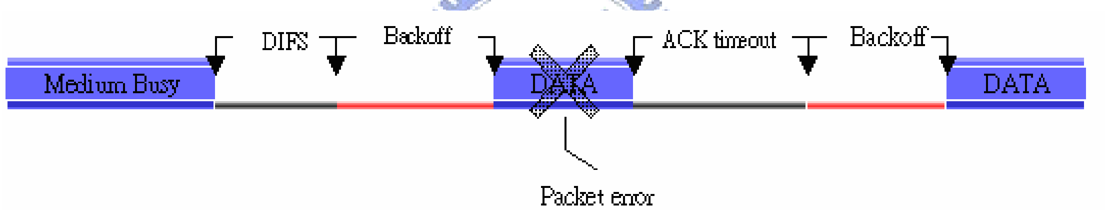

Fig. 2.2 (b)

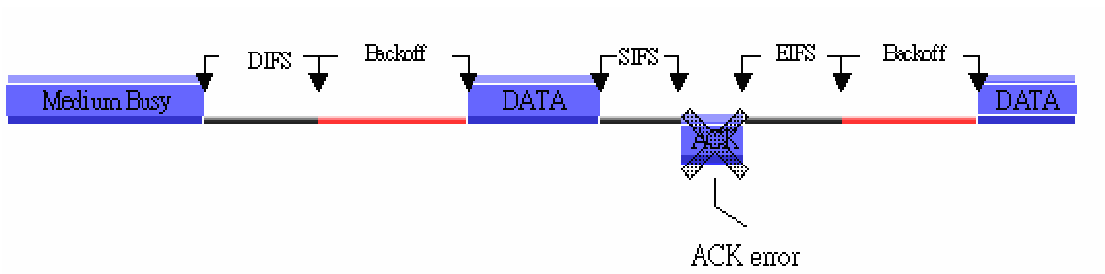

Fig. 2-2 (c)

Fig. 2. 2 Process of packet transmission in IEEE 802.11 basic access mechanism

(a)A success Packet transmission.(b) A collided Packet transmission or erroneous reception of the data packet.(c) Packet transmission success but the transmitter station receives an error ACK frame

When a packet is transmitting, other stations in the network can hear the packet transmission and get the information about the duration of this transmission based on the duration field value contained in the packet. An estimated packet transmission period is attached in the duration field when the packet is sent from the transmitter station. All the other stations hearing the packet adjust their network allocation vector (NAV) based on the estimated packet transmission period. NAV is used for virtual carrier sensing at the MAC layer to indicate the period of time in which the channel will remain busy. Stations would not try to detect channel state during this time period because channel is assumed to be busy.

2.1.1.2 RTS/CTS Mechanism

There is another particular feature of wireless local-area networks (LANs), known as "hidden node" problem. Two stations that are not within hearing distance of each other can lead to collisions at a third node which receives transmission from both sources. So DCF implements RTS/CTS mechanism in contention process to solve this problem. First, when the medium's idle period is longer than the DIFS period and the backoff timer reaches zero, station that wants to use the medium transmit a request-to-send (RTS) frame to the destination station to acquaint that a packet transmission is ready; After receiving RTS frame, the destination station will transmit a clear-to-send (CTS) frame after a SIFS to inform the transmitter station that the destination station is ready to receive the packet. The transmitter station begins a transmission after a SIFS when it receives CTS frame successfully. Otherwise, the transmitter station starts a retransmission procedure when a RTS frame is sent but receives no CTS frame after a "CTS timeout" period. Sometimes the transmitter station may also receive an error CTS frame. An error CTS frame means that it has been decoded in error. When the transmitter station receives an error CTS frame, the transmitter station also begins retransmission procedure after an EIFS period.

Fig. 2.3 shows the transmission procedure in the RTS/CTS mechanism. A transmission period under RTS/CTS access mechanism consists of the following phases: DIFS deferral time, backoff time, RTS transmission, SIFS deferral time, CTS transmission, SIFS deferral time, data transmission, SIFS deferral time, and ACK transmission. The timing schedule of successful packet transmission is illustrated in Fig. 2.3(a). On the other hand, the condition that the transmitter station does not receive the CTS frame, which is possibly because of collision or an erroneous reception of the RTS frame, is shown in Fig. 2.3(b).

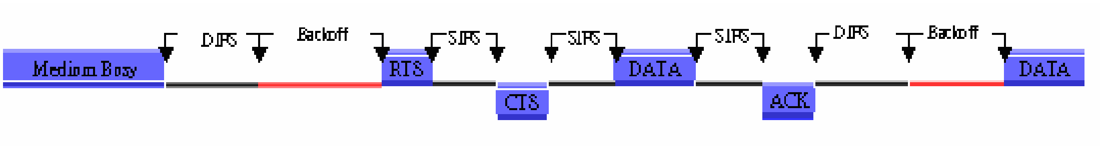

Fig. 2.3 (a)

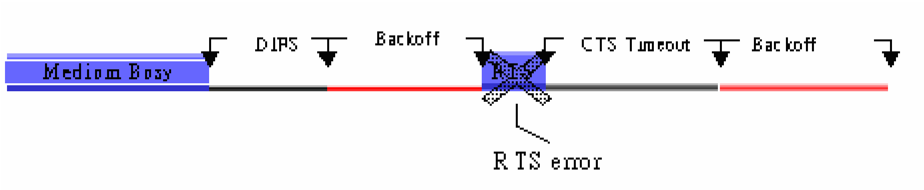

Fig. 2.3 (b)

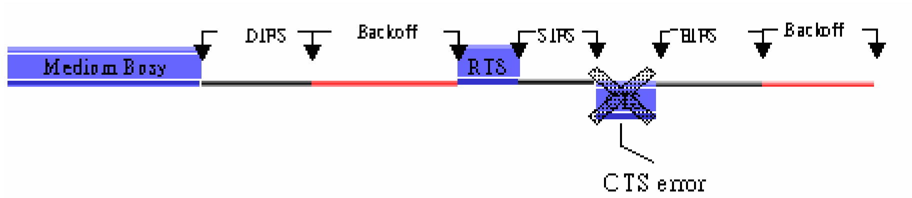

Fig. 2.3 (c)

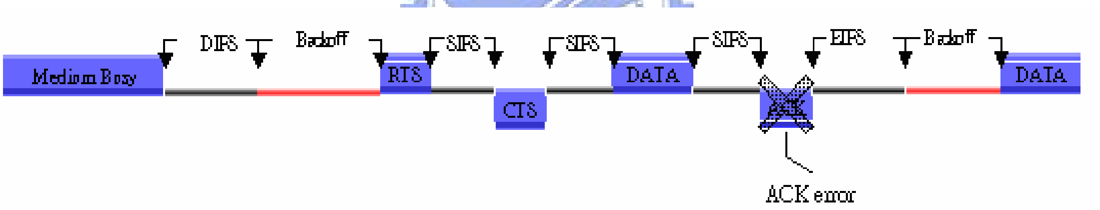

Fig. 2.3 (d)

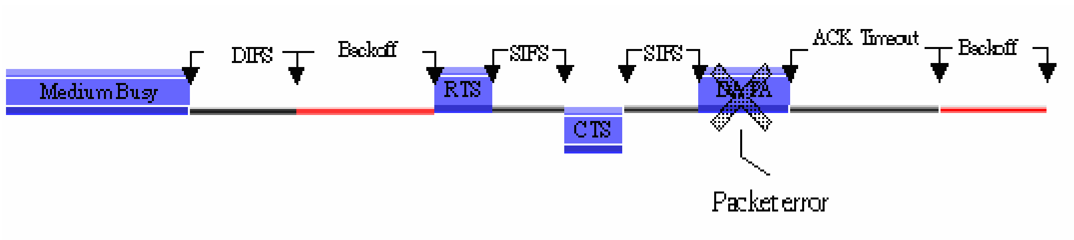

Fig. 2.3 (e)

Fig. 2. 3 Process of packet transmission in the IEEE 802.11 RTS/CTS access mechanism

(a)A successful packet transmission. (b) RTS frame error or collision. (C) CTS frame error. (d) ACK frame error. (e) Packet transmission error.

If CTS frame or ACK frame is received in error, the transmitter station will contend for the medium again after an EIFS interval, as shown in Fig. 2.3(c) and Fig. 2.3(d). If no ACK frame is received after a SIFS and ACK_Timeout period when the transmitter station has transmitted data packet. It is possibly because of an erroneous reception of the data packet. At this time, the transmitter station will contend for the medium again. Timing of this situation is plotted in Fig. 2.3(e).

The RTS and CTS frames carry information of the duration field of this packet which is contained in the data packet based on the definition of basic assess mechanism. This information can be read by any listening station so all of them are able to update their NAV. Therefore, when a station is hidden from either the transmitting or receiving station, it can avoid collision by detecting just one frame among RTS and CTS frames because the hidden station knows that channel is busy during this time period. So RTS/CTS mechanism can guarantee an undisturbed transmission for longer data packet.

2.2 IEEE 802.11e EDCA

EDCA mechanism is an enhanced DCF for supporting service differentiation. In IEEE 802.11e standard, Service prioritization is achieved through eight prioritized "access categories" (ACs). Packets are classified into eight "traffic categories" (TCs). Stations decide priorities of those TCs by mapping to one of the access categories. Among eight ACs, four ACs are reserved for EDCA TCs, so one or more TCs may be mapped to one AC.

Priority differentiation between each TC is achieved by varying the value of CWmin, CWmax, and IFS. For the *i*th TC (TC[i] i=0,1, 2,…,7), the CWmin, CWmax, and AIFS are marked as CWmin[i], CWmax[i], and AIFS[i]. The length of AIFS is set as

$$AIFS = SIFS + AIFS[i] * slottime\ (us) \tag{2-3}$$

Another new feature in IEEE 802.11e standard is the definition of "Transmission Opportunity" (TXOP). TXOP is the period of time that a station has the right to access the medium after a successful contention, with maximum duration defined in TXOPLimit[AC]. An EDCA-TXOP is obtained through contention process. A transmitter station is allowed to transmit packets continuously after waiting a SIFS following a successful completion of a packet exchange sequence, while the total transmission time does not exceed TXOPLimit[AC].

Fig. 2.4 shows different AIFS values of every access categories. Low priority service has a longer backoff timer as compared to high priority services. The contention window of low priority service is larger compared with that of high priority services. The assumption of contention window is also shown in Table2.1.

Table 2.1 lists parameters of different service priorities defined in [2] and Table 2.2 is service priority of different services. Table 2.1 shows that service of high priority contends for channel with short AIFS and CW and EDCA-TXOP of high priority is longer than that of low priority service. So QoS of high priority service can be better than that of low priority service.

Fig. 2.5 shows a EDCA buffer structure of the MAC layer implemented in a station. Under the definition of [2], packets of different priorities from the upper layer are put into different buffers according to their traffic categories. Every buffer has its own backoff timer. Backoff timers of each traffic category start when the medium keeps idle longer than AIFS[i] period. So, every TC in a station contends for the chance to transmit packet just like what a station does in the DCF mechanism. In a station, two or more ACs may try to send packets at the same time. This means collision happens in a station and this phenomenon is called "virtual collision", which means not a really collision. In this situation, packet of the higher AC always gets the right to transmit packet. All of these contention and virtual collision processes will be implemented in the EDCA scheduler.

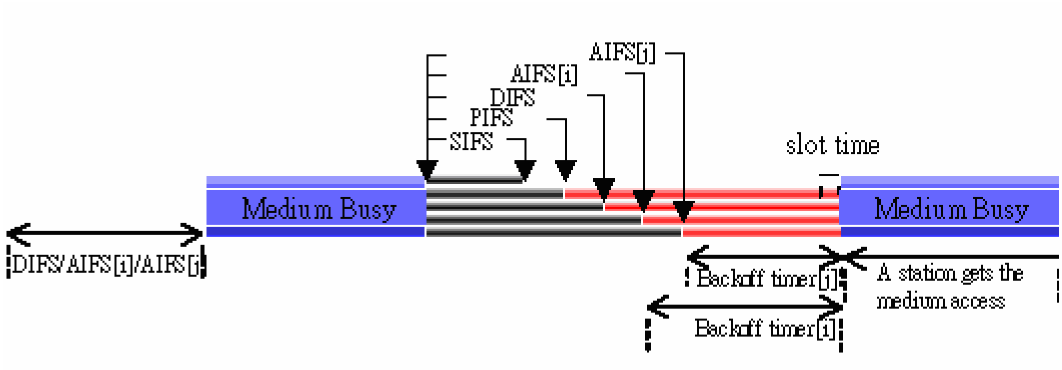

Fig. 2. 4 EDCA mechanism. Different AIFS value are set according to the packet ' s priority

Table2. 1 Default values of different ACs. aCWmin=15 and aCWmax=1023

AC [i]	*CWmin*	*CWmax*	*AIFS*	*TXOP Limit(ms)*
0(best effort)	***aCWmin***	***aCWmax***	***2***	***0***
1(excellent effort)	***aCWmin***	***aCWmax***	***1***	***1.5***
2(Video)	***Floor(aCWmin/2)***	***aCWmin***	***1***	***3***
3(VoIP)	***Floor(aCWmin/4)***	***Floor(aCWmin/2)***	***1***	***1.5***

Table2. 2 Access categories defined in the IEEE 802.11

Service priority	**Traffic class**	**Targeted services**
7	Network Control	Network control service has critical requirements. Although packet size of network control service is very low, but the delay bound and reliability are very important.
6	Voice	Voice service is very sensitive to delay and jitter.
5	Video	QoS requirement of video streaming services are not as critical as voice service. The maximum 100 ms one-way delay is tolerable.
4	Controlled load	Controlled load service is assigned to important applications accessing the LAN.
3	Excellent effort	Excellent service is intended to be better than best-effort service.
2	Best-effort	Best effort service is currently the most utilized traffic class. No delay or throughput guarantee to this kind of service.
1	Not specified	
0	Background	Background service is for applications such as bulk transfer of large files. This kind of service can exist in the background of a LAN.

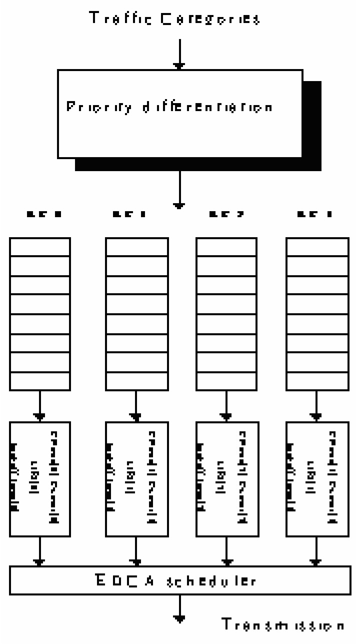

Fig. 2. 5 EDCA buffer structure of the MAC layer implemented in a station

2.3 802.11 PHY Layer

Physical layer defined in IEEE 802.11g [18] is briefly described in this section. Table 2.3 lists parameters that would be used to estimate packet transmission period. In IEEE 802.11 MAC layer, each MAC data frame, or MPDU (MAC protocol data unit), are composed by the following components: MAC header, variable length information frame body, which is called MSDU (MAC service data unit), and frame check sequence (FCS). MAC layer overhead, including MAC header and FCS, is 28 bytes in length. Besides, the information frame body can be up to 2312 bytes when encryption is applied. Based on the definition of [1], data fields of a RTS, CTS and ACK frame is 20 bytes, 14 bytes and 14 bytes, respectively.

Except MAC header, PHY layer also adds its control field to packets. A period of signal preamble and PHY header, which are transmitted with the BPSK modulation and rate-1/2 convolutional coding, takes 20 us to transmit in total. There are also six zero tails bits which is used to make the channel decoder returns to the zero and the pad bits are used to make the resulting bit string into a multiple of OFDM symbols. A 16-bits SERVICE field in the header is transmitted with the same transmission rate of data field. Each OFDM symbol interval, which is labeled T_{SI}, is 4 us. How many data bytes are contained in an OFDM symbol is decided by the modulation and coding scheme.

When the data field are transmitted at a supported data rate, all the control frame, including RTS, CTS, and ACK frame, have to be transmitted at one of the basic rate set{6Mbps, 12 Mbps, 24 Mbps}to make sure that they can be understood by all the stations in the network. In addition, RTS and CTS frame will be transmitted at 6 Mbps while the ACK frame is transmitted at the highest rate in the basic rate set that is less than or equal to the rate of the data packet it is acknowledging.

After the introduction of packet components, the required transmission period is calculated below. First, when the packet transmits B_{data} bytes data payload over IEEE 802.11g PHY using the data rate R_{data}, the transmission period will be

$$T_{data}(B_{data}, R_{data}) = T_{Preamble} + T_{SIGNAL} + \left\lceil \frac{MAC\_Header + Service\_field + tail\_bits + B_{data}}{Data\ bytes\ per\ symbol(R_{data})} \right\rceil * T_{SI} \tag{2-4}$$

Note that the *Data Bytes per Symbol(R_{data})* is shown in Table 2.4. The data bytes per symbol means how many bytes are contained in a symbol which is modulated in the OFDM technique. Similarly, the transmission period of a RTS frame is

$$T_{rts}(B_{data}, R_{data}) = T_{Preamble} + T_{SIGNAL} + \left\lceil \frac{MAC\_Header + Service\_field + tail\_bits + RTS\_Control\_field}{Data\ bytes\ per\ symbol\ (1)} \right\rceil * T_{SI} \tag{2-5}$$

The packet transmission period of CTS frame is calculated in the same way:

$$T_{cts}(B_{data}, R_{data}) = T_{Preamble} + T_{SIGNAL} + \left\lceil \frac{MAC\_Header + Service\_field + tail\_bits + CTS\_Control\_field}{Data\ bytes\ per\ symbol\ (1)} \right\rceil * T_{SI} \tag{2-6}$$

And the transmission duration of an ACK frame using the PHY mode R_{data}* is equal to

$$T_{ack}(D_{data}, R_{data}) = T_{Preamble} + T_{SIGNAL} + \left\lceil \frac{MAC\_Header + Service\_field + tail\_bits + ACK\_Control\_field}{Data\ bytes\ per\ symbol\ (R_{data}*)} \right\rceil * T_{SI} \tag{2-7}$$

From the description above, the transmission period is only related to the MSDU size and the transmission mode in the PHY layer.

Table2. 3 System parameters accepted and parameters used to calculate the packet transmission period

B_{data} (Data length of the data packet)	**0~2312 bytes**
R_{data} (PHY layer transmission rate)	**6~54 Mbps**
R_{data}* (PHY layer transmission rate for ACK frame)	**{6 Mbps,12 Mbps,24 Mbps}**
T_{Slot} (Slot time)	**9** μs
T_{SIFS} (SIFS time)	**16** μs
T_{DIFS} (DIFS time)	**34** μs
T_{SI} (OFDM Symbol interval)	**4** μs
$T_{preamble}$ + T_{SIGNAL} (PHY layer Preamble field & signal field)	**20** μs
MAC_Header + FCS (Mac layer header field and frame check sequence)	**20 bytes**
Service_field (PHY layer service filed)	**2 bytes**
Tail_bits (tail nits for the convolutional codes returns to zero)	**6 bits**
RTS_Control_field(Data field of RTS frame)	**20 bytes**
CTS_Control_field(Data field of CTS frame)	**14 bytes**
ACK_Control_field(Data field of ACK frame)	**14 bytes**
CWmin minimum contention window size	**15**
CWmax maximum contention window size	**1023**

Table2. 4 Modulation and Coding scheme in the 802.11g spec

Mode	***Data rate (Mbits/s)***	***Modulation***	***Coding rate***	***Data bytes per OFDM symbol***	***Data rate for the ACK frame (Mbits/s)***
1	***6***	***BPSK***	***1/2***	***3***	***6***
2	***9***	***BPSK***	***3/4***	***4.5***	***6***
3	***12***	***QPSK***	***1/2***	***6***	***12***
4	***18***	***QPSK***	***3/4***	***9***	***12***
5	***24***	***16-QAM***	***1/2***	***12***	***24***
6	***36***	***16-QAM***	***3/4***	***18***	***24***
7	***48***	***64-QAM***	***2/3***	***24***	***24***
8	***54***	***64-QAM***	***3/4***	***27***	***24***

Chapter 3
Quality of Service and Related Works

In this chapter, QoS controls and prior works are discussed. Section 3.1 introduces the basic concept of QoS. Sections 3.2 describes the related works that how the QoS could be implemented in the WLAN network.

3.1 Definition of QoS

The original attempt of IEEE 802.11 is to extend the wire LAN services in wireless environment. So applications in WLAN are all like web browsing, FTP services, VoIP services, and etc. According to the International Telecommunication Union telecommunication standardization sector (ITU-T), QoS is defined as below [18]:

"Quality of Service is the collective effect of service performances that determine the degree of satisfaction of a user of the service"

Take data transformation for example; the integrity of the file is undoubtedly the most important one in its QoS. But for real-time service such as VoIP or video streaming, they have more stringent requirements than data services because of the sensitivity of human ears and eyes about packet loss. Packet loss means distortion in the speech or video signal and users can only tolerate for a certain amount of distortion. In the WLAN network, packet loss happens more often than the wired network. As a result, lost packets could degrade the service quality.

ITU has proposed an indication of suitable performance requirements for different types of services in the network [19]. A part of the performance targets is shown in Table 3.1. Those requirements are good indicators no matter packets are transmitted in a wired network or wireless network.

Table3. 1 Proposed QoS requirements defined in [13] about some familiar services in the network

Data services					
Application	***Degree of symmetry***	***Typical Data amount***	***QoS parameters and tolerable thresholds***		
			One-way delay	***Delay Jitter***	***PLR***
Web browsing	One-way	~ 15 KB	Preferred<2s Acceptable<4s	N.A.	0
Interactive games	Two-way	< 1 KB	<200 ms	N.A.	0
Bulk data retrieval	One-way	10KB ~ 10MB	Preferred<15 s Acceptable<60s	N.A.	0
Interactive services					
Application	***Degree of symmetry***	***Typical Data amount***	***QoS parameters and tolerable thresholds***		
			One-way delay	***Delay Jitter***	***PLR***
Audio streaming	One-way	128 Kbps	<10s	< 1 ms	< 1%
Video on demand	One-way	480 Kbps	<10s	N.A.	<1%
streaming services					
Application	***Degree of symmetry***	***Typical Data amount***	***QoS parameters and tolerable thresholds***		
			One-way delay	***Delay Jitter***	***PLR***
VoIP	Two-way	64 Kbps	Preferred<150ms Limit 400ms	< 1 ms	< 1%
Video phone	Two-way	384 Kbps	Preferred<150ms Limit 400ms	N.A.	<1%

3.2 Related Work in QoS Management

There are already many studies on how to support QoS in the WLAN. Basically, all related QoS controls can be categorized into three approaches:

- Service differentiation
- Call admission control
- Bandwidth reservation

3.2.1 Service Differentiation

Many studies have addressed on how to achieve service differentiation in the DCF-based access mechanism. A Persistent Factor DCF (P-DCF) is proposed in [7] based on the 802.11 DCF. In this algorithm, each traffic class is associated with a persistent factor and the high priority service has smaller P in this algorithm. Then, a uniformly distributed random number γ is generated in every slot time when the station is decreasing its backoff timer. The station will stop the backoff timer and start to transmit packet only if $\gamma > P$ in the current slot time. So the backoff interval is a geometrically distributed random variable with P and the service of high priority can access the channel easier.

Distributed Fair Scheduling (DFS) is proposed in [8] and [20]. The concept of distributed fair scheduling is to differentiate the backoff interval (BI) based on the packet size and service priority. The length of backoff interval is proportional to the packet size and is inversely proportional to the priority of the service.

Distributed Weighted fair Queue (DWFQ) is proposed in [9] where the size of contention window of any station is adjusted based on the difference between the actual throughput and expected throughput. CW of different stations will be decreased in order to increase the station's priority if the actual throughput of the station is lower than its expected throughput. A ratio of the actual throughput and the corresponding weight of station is used to calculate the priority of the station. A station can adjust its CW by comparing its current ratio value. The station will decrease its CW if its ratio value is smaller than other stations.

Distributed deficit round robin (DDRR) is proposed in [10]. In this algorithm, service priority is assigned with a service quantum rate equal to the throughput it requires and a "service quantum" is utilized to adjust IFS value of service. A deficit counter is assigned to accumulate service quantum rate and the deficit counter will be decreased by the packet length when a packet is transmitted successfully.

3.2.2 Call Admission Control

Service differentiation is a good method to provide better QoS. But no service performs well when the system loading is saturated. So, call admission control is necessary to maintain QoS of existing services by rejecting new services requests. Call admission control can be categorized in two types: calculation-based algorithm and measurement-based algorithm.

3.2.2.1 Calculation-based Algorithm

Calculation-based schemes construct performance criteria for evaluating the status of network and predict the QoS of services in the system. The call admission control in [20] provides guaranteed throughput performances in a statistical sense: A predicted achievable throughput is calculated based on some estimated probability such as the probability of the MAC buffer is empty; the probability of a new packet arrives in one slot time and the probability that the channel is busy, etc. The call admission control tries to satisfy throughput of services in the system. Otherwise, new service will be rejected. Call admission control in [20] has not taken delay and jitter, which are concerned by the QoS of real-time services, into account. Pong, D. and Moors, T. [13] estimates achievable throughput of every active traffic flow for call admission control and scheduling controls. This algorithm first admits the new service to transmit data in a period of time when a new service is coming. Then, the scheduler tries to satisfy the requested throughput of the new service by changing the CWmin and TXOP. But the new service is discarded when the CWmin and TXOP reach the upper limit of the scheduler. The drawbacks of this algorithm are: (1)It limits the throughput of every traffic flow below its estimated “achievable throughput”. The algorithm does not provide a solution when other QoS criteria of the service become intolerable. (2) The algorithm decides to admit a new coming service in a loose way and it does not estimate the influence of this new service to other serving traffic stream. It means that the QoS of other services may be intolerable after the admission of the new service but the algorithm can not prevent it. (3) The algorithm only estimates the throughput of every service but ignores the packet delay. According to the research results in [21], system performance can be divided as “not congested”, “delay limited” and “throughput limited”. “Not congested” state means that channel can probably serve the new traffic flow without severely degrading channel state. Delay limited means that the system can not fulfill the service’s request on the packet delay. Throughput limited means system can not fulfill service throughput. The channel usually becomes delay limited before becoming throughput limited based on the observation in [21]. So, simply applying call admission control based on service’s throughput is not enough.

3.2.2.2 Measurement-based Algorithm

Measurement-based call admission control makes decision based on the measurements of existing system condition. Yang Xiao, and Haizhon Li [6] proposed a simple and effective measurement-based call admission control and bandwidth reservation algorithm. They reserve transmission period of a beacon interval for users of different priority, i.e. voice data can most occupy time period up to 0.4 * beacon interval and video beacon interval can occupy time period up to 0.2 * beacon interval. Except the call admission, [6] uses some parameters Txlimit[i], Txmenory[i] and Txreminder[i] to control the available bandwidth of AC[i] in the network. CWmax[i], CWmin[i] and AIFS[i] are also adjusted dynamically with packet collision rate of every access categories. Zhen-ning *et al* proposed a measurement-assisted model-based call admission control in the 802.11e EDCA [20]. Valaee and Li [12] proposed a measurement-based admission procedure using a sequence of probe packets for ad hoc networks. In [12], it introduces a distributed call admission control in the ad hoc network. The call admission control makes decision based on a “service curve” provisioning. Service curve reflects the status of network and depends on the number of stations, their activity index, and the backoff procedure used for contention. The service curve along with the aggregated services can be used to calculate maximum delay and maximum backlog. Backlog means packets stay in the MAC buffer. The call requests are accepted if the service curve is bounded below by some non-decreasing deterministic function which is called the universal service curve. Universal service curve is independent of the number of stations and services. Universal service curve acts as a worst-case reference curve. All stations want to establish a new service should compare the performance of network to the universal service curve.

3.2.3 Bandwidth Reservation

Call admission control is applied to reject a new service request. Another important topic is how to reserve bandwidth or resources for serving application with varied traffic demands, which is called bandwidth reservation.

Shah et al [15] proposed a bandwidth management method. A scheduler records the minimum bandwidth and maximum bandwidth of this application. The scheduler estimates total bandwidth of the system and perceives bandwidth for every serving application to satisfy their minimum required bandwidth. Scheduler reserves bandwidth by reserving the time period of each application and controlling packet transmission rate of each application. This algorithm may be useful in the 802.11 PCF, which is a centralized coordinate function. But the algorithm in [15] is difficult to be implemented in EDCA. Another drawback is the overhead of this algorithm. Scheduler has to negotiate with stations many times. Each traffic flow re-negotiates its bandwidth once every 14 seconds on average and each of these re-negotiations takes 35 ms delay in maximum.

Ming Li *et al* proposed a call admission control based on estimated throughput [16]. In the assumption of [16], new real-time service sends minimum tolerable bandwidth when it requests to access the system. System accepts a new real-time service based on the First Come First Serve policy until the new bandwidth request cannot be satisfied. After the call admission, there is a priority re-allocation algorithm trying to re-allocate the service priority of real-time services. An existing new service will be dropped when system can not fulfill its minimum bandwidth request. This algorithm only guarantees the QoS of real-time service is better than that of low priority services. Another disadvantage is that it will drop a service even when it is undergoing. But service drop is intolerable for real-time service such as VoIP or video streaming.

Liu *et al* [17] defines a reservation-based MAC access protocols called adaptive acquisition collision avoidance (AACA) protocol with multi-channel supports. AACA adopts the RTS/CTS access mechanism on a common channel solely for reservation purposes. After a successful reservation, the station pair transmits packets in the reserved channel. The AACA mechanism is designed to solve the hidden point in multi-hop networks and it is implemented in a multi-channel system. Nevertheless, it can also be used to achieve bandwidth reservation.

Chapter 4
QoS Based Call Admission Control

Many studies have shown that the packet transmission delay and number of collision times increases rapidly when the WLAN system is saturated. In order to prevent a new requesting service from further degrading system performance, a call admission control is necessary in WLAN network. In this chapter, a calculation-based call admission control is proposed. This call admission control estimates system loading based on a virtual traffic source estimation (VTSE) algorithm. Basic concept of the proposed virtual traffic source estimation algorithm is introduced in Section 4.1. Section 4.2 describes features of proposed VTSE algorithm. Except the proposed call admission control for real-time services, a bandwidth reservation is proposed to steady the QoS of real-time services.

4.1 Virtual MAC Algorithm

The basic concept of virtual MAC algorithm is to estimate system condition, which can be found in the journal proposed by Veres. A, Campbell. A, Barry. M and Li-Hsiang Sun [21-22]. The virtual MAC algorithm is basically composed of a virtual source and virtual MAC. Functions of these algorithms are explained below.

First, when a station decides to serve a new real-time service, the station starts to generate a virtual source. The function of the virtual source is generating "virtual packets" periodically to test the system performance. Parameters of the virtual traffic source are set based on a statistical database. So, stations has to record traffic characters in the network. Except the virtual source, there is a "virtual MAC" mechanism in the requesting station to handle the contention procedure of those virtual packets. Virtual packets contend for the channel in the same way as real packets. Fig.4.1 shows how the virtual MAC estimates the delay time of virtual packets. Packet delay is defined as the duration from the packet is stacked into the buffer to the start successful transmission. Stations utilize virtual MAC algorithm only if the new service is a real-time service. It is because that real-time service generates packets periodically and the QoS requirements of real-time service are strict.

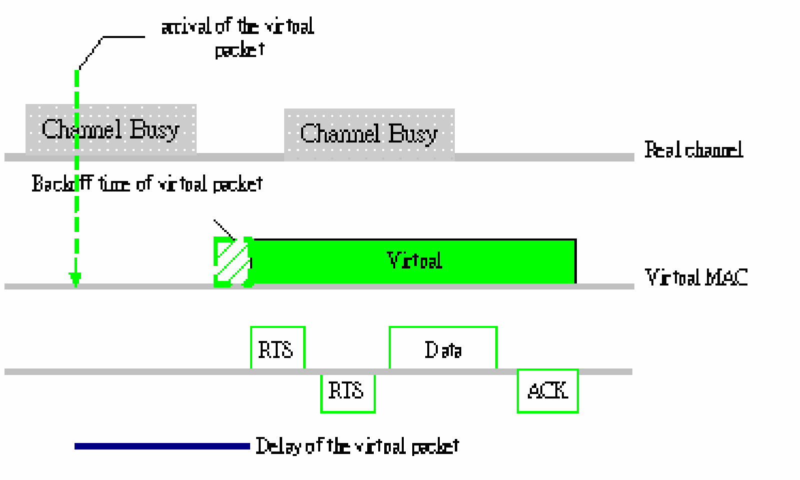

Fig. 4. 1Virtual delay estimation in the virtual MAC algorithm

Advantages of virtual MAC algorithm:

(1) Virtual MAC algorithm can be implemented in the ad hoc network in [21-22]. In the ad hoc network, there is no central controller to control the packet transmission and to monitor the system performance, so the QoS control is hard to realize in the ad hoc network. Virtual MAC algorithm is a good choice because it is easy to be implemented in an ad hoc network.

(2) Virtual MAC algorithm generates a time series of simulated contention and transmission record that can be analyzed by a real test. It does not estimate only a small set of performance measurements, (i.e., estimates of first-order statistics). Take packet delay time as an example, not only the *n*th moments of the delay can be estimated but also delay jitter, packet collision rate, also can be analyzed.

(3) All the procedures in virtual MAC algorithm are the same as a normal MAC layer. No new algorithm needs to be implemented.

4.2 Virtual Traffic Source Estimation Algorithm

Based on the idea of the virtual MAC algorithm, virtual traffic source estimation (VTSE) algorithm is proposed. Section 4.2.1 describes the decision rule of the proposed VTSE algorithm and section 4.2.2 states how VTSE algorithm estimates the delay time of virtual traffic source.

4.2.1 Decision Rule

Differences between the [21-22] virtual MAC algorithm and the proposed VTSE algorithm are listed as follows:

(1) First, the proposed algorithm is implemented in a centralized network, which is called infrastructure network in [1-2]. A centralized network is more convenient than a distributed network for QoS controls. According to the description of [2]. AP in a centralized network can monitor the traffic condition of all of the serving stations in the network and change system parameters dynamically to preserve services' QoS. Research in [23] has shown the downlink direction is the bottleneck of bi-directional service. So a call admission control for real-time services in the downlink will be sufficient.

(2) Since the proposed virtual traffic source estimation algorithm is implemented in the AP and AP can monitor the condition of packets transmission in the downlink direction. So, the third modification of the algorithm is that AP not only estimates the delay time of the virtual packets but also estimates the possible delay time of real transmitted packets of other existing real-time services in the system.

(3) Virtual MAC algorithm estimates virtual delay based on statistical data. A control frame which is called traffic specification element (Tspec) is defined in [2]. Tspec contains a set of parameters that describes characteristics and QoS specifications of the new real-time service.

    Station has to attach the Tspec when it requires a new real-time service and AP makes decision based on the Tspec. AP can also negotiate with the requesting station to change the contents of Tspec. Parameters which are contained in the Tspec are listed in the Table 4.1.The idea of virtual traffic source estimation algorithm is to use the information in the Tspec to generate virtual packets and to estimate the influence of new services towards this system.

Decision rule of the proposed call admission control is described in Fig.4.2:

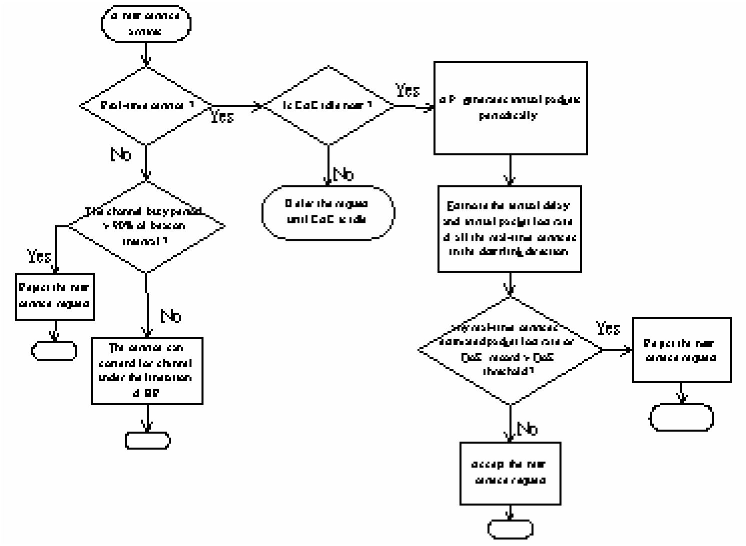

Fig. 4. 2 Decision rule of proposed call admission control

Details of the proposed call admission control algorithm are described below:

(a) When AP receives the new real-time service request, AP begins to set the number of virtual packets based on the interarrival time in the Tspec. For example, the interarrival time of VoIP packets in the Tspec is 20 ms, so the virtual traffic source will generate "virtual packets" every 20 ms. AP utilizes the information of the nominal packet length contained in the Tspec and PHY transmission rate that the AP is available to estimate the transmission period of a virtual packet.

(b) VTSE controller handles the contention process of virtual packets and estimates the influence of virtual packets to other services in AP. Calculation of the propagated delay made by new service is also part of VTSE controller algorithm.

(c) AP makes decision based on the observed virtual packet loss rate in the admission period and the recorded packet loss rate of every existing real-time service in the downlink direction. The new service request will be rejected if the packet loss rate of any real-time services, including the virtual traffic source, exceeds the tolerable threshold. Otherwise, the new service request will be accepted.

(d) The proposed algorithm rejects new non-real time service when the channel is occupied more than 90% of time period in a beacon interval.

Table4. 1 Parameters contained in the Tspec

Tspec information bytes	Description
Periodic traffic	Defines whether the real-time service is periodic or continuous
Bi-directionality	Defines whether the real-time service is bidirectional or unidirectional
ACK policy	Defines the proposed acknowledgement policy
FEC	Enables the use of FEC coding
Inactivity	Defines the minimum time interval that can elapse without any MSDU of this real-time service being transmitted before the AP deletes this stream
Retry interval	Defines the minimum time interval that a station with real-time service waits for a delayed ACK frame before initializing retransmissions
Delivery priority	Defines the utilized delivery(access) priority used for this real-time service
Nominal MSDU size	Defines the nominal MSDUs size belonging to this real-time service
Minimum data rate	Define the lowest data rate that is tolerable for this real-time service
Mean data rate	Defines the nominal sustained data rate for this real-time service (within the delay and jitters)
Maximum burst size	Defines the size of the maximum burst that may occur in this real-time service
Delay bound	Defines the maximum tolerable amount of time to transmit data. The delay of a MSDU is defined as its reception from the local MAC user to the start of the successful transmission to its destination
Jitter bound	Defines the maximum tolerable delay variance for a MSDU between its reception for the local MAC user to the start of the successful transmission to its destination
Interarrival interval	Specifies the nominal interarrival time of MSDUs of this real-time service
Minimum Tx rate	Specifies the minimum PHY rate that is necessary for successful transport of this new real-time service.

4.2.2 Calculation of Virtual Delay

In this thesis, "virtual delay" is used to represent the delay time of the virtual packet and the estimated possible delay of existing real packet. The first thing in the calculation of the virtual delay is to estimate how long a virtual packet transmission will be taken based on the equations (2-1) ~ (2-7) and the packet transmission plotted in the Fig.2.3. Assuming a success virtual packet transmission takes T_p to finish its transmission. If the nominal MSDU size in the Tspec is B_{data} and the PHY data rate the new service requests to transmit packets is R_{data} and the service priority is i, the value of V_p is equal to

$$
\begin{aligned}
&T_p(R_{data}, B_{data}) \\
&= T_{rts} + T_{cts} + T_{data}(B_{data}) + T_{ack} + 3*SIFS\_Period + \mathrm{AIFS[i]} + \mathrm{Backoff\ timer} \\
&= 4*(T_{\mathrm{Preamble}} + T_{\mathrm{SIGNAL}}) \\
&+ \left\lceil \frac{MAC\_Header + Service\_field + tail\_bits + RTS\_Control\_field}{Data\ bytes\ per\ symbol\ (1)} \right\rceil * T_{SI} \\
&+ \left\lceil \frac{MAC\_Header + Service\_field + tail\_bits + CTS\_Control\_field}{Data\ bytes\ per\ symbol\ (1)} \right\rceil * T_{SI} \\
&+ \left\lceil \frac{MAC\_Header + Service\_field + tail\_bits + \mathrm{B}_{data}}{Data\ bytes\ per\ symbol(\mathrm{R}_{data})} \right\rceil * T_{SI} \\
&+ \left\lceil \frac{MAC\_Header + Service\_field + tail\_bits + ACK\_Control\_field}{Data\ bytes\ per\ symbol(\mathrm{R}_{data}*)} \right\rceil * T_{SI} \\
&+ 3*SIFS\_Period + \mathrm{AIFS} + \mathrm{Backoff\ timer}
\end{aligned}
\qquad (4\text{-}1)
$$

After the calculation of packet transmission period of virtual packets, another parameter D_p is used to represent the propagated delay time made by virtual packets. Before the description of D_p, an introduction of how the VTSE controller handles the contention process of virtual packets would help the understanding the estimation of D_p.

Fig.4.3 is plotted to show the state of virtual traffic source in the AP. A real-time service in the WLAN can be seen as a traffic source which generates packets continuously. The buffer of service priority 1 is labeled as buffer[1] in the Fig. 4.3.

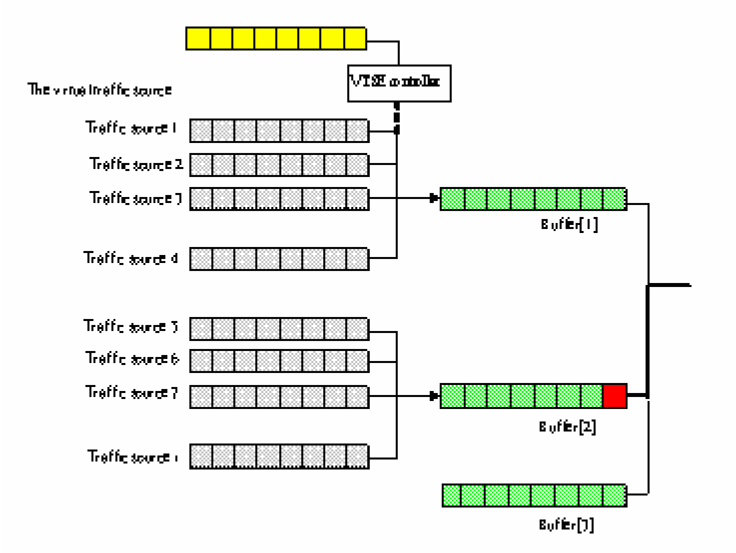

Fig. 4. 3 Virtual packets contend for the channel with other packets

After the generation of virtual traffic source, the VTSE controller is responded to the contention of virtual packets. The VTSE controller estimates packet delays and packet loss rate to make decision. Estimation rules of virtual delay and virtual packet loss rate are described in the Fig.4.4. The mechanism is described as below:

Fig.4.5 plots the condition that virtual traffic source gets the access of channel after a packet R(k, j) is transmitted. R(k, j) means it is the packet of real-time service j which is stacked into the buffer[k]. When there is no real packet in the buffer[k], the VTSE controller handles the virtual packet contention process itself. The contention procedure in the VTSE controller is the same with real packets. In Fig. 4.5, although there is a real packet “R(m, k)” is transmitted. But packet transmission of the packet “R(1, j)” will be affected by the virtual packet if the virtual packet is allowed to be transmitted. It means that the packet delay of the packet “R(1, j)” is increased because of the virtual packet. Assume the transmission period of a virtual packet is V_p. Equation (4-2) lists the update process of $D_p(1)$ and $D_p(2)$. $D_p(k)$ is the estimation of the propagated delay of buffer[k] made by virtual traffic source in the *n*th slottime. dir_new_service is the variable that indicates the direction of the new service. The value of dir_new_service is 2 when the new requesting service is a bi-directional service. Otherwise, dir_new_service would be equal to 1. When the VTSE controller contends success, $D_p(1)$ and $D_p(2)$is updated based on (4-2).

When the buffer[k] is idle, $D_p(k)$ is decreased continuously when buffer[k] keeps idle longer than a DIFS. Otherwise, the $D_p(k)$ will be frozen. So, the formula of updating $D_p(k)$ is listed as the formula (4-3) and (4-4). The VTSE controller estimates the virtual delay of every real packet of different buffers by adding the $D_p(k)$ to the experienced packet delay of real packets.

$$\begin{cases} \text{When a virtual packet is transmitted} & \\ \quad D_p(k) = D_p(k) + V_p * \text{dir\_new\_service} & (4\text{-}2) \\ \text{When the buffer[k] idle longer than a DIFS} & \\ \quad D_p(k) = D_p(k) - \text{idle period} & (4\text{-}3) \\ \text{When the buffer[k] is busy} & \\ \quad D_p(k) = D_p(k) & (4.4) \end{cases}$$

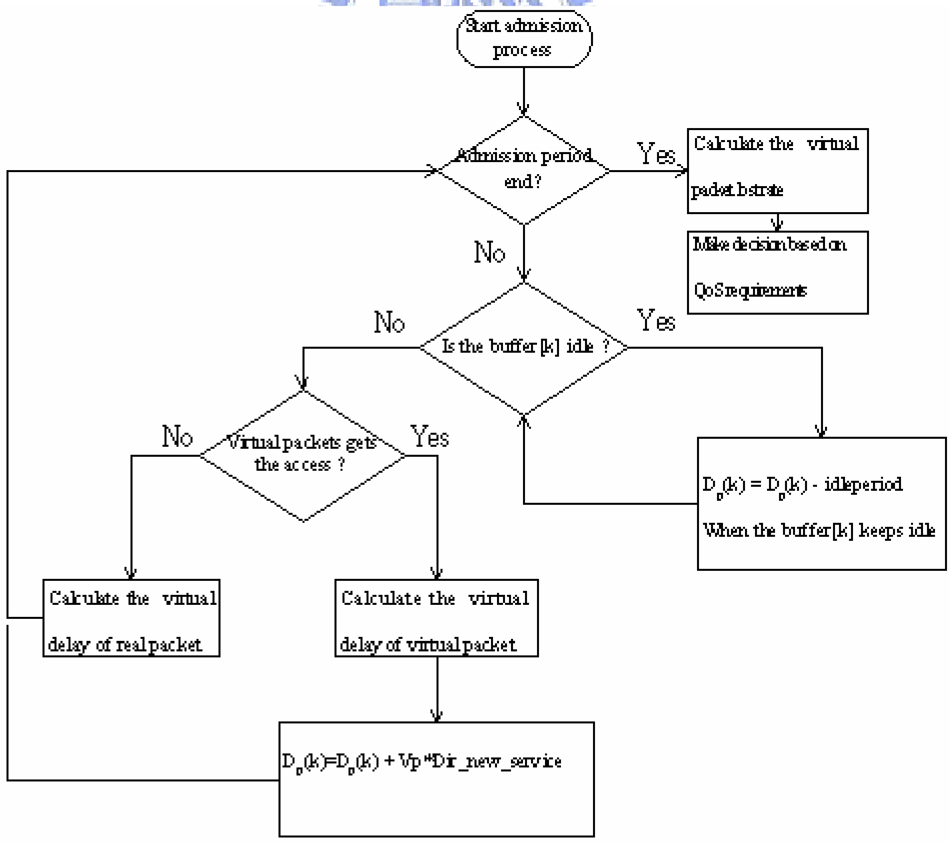

Fig. 4. 4 Estimation rules of virtual delay and virtual packet loss rate

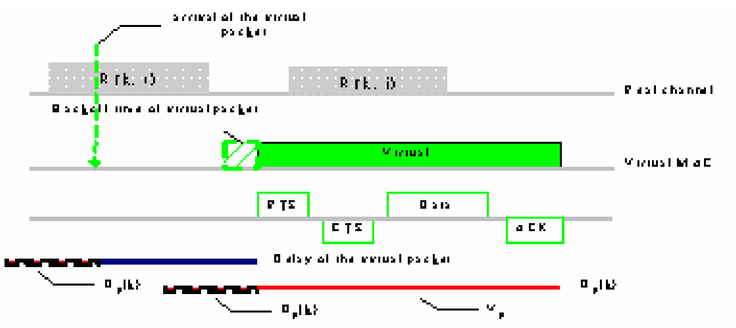

Fig. 4. 5 VTSE controller contends for the channel directly when there is no packet in the buffer[k]

Fig. 4.6 plots the condition that there are packets waiting in the buffer[1] and buffer[1] gets the access to transmit packet. At this time, the VTSE controller has to compare the longest waiting virtual packet to the longest waiting real packet in the buffer[1]. Assume the interarrival time of the eldest waiting virtual packet is V_{int} and the interarrival time of the longest waiting real packet is R_{int}. The virtual packet can be transmitted if $V_{int} < R_{int}$ and the virtual packet can be transmitted directly because of the FIFO feature of buffers. The $D_p(k)$ is updated as (4-2) when the virtual packet is transmitted successfully.

When one real packet of service *i*, which is labeled R(k,i), is transmitted successfully. The VTSE controller estimates the virtual delay of real packet by adding the $D_p(k)$ to its real queuing delay. Suppose the queuing delay of packet the R(k, i), is $D_{original}(R(k, i))$ and the estimated virtual delay is $V_r(R(k,i))$. The estimated virtual delay is calculated as (4.3).

$$V_r(R(k ,i)) = D_{original}(R(k, i)) + D_p(k) \tag{4-5}$$

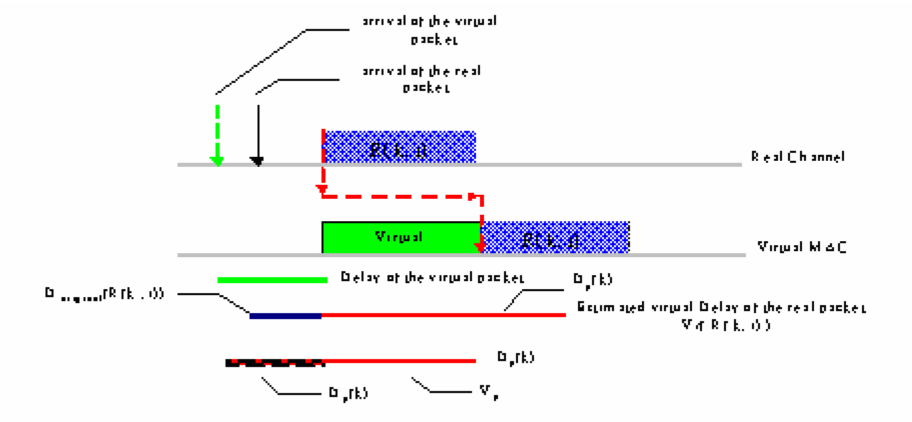

Fig. 4. 6 Virtual packet transmission when the buffer[k] gets the access and the virtual packet's interval time is earlier than the transmitting real packet in the buffer[k]. $D_p(k)$ and $V_r(R(k, i))$ are also updated by the VTSE controller

VTSE controller will simulate the virtual packet transmission during an admission period the QoS of every service is estimated when the admission period is finished. The proposed QoS threshold in the VTSE algorithm is packet loss rate. Assume the delay bound of service *i* is labeled as D_I. A packet of service *i* will be dropped when its queuing delay is larger than D_I. So the virtual packet loss rate of service *i,* which is labeled as V(i), will be calculated. The number of packets that would be dropped are labeled as $V_d(i)$. Assume $V_T(i)$ is the number of packets transmitted in the admission period.

Virtual packet loss rate V(i)

= Number of estimated dump packets / number of estimated transitted packets.

$$=V_d(i)/V_T(i) \qquad (4\text{-}6)$$

The new service request will be rejected when there are any real-time service that its V(i) is larger than its acceptable packet loss rate, which is labeled as D(i). Another rule to reject the new service request is the QoS record packet loss rate of existing services. The new service request will be rejected when the recorded packet loss rate of existing real-time services exceeds its QoS threshold.

4.3 Bandwidth Reservation

The goal of the bandwidth reservation is to protect the QoS of real-time services from being degraded by unstable system loading. The major concept of this algorithm is preserve a period of time that is available for those high priority services. In other words, the proposed bandwidth reservation limits the access of Non-real-time services based on the traffic loading of real-time services.

Procedures of this mechanism are described in Fig.4.8 and they will be explained below:

- Calculate the duration that the real-time services have occupied the channel in the n^{th} beacon interval, which is labeled as $T_{real}(n)$.
- The AP calculates the average time period that real time services occupies during a beacon interval and updates the value after every beacon interval. Assume the average time period is labeled R_{avg}. R_{avg} is updated likes (4-7).

$$R_{avg}(n+1) = \alpha * R_{avg}(n) + (1-\alpha) * T_{real}(n) \quad (4\text{-}7)$$

- In the next $n+1^{th}$ beacon interval, AP limits the duration which all Non-real-time services can occupy under the value of T_{token} calculated in (4-8).

$$T_{token}(n+1) = T_{beacon} - \text{surplus factor} * R_{avg}(n+1) \quad (4\text{-}8)$$

Where T_{beacon} is a constant value, which is equal to 100 ms in [1].
Surplus factor represents a ratio that real-time services require for bandwidth reservation. Surplus factor is contained in the Tspec when a new real-time service requirement is transmitted.

- Based on the specification of [2], AP broadcasts the beacon frame in the beginning of every beacon interval. T_{token} can be attached in the beacon frame so stations in the network can update T_{token} in the beginning of every beacon interval.

After stations in the network receive beacon frame, a bandwidth reservation process starts in this way:

- Stations that have non-real-time services will record the duration that non-real-time services have occupied, which is labeled as $T_{non\text{-}real}$.
  $T_{non\text{-}real}$ is equal to T_{token} in the beginning of every beacon interval.
- When a station transmits a packet of non-real-time service, which is labeled P_k, from its MAC buffer, it calculates the period that P_k will occupy, which is labeled as T_k. According to the description in chapter 2, P_k can be received by other stations in the network and other stations can use T_k to update their NAV. Assume stations can identify the service type of the transmitting packets. It can be achieved because there are many reserved field in the MAC header and the transmitting stations can attach this information in these reserved field. When stations receive the information of P_k, each station updates their $T_{non\text{-}real}$ as:

$$T_{non\text{-}real}(P_k) = T_{non\text{-}real}(P_{k-1}) - T_k \quad (4\text{-}9)$$

- Stations with non-real-time services would halt the packet contention process if $T_{non\text{-}real}(P_k)$ decreased to lower than zero.

Flow chart of thes procedures are posted in Fig. 4.7. Since all the computation are already defined in [1] and [2], the bandwidth reservation would not increase the complexity of WLAN system.

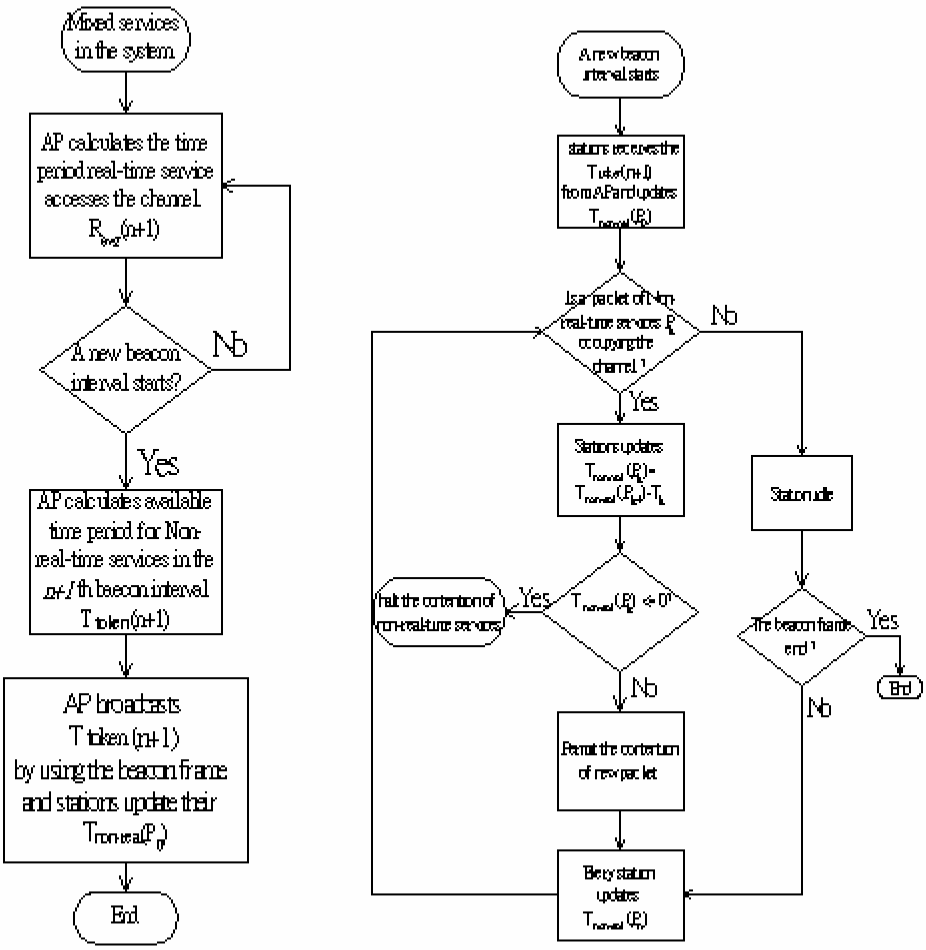

Fig. 4. 7 Flow chart of the proposed bandwidth reservation process towards non-real-time services in the WLAN network

Chapter 5

Simulation Results

This chapter describes the simulation platform and simulation results. Section 5.1 is the introduction of the simulation platform and settings in the platform. Section 5.2 is simulation results of the proposed VTSE algorithm and bandwidth reservation. Section 5.3 is conclusion of simulation results.

5.1 Simulation Platform

An event driven simulation platform is developed based on the specification of [1] and [2]. This platform simulates contention process and packets transmission in the infrastructure network, which means an access point (AP) centralizes the network control. AP and all the stations in the simulation platform contend for the channel in EDCA mode. RTS/CTS mode is implemented in the platform because of the recommendation in [2].

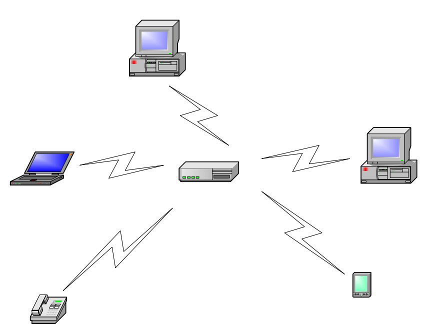

Fig. 5. 2 802.11 WLAN infrastructure network

Table 5.1 lists the assumptions used in the simulation. In order to verify the proposed VTSE algorithm and the bandwidth reservation, real-time services and non-real-time services are simulated in this platform. Traffic models of real-time and non-real-time services are explained in section 5.1.1 and 5.1.2. Section 5.1.3 describes the assumption in the PHY layer and the channel model. Section 5.1.4 explains the QoS criteria of real-time services in the WLAN.

Table5. 1 Settings of parameters of real-time services in the platform

Parameters	**VoIP**	**FTP**	**HTTP 1.1**
Periodic traffic	Continuous	Burst	Burst
	Markov on/off		
Bi-directionality	Bidirectional	User Defined	User Defined
Delivery priority	3	0	1
Delay bound	50 ms	N.A.	N.A.

5.1.1 Real-time Services

The simulation platform selects VoIP to simulate real-time services in the network because VoIP service is common in the network. This platform simulates VoIP traffic in constant bit rate (CBR) mode and Markov on/off model. In the CBR mode, VoIP traffic source generates packets every 20 ms. Payload size of a VoIP packet is 160 bytes and the data rate is fixed at 64 Kbps (160 bytes per 20 ms).

The Markov on/off model [24, 25] is plotted in Fig. 5.2. Traffic source operated in the Markov on/off model generates packets in an uncertain way. There are “on” state and “off” state. Traffic source generates packets periodically when it is in “on” state. The duration of on state is followed by an exponential distribution. Then, the traffic source will change its condition to “off” state and it will keep idle. The duration of traffic sources stay in the off state is also followed by an exponential distribution. The Mean values of the on and off state continues are equal to 1 second and 1.35 seconds [24, 25]. Table 5.2 lists the traffic model of real-time services in the simulation platform.

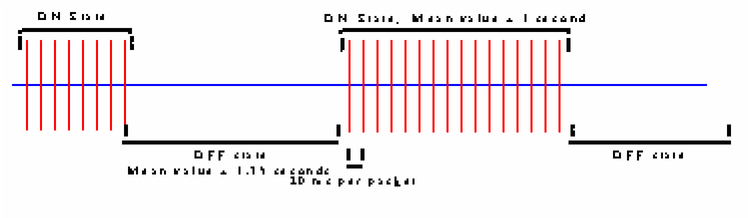

Fig. 5. 3 Traffic model of Markov on/off model

Table5. 2 Simulation model of VoIP services in this platform

Applications	**Interarrival time**	**Packet size**
VoIP	CBR 20 ms per packet.	Fixed packet length 160 bytes
VoIP	Markov on/off model On state $fx = \lambda e^{-\lambda x}, x \geq 0$ $\lambda = 1$ Off state $fx = \lambda e^{-\lambda x}, x \geq 0$ $\lambda = 1.35$	Fixed packet length 160 bytes

5.1.2 Non-real-time Services

Two types of the non-real-time services are simulated in this simulation platform. One is the web browsing service and another is file transport protocol (FTP) service. The traffic model of web browsing and FTP services is explained below.

Fig. 5.3 shows the packet trace of a typical web browsing session. The session is divided into ON/OFF periods representing web-page downloads and the intermediate reading times. In Fig. 5.3, the web-page downloads are referred to as packet calls. Therefore, a packet call, like a packet session, is divided into ON/OFF periods. Unlike a packet session, the ON/OFF periods within a packet call are attributed to machine interaction rather than human interaction. When receiving a page, the web-browser will parse the HTML page for additional references to embedded image files such as the graphics on the tops and sides of the page as well as the stylized buttons. The retrieval of the initial page and each of the constituent objects is represented by ON period within the packet call while the parsing time and protocol overhead are represented by the OFF periods within a packet call. For simplicity, the term “page” will be used in this thesis to refer to each packet call ON period. The initial HTML page is referred to as the “main object” and the each of the constituent objects referenced from the main object are referred to as an “embedded object”.

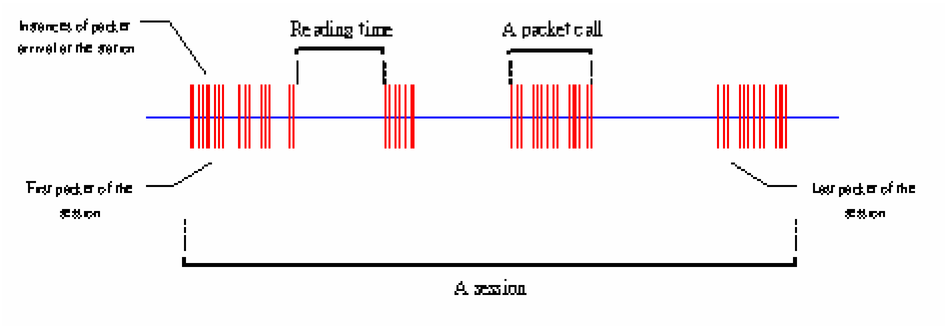

Fig. 5. 4 Packet trace of typical web browsing session

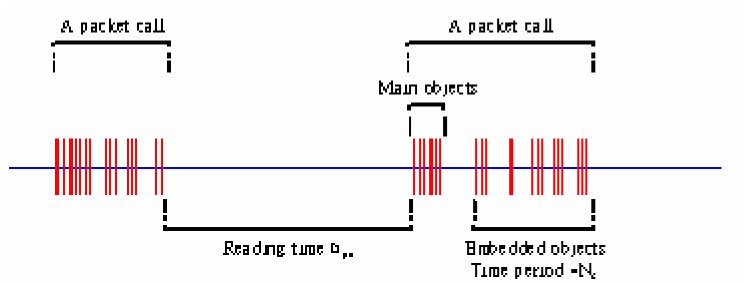

Fig. 5. 5 Contents in a packet call

Parameters for the web browsing traffic are as follows:

- S_M: Size of the main object in a page.
- S_E: Size of an embedded object in a page.
- N_d: Number of embedded objects in a page.
- D_{pc}:Reading time.
- T_p: Parsing time for the main page.

Packet traffic characteristics within a packet call will depend on the version of HTTP used by the web servers and browsers. Currently two versions of the protocol, HTTP/1.0 and HTTP/1.1, are widely used by the servers and browsers. In this platform, the traffic model of HTTP/1.1 is accepted.

Parameters of web browsing service in the platform are listed in Table 5.3. In HTTP/1.1, persistent TCP connections are used to download the objects, which are located at the same server and the objects are transferred serially over a single TCP connection; this is known as HTTP/1.1-persistent mode transfer. The TCP overhead of slow-start and congestion control occur only once per persistent connection. The distributions of the parameters for the web browsing traffic model were determined based on the literature on web browsing traffic characteristics [17].

Table5. 3 HTTP traffic model parameters

Component	Distribution	Parameters	PDF
Main object size (S_M)	Truncated Lognormal	Mean = 10710 bytes Std. dev. = 25032 bytes Minimum = 100 bytes Maximum = 2 Mbytes	$fx = \frac{1}{\sqrt{2\pi}\sigma x}\exp\left[\frac{-(\ln x - u)^2}{2\sigma^2}\right], x \geq 0$ $\sigma = 1.37, u = 8.35$
Embedded object size (S_E)	Truncated Lognormal	Mean = 7758 bytes Std. dev. = 126168 bytes Minimum = 50 bytes Maximum = 2 Mbytes	$fx = \frac{1}{\sqrt{2\pi}\sigma x}\exp\left[\frac{-(\ln x - u)^2}{2\sigma^2}\right], x \geq 0$ $\sigma = 2.36, u = 6.17$
Number of embedded objects per page (N_d)	Truncated Pareto	Mean = 5.64 Max. = 53	$fx = \frac{\alpha_k^{\alpha}}{x^{\alpha+1}}, k \leq x \leq m$ $fx = \left(\frac{k}{m}\right)^{\alpha}, x = m$ $\alpha = 1.1, k = 2, m = 55$ Subtract k from the generated random value to obtain N_d.
Reading time (D_{pc})	Exponential	Mean = 30 sec	$fx = \lambda e^{-\lambda x}, x \geq 0$ $\lambda = 0.033$
Parsing time (T_p)	Exponential	Mean = 0.13 sec	$fx = \lambda e^{-\lambda x}, x \geq 0$ $\lambda = 7.69$

Parameters for the FTP application sessions are described in Table5.4. Fig.5.5 plots the packet trace in a typical FTP session. In FTP applications, a session consists of a sequence of file transfers, separated by reading times. The two main parameters of an FTP session are:

- S : the size of a file to be transferred
- D_{pc} : reading time, i.e., the time interval between end of download of the previous file and the user request for the next file.

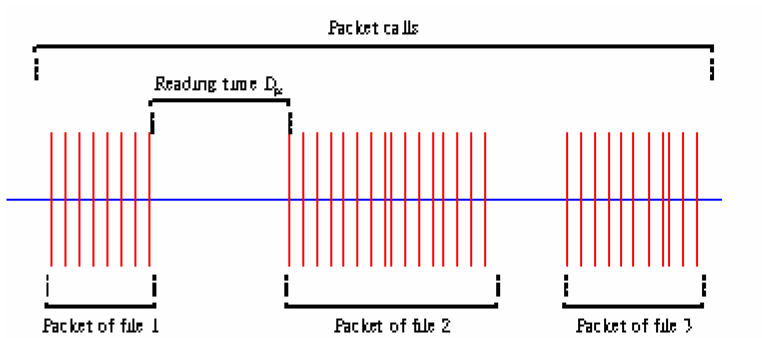

Fig. 5. 6 Packet trace in a typical FTP session

Table5. 4 FTP traffic model parameters

Component	Distribution	Parameters	PDF
File size (S)	Truncated Lognormal	Mean = 2Mbytes Std. Dev. = 0.722 Mbytes Maximum = 5 Mbytes	$fx = \frac{1}{\sqrt{2\pi}\sigma x}\exp\left[\frac{-(\ln x - u)^2}{2\sigma^2}\right], x \geq 0$ $\sigma = 0.35, u = 14.45$
Reading time (D_{pc})	Exponential	Mean = 180 sec.	$fx = \lambda e^{-\lambda x}, x \geq 0$ $\lambda = 0.006$

5.1.3 PHY Layer Assumption

Many values such as SIFS, slottime or the time period each packets occupies the channel are correlated with the assumption of PHY layer. The transmission modes and value assumptions based on [18] are listed in table 2.3 and table 2.4. A simple link adaptation is implemented based on these transmission modes. According to the studies of channel conditions in WLAN [26-28], channel of this platform is set as AWGN channel and the distribution of SNR will be log normal distribution which the mean and variance are equal to 15 dB and 8 dB. Decision rule of rate control is listed in table 5.5. We can assume that error probability of packet transmission is ignorable by using link adaptation since good link adaptations would decrease error probability to very low.

Table5. 5 Decision rule of link adaptation

Transmission Mode	SNR	Data rate(Mbps)
Mode 1	SNR< 7dB	6
Mode 2	7dB < SNR < 10dB	9
Mode 3	7dB < SNR < 10dB	12
Mode 4	10dB < SNR < 14dB	18
Mode 5	14dB < SNR < 17dB	24
Mode 6	17dB < SNR < 22dB	36
Mode 7	22dB < SNR < 24dB	48
Mode 8	24dB < SNR	54

5.1.4 Performance Criteria in Simulation Platform

Following performance criteria are considered:

- **Packet delay:** The packet delay is defined as the time interval from the time that packet arrives at the MAC layer to the beginning of a successful transmission.
- **Delay Jitter:** Delay jitter is the standard deviation of the packet delay. Delay jitter also affects the quality of real-time services when the delay jitter is large.
- **Packet loss rate:** In the simulation platform, packet loss happens because packets stay in the buffer of the local transmitter longer than the delay bound that

the real-time service can tolerate.

- **Capacity:** The number of stations that the WLAN system allows to serve in the system when the QoS is taken into account.
- **Unsatisfied condition:** Unsatisfied condition means that there is at least one active real-time service which its packet loss rate larger than 1%. Unsatisfied condition is defined because the proposed VTSE algorithm reject a new real-time service request when is estimated that the unsatisfied condition will happen or the unsatisfied condition already happens.

5.2 Simulation Results

In order to discuss the traffic features. Some scenarios are created for the convenience of verification. Section 5.2.1 discusses the performance of the proposed virtual traffic source estimation algorithm when all services in the network are VoIP services which generate packets in the CBR mode. In section 5.2.2, best effort services are added to observe the performance of VTSE. A comparison towards a simple algorithm is discussed in section 5.2.3. Observation period of VTSE is 1 second when VoIP services are CBR mode and it will be set 5 seconds when VoIP services are Markov on/off mode.

5.2.1 Scenario I

Discussion in this scenario is focus on the traffic features of VoIP users in WLAN and the performance of the proposed VTSE algorithm. All VoIP users generate packets in CBR mode and the number of active VoIP users is adjusted to examine the performance. Since the VTSE estimates the downlink traffic condition. The observed data is focus on the downlink direction. In scenario I, there are some existing VoIP services begin to generate packets in the beginning of simulation. Then, a new VoIP service request appears during the simulation period. The proposed call admission control will estimate the system condition and make decision according to its estimation and QoS record of existing VoIP services after an observation period equal to 5 seconds.

Fig.5.6 shows the VTSE estimated delay time and the observed packet queuing delay in the downlink direction. It shows the VTSE algorithm can estimated the transmission delay precisely. Fig.5.7 shows the observed conditional probability that unsatisfied condition happens when the number of active VoIP services is adjusted form 21 to 32.The conditional reject probability of VTSE algorithm is also plotted in Fig.5.7. VTSE rejects a new service request when it predicts that the network will be unsatisfied if the system accepts the new VoIP service. Fig.5.7 reveals that the reject probability of VTSE basically close to the probability that unsatisfied condition happens in realistic except when the 27th and 28th VoIP service request. It is because the service performance begins to degrade seriously. This phenomenon can be observed from the Fig.5.8. Fig.5.8 plots the average unsatisfied users in the system when there are fixed number of VoIP services. Fig.5.8 reveals that the average unsatisfied users begin to increase seriously when there are 27 active VoIP users in the system.

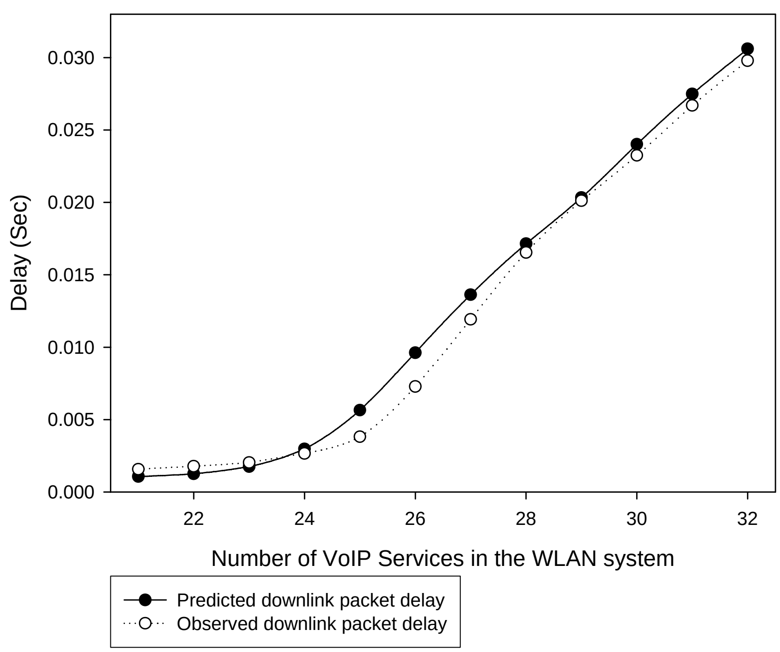

Fig. 5. 7VTSE estimated delay and the observed packet queuing delay in the downlink direction

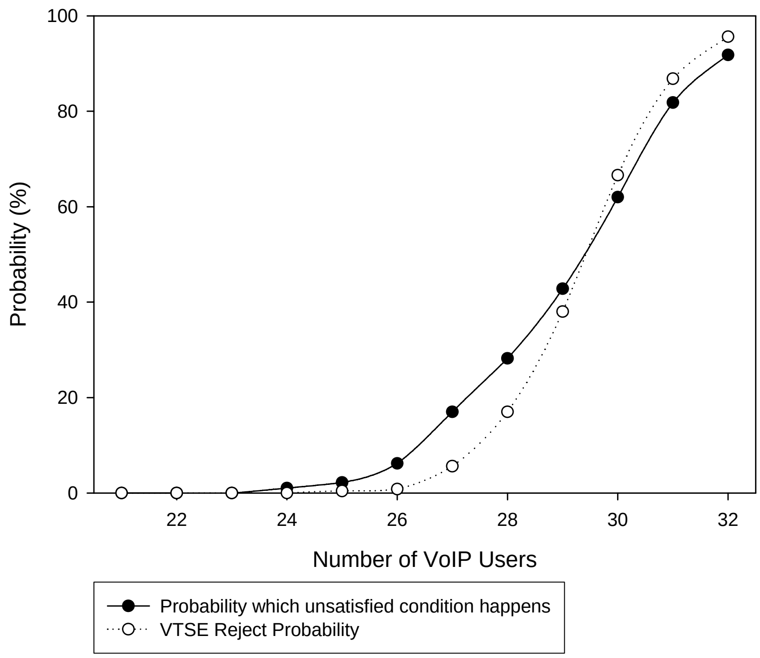

Fig. 5. 8 Observed conditional probability that unsatisfied condition happens and the conditional reject probability of VTSE algorithm

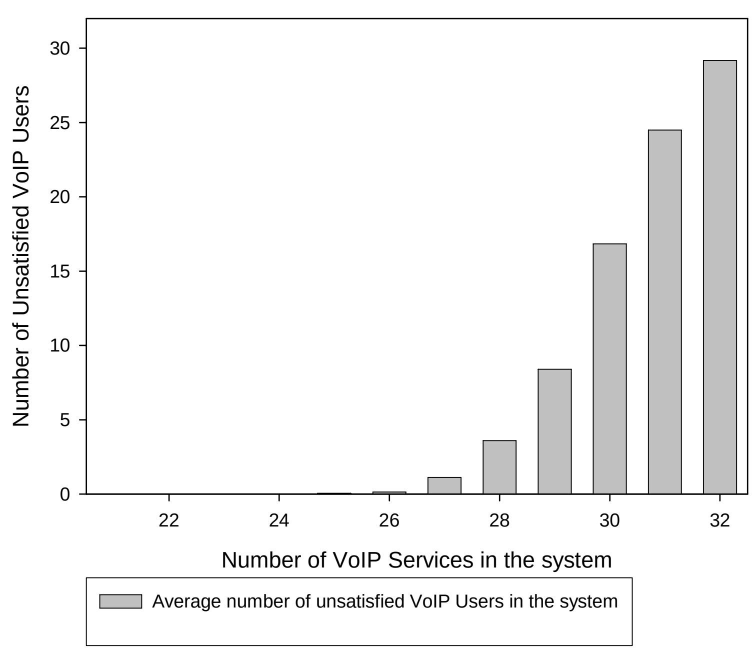

Fig. 5. 9 Average unsatisfied users in the system when the number of VoIP services in fixed

Fig.5.9 plots the VTSE reject probability and the probability that unsatisfied condition happens in the system when VoIP users are fixed and all VoIP services generate packets in Markov on/off mode. It is assumed that the proposed VTSE can estimate the average length of on/off period and generates virtual packets in on/off mode and the VTSE observation period is 5 seconds. The result shows the reject probability is close to the probability that unsatisfied condition happens.

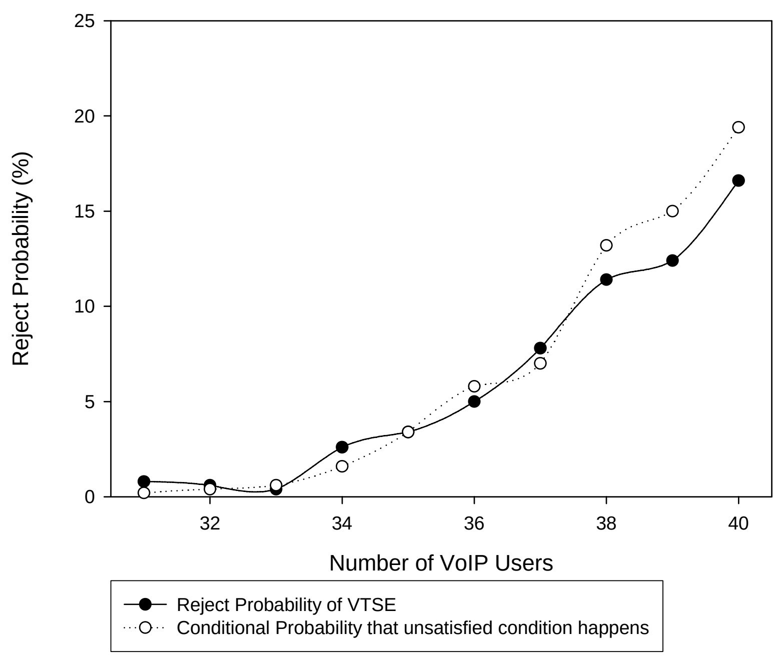

Fig. 5. 10 Observed conditional probability that unsatisfied condition happens and the conditional reject probability of VTSE under Markov on/off mode

5.2.2 Scenario II

In the scenario II, number of best effort services is fixed in 100 web browsing services and 10 FTP services. All the web browsing services generate packets in the downlink direction and FTP services are assumed bi-directional to simulate the uplink best effort services. Then, the number of VoIP services in the platform will be adjusted to examine the performance of VTSE algorithm. The effect of the proposed bandwidth reservation will also be surveyed in this scenario.

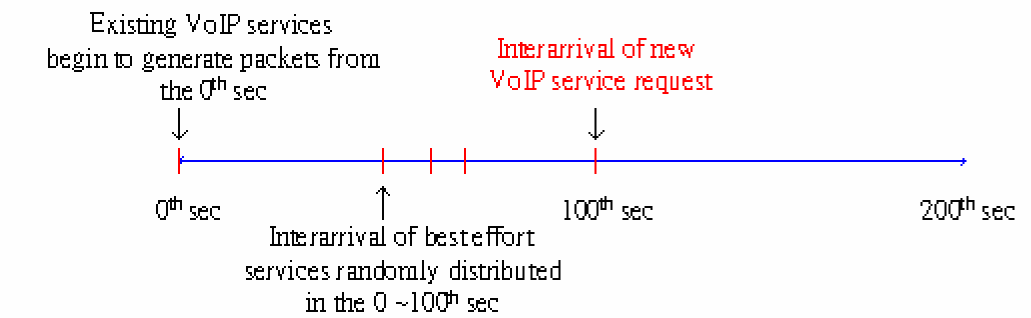

Fig. 5. 11 Assumptions of interarrival of different services in the simulation period

Fig.5.10 describes the assumption of interarrival of different services in the simulation period. Simulation period will last up to 200 seconds. In the beginning, fixed numbers of active VoIP services begin to generate packets in the 0^{th} second. Then, the interarrival time of 100 web browsing services and 10 FTP services are set randomly in the 0 to 100^{th} second. A new VoIP service request happens at the 100^{th} second and it relies on the VTSE to make decision. i.e. When the VTSE makes decision for the 27^{th} VoIP services request, it means that there are 26 existing active VoIP services and 10 FTP and 100 web browsing services in the system. VTSE will reject new VoIP service when it predicts that the system will become "unsatisfied". The probability that "unsatisfied condition" happens after the system accept the new service request will also be observed to examine the difference of predicted unsatisfied probability and real unsatisfied probability.

Fig 5.11 shows the observed conditional probability that unsatisfied condition happens when the number of active VoIP services is adjusted from 15 to 30. The reject probability of VTSE algorithm is also plotted in Fig.5.11. Fig.5.11 reveals that the VTSE algorithm can estimate the unsatisfied condition close to the probability that unsatisfied condition happens when the new VoIP service is allowed to access the system. Fig.5.12 reveals the reject probability and the observed probability that unsatisfied condition happens when the bandwidth reservation is implemented with the VTSE algorithm. In Fig.5.12 the reject probability is larger than the realistic observed probability. It is because BR reserves bandwidth for real-time services adaptively when the new service request is allowed to access the system. But the bonus of BR toward real-time services has not taken into account in the VTSE. Fig.5.13 is the capacity that the system allows how many VoIP services to access the system when the number of best effort service is fixed in 100 web browsing services and 10 bi-directional FTP services. Fig.5.14 plots the influence of the proposed BR towards system throughput; the system throughput would degrade 10%~20% in average when BR is implemented.

Fig.5.15 reveals the comparison of delay jitter of VoIP services whether the BR is implemented or not. BR will keep the delay jitter of VoIP services steady except when there are more than 27 VoIP users in the system. But the system only accepts27 VoIP users in maximum when BR is implemented according the result plotted in Fig.5.13,. So, VTSE algorithm rejects new VoIP service before the BR fail to keep delay jitter steady.

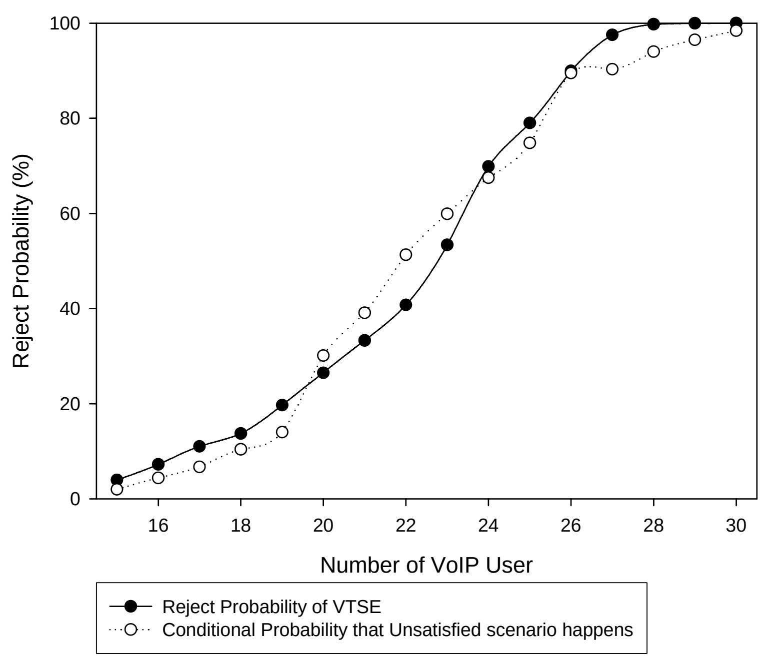

Fig. 5. 12 Conditional probability of unsatisfied condition and the conditional reject probability of VTSE

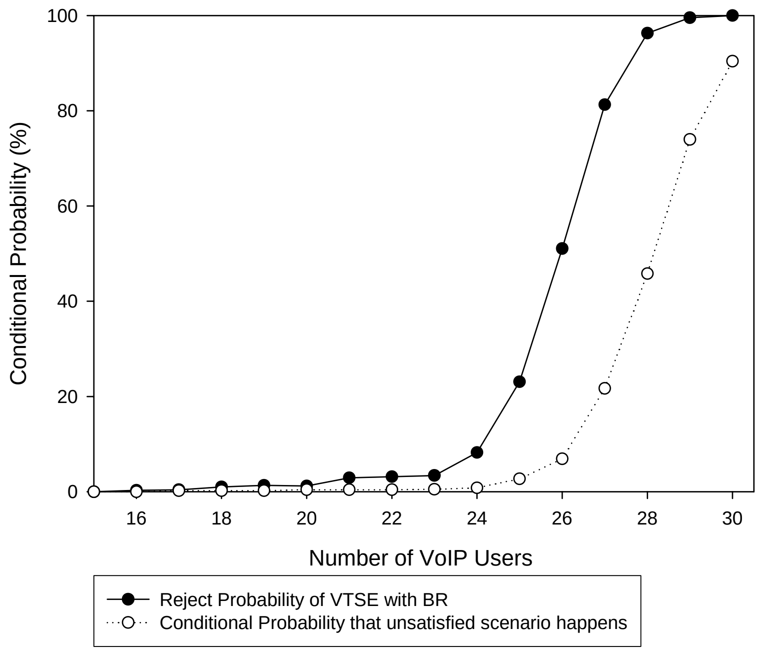

Fig. 5. 13 Conditional reject probability of VTSE and observed conditional probability of unsatisfied condition when BR is implemented with VTSE

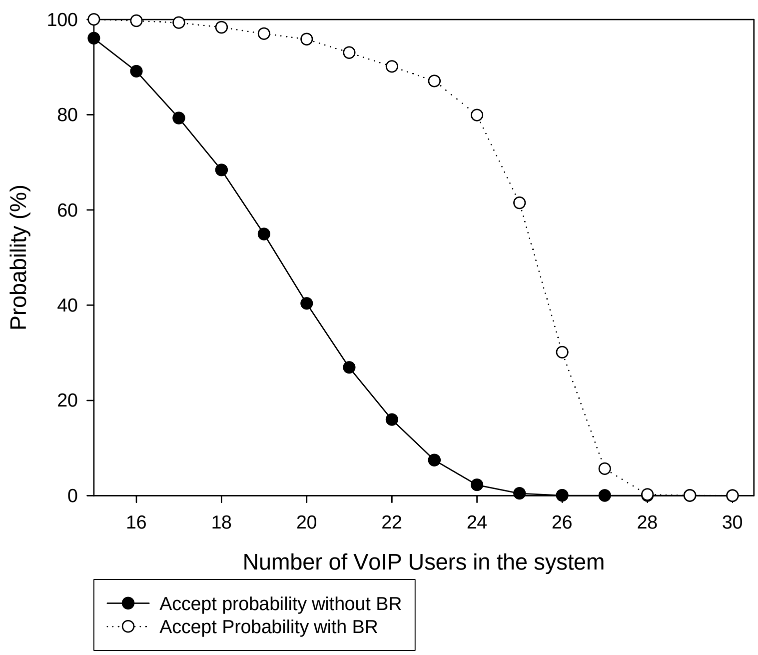

Fig. 5. 14 Capacity of VoIP services with fixed number of non-real-time services

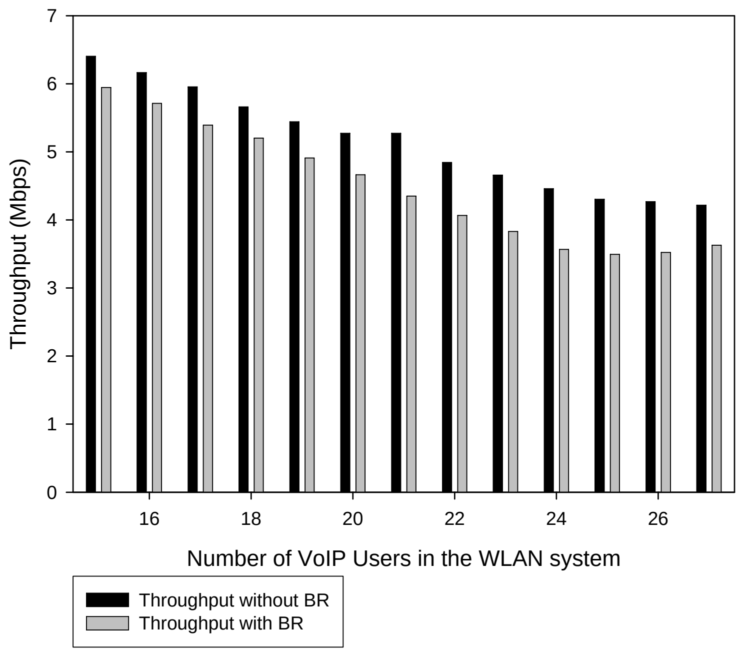

Fig. 5. 15 Comparison of the throughput when BR is implemented or not

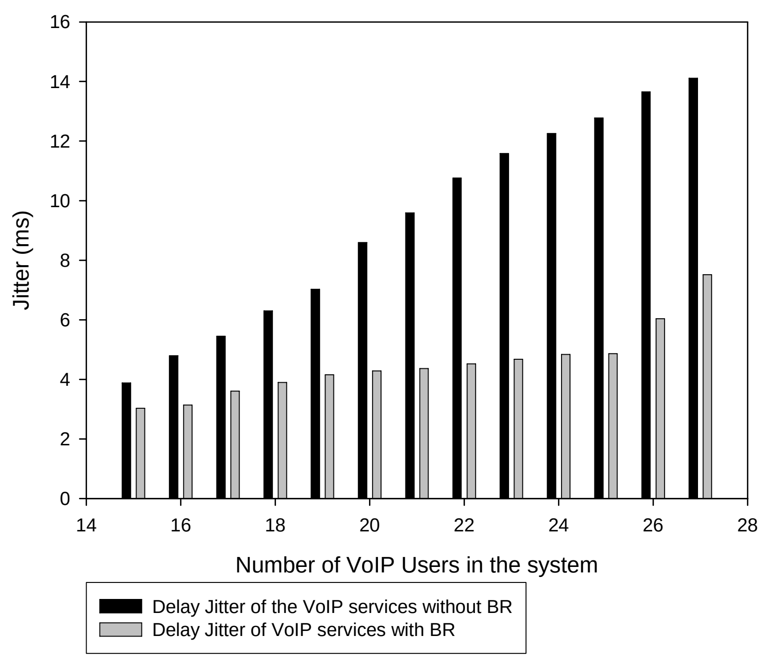

Fig. 5. 16 Comparison of delay jitter of VoIP services when BR is implemented or not

5.2.3 Scenario III

In Section 5.2.3, the VTSE algorithm with BR is compared with a simple and intuitive call admission algorithm proposed in [29], which is proposed by. Daqing Gu & Jinyun Zhang. In this algorithm, the network reject new service request, including best effort service, when the channel is busy more than α % of time period. Algorithm in [29] will halt the contention process of the lowest priority when the channel is busy more than β % of time period. In scenario III, α is equal to 90 and β is equal to 95.

Then, best effort services are simulated with VoIP services to compare the blocking probabilities of both call admission algorithms and the BR algorithm is implemented with VTSE to verify the blocking probabilities of both algorithms. Fig.5.16 describes the assumption of interarrival of different services in the simulation period. Simulation period will last up to 500 seconds in scenario III. Numbers of best effort services requests are assumed 100 web browsing services and 10 bi-directional FTP services and interarrival times of these best effort services are randomly selected in the 0 to 100th second. Both algorithms will reject best effort services when the channel busy period is larger than 90% of a beacon interval. The simulation observation data are sampled form the 400th second to the 500th second. The observation period of VTSE in scenario III is 1 second.

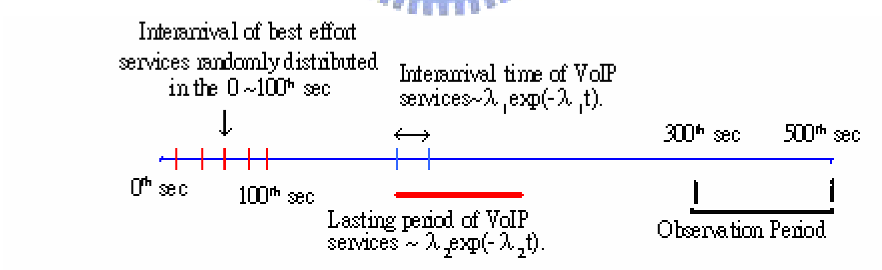

Fig. 5. 17 Assumption of the interarrival of different services

Fig5.17 compares the blocking probabilities of both algorithms when the mean interarrival time is adjusted from 1 to 5 seconds. The interarrival time of new VoIP service request is assumed based on exponential distribution which the mean interarrival time is adjustable. Every active VoIP service existing in the scenario III will last a period of time which is also set based on exponential distribution and the mean value is 120 seconds. VoIP services generate packets in CBR mode. It shows the VTSE can accept more VoIP services when system can support the QoS of all VoIP services.

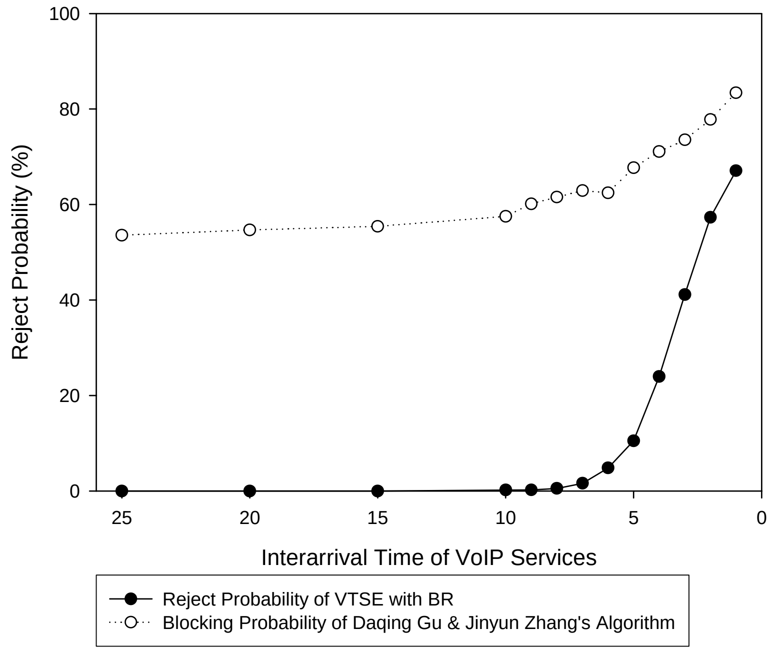

Fig. 5. 18 Reject probability of VTSE with BR and Daqing Gu & Jin yun Zhangs' algorithm

Fig.5.18 plots the erlang and packet loss rate of both algorithms. The erlang is defined as:

$$\frac{\text{Aggregated time period of all VoIP services stays in the system during the observation period}}{\text{observation period}} \quad (5.1)$$

Fig.5.18 shows the VTSE with BR can support more VoIP services under certain packet loss rate.

Fig5.19 plots the throughput of both algorithms. Fig5.19 shows that the throughput of VTSE with BR is close to Daqing Gu & Jinyun Zhangs' algorithm when less VoIP services requires to access the system. When the interarrival time of VoIP services decreases, the VTSE with BR accepts more VoIP services and it decreases the system throughput.

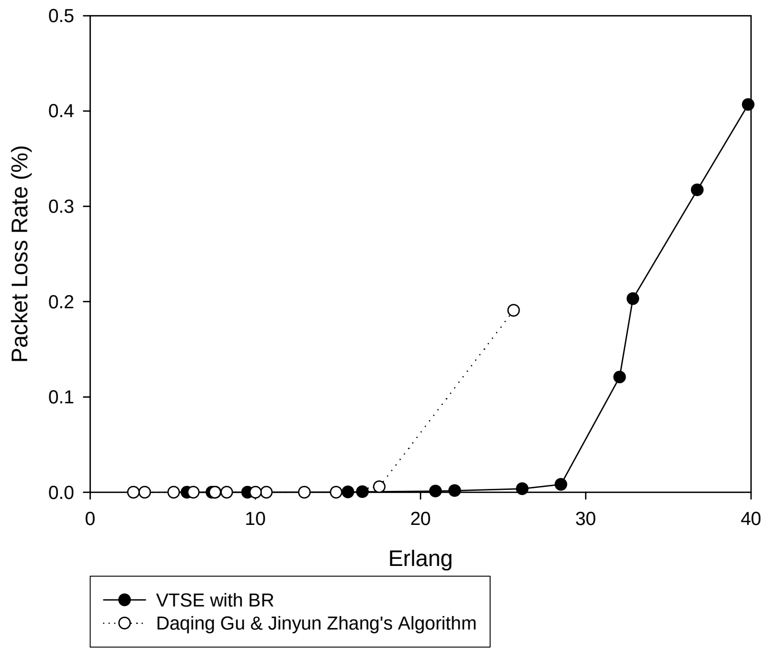

Fig. 5. 19 Erlang and packet loss rate of VTSE with BR and Daqing Gu & Jinyun Zhang' algorithm

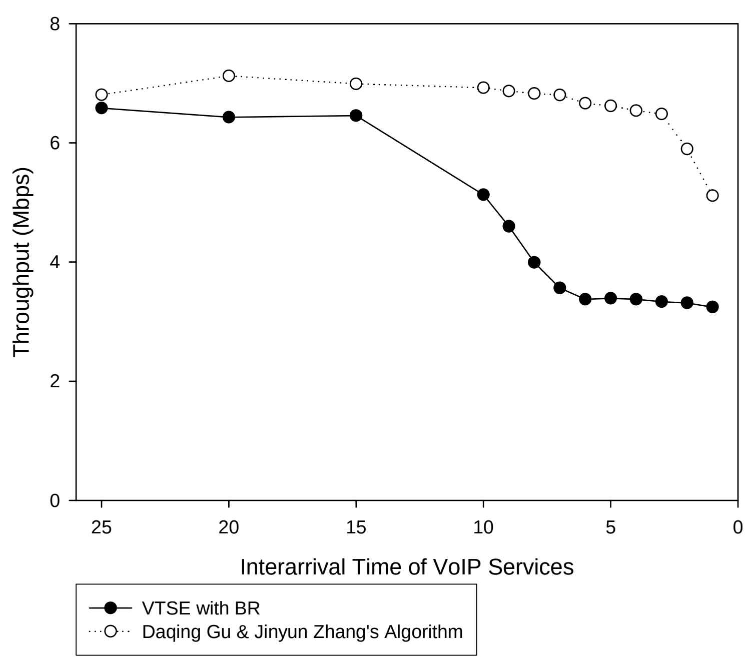

Fig. 5. 20 System throughput of VTSE with BR and Daqing Gu & Jinyun Zhangs' algorithm

5.3 Conclusion

Based on the observation of scenario I and II, VTSE can predict the "unsatisfied condition" closely. In scenario II, the influence of BR is examined. BR can increase the capacity of VoIP services and stabilize the delay jitter of VoIP services. The penalty of BR is the decrease of system throughput. In scenario III, the VTSE with BR can support more VoIP services compared with an intuitive algorithm. System throughput decreases when more VoIP services are allowed to access the system.

Chapter 6
Conclusion

This chapter is the conclusion of the simulation works and the future works. Section 6.1 is Conclusion and section 6.2 is future works.

6.1 Conclusion

This thesis proposed a QoS based call admission control and bandwidth reservation to preserve QoS of real-time services in the system. An event driven simulation platform is constructed to verify the proposed algorithm. The proposed call admission control can prevent the QoS of real-time services from degrading and the bandwidth reservation can stabilize the delay jitter of the existing VoIP services in the system.

6.2 Future Works

Handoff process is an important issue now in the WLAN system. So call admission control in the future has to concern how to resolve resources for handover users. Besides, MAC layer has to provide advanced resources allocation and bandwidth reservation with multiple input multiple output (MIMO) technology in the PHY layer since MIMO is defined in IEEE 802.11n.

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