行政院國家科學委員會專題研究計畫 成果報告

802.11e/802.16 閘道於點對點/網狀/轉傳模式下之服務品質提供
研究成果報告(精簡版)

計 畫 類 別 ： 個別型
計 畫 編 號 ： NSC 95-2221-E-004-005-
執 行 期 間 ： 95 年 08 月 01 日至 96 年 07 月 31 日
執 行 單 位 ： 國立政治大學資訊科學系

計畫主持人： 蔡子傑

計畫參與人員： 碩士班研究生-兼任助理：王川耘、陳彥賓、張志華

報 告 附 件 ： 出席國際會議研究心得報告及發表論文

處 理 方 式 ： 本計畫可公開查詢

中 華 民 國 96 年 10 月 31 日

行政院國家科學委員會補助專題研究計畫成果報告

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成果報告類型(依經費核定清單規定繳交)：■精簡報告　□完整報告

本成果報告包括以下應繳交之附件：
□赴國外出差或研習心得報告一份
□赴大陸地區出差或研習心得報告一份
■出席國際學術會議心得報告及發表之論文各一份
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執行單位：國立政治大學 資訊科學系

中　華　民　國　96　年　10　月　31　日

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計畫參與人員：王川耘、陳彥賓、張志華

此成果報告爲三篇論文的集節：

[1] Tzu-Chieh Tsai and Chuan-Yin Wang, "Routing and Admission Control in IEEE 802.16 Distributed Mesh Networks", in IEEE Fourth International Conference on Wireless and Optical Communications Networks" (WOCN 2007), July 2, 3 and 4, 2007, Singapore. (IEEE Catalog Number: 07EX1696, ISBN: 1-4244-1005-3, Library of Congress: 2007920880), (Engineering Index (EI) and EI Compendex).

[2] Tzu-Chieh Tsai,Chen Yan-Bin, "Modeling the Distributed Scheduler of IEEE 802.16 Mesh Mode", in 2006 National Symposim on Telecommunications, Kaohsiung, Taiwan, Dec 01~02, 2006.

[3] Tzu-Chieh Tsai, Chih-Hua Chang, "QoS Guarantee for IEEE 802.16 Integrating with 802.11e", prepared to submission.

首先我們先整理摘錄此三篇論文在本計畫中的關聯度與重要成果，後附上完整論文內容供參考。其中第三篇還未投稿，暫不附全文，請見諒。

另外，已發表的先前論文有兩篇：

[4] Tzu-Chieh Tsai, and Ming-Ju Wu, "An Analytical Model for IEEE 802.11e EDCA", in IEEE 2005 International Conference on Communications (ICC 2005 Wireless Networking), 16-20 May, 2005, Seoul, Korea. pp. 3474 - 3478.(ISSN: 0536-1486, EI)

[5] Tzu-Chieh Tsai, Chi-Hong Jiang, and Chuang-Yin Wang, "CAC and Packet Scheduling Using Token Bucket for IEEE 802.16 Networks", in Journal of Communications (JCM, ISSN 1796-2021), Volume : 1 Issue : 2, May, 2006. Page(s): 30-37. Academy Publisher.

一、Abstract

本研究主要研究主題為服務品質(Quality of Service, QoS)，以 802.16 與 802.11e 為主，探討 802.16 的三種傳輸模式、與 802.11e 的服務品質對應與保證該如何達成，還有如何對封包做排程的動作；除此之外，第一層調變技術的選擇與第三層的繞徑協定也部分包括在本研究計畫中，以期讓使用者在上網時得到更好的服務品質。

本研究對現有無線網路架構做了少許的修改，而得到一個類似於現有網路架構，但卻更富有挑戰性且成本更低、更方便的新網路架構：使用者的 mobile devices 以 802.11e 的網路卡連上 Access Point(AP)，而 AP 後端以 802.16 連上網際網路，這樣的 AP，我們稱為 802.11e/802.16 閘道器。在這樣的環境中，使用者的移動性(mobility)以及可能需要交遞(handoff)的問題，都可以由現有的 802.11 相關的處理方法來解決，由於關於這個主題的研究已經不勝枚舉，本研究在這部份將不多作著墨，而將專注於 802.16 相關議題與傳輸模式的研究。

在目前的 802.16-2004 版本中，定義了兩種傳輸模式，分別是:點對多點(PMP)與網狀(mesh)兩種，兩種模式各有優缺點，而我們預計會有新的傳輸模式：轉傳模式(relay)將結合兩者的優點，並會被廣泛的使用。為了本研究的一致性與完整性，本研究將對二種模式做個別深入的探討。

上述的二種傳輸模式，都將在我們的閘道器後端選用，因此分別產生了二種如何提供服務品質的問題，我們將對二種模式分別提出如何對應服務品質與保證的方法。

本研究將對所提出的網路架構，提出一套由下而上，跨越 OSI 標準中的第一、二、三層的設

計，其中在第二層，包括了 802.11e 以及 802.16 的所有可能模式，結合了目前炙手可熱的兩種科技:WLAN、WiMax，期望提供更好的服務品質與新的網路使用經驗給使用者，並成就一深度、廣度兼具的完整性研究。

關鍵詞：服務品質、802.11e、802.16、 點對多點、網狀、轉傳模式

二、緣由與目的、結果與討論

在 802.11e/802.16 Gateway 之 QoS 研究，分為幾個部份。

- 802.11e QoS
- 802.16 QoS for PMP Mode
- 802.16 QoS for Mesh Mode
- 802.11e/802.16 Integration

其中 802.11e QoS 的研究，之前就已有成果發表在[4]. 在那篇論文中，我們用 Markov Chain 去 model 各個 Access Category 的 channel access 情形，進而求出各個服務等級的 MAC delay，針對在不同的 loading 下。如此可以供作 CAC(Call admission control)的依據，而達成 QoS 的要求。

而 802.16 QoS for PMP Mode 的研究，也是之前我們就有成果發表在[5]. 在那篇論文中，我們用 token bucket 的方式，去控制 rtPS 等級的 traffic，進而能有效預估在符合 delay 要求條件下所需要的頻寬。藉此也發展出 scheduling algorithm 以及 CAC。在這篇論文，我們的 scheduling 與 CAC 特別設計下，任何等級的服務不會有 starvation 的情況。另外，我們也針對 Poisson traffic，利用 Markov Chain 分析，在 token bucket 模式下，該如何管理這些參數，以致於最有效率的達到 QoS。

接續[5]的 QoS 研究經驗，我們繼續研究如何擴充我們的作法到 Mesh Mode。首先， 必需瞭解 Mesh Mode 的運作模式與 PMP Mode 有何不同。我們發現在 access 方式大大不同，所以會多增加 access delay 的部份。因此我們研究了這個部份在[2]。而 token bucket based CAC and scheduling for Mesh Mode 則在[1]。最後我們再將 802.11e 的部份以及 802.16 作整合的 delay 與 CAC 的研究則在[3]。

接下來，我們就分別摘錄[1][2][3]的重要成果。完整論文則附於後面。

1. "Routing and Admission Control in IEEE 802.16 Distributed Mesh Networks"

1.1 Abstract

QoS provisioning in wireless mesh network has been known to be a challenging issue. In this paper, we propose a fixed routing metric (SWEB) that is well-suited in IEEE 802.16 distributed, coordinated mesh mode. Also, an admission control algorithm (TAC) which utilizes the token bucket mechanism is proposed. The token bucket is used for controlling the traffic patterns for easy estimating the bandwidth used by a connection. In the TAC algorithm, we apply the bandwidth estimation by taking the hop count and delay requirements of real-time traffics into account. TAC is designed to guarantee the delay requirements of real-time traffics, and avoid the starvations of low priority traffics. With the proposed routing metrics, the admission control algorithm and the inherent QoS support of the IEEE 802.16 mesh mode, a QoS-enabled environment can be established. Finally, extensive simulations are carried out to validate our algorithms.

1.2 Main Results

The proposed SWEB is compared with the ETX and the shortest path. The performance and delay of VBR traffics are compared across all three different metrics. The performance is given in figure 1. And figure 2 shows the delay.

As shown in the figure 1 and 2, when number of flows is reaching 25, some VBR flows are preempted by CBR flows. By simulation results, we claim that SWEB is a compromise of delay and throughput. But in figure 3, we can find that SWEB has best performance in jitter of real-time packets.

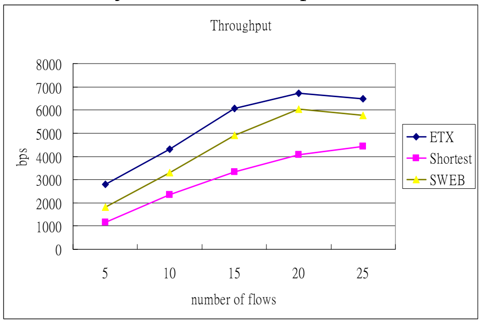

Fig. 1. Throughput of VBR flows.

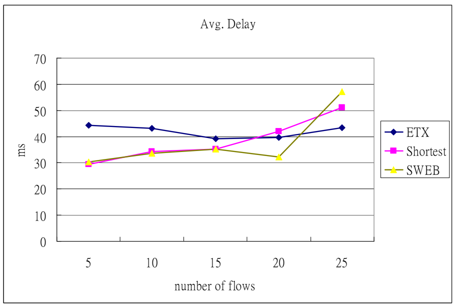

Fig. 2. Delay of VBR flows.

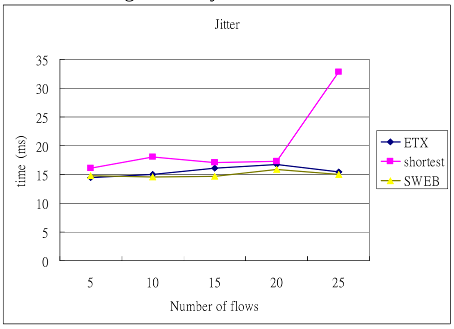

Fig. 3. Jitter of VBR flows.

In TAC algorithm, the minimum usage of each traffic class must be set. In the simulations, the *CBR_min*, *VBR_min* and *BE_min* are set as 10, 40 and 75 timeslots, respectively. Also, the parameters of token bucket are shown in table III.

We compare the throughput in figure 4 and figure 5. In figure 4, BE traffics suffers from preemption from higher priority traffic class, therefore, receiving low throughput when network is heavily-loaded. By applying the CAC algorithm in figure 5, the BE flows has the guaranteed throughput by setting the minimum usage. The preemption occurs only in down-graded flows.

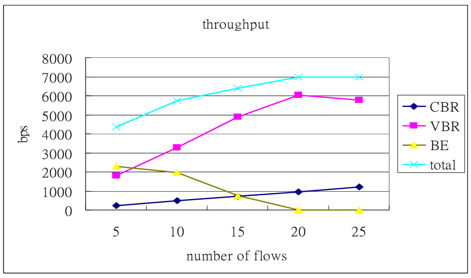

Fig. 4. Throughput for the original IEEE 802.16 mesh mode.

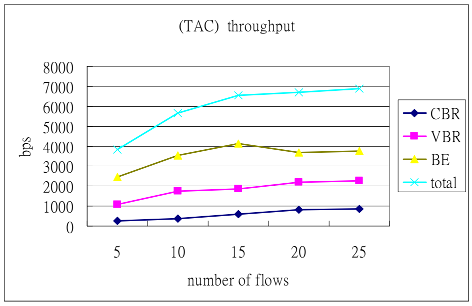

Fig. 5. Throughput when TAC is applied.

2. "Modeling the Distributed Scheduler of IEEE 802.16 Mesh Mode"

2.1 Abstract

The IEEE 802.16 standard is a protocol for wireless metropolitan networks. IEEE 802.16 MAC protocol supports both of PMP (point to multipoint) and Mesh mode. In the mesh mode, all nodes are organized in a similar ad-hoc fashion and calculate their next transmission time based on the scheduling information performed in the control subframe. Each node has to compete with each other to win the time slots of opportunities for the subsequent advertisement of the scheduling messages to its neighbors. This behavior does not depend on all of past history. In other words, it is a "Time Homogeneous" and suitable for being modeled by stochastic process. In this study, we will model this scheduling behavior by queuing process, and apply the Markov Chain to estimate its average delay time which a node keep waiting until it win the competition.

2.2 Main Results

We validate our model by the following. The transmission behavior is simulated by our C code. The mathematic evaluations are computed by the MATLAB 7.0.

The result is shown as Fig.6. With this figure, it shows our mathematical model approaches the simulation result.

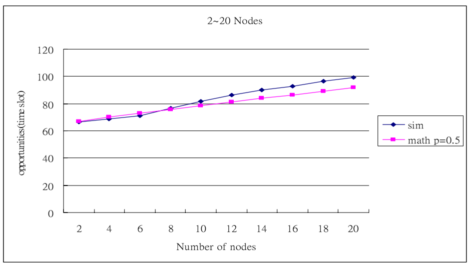

Fig. 6: The delay time of opportunities between simulation and mathematic model

The error rate is shown as Fig.7. It shows the error is under 10% while the nodes of number between 2 to 20.

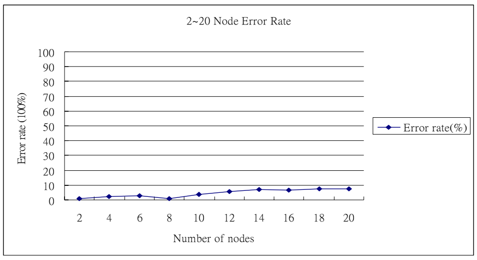

Fig. 7: The error rate between simulation and mathematic model

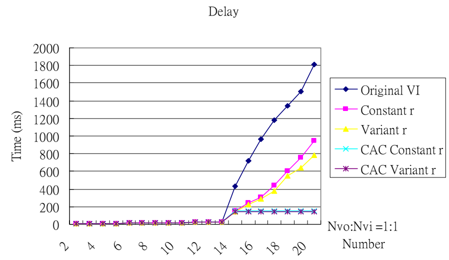

Figure 8: Delay time (N_{VO} : N_{VI} = 1 : 1)

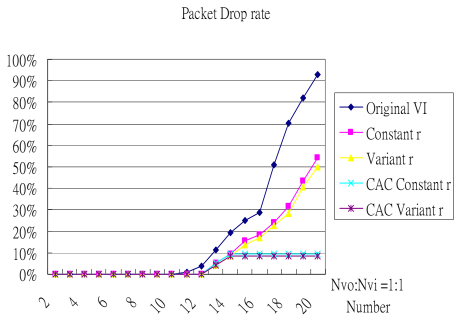

Figure 9: Packet drop rate (N_{VO} : N_{VI} = 1 : 1)

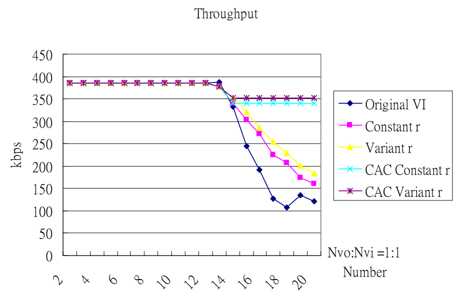

Figure 10: Throughput (N_{VO} : N_{VI} = 1 : 1)

3. "QoS Guarantee for IEEE 802.16 Integrating with 802.11e"

3.1 Abstract

IEEE 802.16 and 802.11e both provide Quality of Service (QoS), but the MAC of betweens is different. Ensuring the QoS guarantee, we use a Markov Chain model to analyze the 802.11e EDCA delay time under variance number of connections. Therefore, we can employ a CAC mechanism constraining the number of connections to guarantee the delay requirement. Further, considering the delay requirement and the bandwidth, we use a Token Bucket mechanism to throttle the traffic output that ensures the delay and bandwidth to be satisfied. And our Token Bucket mechanism can tune the token rate automatically by bandwidth requirement. Finally, we use the Packet Drop mechanism to improve throughput.

3.2 Main Results

We validate our methos through comparing delay, throughput and packet drop rate with the simulator Qualnet. The VI (rtPS) delay is what we mainly concern about. Therefore, we just show the VI (rtPS) delay time and the result.

There are five lines we compare with. Original VI means we just run the scenario without any change in protocol. Constant r represents using Token Bucket mechanism but without tuning the token rate r. Variant r expresses using Token Bucket mechanism and it will tune the token rate r. CAC Constant r and CAC Variant r are similar to Constant r and Variant r but with CAC mechanism. Expect the Original VI, otherwise with our packet drop mechanism. The simulation result is as Figure 8, Figure 9, and Figure 10.

三、計畫成果自評

本計畫的 scope 相當大，但是我們先前已有很紮實的研究經驗與成果。因此不論在 802.11e 的 QoS 或 802.16 以 token bucket based 的 QoS 都可循經驗再作進一步的擴充與整合。

這個計畫花較多的時間是在研讀 802.16 Mesh mode 的 draft。由於是較新的文件，一些其他輔助資料較欠缺，不過經過一番努力，還是把它完成了。我想這個部份的成果應可作為其他機構在研究這個主題時，一個很好的參考。

另外，我們在整合 802.11e 與 802.16 時，用 Markov Chain 去 model delay analysis，碰到一些瓶頸，因為不同的架構下，要整合成一套相容的 QoS 管理機制，本就有困難度。不過，我們還是把它達成了，雖不盡完美但尚可接受。日後可供 WiMAX 業者與 WiFi HotSpot 業者結盟一個非常好的管理平台作參考。

附件二

可供推廣之研發成果資料表

■ 可申請專利　■ 可技術移轉　　　　日期：96 年 10 月 31 日

國科會補助計畫	計畫名稱：802.11e/802.16 閘道於點對點/網狀/轉傳模式下之服務品質提供 計畫主持人：　蔡子傑 計畫編號：　NSC 95-2221-E-004-005　　　學門領域：資訊
技術/創作名稱	**Routing and Admission Control in IEEE 802.16 Distributed Mesh Networks**
發明人/創作人	蔡子傑，王川耘
技術說明	中文：我們 針對 IEEE 802.16 協調分散式之網狀網路提出一允入控制之演算法。在此類網路中，控制子訊框交換各站台之排程訊息，並預留資料子訊框之時槽作為實際資料傳輸之用。我們利用令牌桶機制來控制網路訊流之流量特徵，如此可簡單的估計各訊流所需之頻寬。我們使用了所提出的頻寬估計方法，並一起考慮各訊流之跳接數與延遲時間之需求，提出的允入控制演算法能夠保證即時性串流之延遲時間需求，且可避免低等級訊流發生飢餓情形。模擬結果顯示，所提出的允入控制方法可以有效的把超過延遲時間需求之即時性訊流封包數目降低，並且低等級訊流在網路負載大時仍然可以存取頻道。 英文：We propose a routing metric (SWEB: Shortest-Widest Efficient Bandwidth) and an admission control (TAC: Token bucket-based Admission Control) algorithm under IEEE 802.16 coordinated, distributed mesh networks. In such network architectures, all scheduling messages are exchanged in the control subframes to reserve the timeslots in data subframes for the actual data transmissions. The token bucket mechanism is utilized to control the traffic pattern for easily estimating the bandwidth of a connection. We apply the bandwidth estimation and take the hop count and delay requirements into consideration. TAC is designed to guarantee the delay requirements of the real-time traffic flows, and avoid the starvation of the low priority ones. Simulation results show that TAC algorithm can effectively reduce the number of real-time packets that exceed the delay requirements and low priority flows still can access the channel when the network is heavily-loaded.
可利用之產業 及 可開發之產品	相關 802.16 的產業

技術特點	利用 token bucket 來管理與控制頻寬，並提出 CAC 與 scheduling 機制來達成 QoS 的要求。
推廣及運用的價值	WiMAX 的執照已陸續在發放，業者也會建置越來越多的基地台。然而不管是 PMP 或 Mesh Mode，QoS 的管理是 WiMAX 相當重要的課題。因此本研究成果可供業者一個很重要的參考依據。

※ 1. 每項研發成果請填寫一式二份，一份隨成果報告送繳本會，一份送 貴單位研發成果推廣單位（如技術移轉中心）。

※ **2. 本項研發成果若尚未申請專利，請勿揭露可申請專利之主要內容。**

※ 3. 本表若不敷使用，請自行影印使用。

附論

已發表之論文全文

[1] Tzu-Chieh Tsai and Chuan-Yin Wang, "Routing and Admission Control in IEEE 802.16 Distributed Mesh Networks", in IEEE Fourth International Conference on Wireless and Optical Communications Networks" (WOCN 2007), July 2, 3 and 4, 2007, Singapore. (IEEE Catalog Number: 07EX1696, ISBN: 1-4244-1005-3, Library of Congress: 2007920880), (Engineering Index (EI) and EI Compendex).

[2] Tzu-Chieh Tsai,Chen Yan-Bin, "Modeling the Distributed Scheduler of IEEE 802.16 Mesh Mode", in 2006 National Symposim on Telecommunications, Kaohsiung, Taiwan, Dec 01~02, 2006.

Routing and Admission Control in IEEE 802.16 Distributed Mesh Networks

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***Abstract*—QoS provisioning in wireless mesh network has been known to be a challenging issue. In this paper, we propose a fixed routing metric (SWEB) that is well-suited in IEEE 802.16 distributed, coordinated mesh mode. Also, an admission control algorithm (TAC) which utilizes the token bucket mechanism is proposed. The token bucket is used for controlling the traffic patterns for easy estimating the bandwidth used by a connection. In the TAC algorithm, we apply the bandwidth estimation by taking the hop count and delay requirements of real-time traffics into account. TAC is designed to guarantee the delay requirements of real-time traffics, and avoid the starvations of low priority traffics. With the proposed routing metrics, the admission control algorithm and the inherent QoS support of the IEEE 802.16 mesh mode, a QoS-enabled environment can be established. Finally, extensive simulations are carried out to validate our algorithms.**

***Index Terms*—IEEE 802.16, wireless mesh networks, WiMAX**

I. Introduction

As wireless technology evolves, IEEE 802.16[1], or WiMAX (Worldwide Interoperability for Microwave Access), appears to be a great competitor to IEEE 802.11 or 3G networks for its wide coverage and high data rate. In the IEEE 802.16 standards, meshing functionality is included as an optional mode. We propose a simple routing metrics called Shortest Widest Effective Bandwidth (SWEB) and a Token bucket-based Admission Control (TAC) in the IEEE 802.16 mesh networks.

IEEE 802.16 is a standard that aims at the use of wireless metropolitan area network (WMAN). Two modes are defined in the standard: PMP (Point to Multi-Point) and mesh mode. In PMP mode, the network architecture is similar to the cellular network. That is, one base station (BS) is responsible for all its subscriber stations (SS). The transmission can occur only between BS and SS. In mesh mode, the networks architecture is similar to the ad-hoc networks. In other words, each SS can be a source node and a router at the same time. The transmission can occur between any two stations in the network.

IEEE 802.16 mesh network is time-slotted. Connections must reserve timeslots in advance for the actual transmissions. As IEEE 802.16 is mostly used as the network backhaul, the network traffics mostly occur between the BS and SS. Therefore, with appropriate routing algorithm, the topology can be reduced into a routing tree. QoS of a connection along the path from the source station to the base station can be provided.

The remaining parts of this paper are organized as follows: Section II gives the background knowledge of token bucket mechanism and details of IEEE 802.16 mesh mode. Section III includes the related works. In section IV and V, the SWEB and TAC are proposed. Simulation results are given in section VI. Finally, we conclude this paper in section VII.

II. Background

A. *Token Bucket mechanism*

Token bucket is a mechanism that controls the network traffic rate injecting to networks. It works well for the “bursty” traffics. Token bucket mechanism needs two parameters: token rate *r* and bucket size *b*. Figure 1 shows how the token bucket

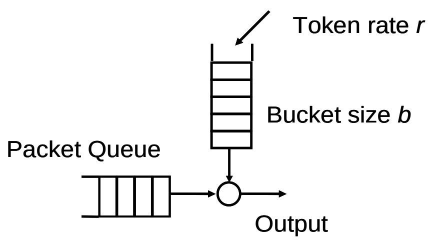

Fig. 1. Token bucket mechanism.

mechanism works.

Each packet represents a unit of bytes or a packet data unit. A packet is not allowed to be transmitted until it possesses a token. Therefore, in the time duration *t*, the maximum data volume to be transmitted will be

$$r \cdot t + b$$

We adopt the token bucket mechanism to estimate the bandwidth required for each connection in IEEE 802.16 mesh networks

B. *IEEE 802.16 mesh mode*

IEEE 802.16 mesh network is time-slotted. That is, the time is divided into equal-length time frames. And each time frame comprises one Control subframe and one Data subframe. Control subframe carries the control messages for the

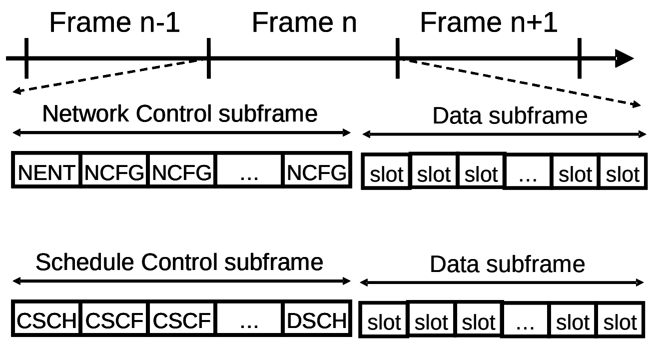

Fig. 2. Frame structures of IEEE 802.16 mesh mode.

scheduling, entry of new stations, exchanging of basic network parameters, etc. The data frame is composed by the time-slots for the actual data transmissions. The frame structure is given in figure 2.

There are two kinds of control subframe: Network Control subframes and Schedule Control subframes. In Network Control subframes, MSH-NENT (Mesh-Network Entry) messages are used to provide the entries of new-coming stations. MSH-NCFG (Mesh-Network Configure) messages are sent by each station periodically to exchange the basic parameters of networks, such as: the identifier of the BS, hops to the BS, and neighbor number of the reporting stations…etc.

Two scheduling modes are defined in IEEE 802.16 mesh mode: centralized and distributed modes. In centralized scheduling mode, the BS is in charge of all the transmissions happening in the mesh network. The resources are allocated by the BS with the MSH-CSCH (Mesh-Centralized Scheduling) messages and MSH-CSCF (Mesh-Centralize Scheduling and Configure) messages. In distributed mesh mode, the scheduling information is carried by MSH-DSCH (Mesh-Distributed Scheduling) messages, whose transmission time is determined by the mesh election algorithm given in the standard. MSH-DSCH has four information elements (IEs): scheduling IE, request IE, availability IE, and grant IE. The Scheduling IE carries the information of next-transmission time used in the mesh election algorithm. The other three IEs are employed in the three-way handshake:

1. The MSH-DSCH:request is made along with MSH-DSCH:availability, which is used to indicates the potential timeslots of the source station.
2. MSH-DSCH:grant is sent in response indicating a subset of the suggested availabilities that fits, in possible, the request.

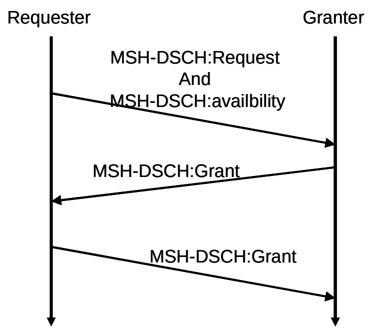

Fig. 3. Three-way handshake.

3. MSH-DSCH:grant is sent by the original requester containing a copy of the grant from the requester, to confirm the schedule.

After the three-way handshake indicated in figure 3, the reservation of timeslots in the data subframe is completed.

In both centralized and distributed mesh modes, the QoS can be supported by the fields of the CID (Connection Identifiers) that is associated with each connection. There are three fields that explicitly define the service parameters:

1. Priority: This field simply defines the service class of the connection
2. Reliability: To re-transmit or not.
3. Drop Precedence: The likelihood of dropping the packets when congestion occurs.

III. Related Work

H. Shetiya and V. Sharma [2] proposed the algorithms of routing and scheduling under IEEE 802.16 centralized mesh networks. The routing metric is based on the evaluation of queue length on each station. The routing is fixed routing, which reduces the topology into a tree. The scheduling algorithm is based on a mathematical model to allocate enough timeslots among the traffic flows. And our previous work [3] that focus on the call admission control and packet scheduling in the IEEE 802.16 PMP mode. In this paper, a mathematical model is proposed to characterize the packets of different traffic flows.

Some other researches also focus on the IEEE 802.16 mesh mode: F. Liu et al. [4] proposed a slot allocation algorithm based on priority, which is to achieve QoS. Z. J. Haas et al. [5] proposed an approach to increase the utilization of IEEE 802.16 mesh mode. They have adopted a cross-layer design in their work. M. Cao et al. [6] proposed a mathematical model and an analysis of IEEE 802.16 mesh distributed scheduler, mostly on the mesh election algorithm.

Douglas, S, J. De Couto et al. [7] proposed a new routing metrics called “ETX”, short for “Expect Transmission Count”. ETX is suitable for wireless networks and is able to fit in any routing algorithms like DSR, DSDV … etc.

IV. Routing Metrics: SWEB

a new routing metrics called SWEB (Shortest-Widest Efficient Bandwidth) is proposed, which considers three parameters: packet error rate, $P_{i,j}$, capacity $C_{i,j}$ over the link (i,j) and the hop count, h, from the source to the destination. The packet error rate can be retrieved by the exchanging of MSH-DSCH messages, which is associated with a unique sequence number. The lost or error MSH-DSCH messages can be detected. And the link capacity can be also known by the burst profile indicated in the MSH-NCFG messages. In MSH-NCFG messages, the hop count for a station to base station is also given. Therefore, we argue that our SWEB metrics is especially suitable for IEEE 802.16 mesh networks.

The efficient bandwidth of a link (i,j) can be calculated as:

$$C_{i,j}(1-p_{i,j})$$
(1)

However, since the flow that comes in and leaves a node shares the bandwidth. Equation (1) should be divided by two to represent the available bandwidth. Therefore, the end-to-end available bandwidth is:

$$\frac{\min(C_{1,2}\cdot(1-p_{1,2}),C_{2,3}\cdot(1-p_{2,3}),\ldots,C_{i,j}\cdot(1-p_{i,j}))}{2}$$
(2)

By using (2), we define our SWEB metrics for all potential paths as:

$$SWEB=\frac{\min(C_{1,2}\cdot(1-p_{1,2}),C_{2,3}\cdot(1-p_{2,3}),\ldots,C_{i,j}\cdot(1-p_{i,j}))}{2}\cdot\frac{1}{h}$$
(3)

The path with the largest path metric will be chosen.

V. Admission Control Algorithm: TAC

Our Token bucket-based Admission Control (TAC) has two essential parts. First, the bandwidth used by a connection must be estimated well. Second, the bandwidth estimation is used for implementing the admission control algorithm.

A. Bandwidth Estimation

If all the connections are under the control of token bucket mechanism, the bandwidth used with a time frame can be estimated as:

$$\frac{r_i\cdot f+b_i}{f}$$
(4)

The r_i and b_i is the token rate and bucket size that associated with a connection i, respectively. f is the frame length. However, (3) is over-estimated since the transmission burst does not happen in every time frame. To better estimate the bandwidth, consider the scenario in figure 4.

Let the hop count and transmission deadline of the flow in Fig. 6 is 3 and 7*f*, respectively. Assume that the transmission burst occurs in time interval [*t*+5*f*, *t*+6*f*] and tokens stored in the

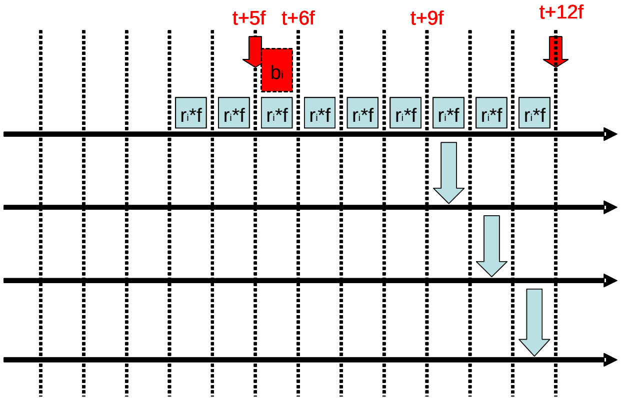

Fig. 4. Transmission Deadline of b_i Bits of Data.

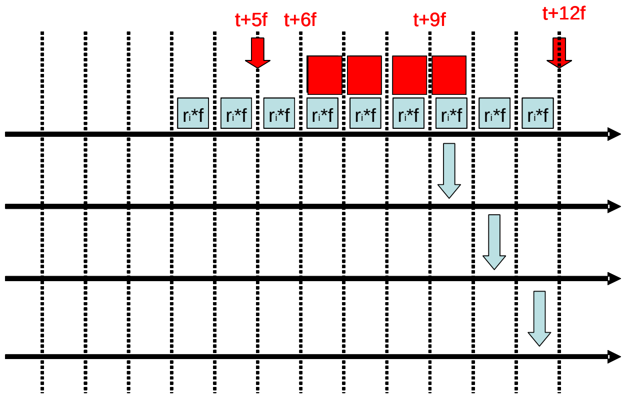

Fig. 5. Sharing b_i Bits of Data.

bucket are completely consumed. In order to satisfy the delay requirement, these bi bits of data must be sent in [*t*+9*f*, *t*+10*f*] at latest. Therefore, the frames from *t*+6*f* to *t*+10*f* can be used for sharing the bi bits, as in figure 5.

Generally speaking, in order to meet the delay requirement, d_i, of real-time traffics, packets generated at time *t* have to be sent after m_i frames after *t*, where

$$m_i=\left\lfloor\frac{d_i}{f}\right\rfloor-h$$
(4)

These m_i frames can be used to share the b_i bits of data. Therefore, the maximum volume of data that can be sent in any given frame is:

$$r_i\cdot f+\frac{b_i}{m_i}$$
(5)

We use (5) as bandwidth estimation of a flow.

B. Admission Control

We use the above-mentioned bandwidth estimation to implement the TAC algorithm. In TAC algorithm, the minimum usage of timeslots by each connection is defined. They are: *CBR_min*, *VBR_min* and *BE_min*. When a station receives a MSH_DSCH:Request, it examines whether the current usage of each class exceeds their minimum usage or not. If it is, the new-coming flow will be marked as downgraded flows. If a MSH-DSCH:Request comes in, the downgraded flows have bigger possibilities to be preempted. On the other

TABLE I
QoS Mappings

	Priority (3 bits)	Reliability (1 bit)	Drop Precedence (2 bits)
CBR	7	0	0
CBR_DG	4	0	1
VBR	6	0	0
VBR_DG	3	0	2
BE	5	1	0
BE_DG	2	1	3

hand, if the current usage does not exceed its minimum usage, the flow will not be downgrade and have bigger change to preempt other downgraded flows.

Since the service levels in IEEE 802.16 mesh mode are identified in the fields of CID (Connection Identifiers), we have the QoS mapping in Table I. With the mappings in table I, the down-graded flows can be marked. And by this information, we develop our TAC algorithm as follows:

1.) A new flow with its *BW_req* (Bandwidth request) in the unit of data timeslots. And set *BW_avail* as the total empty slot number. (*BW_avail* stands for available bandwidth)
2.) The station that handles the request checks if the *BW_req*<*BW_avail* or not. If yes, go to step 3. Or else, go to step 4.
3.) The station determines to downgrade the flow or not, by comparing the current usage and the minimum usage of the traffic class.
4.) The station checks if the current usage exceeds the minimum usage of the traffic class. If yes, the flow shall be rejected. Or else, go to step 5.
5.) Check the timeslots used by downgraded flows in the order of *BE_DG*, *VBR_DG*, and *CBR_DG*. If there is no such timeslots, the request is rejected. Or else, set this timeslots empty, which means to preempt this timeslots. Updating the value of *BW_avail*. Go to step 2.

VI. Simulation Results

The simulations are conducted in a 16-node topology, and the simulation area is a 4 km * 4 km square. The radio range is set as 1.5 km in radius. The frame length is chosen to be 8 ms.

TABLE II
THE PARAMETERS OF QPSK

QPSK coding rate	3/4
OFDM symbols in a frame	676
OFDM symbols in a control subframe	16
OFDM symbols in a data subframe	660
OFDM symbols in a timeslot	4
Number of data timeslots	165
Capacity of a timeslot	144 bytes

In the simulations, QPSK is chosen to be the modulation method. The details of QPSK are given in table II.

The data rate of the CBR traffic is 64 kbps, with the 960-bit packet size in the packet interval of 15 ms. The VBR traffic is sending at the average speed of 400 kbps. The mean packet size is 16000 bits sending at the interval of 40 ms. The packet size of BE traffics is 8000 bits and is sent every frame (8 ms).

A. Routing

The proposed SWEB is compared with the ETX [7] and the shortest path. The performance and delay of VBR traffics are compared across all three different metrics. The performance is given in figure 6. And figure 7 shows the delay.

As shown in the figure 6 and 7, when number of flows is reaching 25, some VBR flows are preempted by CBR flows. By simulation results, we claim that SWEB is a compromise of

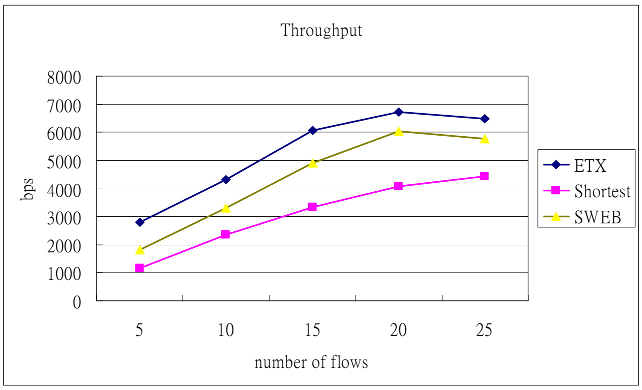

Fig. 6. Throughput of VBR flows.

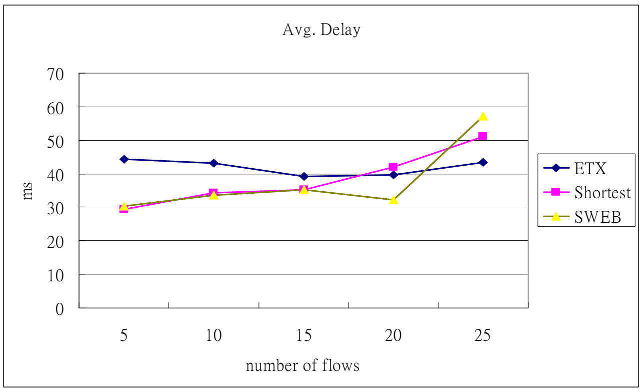

Fig. 7. Delay of VBR flows.

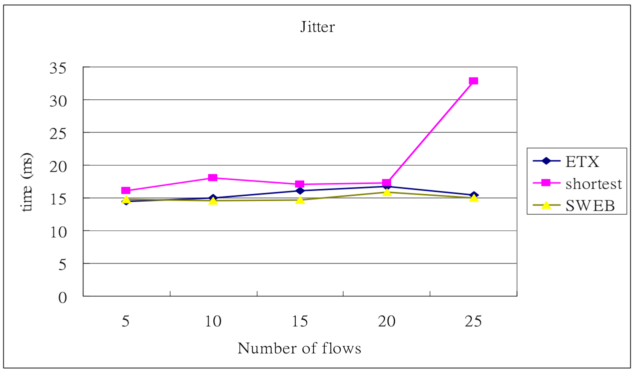

Fig. 8. Jitter of VBR flows.

TABLE III
TOKEN BUCKET MECHANISM PARAMETERS

	Token rate (bytes / frame)	Bucket size (bytes)	Delay requirements
CBR	120	8	40 ms
VBR	1500	500	80 ms
BE	7500	250	--

delay and throughput. But in figure 8, we can find that SWEB has best performance in jitter of real-time packets.

B. Admission Control

In TAC algorithm, the minimum usage of each traffic class must be set. In the simulations, the *CBR_min*, *VBR_min* and *BE_min* are set as 10, 40 and 75 timeslots, respectively. Also, the parameters of token bucket are shown in table III.

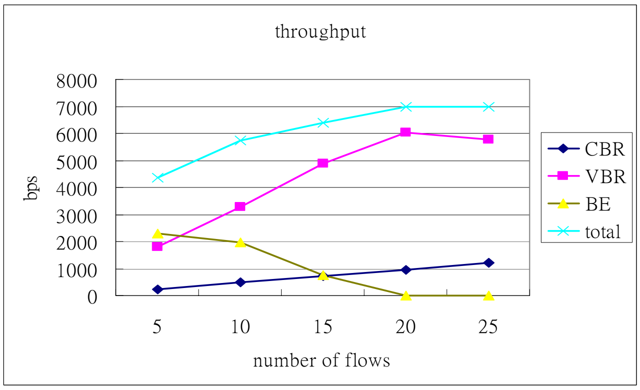

Fig. 9. Throughput for the original IEEE 802.16 mesh mode.

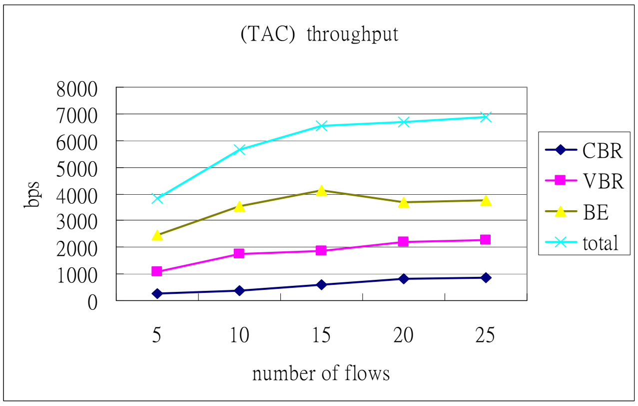

Fig. 10. Throughput when TAC is applied.

We compare the throughput in figure 9 and figure 10. In figure 9, BE traffics suffers from preemption from higher priority traffic class, therefore, receiving low throughput when network is heavily-loaded. By applying the CAC algorithm in figure 10, the BE flows has the guaranteed throughput by setting the minimum usage. The preemption occurs only in down-graded flows.

In figure 11 and figure 12, the statistics is gathered to discuss the percentage of real-time packets exceeds the delay requirements. As in figure 11, around 12% of VBR-packets exceed the delay requirements when the number of flow is 25. However, in figure 12 it is reduced to around 7% for only VBR-downgraded flows. It can be expected that for all VBR flows (VBR and VBR-downgraded), the ratio would be lower than 7%.

VII. Conclusions

In this paper, we proposed a new routing metric, SWEB, and an admission control algorithm, TAC for IEEE 802.16 mesh networks. SWEB is applied in static routing environment and yields the good throughput, delay and jitter performance. The TAC algorithm prevents the starvation of low-priority traffic flows and guarantees the delay requirements of the real-time flows. By SWEB and TAC, a QoS-enabled network environment can be realized with IEEE 802.16 mesh mode in the MAC layer. Thus, end-users will have better experience and convenience in utilizing the networks.

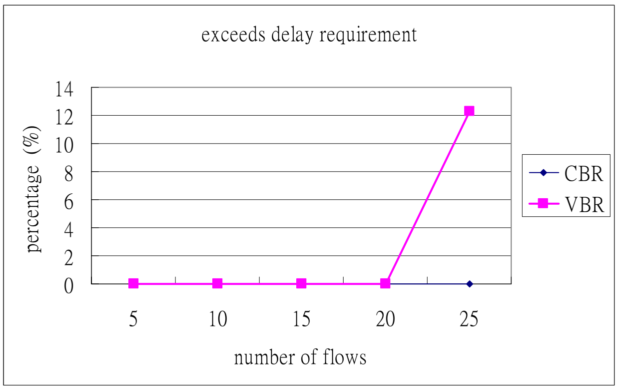

Fig. 11. The ratio of the realtime packets that exceeds the delay requirements for the original IEEE 802.16 mesh mode.

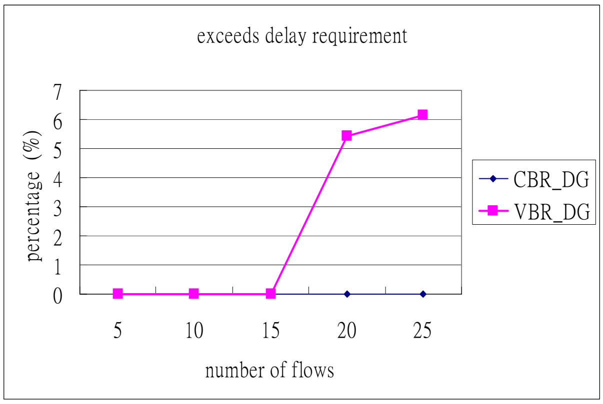

Fig. 12. The ratio of the realtime packets that exceeds the delay requirements when TAC is applied.

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Modeling the Distributed Scheduler of IEEE 802.16 Mesh Mode

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Abstract

The IEEE 802.16 standard is a protocol for wireless metropolitan networks. IEEE 802.16 MAC protocol supports both of PMP (point to multipoint) and Mesh mode. In the mesh mode, all nodes are organized in a similar ad-hoc fashion and calculate their next transmission time based on the scheduling information performed in the control subframe. Each node has to compete with each other to win the time slots of opportunities for the subsequent advertisement of the scheduling messages to its neighbors. This behavior does not depend on all of past history. In other words, it is a "Time Homogeneous" and suitable for being modeled by stochastic process. In this study, we will model this scheduling behavior by queuing process, and apply the Markov Chain to estimate its average delay time which a node keep waiting until it win the competition.

1. Introduction

The IEEE 802.16 standard [1, 2] "Air Interface for Fixed Broadband Wireless Access Systems", also known as WiMAX, targets at providing last-mile wireless broadband access in metropolitan area networks. IEEE 802.16 is a wireless network, which has the high capacity to cover more broad geographic areas without the costly infrastructure development. The technology may prove less expensive to deploy and may lead to more ubiquitous broadband access [3]. The clients also can connect to the IEEE 802.16 by adopting various existing wireless solutions, such as IEEE 802.11 (WiFi). IEEE 802.16 provides a cheaper and more ubiquitous solution to connect home or business to Internet. Much attention was paid to the IEEE 802.16 issues in recent years and a lot of industries formed a WiMAX Forum in order to certify compatibility and interoperability of various 802.16 products.

The 802.16 mesh mode topology is depicted as Fig.1. There are many SSs in this topology which terminals, such as PDAs, notebooks or cellular phones, can be connected to via 802.11 or other protocols. The mesh mode is organized throughout these SSs and BSs. The link coverage is expanded under mesh network. Certain SSs are responsible to connect to the BSs. By these BSs, they connect to the backhaul or internet. Besides, a Markov Chain to model this distributed scheduling of mesh mode as well as a mathematical model are proposed in this paper to evaluate the average delay time.

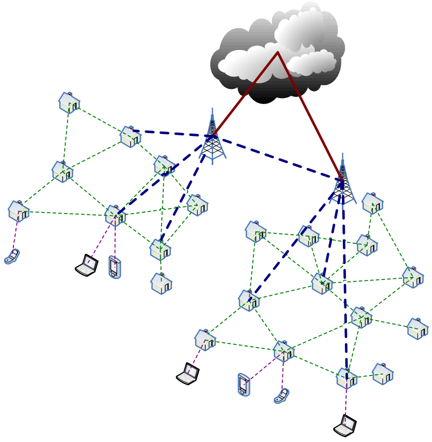

Fig. 1: IEEE 802.16 Mesh Mode Topology

2. Analyze IEEE 802.16 Distributed Scheduling Algorithm

Before we model the Distributed Scheduler in IEEE 802.16 mesh mode, we have to know its behavior.

A. 802.16 Mesh MAC Frame Structure

The IEEE 802.16 defined the mesh frame structure as a convenience to organize the mesh network. The frame is divided into two subframes. One is the data subframe, the other is control subframe. Every control subframe consists of sixteen transmission opportunities, which may be imaged as a "time slot", and every transmission opportunity equals seven OFDM symbols time.

There are two control subframe types in a control subframe. One is network control subframe that creates and maintains the cohesion between different systems. It also provides a new node to gain synchronization and initial network entry into a mesh network. The other is a subframe that to coordinate scheduling of data transfers in system, called schedule control subframe. The scheduling information is encapsulated here. Frames with the network control subframe occur periodically and all the other frames contain schedule control subframes along the network control subframe.

Two messages “MSH-NENT” and “MSH-NCFG” are used in the network control subframe. MSH-NENT means a mesh network entry, which is a message for a new node to gain synchronization and initial network entry into a mesh network; furthermore, MSH-NCFG means a mesh network configuration, provides a basic information of communication between nodes in different nearby networks. On the side, in the schedule control subframe, “MSH-CSCH” and “MSH-DSCH” means the mesh network centralized scheduling and the mesh network distributed scheduling, separately. MSH-DSCH is the key point that this paper will concentrate on.

We have introduced that every control subframe consists of sixteen transmission opportunities. Nevertheless, they are just the opportunities to own these time slots, but the really time slot occupied is indicated by “MSH-CTRL-LEN”. MSH-CTRL-LEN is a field saved in the MSH-NCFG message to express the control subframe length. MSH-DSCH-NUM is also saved in the MSH-NCFG message to express the number of MSH-DSCH opportunities in the schedule control subframe. Of course, what’s left after MSH-DSCH-NUM is subtracted from MSH-CTRL-LEN becomes the number of MSH-CSCH opportunities. All of the parameters we introduced thus far are depicted in Fig. 2.

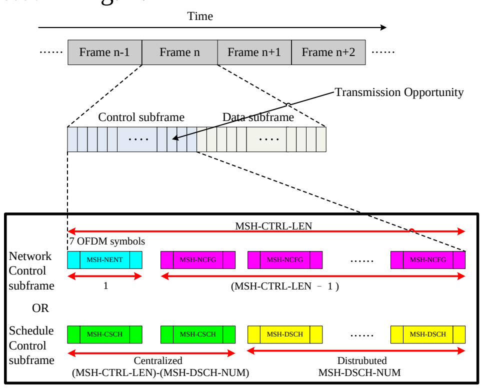

Fig. 2: Network Control subframe and Schedule Control subframe

B. Next Transmission Time and Transmission Holdoff Time

In this section, we will introduce parts of the terminologies and abbreviations in the IEEE 802.16 specification.

The schedule information for each node is described by two parameters Next Xmt Time and Xmt holdoff Time. In the IEEE 802.16 specification, Next Xmt Time is not employed directly. It uses Next Xmt Mx to calculate the Next Xmt Time. It doesn’t use Xmt holdoff Time, neither. It uses Xmt holdoff exponent to calculate the Xmt holdoff Time. As the Fig. 3 shows, Next Xmt Mx and Xmt holdoff exponent are two parameters in the MSH-DSCH message to perform the schedule information. So that whenever a node transmits MSH-DSCH message, every node has the schedule information of its neighbors.

Syntax	Size	Notes
MSH-DSCH_Scheduling_IE() {		
Next Xmt Mx	5 bits	
Xmt holdoff exponent	bits	
No. SchedEntries	bits	
for (i=0; i<No_SchedEntries; ++i) {		
Neighbor Node ID	16 bits	
Neighbor Next Xmt Mx	5 bits	
Neighbor Xmt holdoff exponent	3 bits	
}		
}		

Fig. 3: Next Xmt Mx and Xmt holdoff exponent in the MSH-DSCH (source: IEEE 802.16-2004)

A node has to decide the next transmission time to know when to transmit the next MSH-DSCH message. There is a special terminology employed in the IEEE 802.16 specification to describe this transmission duration named “Eligible Interval”. This next transmission time is denoted as Next Xmt Time and calculated from Next Xmt Mx. Assume “Next” is denoted as Next Xmt Time of an observed node; “Mx” and “x” means its corresponding Next Xmt Mx and Xmt holdoff exponent separately. Duration of Next Xmt Time could be shown as the following formula (1) defined in the standard. By the observation of this formula, we know 2^x is the length of “Next”. “x” is clearly an exponential value to express the length of “Next”.

$$2^x \cdot Mx < Next \le 2^x \cdot (Mx + 1) \tag{1}$$

Xmt Holdoff Time is also a special terminology applied in the IEEE 802.16 specification to indicate that this node is not eligible to transmit messages. Assume “Holdoff” is denoted as Xmt Holdoff Time of an observed node; “x” means its corresponding Xmt holdoff exponent. Then, Xmt Holdoff Time could be shown as the following formula(2) defined in the standard. We know 2^x is the length of “Next”. From this formula, we know the holdoff time is in multiples of sixteen “Next”.

$$Holdoff = 2^{x+4} \tag{2}$$

The following figure shows these variations on time axis. (Fig. 4) Earliest Subsequent Xmt Time is a terminology in the standard to denote the earliest possible transmission time, without been determined.

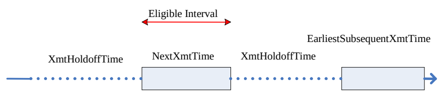

Fig. 4: Next Xmt Time and Xmt Holdoff Time

C. Competing Behavior and Scheduling Algorithm

Distributed scheduling ensures that the transmissions are collision-free. There is an election algorithm named MeshElection defined in the IEEE 802.16 standard to achieve collision-free.

The competing behavior and scheduling algorithm occur in each of nodes which are activating all over the neighborhood in mesh network. For instance, we observe certain node's competing behavior and its scheduling algorithm. We assume this node as an observed node; its neighboring nodes are denoted as neighbors. In the period of the competing behavior happened on this observed node, the scheduling algorithm is been computed. (3) is a formula to get the Earliest Subsequent Xmt Time over its all neighbors. Formula (4) sets Temp Xmt Time equal to this observed node's Xmt Holdoff Time added to the current Xmt Time.

$$\begin{aligned}&\text{Earliest Subsequent Xtm Time}\\&=\text{Next Xmt Time}+\text{Xtm Holdoff Time}\end{aligned} \tag{3}$$

$$\text{Temp Xtm Time}=\text{Current Xmt Time}+\text{Xtm Holdoff Time} \tag{4}$$

Depends on the information obtained previously, the observed node has the sufficient information to judge whether the possible collisions will occur or not. That is, there is a probability that this observed node's Next Xmt Time results in collision with neighbors' Next Xmt Time. The competing nodes are the subset of the neighbors with a Next Xmt Time eligibility interval that includes Temp Xmt Time or which an Earliest Subsequent Xmt Time equal to or smaller than Temp Xmt Time. These collision situations are depicted as Fig. 5 to express the collisions will be occurred between an observed node's Next Xmt Time and its neighbors'. The neighbor i is save. The neighbor j has its Next Xmt Time at the same time with the observed node. Neighbor k owns its Next Xmt Time early but its Earliest Subsequent Xmt Time overlaps the observed node's Next Xmt Time. In brief, observed node has two collisions with neighbor j and neighbor k.

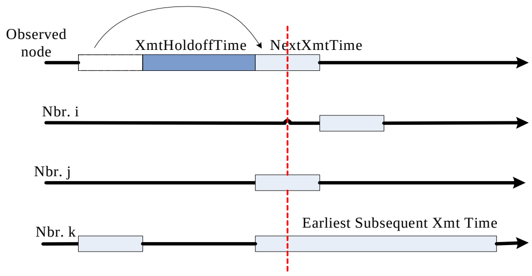

Fig. 5: One node results in collision with neighbors

If the collision will happen on observed node's Next Xmt Time as mentioned previously, the algorithm MeshElection will be executed during this computing period of distributed schedule. MeshElection is a C code function implemented in the standard. The Boolean value will be come out after MeshElection. "TRUE" means that this observed node wins the competing; on the contrary, "FALSE" means not. Corresponding procedures of them are:

- TRUE: Set Temp Xmt Time to Next Xmt Time, and ends off this algorithm.
- FALSE: Temp Xmt Time need to back.

D. Three-Way Handshaking

Thansmiting the MSH-DSCH message to the neighbors shall stable then subsequent data transmission may work better. Before data transmission, both of the coordinated and uncoordinated scheduling employs a three-way handshake to setup the connections with neighbors. This mechanism is used to convey the channel resources for the preparation of consequent data transmission. As follows, the three-way handshaking IEs (information elements) "**Request IE**", "**Availability IE**" and "**Grants IE**" are encapsulated in the MSH-DSCH. Hence it implies that the performance of MSH-DSCH packet traffic influences the three-way handshaking. This is why we concentrate upon the MSH-DSCH performance evaluation in this paper.

3. MATHEMATIC MODEL

So far, the competing behaviors of control subframe in the distributed scheduling of IEEE 802.16 mesh mode are presented. Next, we are going to propose a mathematical analysis to model the MSH-DSCH transmission behavior of IEEE 802.16 mesh mode. The delay time of MSH-DSCH transmission will be evaluated by our proposed mathematical model.

Assume X_n is denoted as a state in our consequent Markov Chain model that a node stays at a certain time to transmit MSH-DSCH. Time unit is an opportunity. A set of random variable $\{X_n\}$ forms a Markov chain if the probability that the next state is X_{n+1} depends only upon the current state X_n and not upon any previous stations. Base on our analysis in previous section, the next state merely depends on the current competing result, neither on the last nor on all of past history. Thus we have a random sequence in which the dependency extends backwards one unit in time. If this node's Temp Xmt Time overlaps with its neighbors, it implies the competing is occurred with them. If it wins or there is no competition, it will set this Temp Xmt Time as its Next Xmt Time. If it loses, it will back one opportunity to run this behavior again until it wins. In order to simplify the notification, we assume integer 1,2,3 … represent each of certain state X_n , the physical concept of our proposed Markov Chain are depicted as Fig.6.

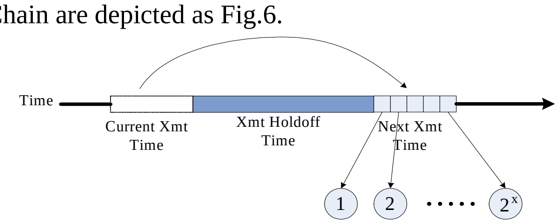

Fig. 6: Each state corresponds to the Next Xmt Time

With this concept of Fig.6, we can model this behavior with a vertical chain as Fig.7. The states and transition definitions are defined as Table 1. From state 1 to state 2^x implies the time duration of one Next Xmt Time. Suppose we have N nodes totaly, the probability which a node wins N-1 nodes can be gotten by expression (5). Oppositely, the probability of a node loses them can be gotten by equation (6).

$$\mathrm{Win} = \mathrm{Prob}_{N-1} \tag{5}$$

$$\mathrm{Lose} = 1 - \mathrm{Prob}_{N-1} \tag{6}$$

TABLE 1: THE NOTATION DEFINITIONS IN THE MARKOV CHAIN

Notation	Description
Integers in the state	The state probability that the transmission time backs to certain opportunity
Prob	The transition probability to indicate the probability that the node wins.
x	Exponent of Xmt Holdoff Time
N	The number of nodes

For example, if our observed node loses, it transfers from state 1 to state 2, the transition probability is $1-\mathrm{Prob}_{N-1}$. If it wins, it stays at state 1, the transition probability is Prob_{N-1}.

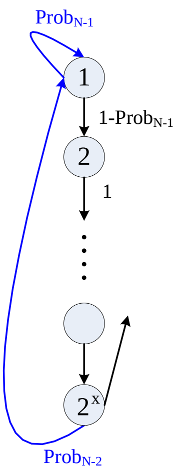

Fig. 7: One vertical chain

In order to model it easily, we assume that as long as the node lose this competition, it does not back one opportunity. It has to back a length of Next Xmt Time. That's why the transition probabilities during the inter-states are always 1 in Fig.7. At last, a Markov chain is organized as Fig8

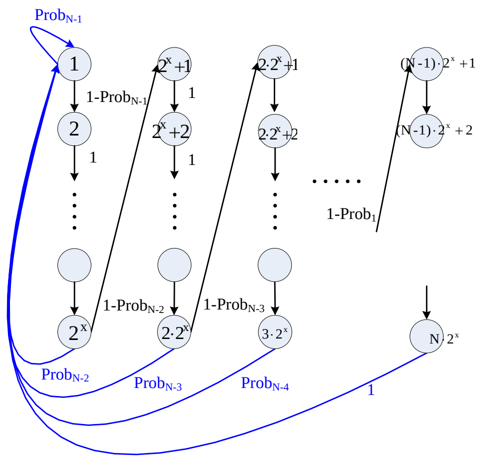

Fig. 8: The Markov Chain

The Markov Chain we proposed presents the variation of state transitions. We hope to induce an equation to evaluate the average delay time. The dependency of delay time relates to a probability of win. Before evaluating the delay time, we have to induce this probability formula initially. Then the expected value which is just the average delay time we target is the product of probability and time.

To begin with, we assume the number of nodes is N. Each observation of a node competes with neighbors is independent and represents one of two outcomes "competing" or "non-competing". So formula (7) is obtained and it is the extension of binomial distribution we know. The P_c in the (7) is a probability that a node competes with one another node. $P_{competing}$ is different from P_c in our assumptions. P_c is the condition happened between one node and one node in a very short time. Nevertheless, $P_{competing}$ is the condition while at least one of the following events is happened: between one node and one node, or between one node and two nodes, or between one node and more another nodes. So formula (7) means one of the following situations is occurred: observed node competes with one neighbor and wins, or observed node competes with two neighbors and wins, or observed node competes with three neighbors and wins …etc.

TABLE 2: NOTATIONS OF EQUATIONS

Notation	Description
Prob	Probability of $(\mathrm{competing} \cap \mathrm{win})$
$P_{competing}$	Probability of competing, at least one of more events happens

P_c	Probability of competing between one node to one of another node.
N	Number of nodes

$$
\begin{aligned}
&Prob_{N-1} = P_{competing \cap win} \\
&= \frac{1}{1} C_0^{N-1} \cdot P_c^{\ 0} \cdot (1-P_c)^{N-1} + \frac{1}{2} C_1^{N-1} \cdot P_c^{\ 1} \cdot (1-P_c)^{N-1-1} + \ldots \\
&= \sum_{k=0}^{N-1} \frac{1}{k+1} \cdot C_k^{N-1} \cdot P_c^{\ k} \cdot (1-P_c)^{N-1-k}
\end{aligned} \tag{7}
$$

In opposition, the losing probability can be derived as (8).

$$
1 - Prob_{N-1} = 1 - \left(\sum_{k=0}^{N-1} \frac{1}{k+1} \cdot C_k^{N-1} \cdot P_c^{\ k} \cdot (1-P_c)^{N-1-k} \right) \tag{8}
$$

If the node wins at state 1, the transition probability of win can be expressed by using (9). If the node wins at the state 2^x that is the end of first vertical chain, the probability of win can be expressed by using (10).

$$
Prob_{N-1} \tag{9}
$$

$$
(1 - Prob_{N-1}) \cdot Prob_{N-2} \tag{10}
$$

This probability distribution gives the trial number of the first success, so it is a geometric distribution. Substitute (7) and (8) into (9) and (10), we can derive the probability (11) and (12).

$$
\begin{aligned}
&Prob_{N-1} \\
&= \left(\sum_{k=0}^{N-1} \frac{1}{k+1} \cdot C_k^{N-1} \cdot P_c^{\ k} \cdot (1-P_c)^{N-1-k} \right)
\end{aligned} \tag{11}
$$

$$
\begin{aligned}
&(1 - Prob_{N-1}) \cdot Prob_{N-2} \\
&= \left(1 - \left(\sum_{k=0}^{N-1} \frac{1}{k+1} \cdot C_k^{N-1} \cdot P_c^{\ k} \cdot (1-P_c)^{N-1-k} \right)\right) \\
&\cdot \left(\sum_{k=0}^{N-2} \frac{1}{k+1} \cdot C_k^{N-2} \cdot P_c^{\ k} \cdot (1-P_c)^{N-2-k} \right)
\end{aligned} \tag{12}
$$

So far, each probability on the corresponding vertical chain has been derived. The expected value can be calculated by the summation of these probabilities and multiplied by time. Then fomular (13) can be obtained. The unit of time in this formula is opportunity.

$$
\begin{aligned}
&E(opportunity) \\
&= \left(\sum_{k=0}^{N-1} \frac{1}{k+1} \cdot C_k^{N-1} \cdot P_c^{\ k} \cdot (1-P_c)^{N-1-k} \right) \cdot 2^x \\
&+ \\
&\left(1 - \left(\sum_{k=0}^{N-1} \frac{1}{k+1} \cdot C_k^{N-1} \cdot P_c^{\ k} \cdot (1-P_c)^{N-1-k} \right)\right) \\
&\cdot \left(\sum_{k=0}^{N-2} \frac{1}{k+1} \cdot C_k^{N-2} \cdot P_c^{\ k} \cdot (1-P_c)^{N-2-k} \right) \cdot (2 \cdot 2^x) \\
&+ \\
&\ldots.
\end{aligned} \tag{13}
$$

Finally, we generalize our equation, as (14). In conclusion, the input parameters are N and x. It means the delay time is affected by the number of nodes and holdoff exponent.

$$
\begin{aligned}
&E[opportunity] \\
&= \sum_{j=2}^{N} \prod_{i=1}^{j-1} \left(1 - \left(\sum_{k=0}^{N-(i-1)} \frac{1}{k+1} C_k^{N-(i-1)} P_c^{\ k} (1-P_c)^{N-(i-1)-k} \right)\right) \\
&.\left(\sum_{k=0}^{N-i} \frac{1}{k+1} C_k^{N-i} P_c^{\ k} (1-P_c)^{N-i-k} \right) (2^x \cdot j)
\end{aligned} \tag{14}
$$

4. SIMULATION RESULT

We validate our model is this section. The transmission behavior is simulated by our C code. The mathematic evaluations are computed by the MATLAB 7.0. Following parameters are applied:
Exponent = 2
Node ID: random number between 1~4095
Probability: Pc= 0.5

And the result is shown as Fig.9. With this figure, it shows our mathematical model approaches the simulation result.

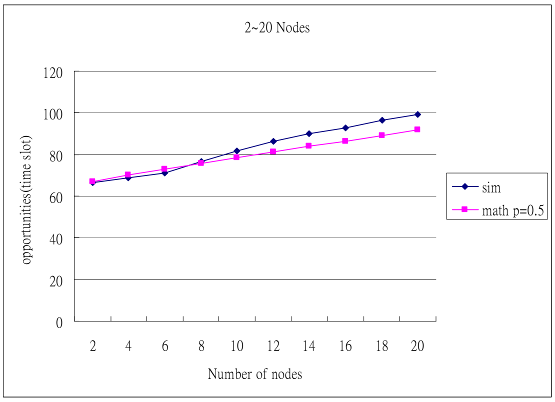

Fig. 9: The delay time of opportunities between simulation and mathematic model

The error rate is shown as Fig.10. It shows the error is under 10% while the nodes of number between 2 to 20.

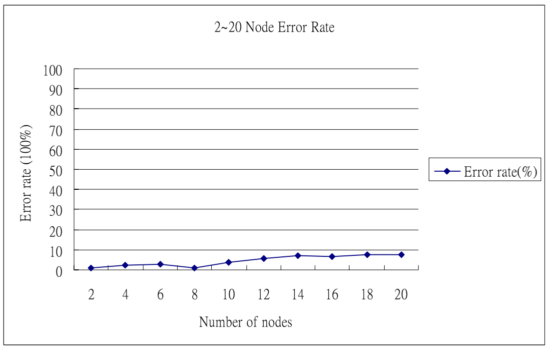

Fig. 10: The error rate between simulation and mathematic model

5. CONCLUSIONS AND FUTURE WORKS

In this paper, we have presented a Markov chain model which can be used to simulate real MSH-DSCH transmission behavior in 802.16 mesh mode. This model considers the competing probability and back behavior of MSH-DSCH. Base on this model, we derive a formula to evaluate an average delay time of MSH-DSCH transmission.

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中文摘要

IEEE 802.16 是一支援都會型無線網路的協定，IEEE 802.16 支援 PMP 模式(點對多點)和網狀模式兩種。在網狀模式中，所有節點的構成仿如 ad-hoc 方式，並依據在控制類子框中的排程資訊來計算下次遞送時間。每一個節點都必須跟鄰節點競爭以贏得時間槽的機會，接下來廣播排程訊息給它的鄰節點。這樣的行為跟它過去的歷史無關。換句話說，它具有”時間同質性”而適合以隨機程序來模擬。在這篇研究中，我們將用排隊程序來建立排程行為的模型，然後以馬可夫鏈來估計它的平均延遲時間，也就是節點持續競爭直到它贏得的這段等待時間。

行政院國家科學委員會補助國內專家學者出席國際學術會議報告

96 年 7 月 13 日

附件三

報告人姓名	蔡子傑	服務機構 及職稱	政大資訊科學系 副教授兼系主任
會議 時間 地點	July 2-4, 2007 Singapore	本會核定 補助文號	NSC 95-2221-E-004 -005 與國立政治大學共同補助
會議 名稱	（中文） （英文）2007 Fourth International Conference on Wireless and Optical Communications Networks (IEEE WOCN 2007)		
發表 論文 題目	（中文） （英文）Routing and Admission Control in IEEE 802.16 Distributed Mesh Networks		

報告內容應包括下列各項：

一、參加會議經過

1. 此次會議在新加坡的 Grand Hyatt 舉行，該地點在新加坡最熱鬧的街道上，與會人士除當地學者外，也有來自印度、印尼、馬來西亞以及中國等地。
2. 來到新加坡，感覺這真是一個國際化的都市，各國人種以及各國語言隨地可見，在路上有不同的文化語言在相互交流，已經成為常態而且相處自然，這正是國內學術界想要國際化的一個學習的榜樣。

二、與會心得

1. 這個會議由 Chief Technology Officer at Infocomm Development Authority IDA of Singapore 的 Dr. TAN Geok Leng 作開場 keynote speech。講的題目是新加坡的 Infocom 計畫 iN2015。從使用人口作出發，2006 Nov 的 HomeBroadBand 使用者有 61.1%，Mobile Phone 有 101.5%，到 2007 的 Home Internet Access 有 71%。另外，3G 到 3.5G 的 speed 增加，但費用卻從$68/month down 到$22/month。這都證明的 Infocom 的使用趨勢以及市場規模。從 2006-2015 的 iN2015 是一個 co-creation effort by People, Private & Public Sectors 的一個國家型計畫。目地希望能(1) Enrigh lives through infocom (2) enhanced economic competitiveness and innovation through infocom (3) increased growth and competitiveness of the infocom industry。最後也提出如何 realize iN2015 vision 的想法。基本上包含 foundation、economic sectors 以及 user side。Foundation 包括 infocom infrastructure infocom industry 以及 infocom manpower，economic sectors 包括 government 以及 society。
2. 接下去就是 panel discussion 以及 technical session 了。我講的題目是利用 token bucket 方式，在 IEEE 802.16 Mesh mode 上來作 traffic control，以及 bandwidth estimation, admission control, and scheduling，以達到 QoS 的要求，包含保證 delay 以及頻寬。
3. 會議中間的 break，碰到在中央念博士班並在萬能兼課的印度學者，聊到互相研究交流事宜，也與新加波南陽理工學院的老師學生們聊起無線網路的研究。
4. 晚餐期間，碰到來自海洋大學、中研院、長庚大學以及武漢大學的教授與研究生一起用餐，聊到雙方交換學生的可行性以及一些兩岸問題，感覺雙方多交流有助彼此瞭解，也感覺到彼此的善意，學術界的交流應該多多益善。

三、攜回資料名稱及內容

會議論文集 CD 一片

Routing and Admission Control in IEEE 802.16 Distributed Mesh Networks

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***Abstract*—QoS provisioning in wireless mesh network has been known to be a challenging issue. In this paper, we propose a fixed routing metric (SWEB) that is well-suited in IEEE 802.16 distributed, coordinated mesh mode. Also, an admission control algorithm (TAC) which utilizes the token bucket mechanism is proposed. The token bucket is used for controlling the traffic patterns for easy estimating the bandwidth used by a connection. In the TAC algorithm, we apply the bandwidth estimation by taking the hop count and delay requirements of real-time traffics into account. TAC is designed to guarantee the delay requirements of real-time traffics, and avoid the starvations of low priority traffics. With the proposed routing metrics, the admission control algorithm and the inherent QoS support of the IEEE 802.16 mesh mode, a QoS-enabled environment can be established. Finally, extensive simulations are carried out to validate our algorithms.**

***Index Terms*—IEEE 802.16, wireless mesh networks, WiMAX**

I. Introduction

As wireless technology evolves, IEEE 802.16[1], or WiMAX (Worldwide Interoperability for Microwave Access), appears to be a great competitor to IEEE 802.11 or 3G networks for its wide coverage and high data rate. In the IEEE 802.16 standards, meshing functionality is included as an optional mode. We propose a simple routing metrics called Shortest Widest Effective Bandwidth (SWEB) and a Token bucket-based Admission Control (TAC) in the IEEE 802.16 mesh networks.

IEEE 802.16 is a standard that aims at the use of wireless metropolitan area network (WMAN). Two modes are defined in the standard: PMP (Point to Multi-Point) and mesh mode. In PMP mode, the network architecture is similar to the cellular network. That is, one base station (BS) is responsible for all its subscriber stations (SS). The transmission can occur only between BS and SS. In mesh mode, the networks architecture is similar to the ad-hoc networks. In other words, each SS can be a source node and a router at the same time. The transmission can occur between any two stations in the network.

IEEE 802.16 mesh network is time-slotted. Connections must reserve timeslots in advance for the actual transmissions. As IEEE 802.16 is mostly used as the network backhaul, the network traffics mostly occur between the BS and SS. Therefore, with appropriate routing algorithm, the topology can be reduced into a routing tree. QoS of a connection along the path from the source station to the base station can be provided.

The remaining parts of this paper are organized as follows: Section II gives the background knowledge of token bucket mechanism and details of IEEE 802.16 mesh mode. Section III includes the related works. In section IV and V, the SWEB and TAC are proposed. Simulation results are given in section VI. Finally, we conclude this paper in section VII.

II. Background

A. Token Bucket mechanism

Token bucket is a mechanism that controls the network traffic rate injecting to networks. It works well for the "bursty" traffics. Token bucket mechanism needs two parameters: token rate *r* and bucket size *b*. Figure 1 shows how the token bucket

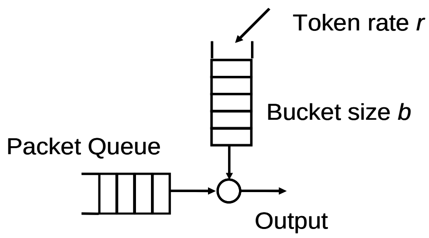

Fig. 1. Token bucket mechanism.

mechanism works.

Each packet represents a unit of bytes or a packet data unit. A packet is not allowed to be transmitted until it possesses a token. Therefore, in the time duration *t*, the maximum data volume to be transmitted will be

$$r \cdot t + b$$

We adopt the token bucket mechanism to estimate the bandwidth required for each connection in IEEE 802.16 mesh networks

B. IEEE 802.16 mesh mode

IEEE 802.16 mesh network is time-slotted. That is, the time is divided into equal-length time frames. And each time frame comprises one Control subframe and one Data subframe. Control subframe carries the control messages for the

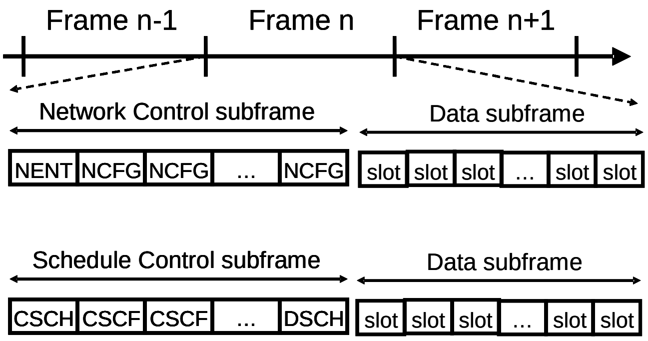

Fig. 2. Frame structures of IEEE 802.16 mesh mode.

scheduling, entry of new stations, exchanging of basic network parameters, etc. The data frame is composed by the time-slots for the actual data transmissions. The frame structure is given in figure 2.

There are two kinds of control subframe: Network Control subframes and Schedule Control subframes. In Network Control subframes, MSH-NENT (Mesh-Network Entry) messages are used to provide the entries of new-coming stations. MSH-NCFG (Mesh-Network Configure) messages are sent by each station periodically to exchange the basic parameters of networks, such as: the identifier of the BS, hops to the BS, and neighbor number of the reporting stations…etc.

Two scheduling modes are defined in IEEE 802.16 mesh mode: centralized and distributed modes. In centralized scheduling mode, the BS is in charge of all the transmissions happening in the mesh network. The resources are allocated by the BS with the MSH-CSCH (Mesh-Centralized Scheduling) messages and MSH-CSCF (Mesh-Centralize Scheduling and Configure) messages. In distributed mesh mode, the scheduling information is carried by MSH-DSCH (Mesh-Distributed Scheduling) messages, whose transmission time is determined by the mesh election algorithm given in the standard. MSH-DSCH has four information elements (IEs): scheduling IE, request IE, availability IE, and grant IE. The Scheduling IE carries the information of next-transmission time used in the mesh election algorithm. The other three IEs are employed in the three-way handshake:

1. The MSH-DSCH:request is made along with MSH-DSCH:availability, which is used to indicates the potential timeslots of the source station.
2. MSH-DSCH:grant is sent in response indicating a subset of the suggested availabilities that fits, in possible, the request.

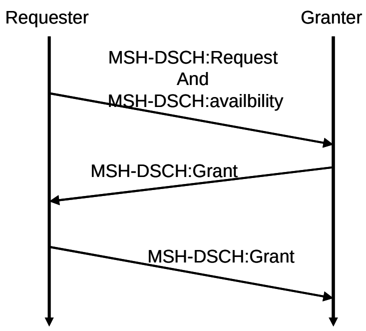

Fig. 3. Three-way handshake.

3. MSH-DSCH:grant is sent by the original requester containing a copy of the grant from the requester, to confirm the schedule.

After the three-way handshake indicated in figure 3, the reservation of timeslots in the data subframe is completed.

In both centralized and distributed mesh modes, the QoS can be supported by the fields of the CID (Connection Identifiers) that is associated with each connection. There are three fields that explicitly define the service parameters:

1. Priority: This field simply defines the service class of the connection
2. Reliability: To re-transmit or not.
3. Drop Precedence: The likelihood of dropping the packets when congestion occurs.

III. Related Work

H. Shetiya and V. Sharma [2] proposed the algorithms of routing and scheduling under IEEE 802.16 centralized mesh networks. The routing metric is based on the evaluation of queue length on each station. The routing is fixed routing, which reduces the topology into a tree. The scheduling algorithm is based on a mathematical model to allocate enough timeslots among the traffic flows. And our previous work [3] that focus on the call admission control and packet scheduling in the IEEE 802.16 PMP mode. In this paper, a mathematical model is proposed to characterize the packets of different traffic flows.

Some other researches also focus on the IEEE 802.16 mesh mode: F. Liu et al. [4] proposed a slot allocation algorithm based on priority, which is to achieve QoS. Z. J. Haas et al. [5] proposed an approach to increase the utilization of IEEE 802.16 mesh mode. They have adopted a cross-layer design in their work. M. Cao et al. [6] proposed a mathematical model and an analysis of IEEE 802.16 mesh distributed scheduler, mostly on the mesh election algorithm.

Douglas, S, J. De Couto et al. [7] proposed a new routing metrics called “ETX”, short for “Expect Transmission Count”. ETX is suitable for wireless networks and is able to fit in any routing algorithms like DSR, DSDV … etc.

IV. Routing Metrics: SWEB

a new routing metrics called SWEB (Shortest-Widest Efficient Bandwidth) is proposed, which considers three parameters: packet error rate, $P_{i,j}$, capacity $C_{i,j}$ over the link (i,j) and the hop count, h, from the source to the destination. The packet error rate can be retrieved by the exchanging of MSH-DSCH messages, which is associated with a unique sequence number. The lost or error MSH-DSCH messages can be detected. And the link capacity can be also known by the burst profile indicated in the MSH-NCFG messages. In MSH-NCFG messages, the hop count for a station to base station is also given. Therefore, we argue that our SWEB metrics is especially suitable for IEEE 802.16 mesh networks.

The efficient bandwidth of a link (i,j) can be calculated as:

$$C_{i,j}(1-p_{i,j}) \tag{1}$$

However, since the flow that comes in and leaves a node shares the bandwidth. Equation (1) should be divided by two to represent the available bandwidth. Therefore, the end-to-end available bandwidth is:

$$\frac{\min(C_{1,2}\cdot(1-p_{1,2}),C_{2,3}\cdot(1-p_{2,3}),\ldots,C_{i,j}\cdot(1-p_{i,j}))}{2} \tag{2}$$

By using (2), we define our SWEB metrics for all potential paths as:

$$SWEB = \frac{\min(C_{1,2}\cdot(1-p_{1,2}),C_{2,3}\cdot(1-p_{2,3}),\ldots,C_{i,j}\cdot(1-p_{i,j}))}{2}\cdot\frac{1}{h} \tag{3}$$

The path with the largest path metric will be chosen.

V. Admission Control Algorithm: TAC

Our Token bucket-based Admission Control (TAC) has two essential parts. First, the bandwidth used by a connection must be estimated well. Second, the bandwidth estimation is used for implementing the admission control algorithm.

A. Bandwidth Estimation

If all the connections are under the control of token bucket mechanism, the bandwidth used with a time frame can be estimated as:

$$\frac{r_i\cdot f+b_i}{f} \tag{4}$$

The r_i and b_i is the token rate and bucket size that associated with a connection i, respectively. f is the frame length. However, (3) is over-estimated since the transmission burst does not happen in every time frame. To better estimate the bandwidth, consider the scenario in figure 4.

Let the hop count and transmission deadline of the flow in Fig. 6 is 3 and $7f$, respectively. Assume that the transmission burst occurs in time interval [t+5f, t+6f] and tokens stored in the

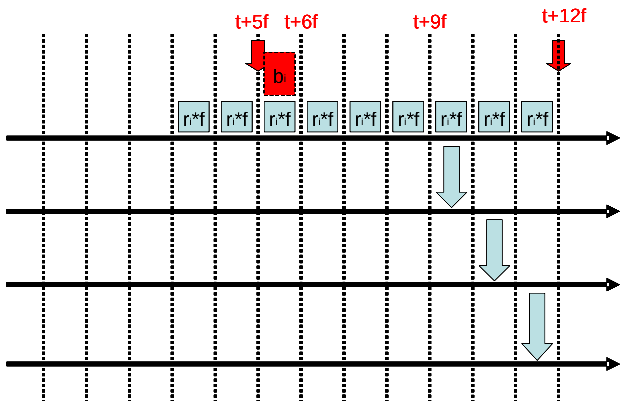

Fig. 4. Transmission Deadline of b_i Bits of Data.

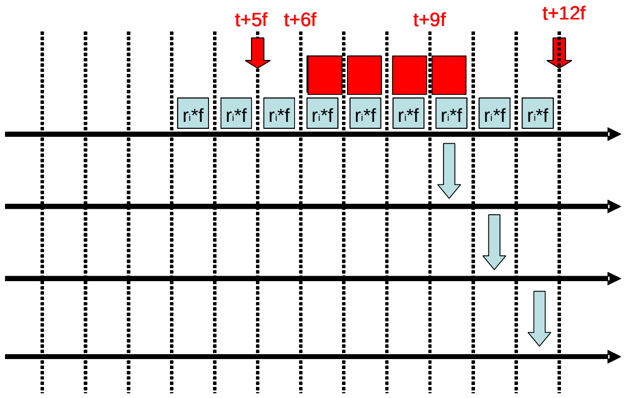

Fig. 5. Sharing b_i Bits of Data.

bucket are completely consumed. In order to satisfy the delay requirement, these bi bits of data must be sent in [t+9f, t+10f] at latest. Therefore, the frames from t+6f to t+10f can be used for sharing the bi bits, as in figure 5.

Generally speaking, in order to meet the delay requirement, d_i, of real-time traffics, packets generated at time t have to be sent after m_i frames after t, where

$$m_i = \left\lfloor \frac{d_i}{f} \right\rfloor - h \tag{4}$$

These m_i frames can be used to share the b_i bits of data. Therefore, the maximum volume of data that can be sent in any given frame is:

$$r_i\cdot f+\frac{b_i}{m_i} \tag{5}$$

We use (5) as bandwidth estimation of a flow.

B. Admission Control

We use the above-mentioned bandwidth estimation to implement the TAC algorithm. In TAC algorithm, the minimum usage of timeslots by each connection is defined. They are: *CBR_min*, *VBR_min* and *BE_min*. When a station receives a MSH_DSCH:Request, it examines whether the current usage of each class exceeds their minimum usage or not. If it is, the new-coming flow will be marked as downgraded flows. If a MSH-DSCH:Request comes in, the downgraded flows have bigger possibilities to be preempted. On the other

TABLE I
QoS Mappings

	Priority (3 bits)	Reliability (1 bit)	Drop Precedence (2 bits)
CBR	7	0	0
CBR_DG	4	0	1
VBR	6	0	0
VBR_DG	3	0	2
BE	5	1	0
BE_DG	2	1	3

hand, if the current usage does not exceed its minimum usage, the flow will not be downgrade and have bigger change to preempt other downgraded flows.

Since the service levels in IEEE 802.16 mesh mode are identified in the fields of CID (Connection Identifiers), we have the QoS mapping in Table I. With the mappings in table I, the down-graded flows can be marked. And by this information, we develop our TAC algorithm as follows:

1.) A new flow with its *BW_req* (Bandwidth request) in the unit of data timeslots. And set *BW_avail* as the total empty slot number. (*BW_avail* stands for available bandwidth)
2.) The station that handles the request checks if the *BW_req*<*BW_avail* or not. If yes, go to step 3. Or else, go to step 4.
3.) The station determines to downgrade the flow or not, by comparing the current usage and the minimum usage of the traffic class.
4.) The station checks if the current usage exceeds the minimum usage of the traffic class. If yes, the flow shall be rejected. Or else, go to step 5.
5.) Check the timeslots used by downgraded flows in the order of *BE_DG*, *VBR_DG*, and *CBR_DG*. If there is no such timeslots, the request is rejected. Or else, set this timeslots empty, which means to preempt this timeslots. Updating the value of *BW_avail*. Go to step 2.

VI. Simulation Results

The simulations are conducted in a 16-node topology, and the simulation area is a 4 km * 4 km square. The radio range is set as 1.5 km in radius. The frame length is chosen to be 8 ms.

TABLE II
THE PARAMETERS OF QPSK

QPSK coding rate	3/4
OFDM symbols in a frame	676
OFDM symbols in a control subframe	16
OFDM symbols in a data subframe	660
OFDM symbols in a timeslot	4
Number of data timeslots	165
Capacity of a timeslot	144 bytes

In the simulations, QPSK is chosen to be the modulation method. The details of QPSK are given in table II.

The data rate of the CBR traffic is 64 kbps, with the 960-bit packet size in the packet interval of 15 ms. The VBR traffic is sending at the average speed of 400 kbps. The mean packet size is 16000 bits sending at the interval of 40 ms. The packet size of BE traffics is 8000 bits and is sent every frame (8 ms).

A. Routing

The proposed SWEB is compared with the ETX [7] and the shortest path. The performance and delay of VBR traffics are compared across all three different metrics. The performance is given in figure 6. And figure 7 shows the delay.

As shown in the figure 6 and 7, when number of flows is reaching 25, some VBR flows are preempted by CBR flows. By simulation results, we claim that SWEB is a compromise of

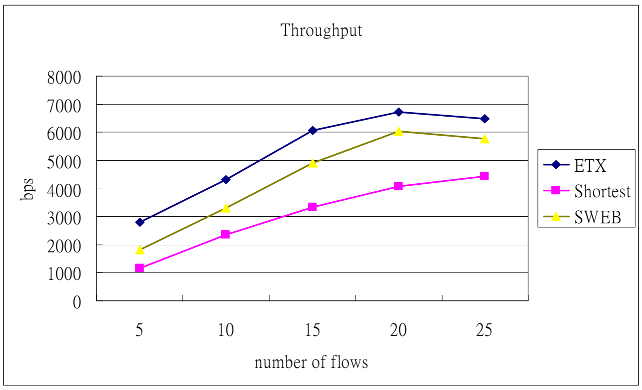

Fig. 6. Throughput of VBR flows.

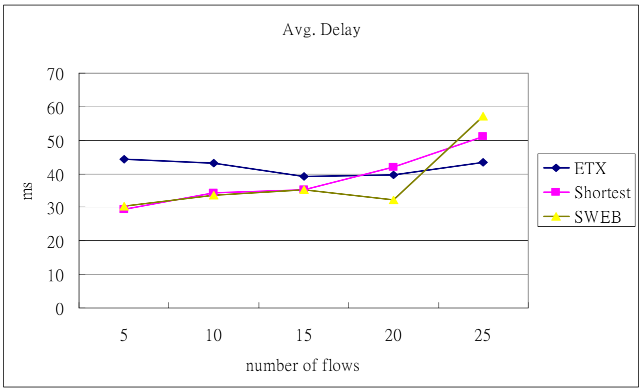

Fig. 7. Delay of VBR flows.

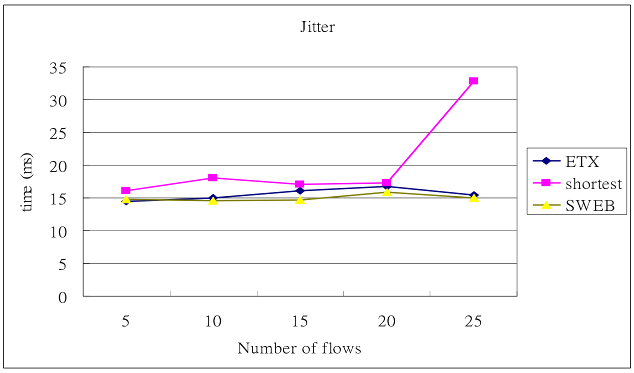

Fig. 8. Jitter of VBR flows.

TABLE III
TOKEN BUCKET MECHANISM PARAMETERS

	Token rate (bytes / frame)	Bucket size (bytes)	Delay requirements
CBR	120	8	40 ms
VBR	1500	500	80 ms
BE	7500	250	--

delay and throughput. But in figure 8, we can find that SWEB has best performance in jitter of real-time packets.

B. Admission Control

In TAC algorithm, the minimum usage of each traffic class must be set. In the simulations, the *CBR_min*, *VBR_min* and *BE_min* are set as 10, 40 and 75 timeslots, respectively. Also, the parameters of token bucket are shown in table III.

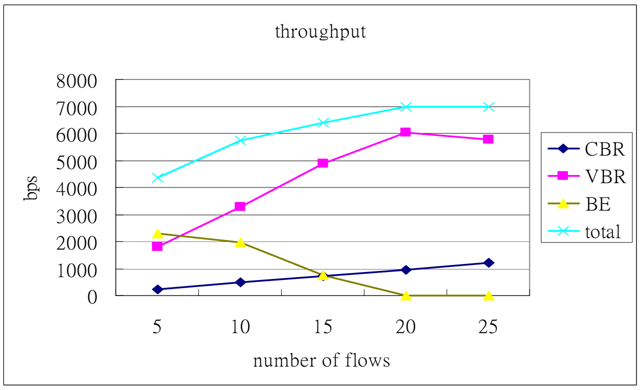

Fig. 9. Throughput for the original IEEE 802.16 mesh mode.

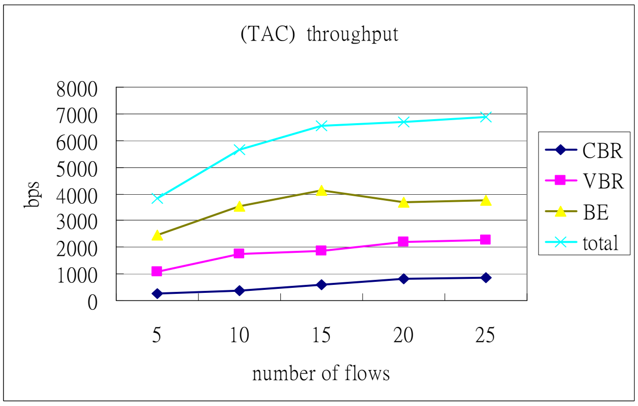

Fig. 10. Throughput when TAC is applied.

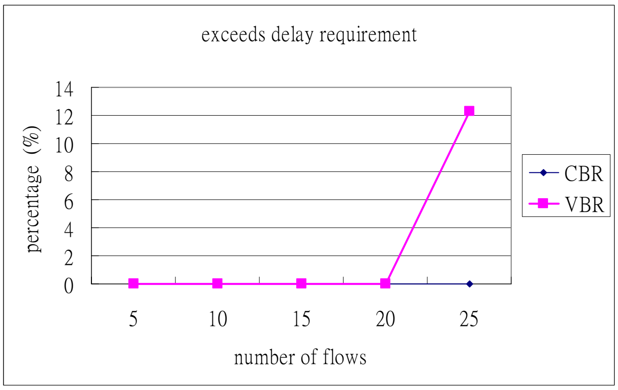

Fig. 11. The ratio of the realtime packets that exceeds the delay requirements for the original IEEE 802.16 mesh mode.

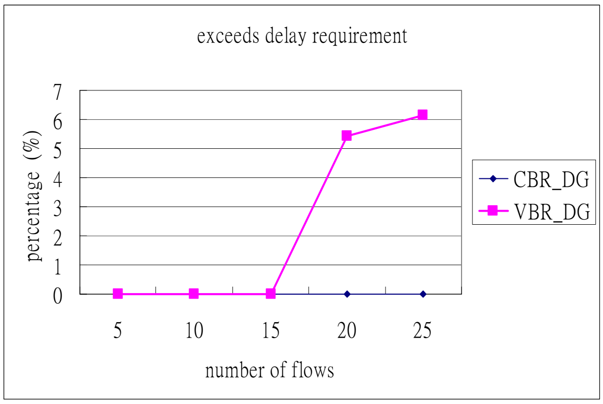

Fig. 12. The ratio of the realtime packets that exceeds the delay requirements when TAC is applied.

We compare the throughput in figure 9 and figure 10. In figure 9, BE traffics suffers from preemption from higher priority traffic class, therefore, receiving low throughput when network is heavily-loaded. By applying the CAC algorithm in figure 10, the BE flows has the guaranteed throughput by setting the minimum usage. The preemption occurs only in down-graded flows.

In figure 11 and figure 12, the statistics is gathered to discuss the percentage of real-time packets exceeds the delay requirements. As in figure 11, around 12% of VBR-packets exceed the delay requirements when the number of flow is 25. However, in figure 12 it is reduced to around 7% for only VBR-downgraded flows. It can be expected that for all VBR flows (VBR and VBR-downgraded), the ratio would be lower than 7%.

VII. Conclusions

In this paper, we proposed a new routing metric, SWEB, and an admission control algorithm, TAC for IEEE 802.16 mesh networks. SWEB is applied in static routing environment and yields the good throughput, delay and jitter performance. The TAC algorithm prevents the starvation of low-priority traffic flows and guarantees the delay requirements of the real-time flows. By SWEB and TAC, a QoS-enabled network environment can be realized with IEEE 802.16 mesh mode in the MAC layer. Thus, end-users will have better experience and convenience in utilizing the networks.

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行政院國家科學委員會補助國內專家學者出席國際學術會議報告

95 年 10 月 31 日

附件三

報告人姓名	蔡子傑	服務機構 及職稱	政大資訊科學系 副教授兼系主任
會議 時間 地點	Oct 25-27, 2006 Beijing, China	本會核定 補助文號	NSC 95-2221-E-004 -005 與國立政治大學共同補助
會議 名稱	(中文) (英文) First International Conference on Communications and Networking in China (Chinacom2006)		
發表 論文 題目	(中文) (英文) Reducing Calibration Effort for WLAN Location System Using Segment Technique with Autocorrelation		

報告內容應包括下列各項：

一、參加會議經過

1. 此次會議在北京的九華山莊舉行，該地點在北京北方約一個小時的車程，北京人士喜歡選在這個山莊開會，原因是他們認為這兒空氣較好遠離塵囂。所以同樣這個期間也有其他會議在這舉行。
2. 初次來到北京，感覺他們的服務還不錯，但就是管理非常的差，光在 hotel check in 就等了一個小時，又要 deposit 200 元才能開通電話，真是有點莫名其妙，而且會場佈置也不明顯，找了一會才找到會場。

二、與會心得

1. 這個會議由 Georgia Institute of Technology 的 Dr Ian F. Akyildiz 開場 keynote speech。他是很多知名期刊的主編，也是 IEEE, ACM fellow。講的題目是"Key Technologies for Wireless Networking in the Next Decade"，包括 WLANs, Satellite Networks, Sensor & Actor Networks, Wireless Mesh, WiMAX, Dynamic Spectrum Access Networks/Congitive Radio Networks/xG Wireless Systems。他研究了蠻多 Wireless Sensor Networks，所以講了非常多的議題都圍繞在這方面，從 Transport layer, Network layer 到 MAC layer 都有涉獵到，最後得出的結論是，必須有 cross-layer 的 design 才能面面俱到的符合需求。他並勉勵年輕學者，不要一味寫無用的 paper，應該更從應用需求去找尋與技術之間的 mapping，才能真正推動科技的進步。
2. 接下去的 panel 是 Research in the European Seventh Framework Program and the Chinese Eleventh Five-Year Plan on Telecommunications。與會人士有歐洲與中國的官方代表，學界代表與業界代表。介紹歐洲的 FP7 計畫以及中國的 863 & 976 project，討論雙方如何交流與合作，一些文件是否可英文化以方便交換文件和互相瞭解。這兩個計畫都很大而且一個是五年期一個是六年期，雙方都投入相當多的資源與規劃在通訊領域，以尋求未來在這個領域的領先地位。FP7 是從 2007~2013，全部經費為 72,726 million，而中國的經費則為第一年(2006) 71.6 billion RMB。
3. 下午就是 technical session 了。我講的題目是利用減少 calibration effort 的角度去實作無線定位系統，講完之後立刻有一位澳洲來的教授來找我，表達他們也正在作這樣的系統，很高興看到我們已經有作出這樣的系統了，讓他非常的放心，並表示有機會可以跟我們合作，我也表達有機會可以去他們學校訪問作交流。他們的出發點為無線安全，必須判斷出使用者是在建築物內部的某個房間裡，才可允許存取無線網路，因此系統必須有能力去定位使用者的位置。除此之外，也要防範攻擊者利用較大發射功率，誤導系統認為它較接近 access point 而誤判。
4. 晚餐期間，碰到來自新加坡大學以及山東濟南大學的教授與研究生一起用餐，聊到雙方交換學生的可行性以及一些兩岸問題，感覺雙方多交流有助彼此瞭解，也感覺到彼此的善意，學術界的交流應該多多益善。

三、攜回資料名稱及內容

會議論文集 CD 一片

REDUCING CALIBRATION EFFORT FOR WLAN LOCATION SYSTEM USING SEGMENT TECHNIQUE WITH AUTOCORRELATION

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ABSTRACT

Context-aware applications become more and more popular in today's life. Location-aware information has derived a lot of research issues. This paper presents a precise indoor RF-based WLAN (IEEE 802.11) locating system named Precise Indoor Locating System (PILS). Most proposed location systems acquire well location estimation results but consume high level of manual effort to collect a huge amount of signal data. As a consequence, the system becomes impractical and manpower-wasted. In this paper, we aim to reduce the manual effort in constructing radio map and maintain high accuracy in our system. We propose models for data calibration, interpolating, and location estimation in PILS. In data calibration and location estimation models, we consider of autocorrelation of signal samples to enhance the accuracy. Wireless Channel Propagation model is also in our concern. Large scale and small scale fading are involved in the wireless channel propagation.

1. INTRODUCTION

It's critical to develop a locating system indoors with high accuracy. Several studies have been conducted to offer some locating theorems. Many of above implements are based on signal strengths (SS) received from access points (AP). Although only using SS information makes location system workable, it's better to consider with physical architecture property. Real world wireless channel can be decomposed into two parts [1,3]: large-scale model and small-scale fading model. Large-scale model considers the path loss which represents the local mean of the channel gain and is therefore dependent on the distance between the transmitter and receiver. Small-scale fading is a characteristic of radio propagation resulting from the presence reflectors and scatters that cause multiple versions of the transmitted signal to arrive at the receiver, each distorted in amplitude, phase and angle of arrival.

Most RF-based location systems are operated in two phases: offline calibration phase and online estimation phase. In offline calibration phase, location system calibrates a great deal of data on specific location and stores these data labeled with location information into database, which is called radio map. In online estimation phase, system calibrates some data in real time and substitutes them into radio map for estimating people's location. The greater part of RF-based location systems described above demand large amount of calibration data. It's manpower-wasted to calibrate a lot of data for each building. Reducing offline calibration [4,11] effort and making estimation result acceptable are our goals in this paper. We expect to train data on one location in small room and to train less than three locations in bigger ones.

We propose a methodology called Segment Process. Both offline and online phases refer to the concept of Segment Process. In offline phase we use Segment Process to gain more useful data in radio map, and in online we use it for making estimation result more accurate. Segment Process only consumes computer calculation frequency but needn't more manual effort. The detail of Segment Process will introduce below. Autocorrelation of signal samples [8] is concerned to enhance the utility of Segment Process. Our approach is to calculate autocorrelation dynamically, and slice a segment when the value of autocorrelation is less than a threshold.

The result of PILS [11] is very useful regarding context-aware applications. PILS is able to evaluate mobile terminals' location within few meters to their actual location. If the calibration node rate is 9.7%, the average error distance of PILS is 1.5370 m with autocorrelation and 1.7731 m without autocorrelation. The use of autocorrelation for a sequence of consecutive signal samples enhances the accuracy by 13.3%. And for 100% calibration node rate, accuracy has been enhanced by 13.4%. It's also remarkable that PILS decreases large amount of manual effort when calibrating data. PILS is more practical than other location system and is with great estimation accuracy.

The remainder of this paper is organized as follows. In Section 2, we lay out our concept in estimating user's location and present our locating system in Section 3. Experiment results are described in Section 4. In Section 5, we summarized the conclusion of this paper.

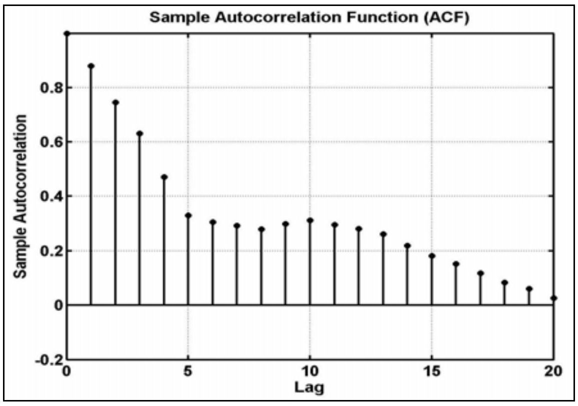


Figure 1.Sample Autocorrelation

2. LOCATION ESTIMATION CONCEPT

2.1. Samples Correlation

Figure 1 illustrates the autocorrelation function of the samples calibrated from one access point (20 samples per second) at a fixed position and our system calibrates 20 samples for each AP. The autocorrelation of a sequence of signals varies on different environment. When the curve is smooth, it means that the signal strength received from an AP at a particular position is relatively stable. We expect to improve our Segment Process by considering with samples autocorrelation.

2.2. Segment Process

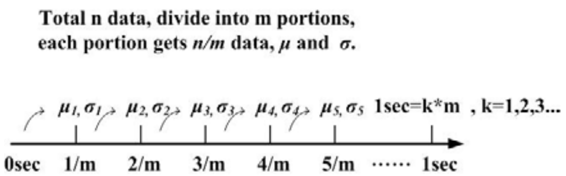


Figure 2.Segment Process

Due to the random nature of the wireless medium, wireless channel signal strengths are changeful and the sequence of signal samples can be viewed as random variables. Even immovable calibrating, the signal strengths still pulse up and down. Inconstant signal strengths on one position make estimating location difficult and inaccurate. Most research adopts mean value to solve this problem, but we think it's insufficient due to the inaccurate location estimation results. Using one mean value to stand for one position's wireless channel information is not enough. We present a Segment Process to make some improvement as showed in figure 2. We consider the autocorrelation of a sequence of samples received from the same AP. Then make a rule when the autocorrelation of consecutive samples is less than a threshold, we slice a segment.

Alg. 1 autocorr[] = Auto_Correlation(N , S)

Input:
N : Number of input samples
S : Signal strengths vector input(S = S[1],S[2],...,S[N])

Ouput:
The autocorrelation vector autocorr[N] for input signal strength vector S.

```
nFFT ← 2^(nextpow2(N) + 1)
F ← fft(mean-value(S) , nFFT)
F ← F * conj(F)
autocorr ← ifft(F)
autocorr ← autocorr / autocorr[1]
autocorr ← real(autocorr)
```

Table1. Auto_Correlation Function

Assume one training process gains 100 signal strengths, we divide up these 100 signal strengths into m parts dynamically. Table1 shows the algorithm to calculate autocorrelation of a sequence of signal samples. When autocorrelation between the first and the last sample in one sequence of signal strengths is less than our threshold, we slice one segment up. Then calculate mean and variance value of each part and store them into radio map to substitute for original data. Now we acquire m slices of mean and variance values and have more information to estimate location. We call the divide procedure as Segment Process. Not only in offline calibration phase we do Segment Process, but in online estimation phase we execute it, as will introduce below.

2.3. Reducing Training Location Numbers

A lot of proposed research show high accuracy of location estimation. But they are all impractical due to high manual effort. We are incapable to train radio map for all buildings with high manual effort. Construct a location system that only requires little offline training effort is significant. These kinds of methods are called reducing calibration effort. Recent research which achieve high accuracy within 1 meter error distance calibrate data on all grid points. In our model we train data on one grid point in small room and within three in bigger one, and then interpolate data for all the other grid points. We will describe it in section 3.3.

3. LOCATION ESTIMATION MODEL

3.1. Variable Definitions

Assume one training we detect I APs, and gain N data for each AP. Let S_i be N consecutive signal samples received from access point b_i.

$$S_i = \{ss_{[1]}, ss_{[2]}, ss_{[3]}, \ldots, ss_{[N]}\}$$

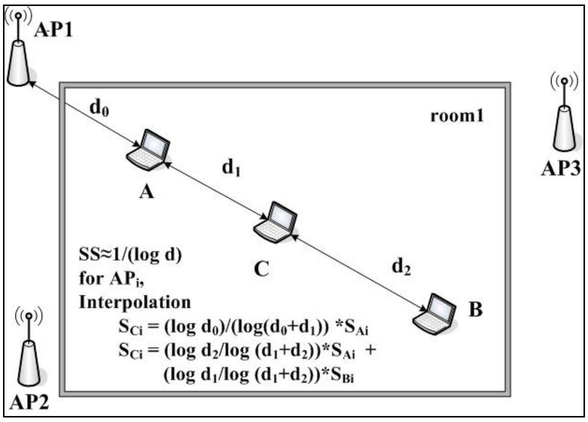


Figure 3.Interpolation Model

Assume there are J positions in radio map RM. Let l_j be jth location.

$$\mathbf{X_j = \{(S_1,S_2,S_3,\ldots,S_I) \mid l_j\}} \quad \text{and}$$
$$\mathbf{RM = \{X_1,X_2,\ldots,X_J\}}$$

We use Segment Process to divide up n data into m parts, then we get values of mean and variance as follows:

$$\mathbf{M_i=\{(\mu_{i1},\mu_{i2},\mu_{i3},\ldots,\mu_{im}) \mid X_j\}} \quad \text{and}$$
$$\mathbf{\Sigma_i=\{(\sigma_{i1}^2,\sigma_{i2}^2,\sigma_{i3}^2,\ldots,\sigma_{im}^2) \mid X_j\}}$$

3.2. Probability Model

A lot of WLAN location systems propose probability model [7,8,11]. As small-scale fading investigates signal fluctuation in short period of time, our Segment Process divide one calibration into m small pieces of time. Definitely Segment Process matches the claim of small-scale fading. Therefore, we adopt Rayleigh-like fading probability density function as our probability model. The model returns the location among the set of radio map RM that maximizes $P(S_i \mid X_j)$, $1 \leqq i \leqq I$, $1 \leqq j \leqq J$, i.e.

$$\mathbf{argmax[P(S_i \mid X_j)]} \tag{1}$$

$$P(S_i \mid X_j) = \{R(S_i,\Sigma_{i1} \mid X_j)_1, R(S_i,\Sigma_{i2} \mid X_j)_{2,\ldots,} R(S_i,\Sigma_{im} \mid X_j)_m\} \tag{2}$$

where

$$R(\mathbf{S_i},\Sigma_i \mid \mathbf{X_j}) = \frac{\mathbf{S_i}}{\Sigma_i} \times \mathbf{exp\{}-\frac{\mathbf{S_i}^2}{2\Sigma_i}\mathbf{\}, S_i} \geq 0 \tag{3}$$

3.3. Interpolation Model

For reducing calibration number of locations, most of the data of grid points on the radio map are gained by interpolating. Large-scale fading represents the relationship between path loss and signal attenuation. Equation 3 and 4 are our interpolating models and we use them to calculate data for un-calibrated grid points on

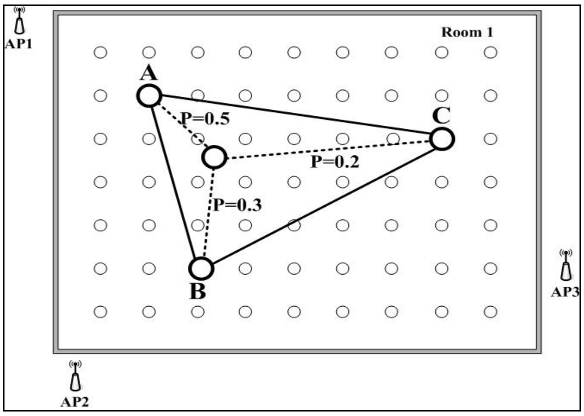


Figure 4.Weighted Triangulation

radio map. An illustration is shown in figure 3, location A and B have been calibrated and location C need to be calculated. When only location A is observable to location C, we use equation 3 to infer signal strength on location C, and when both location A and B are visible, we use equation 4. All the other un-calibrating grid points are computed with the same way.

$$S_{Ci} = \frac{\log d_0}{\log(d_0 + d_1)} \times S_{Ai} \tag{4}$$

$$S_{Ci} = \frac{\log d_2}{\log(d_1 + d_2)} \times S_{Ai} + \frac{\log d_1}{\log(d_1 + d_2)} \times S_{Bi} \tag{5}$$

3.4. Location Deterministic Model

3.4.1. Location Estimation

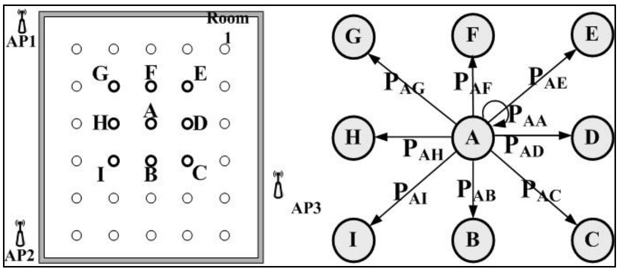
In location deterministic model [2,4,5,6,9], two location models are proposed to estimate locations. We make use of Segment Process to improve the accuracy of estimation results. Before using two location deterministic models, we can estimate locations only by probability model. First we use Segment Process to slice our offline data into m parts, then calculate $P(S_i \mid X_j)$, $1 \leqq i \leqq I$, $1 \leqq j \leqq J$, for each part. Second, we multiply these $P(S_i \mid X_j)$ for final result.

$$\mathbf{P(S_i \mid X_j)_{final} = \prod_{k=1}^{m} P(S_i \mid X_j)_k, 1 \leq k \leq m} \tag{6}$$

Third, we find out $\text{argmax}[P(S_i \mid X_j)_{final}]$ for output result.

3.4.2. Weighted Triangulation Model

We choose three locations X_A, X_B, X_C, which own the max probability values of $P(S_i \mid X_j)_{final}$, and use weighted triangulation to determine the outcome location. Figure 3 illustrates weighted triangulation. Being normalized, P_A equals to 0.5, P_B equals to 0.3, P_C equals to 0.2. As a consequence, the order of distance to estimation location from far to near is C,B,A, respectively.



(a) (b)

Figure 5.(a) The cluster and (b) corresponding Markov Chain

3.4.3. Time Correlation-based Model.

The location system is more appropriately to be implemented on the dynamic environment. We consider referring the correlation between locations. In each estimation, we form a cluster which consists with eight gird points surrounding the prior location and calculate correlation between prior location and all the other grid points in the cluster. The cluster is shown as figure 5(a), and corresponding Markov Chain is illustrated as figure 5(b). There is a limit that user moving can't exceed average human velocity, otherwise the cluster space won't be enough to afford the estimation.

As presenting in section 3.4.1, we use Segment Process to divide one estimation into m slices of brief results and combine them for ultimate output. In all brief calculation, system computes the correlation between prior estimation result and other gird points in the cluster. As figure 5(b) showing, assume prior result is location A, then P_{AA}, P_{AB},…, P_{AI} are calculated. Let prior result, state A be π^k, $\pi^k = P(X_A) = \prod_{i=1}^{l} \mathbf{P}(\mathbf{S}_i \mid \mathbf{X}_A)$. Let all grid points in cluster be π^{k+1}, (P_{AA},P_{AB} ,…, P_{AI}) be state transition probability distribution A.

$$\pi^{k+1} = A\pi^k$$

$$\text{output} = \arg\max \frac{\prod_{k=1}^{m} \pi^k}{\eta}, \ \eta \text{ is a normalizer}$$

4. EXPERIMENTAL RESULT

In this section we discuss the experimental testbed and evaluate the performance of PILS with all the models in this paper.

4.1. Testbed

We performed our experiment in the second floor of the Dept of CS, National Cheng-Chi University. This building has a dimension of 11 × 52 meters. The building is equipped with 802.11b wireless LAN environment. Our experiment devices are IBM X21, six PCI GW-APIIT and six BLW-04G APs. To form the radio map, the environment was modeled as a space of 235 locations, 215 locations inside the rooms and 20 along the corridors, each representing a 1.5×1.5 meter square grid cell.

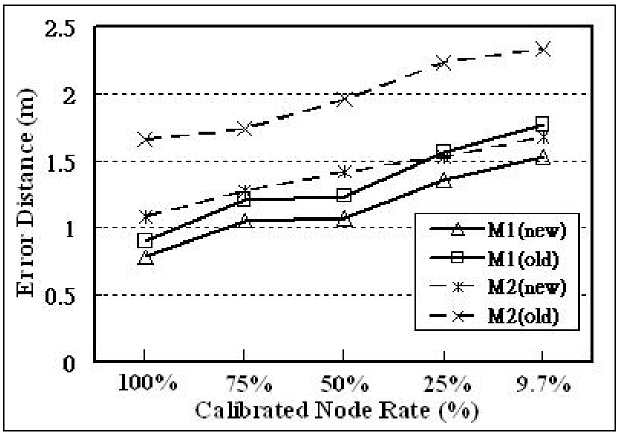


Figure 6.Location Estimation Results

4.2. Data Collection

Since we deem that train data on too many locations is impractical, we attempt to calibrate on few locations and interpolate all the other points by our model. In our experimental environment, we collected 1 point in small rooms, 2 points in bigger ones, and 2 points in corridor. Totally, we selected 23 locations, 9.7% in whole environment locations. On the other hand, system interpolated the rest 90.3%. In our opinion, newly computer owns high performance and can sustain vast data calculation.

We collected 300 samples at these 23 grid points and expected an error of about 10~15 cm due to the inaccuracies when clicking the interface of radio map.

4.3. Experimental Result

First we define M1 as weighted triangulation model, M2 as time correlation-based model. Figure 6 illustrates the effect of reducing the number of calibrated locations. The X axis represents calibrated node rate, and the Y axis shows the error distance. The four curves are our two location estimation models, and each model separates into two parts: old and new versions. Old version [10] is our former work and new is the progress in this paper. In most case, M1 performs better than M2. Since M2 gains worse results, but it owns faster calculation speed because M2 computes only in the cluster it formed. The other advantage of M2 is that the variance of error distance is smaller than M1, and it means estimation results are more stable .If we calibrate data on 100% points, our models can achieve average error distance around 1 meter. Once we only calibrate data on 9.7% points, the results are within 2 meters, and it's still great acceptable. We won't show the figure to illustrate the results of varying threshold of autocorrelation due to the page limit. In our experiments, we gain best accuracy when the threshold is set to 0.5 and in average we slice one segment for 4 signal samples.

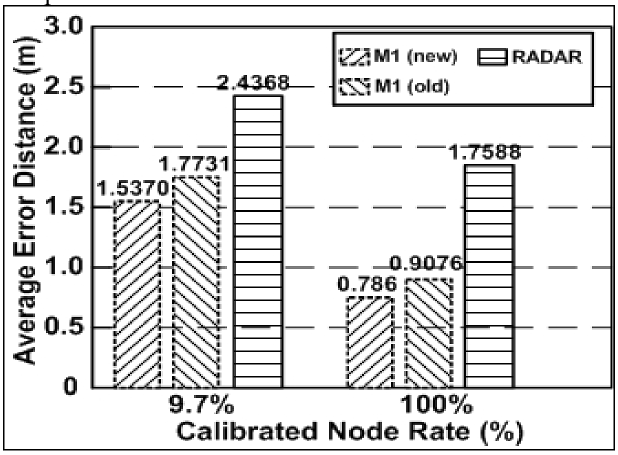


Figure 7.Comparison between M1(new), M2(old) and RADAR

In figure 7 we compare our models (with and without autocorrelation) with well-known WLAN location system RADAR [9] which is designed by Microsoft. Two experiments were held, one calibrated data on 9.7% grid points and interpolated the rest 90.3%, and the other calibrated data on 100% points. The results show that average error distances in our model are 1.5370 and 1.7731 m with 9.7% points and 0.786 and 0.9076 m with 100% points. The average error distances in RADAR are 2.4368 m with 9.7% points and 1.7588 m with 100% points. It's obvious that our system owns better location estimation result than RADAR because more information are acquired with the Segment Process and novel location estimation models are presented by us.

5. CONCLUSIONS

In this paper, we proposed our RF-based WLAN location and system named PILS. We analyze the variations of wireless channel propagation and develop models to solve them. Most variations of wireless channel propagation are caused by wireless channel fading which can be categorized into large-scale fading and small-scale fading.

The main idea of this paper is to reduce the calibration effort and keeps the result won't decline too much. We proposed a methodology called Segment Process. When we do Segment Process, autocorrelation of signal samples is concerned. Autocorrelation model is presented to slice each segment dynamically. And the experimental results show that the use of autocorrelation model indeed enhance our Segment Process. Both offline and online phases refer to the concept of Segment Process. In location deterministic model, we proposed weighted triangulation model and time correlation-based model. In online phase, weighted triangulation model chooses three deterministic locations which own higher probabilities and triangulate them for final output. Time correlation-based model considers the correlation of former output location for each determination and a Markov Chain is operated for calculating the correlation. We had taken a lot of experiments for each model and combination of them. Details of experiments are shown in section 4.3, and the result is well enough for constructing a really impractical location deterministic system.

Finally we compare our system with famed location system RADAR and demonstrate that better accuracy is presented by our system.

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