行政院國家科學委員會補助專題研究計畫 ☑ 成 果 報 告
□ 期中進度報告

多天線多通道多模多速率無線網狀網路之設計與實作-子計畫一：M4 無線網狀網路之網路規劃及資源分配問題(3/3)

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中英文摘要

在本計畫中我們將分別針對 IEEE 802.11-based 的無線網狀網路以及 IEEE 802.16-based 的無線網狀網路設計一些實用的通訊協定。本計畫在第一年提出過一個適用於多重頻道網路的鏈結層頻道管理協定，在第二年我們考量網路層設計(尤其著重在繞路協定)，我們認為多重頻道網路環境下必須搭配多重路徑繞路協定，才能將整體的效能提升至最高，因此我們提出了一個結合了鏈結層與網路層的通訊協定用於 IEEE 802.11-based 無線網狀網路上，此外我們也發現資源分配的好壞會跟使用的繞路樹有密切的關係，因此我們也提出了一個建繞路樹的演算法結合前述的資源分配方法，以改善 IEEE 802.16-based 網狀網路的效能。在第三年我們發現在隨意式網路下，大多數路由協定都沒有考慮現今無線網路產品對於多速率通訊的支援。在建立完路由路徑的條件下，我們為隨意式網路提供了一個吞吐量分析的工具，經由模擬的驗證，我們相信路徑吞吐量會是一個比傳統方法，如節點計數(hop count)，更好的路徑度量方式。

關鍵字：無線網狀網路、繞路協定、鏈結層協定、多重頻道、頻道分配、資源分配、IEEE 802.11、IEEE 802.16、隨意式網路

In this project, we develop some practical communication protocols for IEEE 802.11-based wireless mesh networks (WMNs) and IEEE 802.16-based WMNs respectively. We have designed a multi-channel link-layer protocol for this project at the first year. In second year, we further focus on the protocol design of the network layer (especially on the routing protocols). We point out that multi-path routing has to be used in concert with multi-channel design to improve end-to-end throughput. Thus, we propose a novel protocol for IEEE 802.11-based WMNs, which combines multi-channel link layer with multi-path routing. Beside above works in IEEE 802.11-based WMNs, we exploit spectral reuse in IEEE 802.16 mesh networks in the sense that it takes dynamic traffic loads of SSs into account and integrates not only a hierarchical bandwidth scheduling scheme for bandwidth adaptation and timeslot allocation, but also a routing algorithm with a tree optimization scheme. Our goal is to improve the performance of IEEE 802.16-based mesh networks. At third year, we find that many routing protocols have been proposed for mobile ad-hoc networks (MANETs) based on different criteria, few have considered the impact of multi-rate communication capability that is supported by many current WLAN products. Give a routing path, we provide an analytic tool to evaluate the expected throughput of the route with spectral reuse, assuming that hosts move following the discrete-time, random-walk model. Through verification with simulation, we believe that the path throughput is a better metric than

the traditional metrics, such as the hop count.

Keywords: Wireless Mesh Network, Routing Protocol, Link-layer Protocol, Multi-Channel, Channel Assignment, Resource Allocation, IEEE 802.11, IEEE 802.16, mobile ad-hoc networks.

目錄

附錄一：

"An Efficient MAC Protocol for Multi-Channel Mobile Ad Hoc Networks Based on Location Information"

論文全文，詳細資料如下：

Y.-C. Tseng, S.-L. Wu, C.-M. Chao, and J.-P. Sheu, "An Efficient MAC Protocol for Multi-Channel Mobile Ad Hoc Networks Based on Location Information", Int'l Journal Communication Systems, Vol. 19, 2006, pp. 877-896. (SCI)

附錄二：

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附錄三：

"Joint Multi-Channel Link Layer and Multi-Path Routing Design for Wireless Mesh Networks"

論文全文，詳細資料如下：

W.-H. Tam and Y.-C. Tseng, "*Joint Multi-Channel Link Layer and Multi-Path Routing Design for Wireless Mesh Networks*", IEEE INFOCOM, 2007.

附錄四：

"Exploiting Spectral Reuse in Resource Allocation, Scheduling, and Routing for IEEE 802.16 Mesh Networks"論文全文，詳細資料如下：

L.-W. Chen, Y.-C. Tseng, D.-W. Wang, and J.-J. Wu, "*Exploiting Spectral Reuse in Resource Allocation, Scheduling, and Routing for IEEE 802.16 Mesh Networks*", IEEE VTC, 2007-Fall.

附錄五：

"Route Throughput Analysis with Spectral Reuse for Multi-Rate Mobile Ad Hoc Networks"論文全文，詳細資料如下：

L.-W. Chen, W. Chu, Y.-C. Tseng, and J.-J. Wu, "Route Throughput Analysis with Spectral Reuse for Multi-Rate Mobile Ad Hoc Networks", Journal of Information Science and Engineering, to appear. (SCIE, EI)

附錄六：

附大陸出差或研習心得報告一份

會議/訪問時間地點	2007/8/8~2007/8/11 北京,中國
會 議 名 稱	互聯網服務主題研討會
發 表 論 文 題 目	Mobile GeoWeb Services

附錄七：

出席國際學術會議心得報告兩份

會議/訪問時間地點	第5屆一年一期網路計算與應用研討會 The 5th IEEE International Symposium on Network Computing and Applications (IEEE NCA06) 2006/07/24~2006/07/26, Boston, USA
會 議 名 稱	第5屆一年一期網路計算與應用研討會(IEEE NCA 2006)
發 表 論 文 題 目	Implementation of an Emergency Guiding System by Wireless Sensor Networks

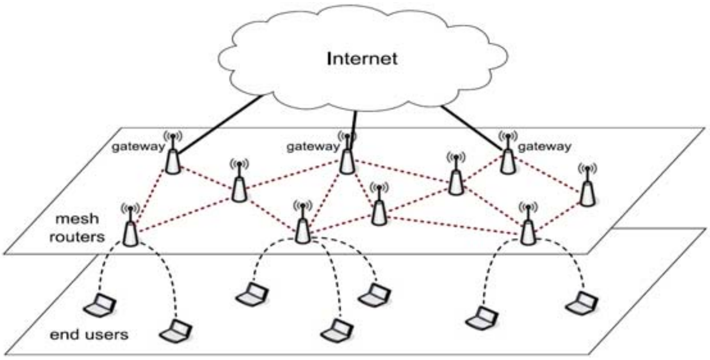
會議/訪問時間地點	16th International Conference on Computer Communications and Networks, Aug. 13 - 16, 2007, Honolulu, Hawaii USA
會 議 名 稱	ICCCN 2007
發 表 論 文 題 目	Exploring Load-Balance to Dispatch Mobile Sensors in Wireless Sensor Networks

一、前言

本計畫之主要目的，為實現一個多天線多模多頻道多速率之無線網狀網路(Multi-antenna Multi-mode Multi-channel Multi-rate Wireless Mesh Network, 簡稱M4無線網狀網路)，此新世代之網路架構可將無線區域網路(WLAN)之涵蓋範圍，以高彈性低建置成本的方式，延伸至企業、校園以至於更大規模之無線都會網路(Wireless Metropolitan Networks)，提供高承載、高速率、高可靠度之無線網路存取，並可達成服務品質(QoS)、無線漫遊(wireless roaming)、VoIP等整合數據、語音、多媒體之即時網路服務。無線網狀網路(Wireless Mesh Network)計畫已於2004年7月正式由IEEE通過，成立802.11s專案研究團隊，在此架構下，每一個擷取點(access point)的涵蓋範圍彼此重疊，並同時扮演資料收送及路由(routing)的角色，封包以多步跳躍 (multi-hop)的方式轉送(relay)至目的端，亦可透過閘道器(gate host)將整個無線網狀網路連接到鄰近的網際網路。

而本子計劃則著重於如何妥善分配天線至中繼節點、設定天線傳輸模式、配置資料傳送頻道、以及決定繞徑方式等問題，以滿足每一個中繼節點存取網際網路之資料傳送需求(此需求可能由連上中繼節點的行動主機Mobile Host 而來)，使行動主機能穩定、快速、順暢無礙地使用網際網路的服務。此外，通訊頻寬是無線網路的寶貴資源，而空間再利用(Spatial Reuse)是重要的資源分配原則，再加上無線通訊頻寬容易受到外在環境因素影響，例如天氣變化、電波干擾、以及發送競爭等，如何在M4 無線網狀網路上，做好通訊資源分配以增加中繼節點的累積頻寬(Aggregate Bandwidth)，使得閘道節點(Gateway Node)的資料傳輸量(Throughput)總和最大，是一個具有挑戰性的問題。

在第一年中我們主要在設計適用於此種網路的媒介存取層協定，設計的重點在於頻道的分配與媒介存取的機制。我們也已實作一個適用於多頻道無線網狀網路的鏈結層協定。在第二年，我們分別針對 IEEE 802.11-based [1] 無線網狀網路以及 IEEE 802.16-based [2] 無線網狀網路設計一些實用的通訊協定。首先我們先針對 IEEE 802.11-based 網狀網路做一個簡單的介紹，圖一為一個 IEEE 802.11-based 網狀網路的架構圖，一個網狀網路是由許多的 Mesh Router 所組成的，它們構成了網路的骨幹，並且可提供給 End Users 連接上網路，通常我們所討論的網狀網路是由 Mesh Router 所組成的部份，由於網狀網路下可能有許多的 End Users 連接上，因此網狀網路必須要能夠提供足夠的效能滿足所有使用者的需求。此外通常網狀網路中會有閘道器連接至有線網路上。

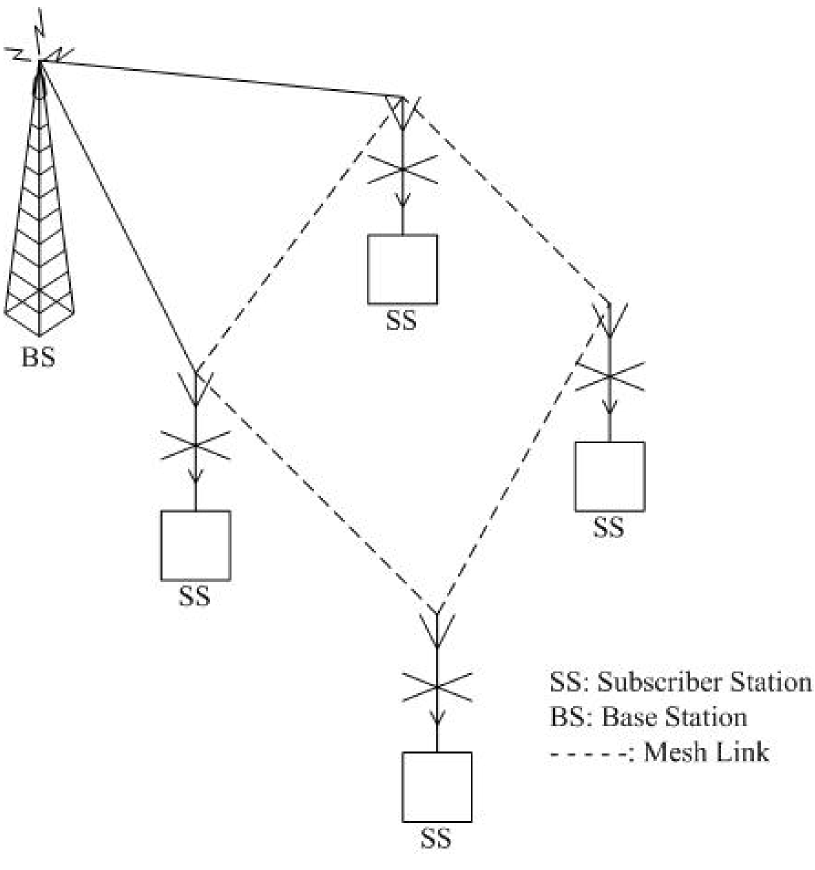


圖一、IEEE 802.11-based 無線網狀網路示意圖

而為了要提升無線網狀網路的效能，一個解決的方法是使用多重頻道，在 IEEE 802.11 網路

中有存在著數個不與其他頻道重疊(non-overlap)的頻道，雖然在 IEEE 802.11 Infrastructure mode 中，在 IEEE 802.11 Ad-hoc mode 中如何有效的利用複數頻道增進網路效能仍是一個值得研究的領域,例如 IEEE 802.11 的媒介存取協定(Medium Access Control Protocol)的設計只針對單一頻道的使用，因此造成效能無法進一步地提升。當複數個頻道被利用時，可預期的是網路的吞吐量(Throughput)會增加,干擾可降低,空間中頻道的再使用率(Spatial Reuse)能提高。在本計畫中，去年我們即針對多重頻道的網路環境提出了一個適用於多重頻道網路的鏈結層頻道管理協定，今年我們更進一步地去考量網路層設計(尤其著重在繞路協定)，我們認為多重頻道網路環境下必須搭配多重路徑繞路協定，才能將整體的效能提升至最高，因此我們提出了一個結合了鏈結層與網路層的通訊協定 JMM (Joint Multi-channel and Multi-path control)用於 IEEE 802.11-based 無線網狀網路上。我們並會以模擬實驗來證明跨越不同層(Cross-Layer)所設計出的通訊協定能夠改善使用多重頻道的 IEEE 802.11-based 網狀網路。

此外，適用於都會網路(Metropolitan Area Networks)的一代無線寬頻通訊標準 IEEE 802.16 在這幾年也開始被廣泛地討論，IEEE 802.16 可支援兩種模式，一是點對多點模式(Point-to-Multipoint mode, PMP)，另一則是網狀模式(Mesh Mode) ，在點對多點模式下，Subscriber Stations (SSs) 只可以直接和 Base Station(BS)進行通訊，因此 SS 必須要在 BS 的傳輸範圍內，而在 Mesh Mode 下，SS 則可扮演一個終端點或是扮演路由器幫助其它的 SS 將資料傳送給 BS，因此一個網狀網路的服務範圍可因此擴大，圖二顯示了一個 IEEE 802.16-based 網狀網路的架構圖。



圖二、IEEE 802.16-based 無線網狀網路示意圖

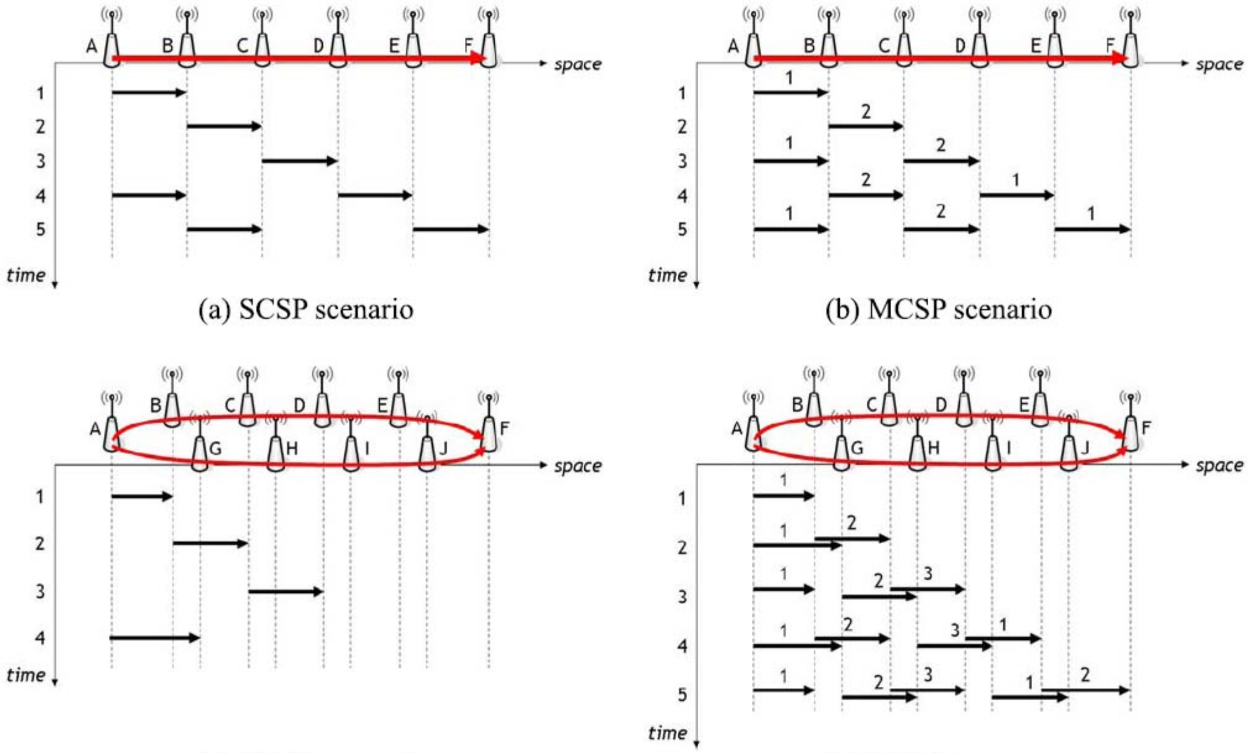
在 IEEE 802.16 網狀網路中每個 SS 會依其所分配到的時間去做資料的傳送，而分配可由兩種方式來達成，一是由每個 SS 以分散式的方式來進行，另一則是由 BS 收集 SS 的資料以集中式的方式來進行，在本計畫中我們著重在集中式的方法，我們發現原本的設計中完全沒有考量資源再利用(Reuse)的議題，因此在本計畫中我們提出了一個資源分配的方法，而我們也發現資源分配的好壞會跟使用的繞路樹有密切的關係，因此我們也提出了一個建繞路樹的演算法結合前述的資源分配方法，以改善 IEEE 802.16-based 網狀網路的效能。

在第三年的計劃中，我們針對多速率的隨意式網路的吞吐量分析提供一個量化的工作。我們相信路徑吞吐量會是一個比節點計數(hop count)更好的路徑度量方式。

二、研究目的

本計畫的研究目的要是增進網狀網路的效能，我們分別討論了 IEEE 802.11-based 網狀網路與 IEEE 802.16-based 網狀網路。首先我們討論 IEEE 802.11-based 網狀網路，為了達到我們的提升網狀網路效能的目的，我們考慮了多重頻道(Multiple-Channel)的使用，由於在 IEEE 802.11 網路中原本就存在著數個不與其他頻道重疊(non-overlap)的頻道，因此只要能善加利用，即可提升網路的效能，而我們更進一步地發現多重頻道網路環境下必須搭配多重路徑繞路協定，才能將整體的效能提升至最高，在這一節中我們將簡單地敘述我們所做的觀察。

首先我們觀察使用單一頻道及單一路徑(Single-Channel, Single-Path)的例子，如圖三(a)所示，我們假設在 Time 1 節點 A 傳送第一個封包給 B，而在 Time 2，由於 B 不可能同時傳送與接收的關係，A 無法傳送資料給 B，而在 Time 3，A 和 C 是無法同時傳送的，因為若 A 和 C 同時傳送，在節點 B 會發生碰狀，只有在 Time 4，A 可再進行傳輸，因此我們可預估最大的效能約只有 1/3。而在圖三(b)，我們分析多重頻道下單一路徑的例子，我們發現在 Time 2，A 仍然無法傳送，因為 B 無法同時進行傳送與接收，但是在 Time 3，如果 A 和 C 能夠使用不同的頻道則 A 可在 Time 3 再一次地進行傳送，因此效能可達 1/2。而在圖三(c)中，我們分析單一頻道多重路徑的例子，在 Time 2，我們發現 A 和 B 無法同是進行傳送，這是由於 A 和 B 是競爭者，當其中一個節點成功競爭到頻道使用權後，另外一個人會聽到贏家的訊號而抑制住自己的傳輸，這裏我們假設 B 取得頻道的使用權，而在 Time 3，A 仍然可能無法傳送資料給 G 因為 C 所發出的訊號可能會干擾到 G。最後我們分析我們所提出的多重頻道多重路徑，A 可在 Time 1 利用 Channel 1 傳資料給 B，而繼續地在 Time 2 傳送給 G，此時只要 B 使用和 A 不同的頻道即可在 Time 2 和 A 一起傳送，因此我們可預期使用多重頻道多重路徑將大幅地提升效能。因此在 IEEE 802.11-based 網狀網路下，我們的目的便是要設計中一個能夠使用多重頻道的多重路徑通訊協定。



(a) SCSP scenario (b) MCSP scenario (c) SCMP scenario (d) MCMP scenario

圖三、理想的封包排程於(a)單一頻道單一路徑，(b)多重頻道單一路徑，(c) 單一頻道多重路徑，(d) 多重頻道多重路徑的網路環境

而在 IEEE 802.16-based 網狀網路中，我們為了要達到提升網路效能的目的也做了一些觀察，我們發現在原本 IEEE 802.16 的設計中，每個 SS 會被分配到一個時間傳送，其傳送時整個網狀網路上的其它 SS 就無法進行傳送，然而這是不必要的，如果一個 SS 其 One-Hop 及 Two-Hop 的其它 SS 都沒有在傳送的話，此 SS 其實是可以傳送的，因此我們需要一個資源再利用的方法，在此方法下 SS 可盡量地去使用可傳送的時間，以達到提升網路效能的目的，而我們的目的便是設計出一個資源再利用的方法及相對應的繞路樹建構演算法。

三、文獻探討

此節我們同樣地分成 IEEE 802.11-based 網狀網路與 IEEE 802.16-based 網狀網路來討論。首先我們先討論 IEEE 802.11-based 網狀網路下一些既有的文獻，使用多重頻道的鏈結層協定（含媒介存取層協定）已經被廣泛地討論，相關學術研究發整列表如下，並簡單地分析它們的優缺點。

協定	[3]	[4]	[5]	[6][7]	[8]
頻道數	有限	有限	有限	有限	無限
MAC 協定	需新 MAC	802.11 相容	802.11 相容	802.11 相容	需新 MAC
網路卡數	兩張	均可	單張	多張	單張
時間同步	不需要	不需要	需要	不需要	不需要
演算法	分散式	集中式	分散式	分散式	分散式
應用網路	Ad-Hoc	Mesh	Mesh	Mesh	Ad Hoc

表一、使用複數頻道的通訊協定比較

上表列出了相關複數頻道應用在無線網路的範疇，在[8]中假設可用頻道數無限多，此假設在現實環境中並不合理，[3]為我們於多年前針對隨意網路所提出的媒介存取協定，在[3]中需要設計一個新的媒介存取層協定(Medium Access Protocol, MAC)，如此就無法使用已經被廣範使用的 IEEE 802.11 網路介面卡，在[6][7]中網路卡數必須多於一張才能實行，[5]為單張網路卡，利用不停的切換頻道來提高空間的再使用率，但必需時間同步，是其缺點，[4]中的演算法是集中式演算法，必須知道整個網路的拓撲。此外適用於多重頻道網路環境下的繞路協定也被許多文獻所探討，文獻[9]針對多重頻道複數收發器的 IEEE 802.11-based 網狀網路提出了一個樹狀的繞路演算法，然而使用複數收發器有時在成本的考量下是不可行的，而文獻[10]結合了頻道分配的議題提出了一個用於多頻道環境的繞路協定 CA-AODV，然而系統假設一個控制頻道是需要的，因此至少需要兩張網路卡是其缺點。另一個設計適用於多頻道環境的繞路協定的方法為將原有針對單一頻道環境設計的繞路演算法改良其所使用的 Metric 使其能適用於多頻道環境，在文獻[11]提出 WCETT (Weighted Cumulative Expected Transmission Time) metric 就有考量繞路路徑上所使用的頻道，然而其並沒有討論頻道分配的問題。

而在多重路徑的使用上，也已經被許多文獻探討[12][13][14][15][16]。根據有向無迴圈圖(Directed Acyclic Graph) ，文獻[12]提出的 TORA 可以支援多重路徑繞路，然而其並不保證路徑的分離。文在文獻[14]中提出的 SMR (Split Multi-path Routing)由於其不會將重複的 RREQ (Route Request)所丟棄，可以做到路徑的分離，然而代價就是會有較多的 RREQ 在網

路上傳送。[15]所提出的 AODVM 可將 AODV 做延伸，其也是依靠較多的 RREQ 來達成，因此同樣有較高的負擔。文獻[16]所提出的 AOMDV 則使用"Advertised hop count"的概念去保証不會有迴圈的產生並使用 Flooding 的一個特殊性質去達到鏈結的分離。然而在這些文獻中都沒有針對多重頻道的網路環境做討論。我們的方法同時則是同時考量了多重頻道與多重路徑的使用。

而在 IEEE 802.16 網狀網路中，也有許多的議題被討論著，例如文獻[17]討論了拓樸的設計問題，而文獻[18]則討論了封包的排程問題，文獻[19]則討論了 QOS 的支援問題。在本計畫中我們是討論 Spectral Reuse 的問題，同時也涉及了繞路與排程的問題，跟我們的方法較為相關的文獻說明如下。在文獻[20]中一個考慮了干擾的讓路與排程演算法被提出來，但是此演算法並沒有完全利用到 Spectral Reuse 並且沒有考量使用者 Traffice 的變化（其繞路樹是固定不變的）。文獻[21]討論一個新的 SS 如何加入現有的樹中，但是並沒有討論排程問題。而文獻[22]中指出繞路樹的形成會跟網路的效能有極大的關係，因此其考慮了繞路樹的重建，然而其沒有考量使用者的需求。相較於之前的文獻，我們提出了一個資源分配的方法，而我們也發現資源分配的好壞會跟使用的繞路樹有密切的關係，因此我們也提出了一個建繞路樹的演算法結合前述的資源分配方法，以改善 IEEE 802.16-based 網狀網路的效能。各方法的比較可參考表二。

	Scheduling		Routing	
Features	Reuse Quantification	Slot Assignment	Route Reconstruction	Load Awareness
Wei et al. [20]	N/A	Yes	N/A	N/A
Tao et al. [22]	N/A	Yes	Yes	N/A
Fu et al. [21]	N/A	N/A	N/A	N/A
Our work	Yes	Yes	Yes	Yes

表二、既有文獻與所提出資源分配方法的比較

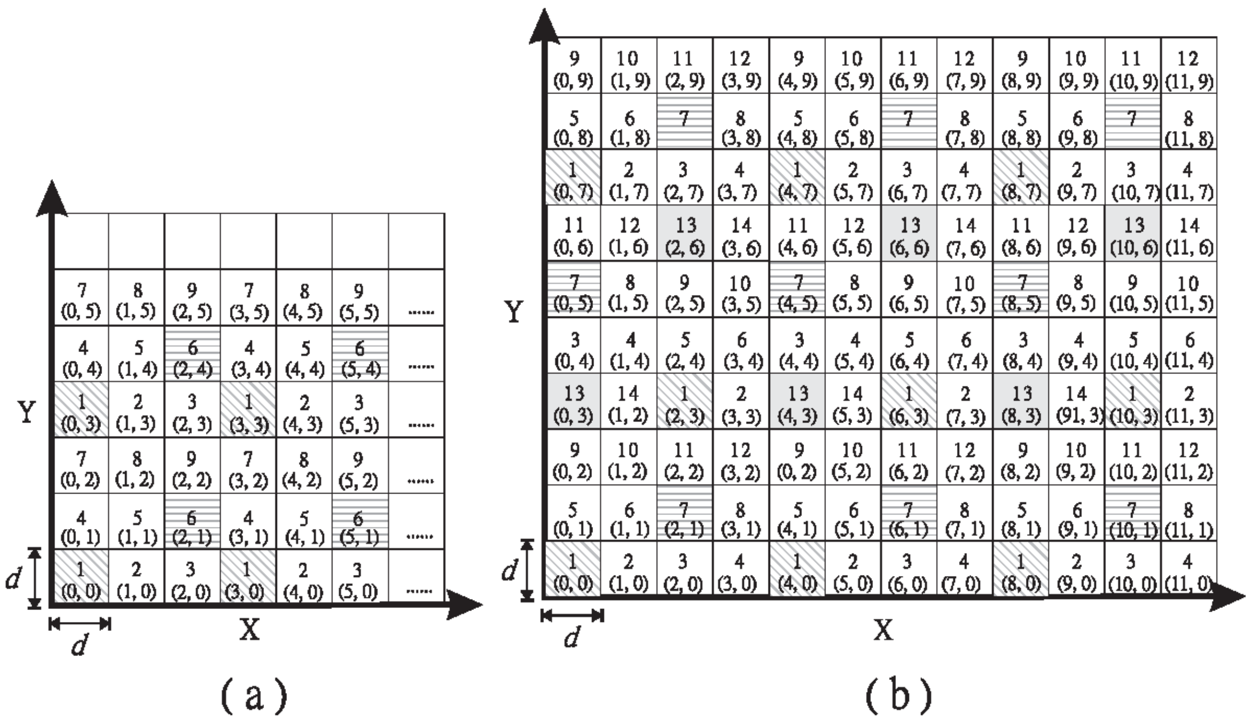
四、研究方法

在第一年計畫中我們針對多重頻道的使用，討論兩個相關的議題：（一）頻道分配(Channel Assignment)問題，（二）IEEE 802.11 相容的媒介存取機制。我們並且實作出了鏈結層(Link-layer)的通訊協定，實作的細節或在下面的報告以及附錄中說明。

以格子狀為基礎(Grid-based)的頻道分配法

在第一個議題方面，我們觀察到頻道分配的原則是要盡量的提高空間中頻道的再使用率(Spatial Reuse)，因此可知頻道的使用和空間是有相互的關係，基於上面的觀察，在我們提

出來的將空間切割成格子狀(Grid) ，一個 Grid 都會被分配到一個頻道，如圖五所示：



圖五、Grid-based 頻道分配法，圖 a 的總頻道數為 9，圖 b 的總頻道數為 14

藉由事先就將空間上所應該使用頻道分配好，我們可以盡可能地確保空間中頻道的再使用率(Spatial Reuse)，而這個方法有一個很重要的議題就是 Grid 的大小問題，我們認為 Grid 的大小會跟節點的傳輸範圍 (Transmission Range) 有關，我們假設一個 Grid 的大小為 ($d \times d$) ，節點的傳輸範圍為 r。我們針對不同的 r/d ratio 測量其效能。實驗的結果可於附錄一中查詢到。

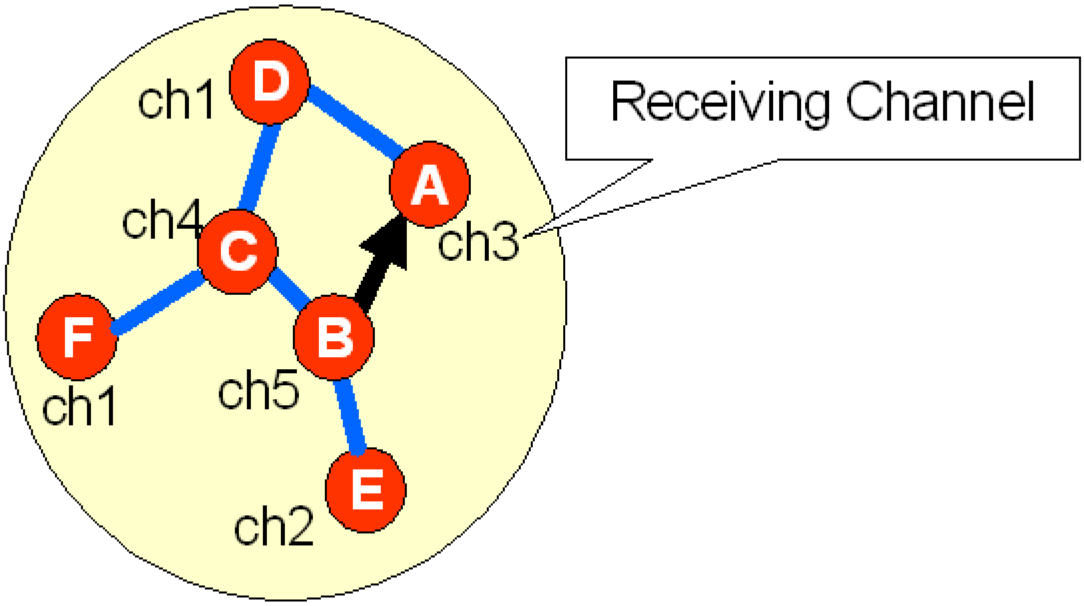
在解決完 Channel Assignment 的問題之後，我們也提出了一個相對應的媒介存取層協定(MAC Protocol)，此協定有點類似於我們之前所提出的 Multi-Channel Mac Protocol[3] ，然而在頻道的選擇方面，我們採用了直接在空間上分配好頻道使用的方法，以提高空間中頻道的再使用率(Spatial Reuse) 。此方法的細結與效能可於附錄一中查詢。

相容於 IEEE 802.11 的鏈結層媒介存取機制

由於我們之前所提出適用於多頻道環境的通訊協定（包含前面所提的 Grid-based Channel Assignment），大多需要修改到媒介存取層協定，也就是網路介面可能需要重新的設計，這和已經相當普及的 IEEE 802.11 WLAN 是有所抵觸的（因為使用者無法在使用原本的網路卡，而必須再另外購買新的網路介面卡），因此我們也試著提出一個能相容於 IEEE 802.11 的鏈結層媒介存取機制，並試著將此協定透過只需要更改網路卡驅動程式的方式於 Linux 平台上實作此機制。下面將先簡單地敘述我們所設計的通訊協定，之後會說明設計此協定時，哪些議題是要考量的，最後我們會敘述一下我們實作的方法。

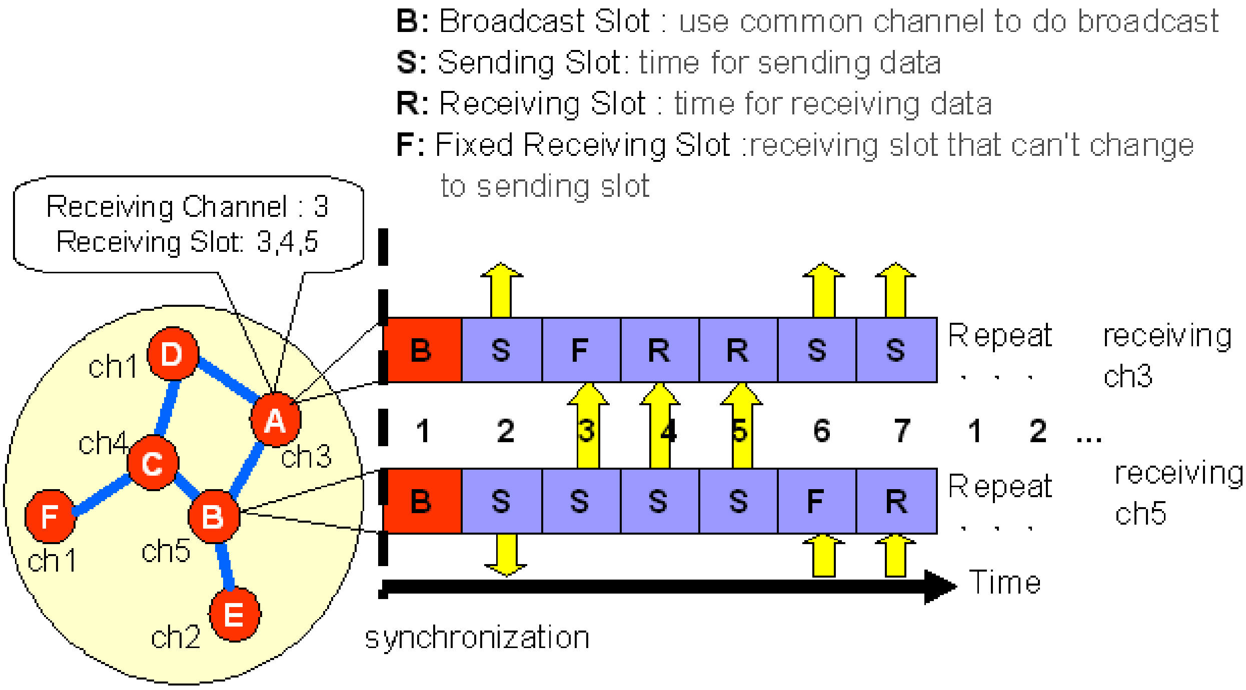
我們發展了一套適用於無線網狀網路(Wireless Mesh Network)，而可實作在鏈結層(Link layer)上使用複數頻道的頻道管理協定(Channel Management Protocol)，這是一個以接收端的

頻道(Receiver-based)做為傳輸頻道的方法，主要想是假設當有一個存取點(Access Point)A 要傳送資料給另一個存取點 B 時，A 就要切換到 B 所使用的頻道上進行通訊。我們假設每個存取點都會分配到一個接收頻道(receiving channel)，此頻道應該與存取點的鄰居(Neighbors)所使用的接收頻道(receiving channel)要盡量的不同。如圖六，存取點 A,B,C,D,E,F 都盡量的使用不同的頻道做為接收頻道(receiving channel)，例如 A 使用 Channel 3，B 使用 Channel 5，C 使用 Channel 4 等，假如 B 要傳輸資料給 A，會用 Channel 3 去傳輸資料，同時 C 要傳資料給 D，會用 Channel 1 去傳輸，所以在同一個鄰居區內，CD 和 AB 的傳輸會用不同的頻道，而降低干擾使網路吞吐量(Throughput)增加。



圖六、接收頻道示意圖

接收端為主(receiver based)的設計會產生一個問題，如果存取點 A 要傳送資料給存取點 B，A 會切換到 B 的接收頻道(receiving channel)上，但假設此時 B 也正要傳送資料給 C，則 B 會切換到 C 的接收頻道(receiving channel)上，如此有可能形成死結(deadlock)的情形，為此我們加進“分時”的概念，我們把每個存取點的時間軸切成一個一個的時槽(time slot)，我們假設每個存取點的時槽的開始是同步的，並設定每 k 個時槽為一個週期(cycle)，然後重覆這 k 個時槽。在這 k 個時槽裡，每個存取點（假設為 A）需指定好那些時槽是用來傳送資料給其它存取點，那些時槽是用來接收其它存取點所送過來的資料，然後把這個資訊廣播給其鄰居（假設為 B），當 B 收到這個資訊時，B 就能知道 A 的時槽使用情況，如此 B 有資料要傳送給 A 時，B 能夠知道 A 何時可接收資料，何時不能接收資料，當然 B 會選擇 A 可接收資料的時槽傳送資料給 A，除了傳送時槽、接收時候和廣播時槽外，我們還選了一個接收時槽當固定式接收時槽(Fixed Receiving Slot)，因為我們可以動態改變排程，所以接收時槽可能轉變為傳送時槽，為了不讓所有的時槽都變成傳送時槽，固定式的接收時槽是不能變成傳送時槽的，至於固定式接收時槽的選取，也是在鄰居區內要盡量不同。



圖七、Channel Model

如圖七：在這個例子中 k 值為 7，當存取點 B 要傳送資料給 A，因為 B 有收到 A 的廣播說 A 的時槽 3,4,5 是用來接收資料的時槽，因此 B 要送資料給 A 時，B 會利用時槽 3,4,5 將頻道切換到 A 的接收頻道(receiving channel)，在此例中為 Channel 3，來進行資料的傳輸。

另外要解決的問題是廣播(Broadcast)的問題，在複數頻道(Multi-channel)的網路環境下，因為每個存取點可能正使用不同的頻道，如何做有效率的廣播就是一個問題，相關研究中廣播的方法大多是複製多個廣播封包(Broadcast Packet)在每個頻道都廣播出去，以便確保鄰居都能接收到此廣播封包，我們所採取的做法是選擇第一個時槽裡當成廣播時槽(broadcast slot)，在這個廣播時槽裡，頻道會切換到一個大家共同的頻道，其目的就是要把所有的存取點在這個時候同時切換至此共同頻道上，如此一來，所有的存取點便能同時接收或傳送廣播封包，因為我們並沒有改變 IEEE 802.11 的 MAC 協定，所以這時的接收和傳送是經由 IEEE 802.11 的競爭機制在傳送。這樣做的好處在於每一次的廣播只需廣播一次便所有的存取點都接收的到，並不需要在每個頻道上做廣播的動作，也不需要複製多個廣播封包。有人會問:「這樣不是會造成頻道擁塞(channel congestion)？因為這個時候所有的存取點都切換到這個頻道上，造成封包過多超過這個頻道所能負荷的量。」我們想這是可以避免的，因為我們定義的一個週期的時槽數可經由廣播封包和一般封包的比例來設定，廣播時槽可以在一個週期不一定只有一個，可以有兩個或三個，可視這個網路的特性去調整這個參數。廣播時槽帶來的好處還不只這些，在複數頻道上同步是有困難的，因為所有的人不在相同的頻道上，有了這個廣播時槽，順使可以在這個時槽發送信號彈(beacon)來達成時間同步的效果。

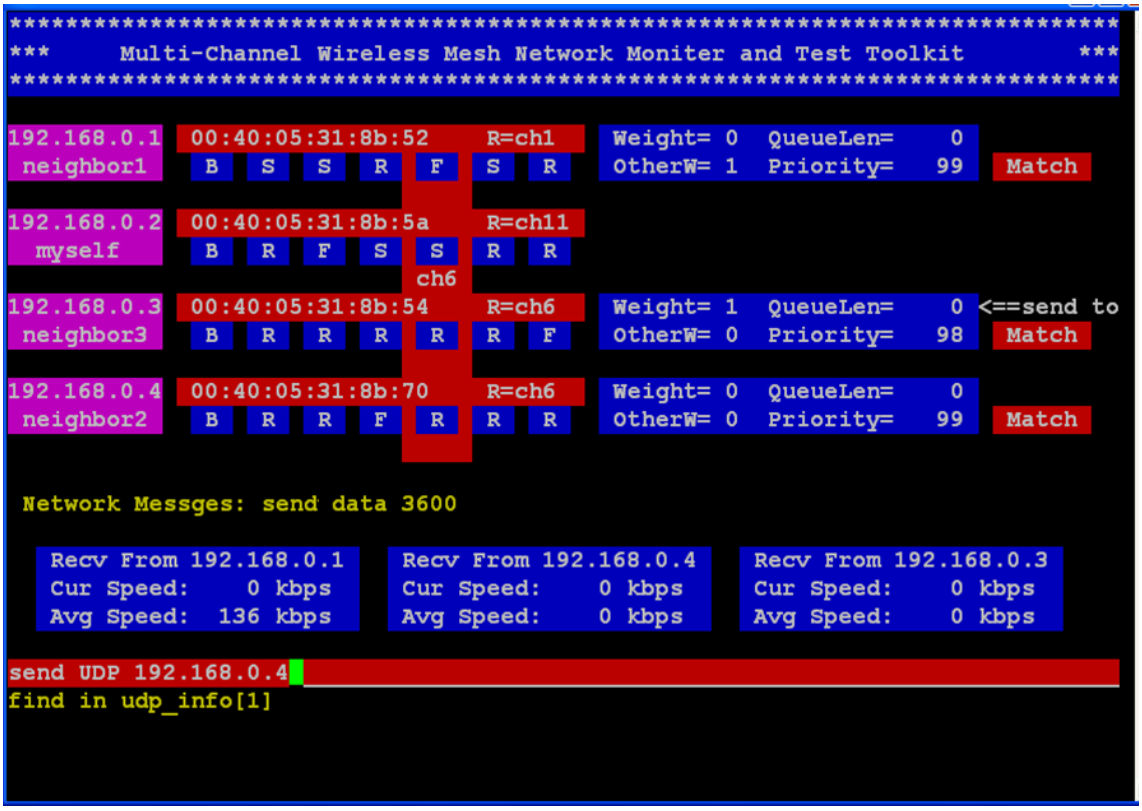
我們所提出的頻道管理協定，會衍生出一些待解決的議題如下：第一，如何決定每個存取點的接收頻道(receiving channel)？第二，如何分配每個存取點傳送時槽(sending slot)和接收時槽(receiving slot)的比例？第三，如何分配傳送時槽和接收時槽的順序？第四，進入某個傳送時槽時，要選擇傳送給那一個鄰居才不會造成不公平？針對這些議題我們分別設計了一些簡單的演算法去解決，詳細的演算法可在附錄二中查閱。而我們將演算法設計得較為

簡單的目的是為了可實作的考量，我們去修改網路卡的驅動程式以便將我們所設計的通訊協定實作出來，下面將針對我們實作的部份做一個簡單的描述。

為了在真的環境上面實作，我們去找尋有公開原始碼的驅動程式(Open Source Driver)，Atheros 有公開晶片 Linux 驅動程式的原始碼，使用 Atheros 晶片的網卡都可以使用這個驅動程式來驅動。所以我們選擇了幾台筆記型電腦，每台上面均裝有 D-link DWL-AG650 的網路卡，它是使用 Atheros 的晶片，這讓我們可以修改公開的原始碼來把我們的管理協定實作在上面。

圖八、實測環境

圖八為我們實作的環境，我們利用這些筆記型電腦當成是一個個的存取點，讓它們形成隨建即連網路(Ad-Hoc Network)，並固定其位置模擬無線網狀網路(Mesh Network)，無線網狀網路和隨建即連網路有很大的共通性，差別在於無線網狀網路有閘道可以連上網際網路(Internet)，且無線網狀網路沒有行動性(Mobility)，我們讓這些筆記型電腦模擬一個無線網狀網路的雛形(Prototype)來當成我們要的環境。



圖九、測試監控程式介面

圖九為我們的介面，為了便於展示我們的系統，我們開發了此一介面，利用此介面我們可以觀看每台筆記型電腦使用頻道的狀況、收送封包的狀況等等，同時我們也可利用此介面去做送封包的動作，次外我們也做了一些簡單的實驗來證明使用複數頻道的確可以帶來好處，詳細的實驗方法與數據可參閱附錄二。

在第二年我們針對 IEEE 802.11-based 網狀網路與 IEEE 802.16-based 網狀分別了提出了改進網路效能的方法，因此我們會分小節來分別討論針對 IEEE 802.11-based 網狀網路提出的多重頻、道多重路徑繞路協定與針對 IEEE 802.16-based 網狀網路提出的資源分配方法與繞路樹建構演算法。

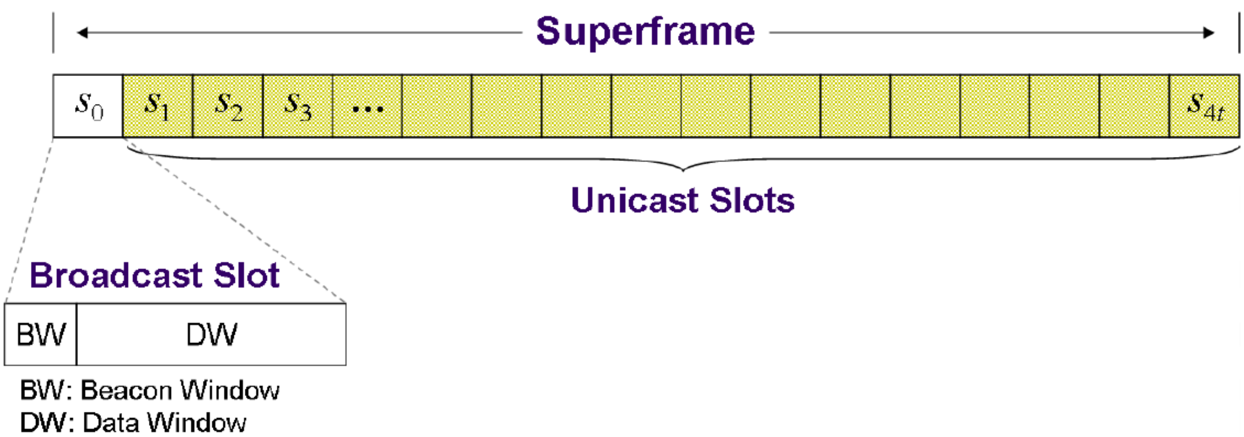
<u>IEEE 802.11 網狀網路下利用多重頻道的多重路徑繞路協定</u>

我們所提出的多重頻道多重路徑繞路協定是一跨層的設計，其結合了鏈結層與網路層，而此繞路協定主要有四個功能：

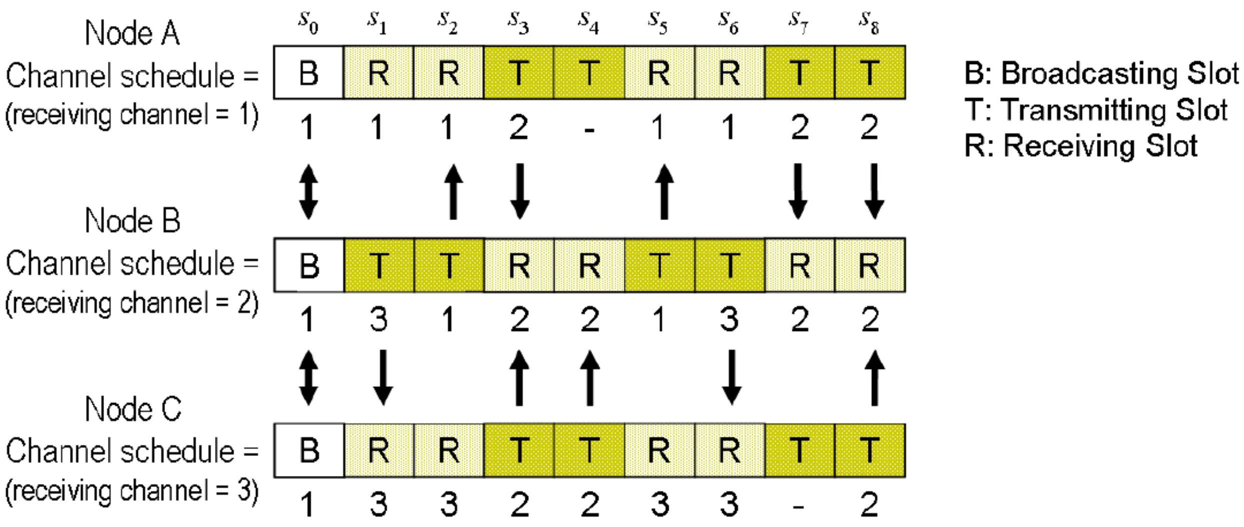
1. 此協定會決定每個節點所使用的接收頻道，以配合多重頻道環境
2. 此協定會幫每個節點建立兩條到閘道器的路徑
3. 此協定會幫每個節點決定其 Super-frame 的時槽分配
4. 此協定會依具每個節點的需求，動態調整傳送時槽與接收時槽的比例

其中(3)和(4)屬於鏈結層的問題，(1)和(2)屬於繞路的問題。我們將先介紹鏈結層部份的設計。

在鏈結層部份我們假設封包在傳送時，是類似於 TDMA 的方式，以一個時槽一個時槽的方式進行，因此我們定義了 Super-frame 如圖十所示，第一個時槽是用來做廣播之用，後面的部份則是用來傳送資料，其中後面的時槽又分為傳送時槽與接收時槽，每個節點會分配到一個接收頻道，它會在自己的接收時槽使用接收頻道去接收封包，而當它想要傳送資料給其它節點時，它會利用自己的傳送時槽及對方的接收頻道傳送資料給對方，圖十一為一個例子，T 表示傳送時槽，R 表示接收時槽，數字則為所使用的頻道。

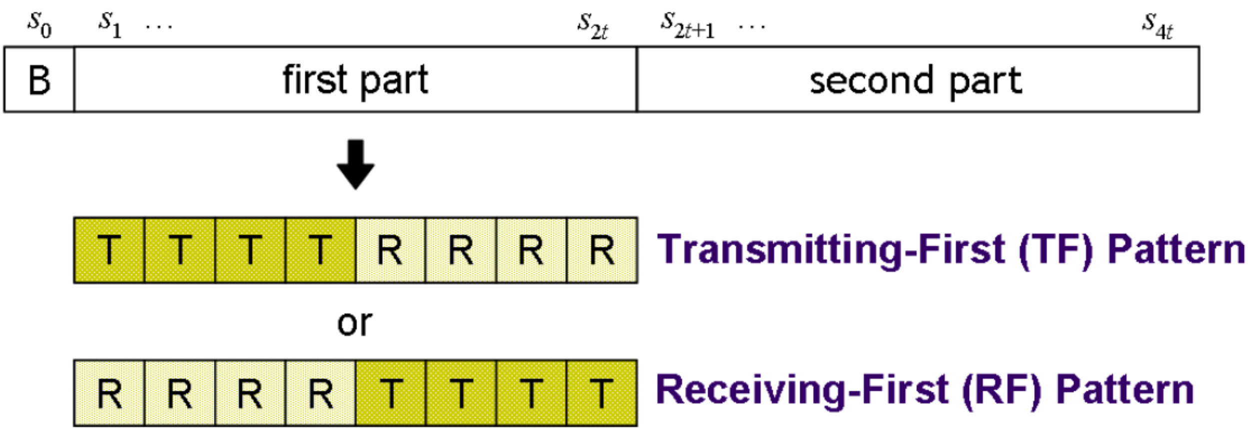


圖十、Super-frame



圖十一、傳送時槽、接收時槽與接收頻道的例子

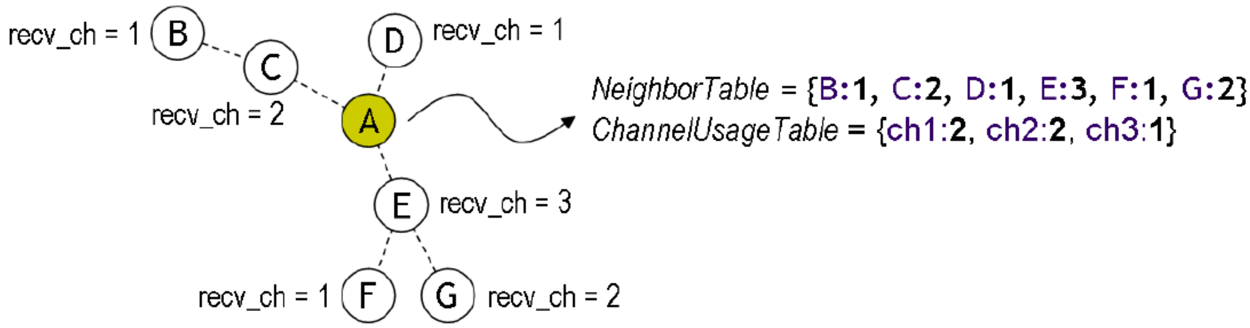
為了要和之後的繞路協定配合，我們將 Super-frame 中傳單播資料更進一步地分成兩部份，每一部份會有一個樣式，TF 指的是傳送時槽會在接收時槽之前，RF 指的是接收時槽會在傳送時槽之後，如圖六所示，因此整個 Super-frame 會有四個樣式，分別為 TF-TF, RF-RF, TF-RF, RF-TF，之後繞路協定的部份會說明每個節點如何決定自己的樣式。



圖十二、TF 與 RF 樣式

此外根據上下傳流量的不同，我們也設計了一個調整傳送時槽與接收時槽個數的演算法，並討論如何重排傳送與接收時槽，使得傳輸延遲可以降低，最後並討論如何在鏈結層建構一個佇列來進行封包的傳送，這些都是較細節的問題，詳細的演算法可在附錄三中查知。

而在繞路協定部份，首先我們要討論的是接收頻道的選擇，一個節點選頻道的原則是採用 Two-Hop Neighbors 中最少人用的頻道，如圖十三所示，節點 A 會選擇 Channel 3 去使用。



圖十三、接收頻道的選擇例子

而在繞路協定最重要的就是路徑的尋找，我們設計了一個 GREQ (Gateway REQuest) 封包做路徑的搜尋，GREQ 的格式如圖十四所示，而其它節點在收到 GREQ 後會根據自己是否知道通往閘道器的路以及自己距離閘道器的距離來決定該如何處理此 GREQ，處理的演算法為圖十五所示，而演算法的細節可參閱附錄三。而圖十六則顯示了一個例子。

STRUCTURE OF THE GREQ MESSAGE (S IS THE SOURCE NODE).

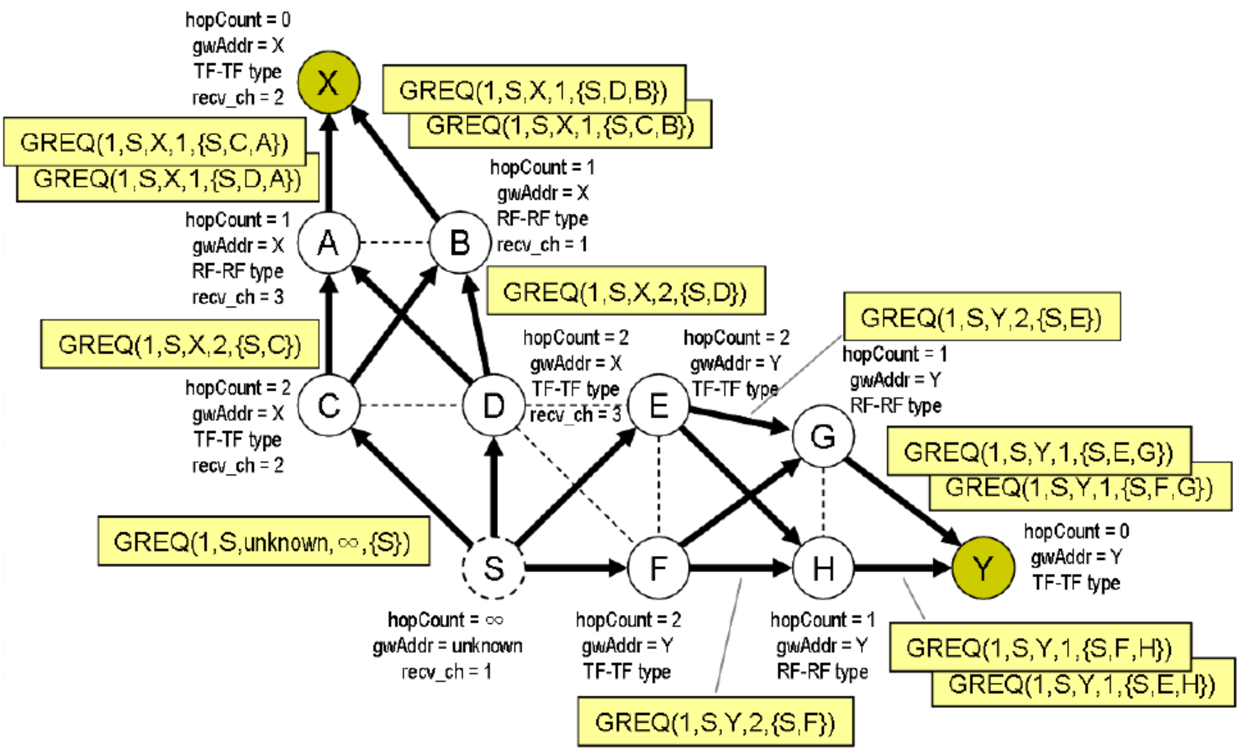
Field	Initial value	Meanings
$seqNum$	seqNum at S	the sequence number
$srcAddr$	S	the source address
$gwAddr$	unknown	the gateway address of the mesh network
$hopCount$	∞	the smallest number of hops to the gateway
$pathRecord$	$\{S\}$	the list of node records on the path

圖十四、GREQ 格式

當 GREQ 最後被閘道器收到時，一個節點可能可以發現有超過兩條的路徑到閘道器，閘道器所選出兩條，一條為主路徑(Master Path)，另一條則為副路徑(Slave Path) ，主路徑與副路徑的選擇會考慮 (1) 兩條路徑的共同節點數，共同節點數當然是要越少越好， (2) 頻道競爭狀況，由於多重頻道的使用，頻道的競爭應該要越小越好，(3) 訊號的強度，跟一般的繞路協定一樣，我們仍需考慮路徑的穩定度。最後閘道器會選出最好的兩條。

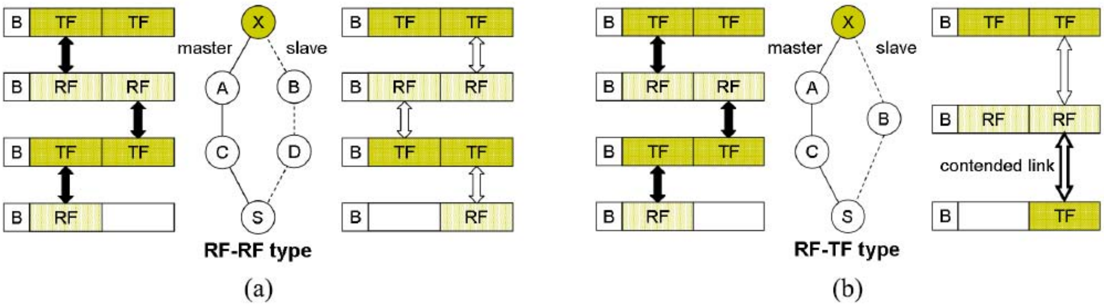
```
/*Executed when a non-gateway node R receives a GREQ from a node T */
begin
    if GREQ.seqNum < R.seqNum[srcAddr] then
        discard and exit;
    else
        R.seqNum[srcAddr] ← GREQ.seqNum;
    endif
    if the slot schedules of R and T mismatch then
        discard and exit;
    endif
    if GREQ.gwAddr ≠ unknown and
       GREQ.gwAddr ≠ R.gwAddr then
        discard and exit;
    endif
    /* Ensure that hopCount progressively decreases */
    if GREQ.hopCount < R.hopCount then
        discard and exit;
    elseif GREQ.hopCount = R.hopCount then
        if R ∈ GREQ.pathRecord then
            discard and exit;
        endif
    endif
    send GREQ(GREQ.seqNum, GREQ.srcAddr, R.gwAddr,
                  R.hopCount, GREQ.pathRecord ∪ {R});
end
```

圖十五、GREQ 處理演算法



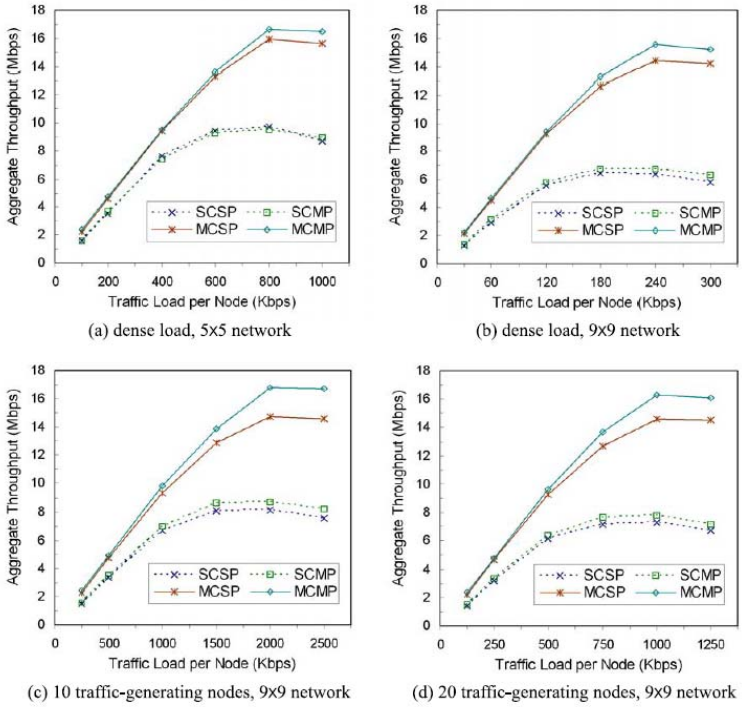
圖十六、多重路徑繞路協定例子

在繞路協定的最後部份就是要決定 Super-frame 的樣式，樣式要依據所選出的主路徑與副路徑來決定，例如如圖十七所示，當主副路徑的長度差是偶數時，與長度差是奇數時，我們需採用不同的樣式，以確保一個節點可和其路徑上的父節點與子節點通訊正常，此部份的詳細細節可參閱附錄三。



圖十七、Super-frame 樣式的決定，副路徑長度差為(a)偶數(b)奇數

針為 IEEE 802.11-based 網狀網路所提出的多重頻道多重路徑協定，我們也使用了 NCTUns Network Simulator 2.0 [23]來進行模擬，圖十八為模擬的結果之一，我們可發現多重頻道搭配多重路徑的確可提升網狀網路的效能。

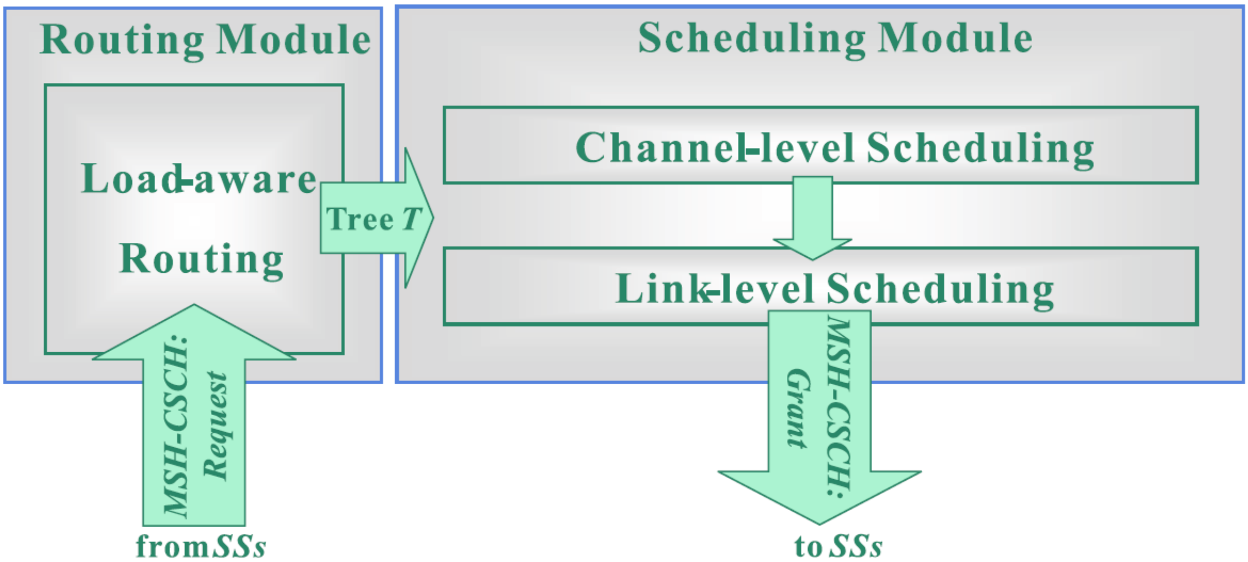


圖十八、多重頻道多重路徑協定模擬結果

<u>IEEE 802.16 網狀網路下資源分配問題與繞路演算法設計</u>

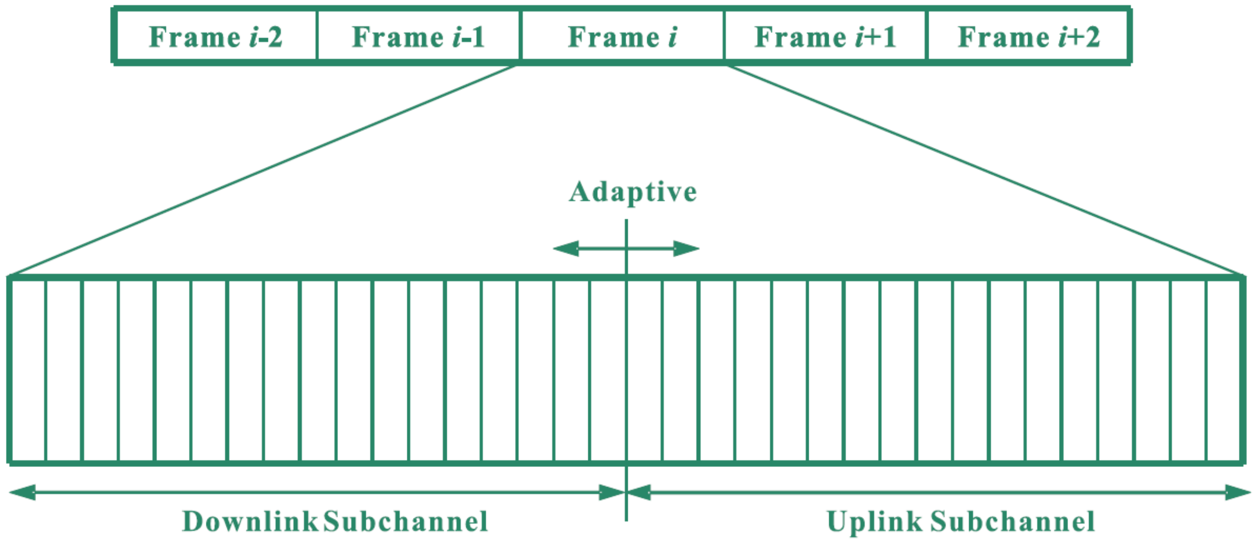
在前一節中我們討論 IEEE 802.11-based 網狀網路，而在這一節中我們則將重心放在 IEEE 802.16-based 網狀網路，我們提出了一個 Spectral Reuse 的架構，圖十九顯示了在 BS 中我們所提出的系統模組，此模組包含了一個繞路(Routing)模組與排程(Scheduling)模組，繞路模組會去收集頻道的狀況與使用者對於頻寬的需求而建構出一個繞路樹，而排程模組會接著進行資源的分配，包含了 Channel-Level 的排程與 Link-Level 的排程，Channel-Level 的排程主要是討論上傳與下載的頻寬比該如何決定，而 Link-Level 的排程則是說明時槽如何分

配給 SSs。



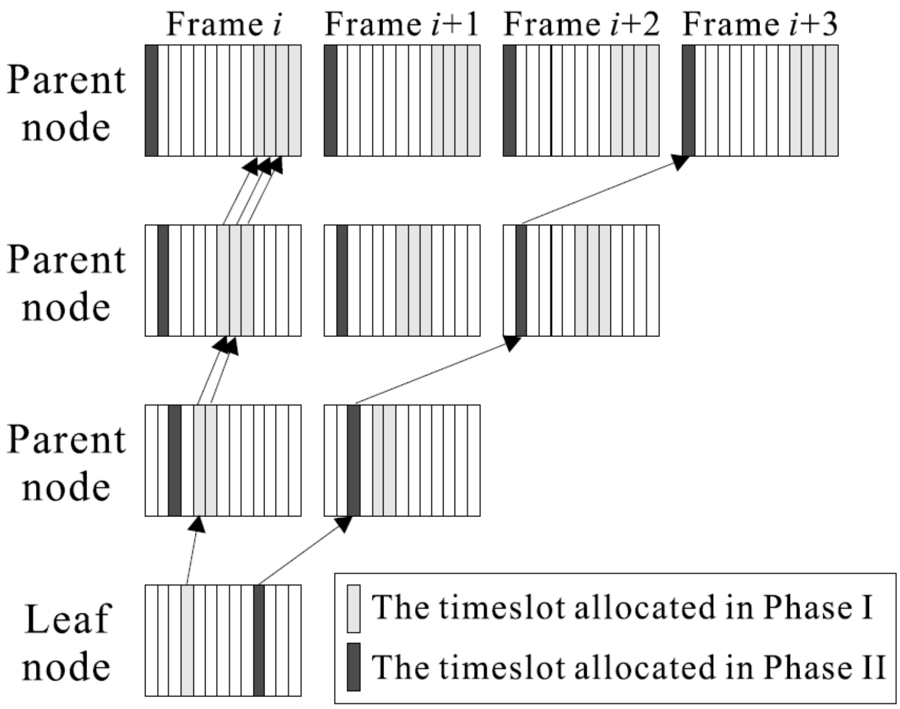
圖十九、BS 中的系統模組

資源分配與排程：這部份需分成兩方面討論，首先討論 Channel-Level 排程，由於 Channel 的存取使採用 TDD (Time Division Duplex)方式，我們必須決定上傳(Uplink)與下載(Downlink)的比例，如圖二十所示，我們發現在原本 IEEE 802.16 的設計中，每個 SS 會被分配到一個時間傳送，其傳送時整個網狀網路上的其它 SS 就無法進行傳送，當資源無法重複利用時，上傳下載的比例就是採用 $C^{u}_{Total}/C^{d}_{Total}$（此式考慮了網路上所有的點），然而這是不必要的，如果一個 SS 其 One-Hop 及 Two-Hop 的其它 SS 都沒有在傳送的話，此 SS 其實是可以傳送的，因此我們需要一個資源再利用的方法，在此方法下 SS 可盡量地去使用可傳送的時間，以達到提升網路效能的目的，而我們的目的便是設計出一個資源再利用的方法及相對應的繞路樹建構演算法。，而當資源能夠在不被干擾情況下被再利用時，我們可將比例調整成 $C^{u}_{\max}/C^{d}_{\max}$，(其中 $C^{u}_{\max}$ 與 $C^{d}_{\max}$ 就是考慮 One-Hop 與 Two-Hop 的 SS 的情況下會遇到瓶阱的點的最大傳送時間，由於不是考慮所有網路的點，因此代表資源可被再利用)，詳細的符號定義與數學式可參閱附錄四。



圖二十、Time Division Duplex (TDD) Framing

接著是討論 Link-Level 排程，我們將一個 SS 可用的時槽分成兩部份，第一部份是原本就要分配給此 SS 的時槽也就是 $N \times (T_i^u / C_{Total}^u)$ 這部份，我們稱為 Phase I，這部份在分配時有一個原則是子 SS (Child)所分配的時槽必須在父 SS (Parent)所分配時槽前面，如圖二十一的例子，由於有子節點需要在父節點前的這個特性，資料在此部份時槽傳送可降低延遲，因此此部份適合用來傳送即時性(Real-Time)資料，另一部份由於 SS 可在其 One-Hop 與 Two-Hop 鄰居都沒傳送時獲得一些額外的時槽，也就是 $N \times (T_i^u / C_{\max}^u - T_i^u / C_{Total}^u)$ 的部份，這部份(稱為 Phase II)在安排上就沒有子父節點的順序關係，適合拿來送送 Best-Effort 或非即時性資料。圖二十二顯示 Link-Level 排程演算法，詳細的數學式則可參閱附錄四。



圖二十一、時槽分配的例子

Link-level scheduling algorithm

Phase I:

Allocate $N \times (T_i^u / C_{total}^u)$ timeslots to each SS i according to the transmission order of MSH-CSCH:Request until all SSs have been allocated.

Phase II:

(1) Construct the frame allocation list L_i of E_i for each SS i in the network.

(2) According to the transmission order of MSH-CSCH:Request, assign the first $N \times (T_i^u / C_{max}^u - T_i^u / C_{total}^u)$ free timeslots in L_i to SS i.

(3) Update all frame allocation lists L_j that E_j includes SS i.

(4) Repeat steps (2) and (3) until all SSs have been assigned.

圖二十二、Link-Level 排程演算法

繞路模組：前面所提的排程模組，其效能會由一個參數($C_{\max}^u$)決定，而我們發現繞路樹的好壞會影響此參數的值，因此我們亦提出一個繞路樹的建構演算法，首先我們定義了此問題下，我們並證明其為 NP-Complete。

The Problem

Given a directed mesh network graph $G = (V, E)$, the traffic demand b_i requested by vertex $i \in V$, the uplink data rate r_j^u of edge $j \in E$, and a real number R, determine whether G has a routing tree such that its $C_{max}^u \leq R$.

Theorem: *The routing tree construction problem is NP-complete.*

證明的過程可參閱附錄四，因此我們提出一個以 Bottom-up 的方式的 Heuristic 演算法，在此演算法中我們將使用者的資料量考量進去，因此是一個 Load-aware 的繞路演算法，整個演算法如下：(細節可參閱附錄四)

Load-aware routing algorithm

(1) Let A be the set of SSs without a parent node that have the largest hop count, and B the empty set

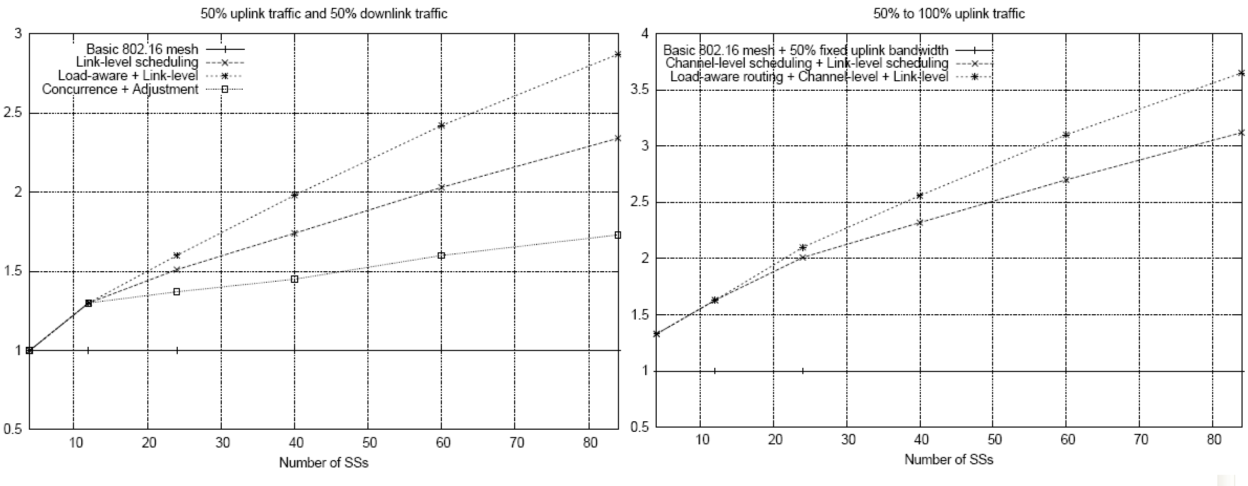
(2) Estimate each C_k^u for all neighbors k with less or equal hop count when SS i in A becomes the child of SS k, and the SS with the smallest C_k^u will be chosen as the parent node of SS i

(3) Remove SS i from A, add SS i into B, and update C_l^u for all SS $l \in E_i \cup E_k$

(4) Repeat steps (2) and (3) until there is no SS in A

(5) Repeat steps (1) $\sim$ (4) until each SS has a parent node

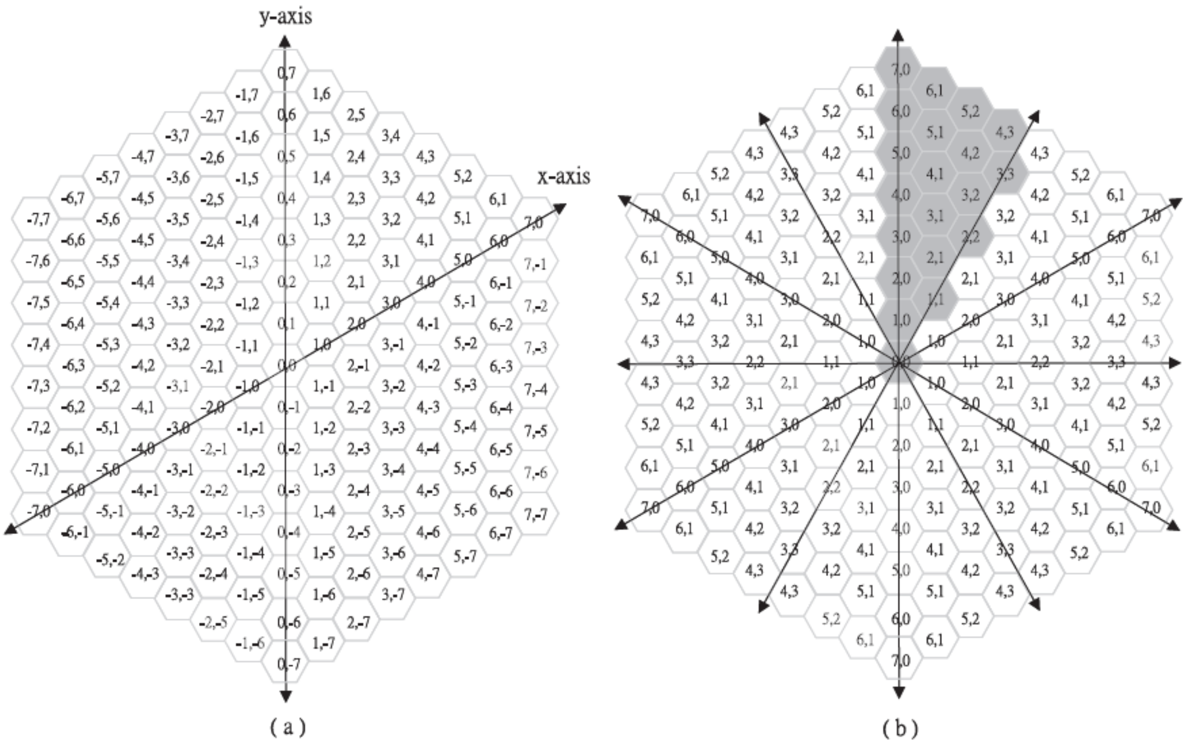
最後我們也用 NS-2 [24]證明我們所提出的方法的確可改善 IEEE 802.16-based 網狀網路的效能。圖二十三顯示了部份實驗結果。



圖二十三、所提出資源再利用方法的模擬結果

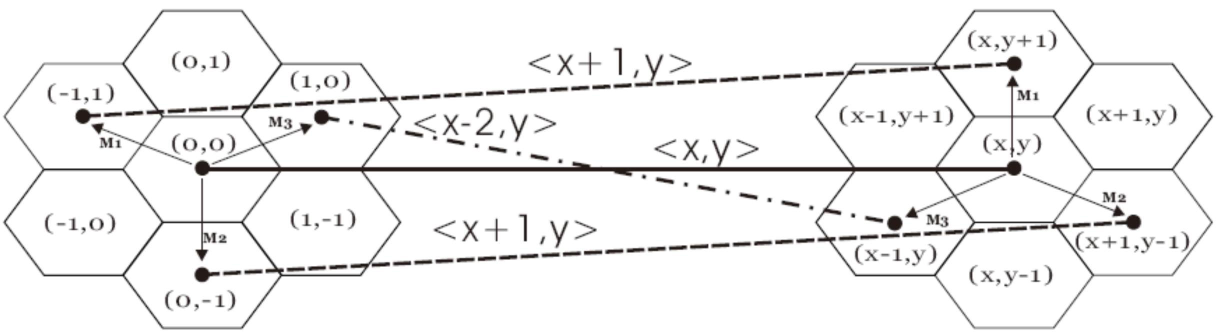
多速率隨意式網路下頻譜再使用的路由吞吐量分析

首先，我們提出我們探討的系統模型，如圖二十四(a)所示:



圖二十四、系統概觀

整個網路由六邊形的細胞(cell)網路所構成，在 x 軸座標上的節點標示成(x,0)，在 y 軸座標上的節點標示成(0,y)，我們把時間切割成適當的小單位，讓每個節點的移動看成可分解的動作，節點從某一個細胞節點開始移動時，往任何方向移動的機率都是相同的 (也就是說往六邊形每個方向的機率都是 1/6)。我們更進一步將網路做切割成多個層次，細胞節點(0,0)代表 layer-0，細胞節點(0,0)代表 layer-1，其它以此類推，對於一個 layer n 來說，總共有 $3n^2+3n+1$ 個節點屬於同一個 layer。



圖二十五、移動向量

我們利用移動向量來表示一個節點的移動情況，移動向量如圖二十五所示，當一個節點從(x,y) 移動到 (x’,y’)的時候，我們用一個向量 <x’-x, y’-y> 來表示節點移動，例如一個節點由座標(0,1) 移動到 (3,1) 再移動到(7,-3) 時，此節點的移動向量分別表示成

[<3,0>,<4,-4>]。經由上述的移動向量模組，我們可以計算出當一個節點初始位置在(x,y)，經過一個時間單位後，此節點移動到(x',y')的機率，整個機率分佈如圖二十六所示:

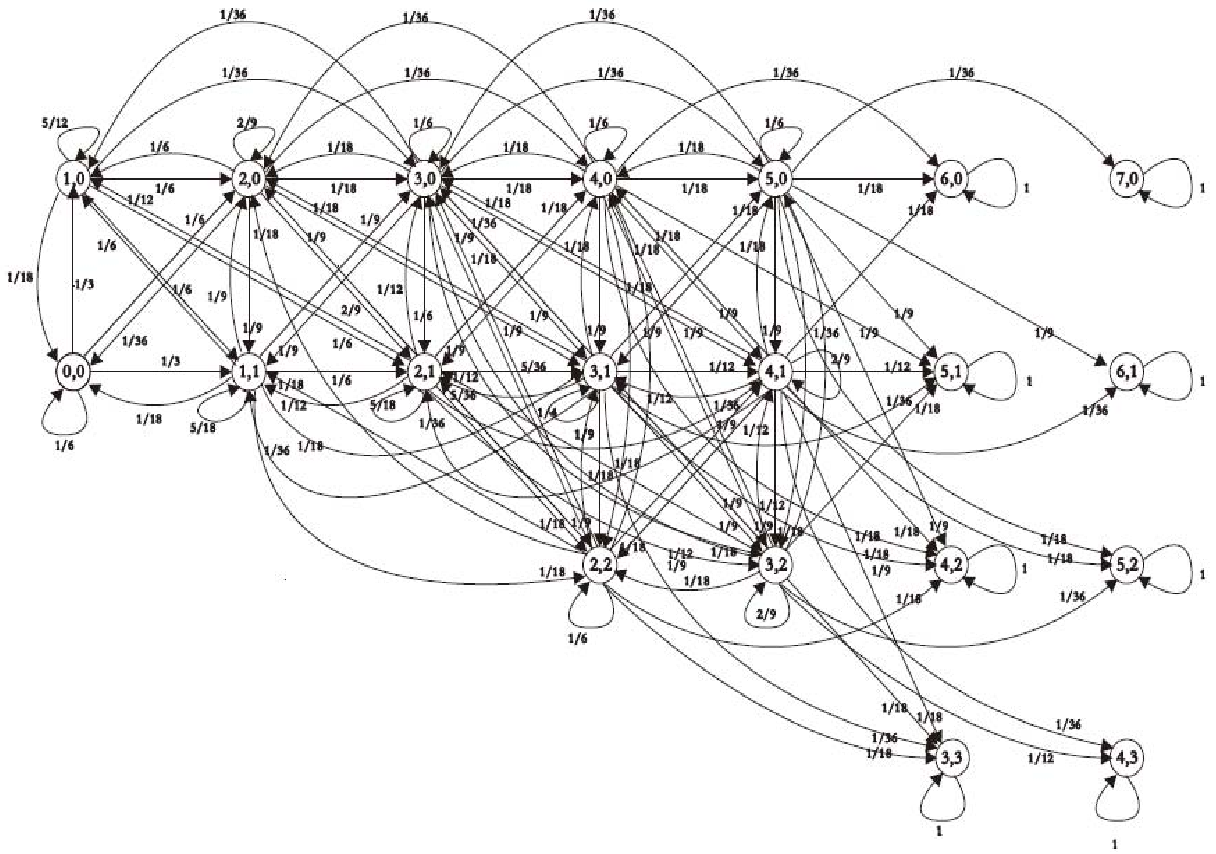
<x',y'>	<x,y>	<x-1,y>	<x-1,y-1>	<x,y-2>	<x+1,y-2>	<x+1,y-1>	<x+1,y>	<x,y-1>	<x+2,y-2>	<x+2,y-1>
Probability	6/36	2/36	2/36	1/36	2/36	2/36	2/36	2/36	1/36	2/36

<x',y'>	<x+1,y+1>	<x,y+1>	<x+2,y>	<x,y+2>	<x-1,y+2>	<x-1,y+1>	<x-2,y+2>	<x-2,y+1>	<x-2,y>
Probability	2/36	2/36	1/36	1/36	2/36	2/36	1/36	2/36	1/36

圖二十六、節點從(x,y)移動到(x',y')的機率

我們可以發現當某一節點從(x,y)經過一個時間點之後，移動到(x',y')時，共有 19 種可能的(x',y')。

接著我們開始討論我們提出的路由吞吐量分析，我們的目標是希望在路由已由的狀況下，去計算從起點到終點的路由吞吐量。首先我們先假設在一個 7 layer 的網路環境下，節點的傳輸半徑為 5 layer，剛開始節點是位於(0,0)的位置，節點的狀態傳輸圖如圖二十七所示:



圖二十七、狀態傳輸圖

圖二十七的整個狀態轉換可以用一個矩陣 M 來表示，如圖二十八所示:

$$M = \begin{array}{c} \\ \langle 0,0\rangle \\ \langle 1,0\rangle \\ \langle 2,0\rangle \\ \langle 1,1\rangle \\ \vdots \\ \langle 5,2\rangle \\ \langle 4,3\rangle \end{array} \begin{array}{c} \begin{array}{ccccccc} \langle 0,0\rangle & \langle 1,0\rangle & \langle 2,0\rangle & \langle 1,1\rangle & \cdots & \langle 5,2\rangle & \langle 4,3\rangle \end{array} \\ \begin{bmatrix} \frac{6}{36} & \frac{12}{36} & \frac{6}{36} & \frac{12}{36} & \cdots & \frac{0}{36} & \frac{0}{36} \\ \frac{2}{36} & \frac{15}{36} & \frac{6}{36} & \frac{6}{36} & \cdots & \frac{0}{36} & \frac{0}{36} \\ \frac{1}{36} & \frac{6}{36} & \frac{8}{36} & \frac{4}{36} & \cdots & \frac{0}{36} & \frac{0}{36} \\ \frac{2}{36} & \frac{6}{36} & \frac{4}{36} & \frac{10}{36} & \cdots & \frac{0}{36} & \frac{0}{36} \\ \vdots & \vdots & \vdots & \vdots & \ddots & \vdots & \vdots \\ \frac{0}{36} & \frac{0}{36} & \frac{0}{36} & \frac{0}{36} & \cdots & \frac{36}{36} & \frac{0}{36} \\ \frac{0}{36} & \frac{0}{36} & \frac{0}{36} & \frac{0}{36} & \cdots & \frac{0}{36} & \frac{36}{36} \end{bmatrix} \end{array}$$

圖二十八、狀態矩陣

接著我們考量上述的機率分佈、鏈結存活機率跟路徑存活機率等等因素，推導出對一個路徑 R 的預期吞吐量函式，函式如圖二十九所示，詳細的推導請參考附錄五。

$$E(R) = \sum_{t_1=1}^{\infty} \left(P_5(R, t_1) \times \sum_{t_2=1}^{t_1} \frac{B(R, t_2)}{t_1} \right). \tag{1}$$

圖二十九、對某一路徑 R 的預期吞吐量

在圖二十九中，B(R,t) 這個 function 包含了整條 path 上每個節點的傳輸速率等參數，整個 B(R,t)如圖三十所示:

$$B(R,t) = \sum_{i_1=1}^{m} \sum_{i_2=1}^{m} \cdots \sum_{i_\alpha=1}^{m} P_6(s_1, R_{i_1}, t)$$
$$\times P_6(s_2, R_{i_2}, t) \times \cdots \times P_6(s_\alpha, R_{i_\alpha}, t) \times f(R_{i_1}, R_{i_2}, \cdots, R_{i_\alpha})$$

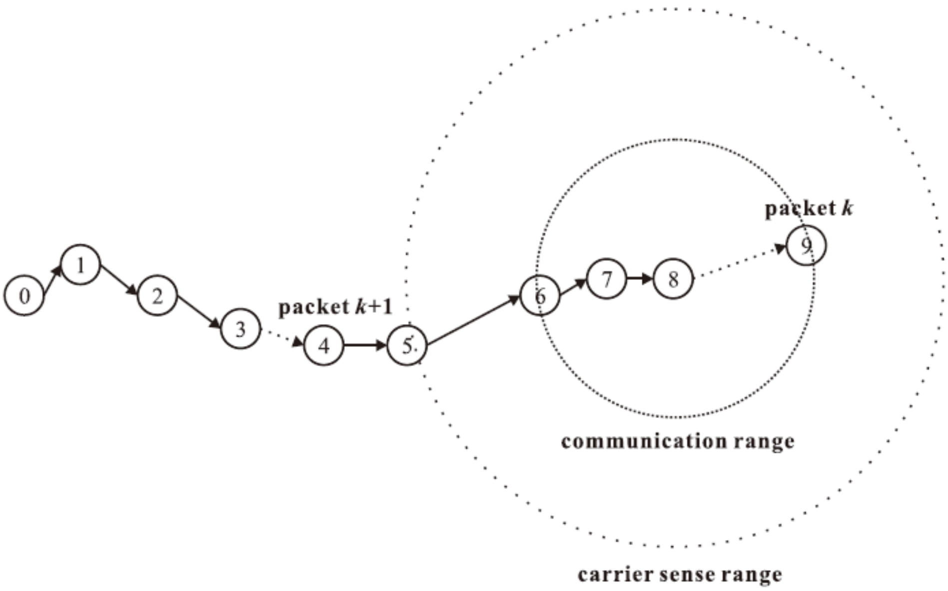
(a)

$$f(r_1, r_2, \cdots, r_\alpha) = \frac{1}{\max_{i=1\sim(\alpha-1)}\{\frac{1}{r_{(i+1)}} + \sum_{j \in G_i} \frac{1}{r_j}\}},$$

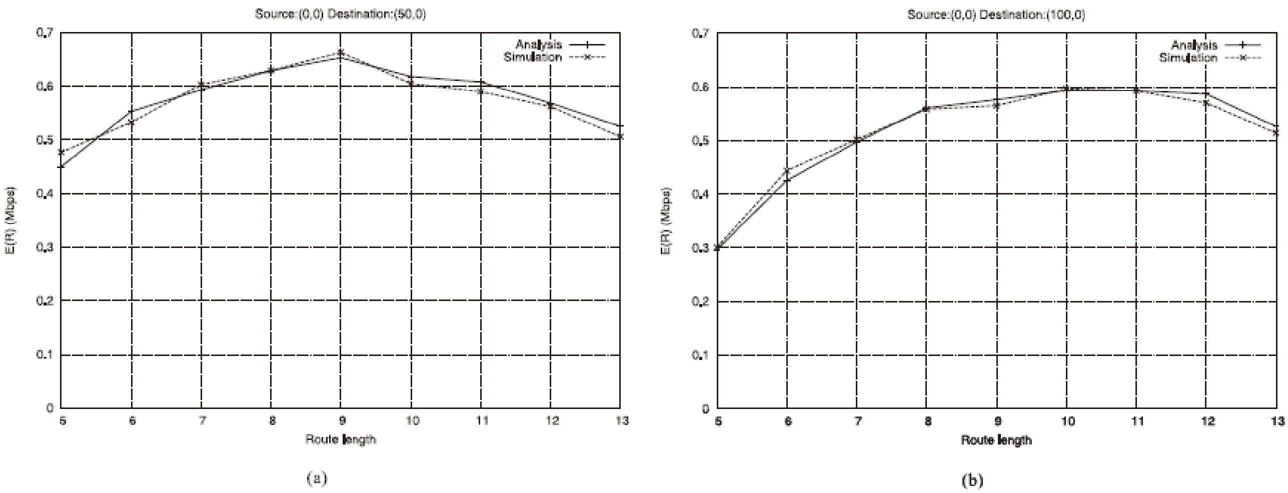
(b)

圖三十、(a) B(R,t) 函式 (b)B(R,t)函式中對多 hop 吞吐量估計

除了對路徑的吞吐量提出了一個預估的模組，我們還有分別考量傳輸半徑與干擾半徑等議題，如圖三十一所示:



圖三十一、路徑的傳輸與干擾區域



圖三十二、部分模擬結果

圖三十二是我們部分的模擬結果。

五、結果與討論

無線網狀網路融合了無線區域網路(Wireless LAN)和隨建即連(Ad Hoc)網路的優勢，同時低佈建成本也可進一步地促進無線網路的普及，然而其必須能提供足夠的頻寬以滿足使用者的需求，在此計畫中我們利用多重頻道(multi-channel)、多重天線(multi-antenna)等的方式設法提高無線網狀網路的效能。

在第一年中我們著重在設計適用於多重頻道環境的通訊協定，首先我們設計了一個 Grid-based 的 Channel Assignment 及其對應的媒介存取層協定，藉由實驗我們發現適當地在空間上做切割之後再分配頻道的方法的確可提高空間中頻道的再使用率(Spatial Reuse) ，如此可讓使用多重頻道的好處更進一步地展現出來。然而此方法需要修改原有的網路卡設計，並不太符合現實面上的考量，因此我們在今年發展了一個相容於 IEEE 802.11 的媒介存取機制，並在鏈結層上實作出來，最後我們也以實作的平台驗證使用多重頻道的確可提升網路的效益。

在第二年中我們分別針對 IEEE 802.11-based 網狀網路與 IEEE 802.16-based 網狀網路提出一個改進的方法。在 IEEE 802.11-based 網狀網路中，為了要提升網狀網路的效能，我們考量一個使用多重頻道的網路環境，而為了搭配此多重頻道環境，我們認為多重路徑的繞路協定是需要的，因此我們提出了一個跨越了鏈結層與網路層的多重路徑繞路協定，並用模擬去證明其效能，而相關的論文也發表在今年國際相當頂級的會議 INFOCOM(26-th Annual IEEE Conference on Computer Communications, 2007)中。在 IEEE 802.16-based 網狀網路中，我們則提出了一個資源分配方法並提出一個搭配的繞路樹建構演算法，藉由我們所提出的方法，一個 SS 可在其 One-Hop 及 Two-Hop 的 SS 不傳送資料時，去利用可用的資源。我們同樣用模擬證明其效能，而相關的論文也發表在今年國際有名的會議 VTC(66-th Semi-Annual IEEE Vehicular Technology Conference, 2007-Fall)中。

在第三年中我們針對隨意式網路 (MANET) 設計了一個度量路徑吞吐量的工具，經由模擬可以證實我們所估計的路徑吞吐量跟實際的吞吐量相當地接近。我們相信用路徑吞吐量來衡量一條傳輸路徑的好壞，比選用節點數目(hop count)來的更有好處，因為節點數目來建立路徑的方式無法考量封包遺失率跟每個節點所剩的可用頻寬。這篇論文將發表在期刊 ”Information Science and Engineering” 上。

本計畫的研究成果如下：

第一年:

Y.-C. Tseng, S.-L. Wu, C.-M. Chao, and J.-P. Sheu, “An Efficient MAC Protocol for Multi-Channel Mobile Ad Hoc Networks Based on Location Information”, *Int’l Journal Communication Systems*, Vol. 19, 2006, pp. 877-896. (SCI)

陳威碩，“IEEE 802.11 無線網狀網路之分散式時槽分割式多重頻道協定”， 碩士論文(指導教授：曾煜棋教授)，民國九十五年六月

第二年

W.-H. Tam and Y.-C. Tseng, “*Joint Multi-Channel Link Layer and Multi-Path Routing Design for Wireless Mesh Networks*”, IEEE INFOCOM, 2007.

L.-W. Chen, Y.-C. Tseng, D.-W. Wang, and J.-J. Wu, “*Exploiting Spectral Reuse in Resource Allocation, Scheduling, and Routing for IEEE 802.16 Mesh Networks*”, IEEE VTC, 2007-Fall.

第三年:

L.-W. Chen, W. Chu, Y.-C. Tseng, and J.-J. Wu, "Route Throughput Analysis with Spectral Reuse for Multi-Rate Mobile Ad Hoc Networks", Journal of Information Science and Engineering, to appear. (SCIE, EI)

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附錄一：

An Efficient MAC Protocol for Multi-Channel Mobile Ad Hoc Networks Based on Location Information

Y.-C. Tseng, S.-L. Wu, C.-M. Chao, and J.-P. Sheu, “An Efficient MAC Protocol for Multi-Channel Mobile Ad Hoc Networks Based on Location Information”, ***Int’l Journal Communication Systems***, Vol. 19, 2006, pp. 877-896. (SCI)

An Efficient MAC protocol for Multi-Channel Mobile Ad Hoc Networks Based on Location Information *

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Abstract

This paper considers the *channel assignment* problem in a multi-channel MANET environment. We propose a scheme called *GRID*, by which a mobile host can easily determine which channel to use based on its current location. In fact, following the GSM style, our GRID spends no communication cost to allocate channels to mobile hosts since channel assignment is purely determined by hosts' physical locations. We show that this can improve the *channel reuse* ratio. We then propose a multi-channel MAC protocol, which integrates GRID. Our protocol is characterized by the following features: (i) it follows an "on-demand" style to access the medium and thus a mobile host will occupy a channel only when necessary, (ii) the number of channels required is independent of the network topology, and (iii) no form of clock synchronization is required. On the other hand, most existing protocols assign channels to a host statically even if it has no intention to transmit [3, 10, 12], require a number of channels which is a function of the maximum connectivity [3, 8, 10, 12], or necessitate a clock synchronization among all hosts in the MANET [12, 27]. Through simulations, we demonstrate the advantages of our protocol.

Keywords: channel management, communication protocol, location-aware protocols, medium access control (MAC), mobile ad hoc network (MANET), mobile computing, wireless communication.

1 Introduction

A *mobile ad-hoc network (MANET)* is formed by a cluster of mobile hosts without the infrastructure of base stations. Two mobile hosts can communicate with each other indirectly in a multi-hop manner. Since no base station is required, one of its main advantages is that it can be rapidly deployed. The applications of MANETs appear in places where pre-deployment of network infrastructure is difficult or unavailable (e.g., fleets in oceans, armies in march, natural disasters, battle fields, festival field grounds, and historic sites).

A *MAC (medium access control)* protocol is responsible of resolving the communication contention and collision among hosts. Many MAC protocols have been proposed for wireless networks [4, 7, 13, 15, 20, 21], which assume a common channel shared by mobile hosts. We call such protocols *single-channel MAC protocols*. The widely accepted standard IEEE 802.11 [1] follows such model. One common problem with such protocols is that the network performance will degrade quickly as the number of mobile hosts increases, due to higher contention/collision.

One approach to relieving the contention/collision problem is to utilize multiple channels. The idea of using separate control and data channels was first proposed in [28]. We thus define a *multi-channel MAC protocol* as one which allows mobile hosts to dynamically access more than one channel in a MANET environment. Using multiple channels has several advantages. First, while the maximum throughput of a single-channel MAC protocol will be limited by the bandwidth of the channel, the throughput may be increased immediately if a host is allowed to utilize multiple channels. Second, as shown in [2, 25], using multiple channels will experience less *normalized propagation delay* per channel than its single-channel counterpart, where the normalized propagation delay is defined to be the ratio of the propagation time over the packet transmission time. Therefore, this reduces the probability of collisions. Third, QoS routing may be supported [22].

Here, we use “channel” upon a logical level. Physically, a channel can be a frequency band (under FDMA), or an orthogonal code (under CDMA). How to access multiple channels is thus technology-dependent. Disregard of the transmission technology, we categorize mobile hosts' channel access capability as follows:

- *single-transceiver:* A mobile host can only access one channel at a time. The transceiver can be simplex or duplex. Note that this is not necessarily equivalent to the single-channel model, because the transceiver is still capable of switching from one channel to another.

- *multiple-transceiver:* Each transceiver could be simplex or duplex. A mobile host can access multiple channels simultaneously.

In this paper, we propose a new multi-channel MAC protocol for a MANET in which each mobile host is equipped with a positioning device, such as GPS. A multi-channel MAC typically needs to address two issues: *channel assignment* and *medium access*. The former is to choose proper channels to send/receive for hosts, while the later is to resolve the contention/collision problem when using a particular channel. These two issues are sometimes addressed separately, but eventually one has to integrate them to provide a total solution. Our channel assignment, called *GRID*, is characterized by two features: (i) it exploits location information by partitioning the physical area into a number of squares called *grids*, and (ii) it does not need to transmit any message to assign channels to mobile hosts since channel assignment is purely determined by a host's physical location. Several channel assignment schemes have been proposed earlier [8, 9, 12, 25, 27], but none of them try to exploit the location information. Our medium access protocol is characterized by the following features: (i) it follows an “on-demand” style to access the medium and thus a mobile host will occupy a channel only when necessary, (ii) the number of channels required is independent of the network topology, and (iii) no form of clock synchronization is required. On the other hand, most existing protocols assign channels to a host statically even if it has no intention to transmit [3, 10, 12], require a number of channels which is a function of the maximum connectivity [3, 8, 10, 12], or necessitate a clock synchronization among all hosts in the MANET [12, 27]. A centralized scheme is proposed in a recent work [34]. Similar to hexagonal cellular systems, all channel assignment in a cell is controlled and allocated by the cell leader located at this cell. Since a cellular structure is assumed, location information is needed by each station. Contrary to [34], our GRID scheme is fully distributed and no traffic overhead is incurred for channel allocation. A detail review will be given in Section 2.1. For an overview, Table 1 gives a comparison on existing and our protocols.

Since a MANET should operate in a physical area, it is very natural to exploit location information in such an environment. Indeed, location information has been exploited in several issues in MANET (e.g., routing [11, 14, 16, 17, 18, 19, 24, 33], broadcasting [26], and power saving [30]), but not in channel assignment. GSM (Global System for Mobile Communications) is an instance which uses location information to exploit channel reuse, but MANET has quite different features — there is no base station, and thus channel assignment has to be done more dynamically in an in-band manner. Since the concept of "channel reuse" is highly related the area where a channel is used, exploiting location information, as we do in this work, on channel assignment could effectively solve this problem.

Outdoor positioning can be solve satisfactorily by GPS (global positioning systems) or DGPS (differential GPS). Both the price drop of GPS and the recent discontinuation of SA (Selective Availability) motivate us to conduct this research. However, for indoor positioning there is no satisfactory solution at this point.

The rest of this paper is organized as follows. Section 2 discusses some existing channel assignment schemes and our GRID scheme. Section 3 presents our MAC protocol by integrating the GRID assignment. Analysis and simulations are in Section 4. Conclusions will be drawn in Section 5.

2 Channel Assignment

As mentioned earlier, a multi-channel MAC needs to address two issues: channel assignment and medium access. In this section, we will consider the channel assignment problem. We will first review some existing protocols, which are all non-location-aware. Then we will present our location-aware channel assignment.

2.1 Non-Location-Aware Schemes

In this section, we review some channel assignment schemes that do not utilize the location information of mobile hosts. These schemes can be further divided to *static* and *dynamic*. The simplest static approach is to assign channels to mobile hosts when the system is first set up. For instance, channel i can be statically assigned to those hosts with ID's such that $i = ID \bmod n$ (supposing that we number channels as 0, 1, $\ldots$, $n-1$).

A scheme based on *Latin square* is proposed in [12], which assumes a TDMA-over-FDMA technology. Each channel is divided into fixed-length frames. Each host is statically assigned to a time slot in each frame belonging to a frequency band. Since TDMA is used, clock synchronization among all hosts is necessary. Furthermore, each host has to be equipped with a number of transceivers equal to the number of frequency bands, so this approach is quite costly. Also, this scheme needs to know in advance the maximum number of mobile hosts as well as the maximum degree of the topology formed by the MANET.

The schemes in [3, 5, 6, 10, 23] are for channel assignment in the traditional packet radio network. Partial or even complete network topology has to be collected to perform channel assignment. These approaches can basically be classified as static, although some can handle dynamic failure of base stations. Since these schemes are not designed for MANET, which is typically characterized by high host mobility, they do not fit our need.

A protocol based on dynamic channel assignment is in [8]. It is assumed that the channel assigned to a host must be different from those of its two-hop neighbors. To maintain this condition, a large amount of update messages will be sent whenever a host determines any change on channel assignment in its two-hop neighbors. This is inefficient in a highly mobile system. Further, this protocol is "degree-dependent" in the sense that it dictates a number of channels equal to an order of the square of the maximum degree of the MANET. So the protocol is inappropriate for a crowded environment.

A "degree-independent" protocol called *multichannel-CSMA* protocol is proposed in [25]. Suppose that there are n channels. The protocol imposes that each mobile host must have n receivers which concurrently listen on all n channels. Also, there is only one transmitter which will hop from channel to channel and, if necessary, will send on any detected idle channel. Again, this protocol has high hardware cost. Further, since no RTS/CTS is used, the hidden-terminal problem may easily occur. A hop-reservation MAC protocol based on very-slow frequency-hopping spread spectrum is proposed in [27]. Its channel assignment employs RTS/CTS dialogue to reserve a channel. The protocol is also degree-independent but requires clock synchronization among all mobile hosts, which is difficult when the network is dispersed in a large area.

Recently, Wu et al. [31] propose a new protocol, called *Dynamic Channel Assignment (DCA)*, which possesses the following characters: (i) it follows an "on-demand" style to access the medium and thus a mobile host will occupy a channel only when necessary, (ii) the number of channels required is independent of the network topology, and (iii) no form of clock synchronization is required. DCA uses one dedicated channel for control packets, and other channels for data. The purpose of the control channel is to assign data channels to mobile hosts or schedule the use of data channels among hosts' while data channels are used to transmit data packets and acknowledgements. Reference [32] combines DCA and power control to further improve channel reuse. However, because there is no location information, DCA cannot maintain an efficient channel reuse pattern.

In Table 1, we summarize and compare existing schemes with our yet-to-be-presented GRID scheme.

Table 1: Comparison of channel assignment schemes (n is the number of hosts, and m is the maximum network degree.

scheme	assignment	no. channels	info. collected	loc.-aware	assgn. cost	transceivers
[3, 5, 6, 10, 23]	static	deg.-dep.	global	no	$O(n^k), k \geq 2$	1
[12]	static	deg.-dep.	none	no	0	m
[8]	dynamic	deg.-dep.	2-hop	no	$O(n^3)$	2
[25]	dynamic	deg.-indep.	none	no	0	m
[27]	dynamic	deg.-indep.	none	no	$O(n)$	1
ours	dynamic	deg.-indep.	none	yes	0	2

2.2 Our Location-Aware Channel Assignment: GRID

Next, we introduce our location-aware channel assignment scheme. The MANET environment is the same, except that each mobile host must be installed with a positioning device, such as GPS receiver. So our protocol is more appropriate for outdoor environment. As will be seen later, our approach will assign a channel to a host once the host knows its current location. As a result, in addition to the positioning cost, there is no communication cost for our channel assignment (no message will be sent for this purpose).

We will refer to our scheme as *GRID*. The MANET is assumed to operate in a pre-defined geographic area. The area is partitioned into 2D logical grids as illustrated in Fig. 1. Each grid is a square of size $d \times d$. Grids are numbered (x, y) following the conventional xy-coordinate. To be location-aware, a mobile host must be able to determine its current grid coordinate. Thus, each mobile host must know how to map a physical location to the corresponding grid coordinate.

Our channel assignment works as follows. We assume that the system is given a fixed number, n, of channels. For each grid, we will assign a channel to it. When a mobile host is located at a grid, say (x, y), it will use the channel assigned to grid (x, y) for transmission. One can easily observe that if we assign the same channel to two neighboring grids, then there will be high chance that the transmission activities on these two neighboring grids will contend, or even interfere, with each other. Thus, we should assign the same channel to grids that are spatially separated by some distance, but will exploit the largest frequency reuse.

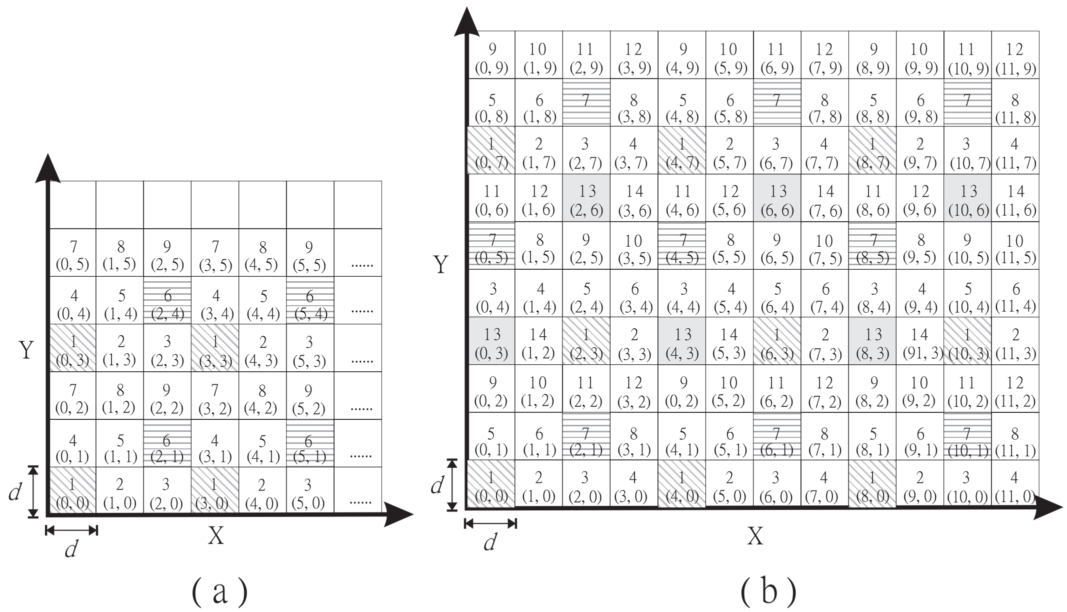


Figure 1: Assigning channels to grids in a band-by-band manner: (a) $n = 9$ and (b) $n = 14$. In each grid, the number on the top is the channel number, while those on the bottom are the grid coordinate. Here, we number channels from 1 to n.

The above formulation turns out to be similar to the channel arrangement in the GSM system. In the following, we propose a way to assign channels to grids. Let $m = \lceil \sqrt{n}\ \rceil$. We first partition the grids vertically into a number of *bands* such that each band contains m columns of grids. Then, for each band, we sequentially assign the n channels to each row of grids, in a row-by-row manner. In Fig. 1, we illustrate this assignment when $n = 9$ and $n = 14$. It can readily be seen that when n is a square of some integer, each channel will be regularly separated in the area.

2.2.1 Grid Size vs. Transmission Range

Let r be the transmission range of an antenna. Suppose the value of r is fixed. In this section, we discuss an important design issue: the relationship between r and the side length of grids, d. Below, we discuss several possibilities. For simplicity, let's assume that $m = \sqrt{n}$ is an integer.

- $d \gg r$: This means many hosts will stay in a grid and thus contend with each other on one channel. When $d = \infty$, this degenerates to the case of one single channel.

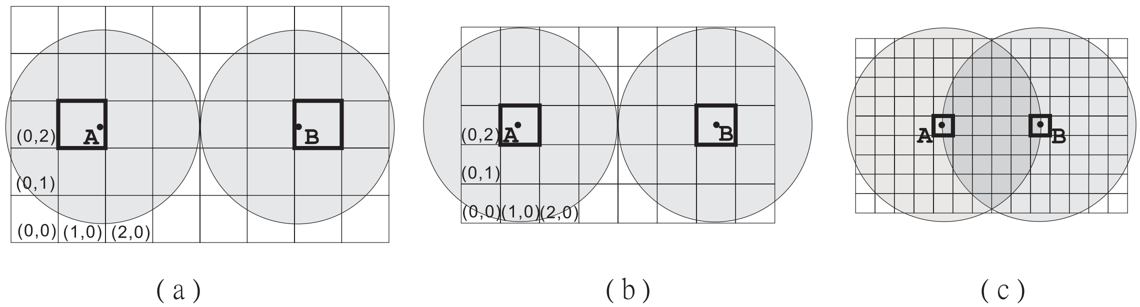


Figure 2: The effect of r/d ratio on channel co-interference when $n = 25$.

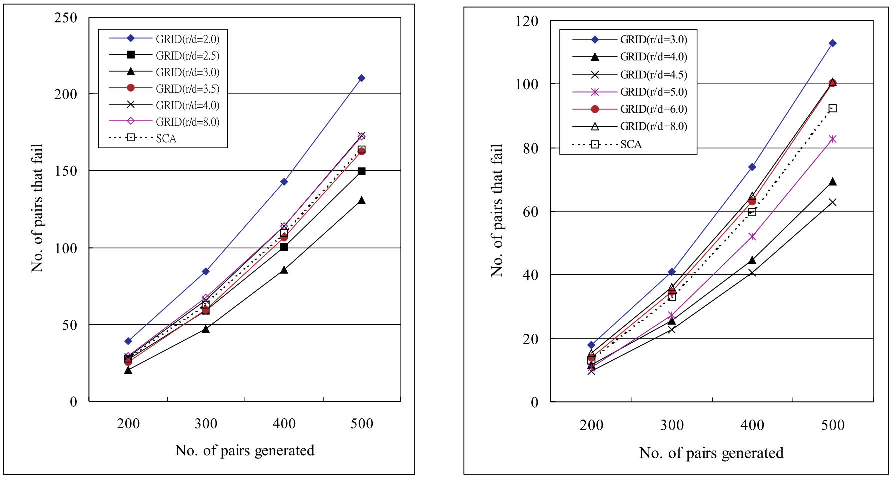
- $d > 2r/(m-1)$: This is the case that the transmission activities from two hosts choosing the same channel will never interfere with each other. As illustrated in Fig. 2(a), hosts A and B (both choosing the same channel) are located in the nearest possible locations, but their signals will not overlap in any location.

- $d = 2r/m$: This is the case that the transmission activities from two hosts which choose the same channel and which are each located in the center of a grid will not interfere with each other. This is illustrated in Fig. 2(b).

- $d = r/m$: This represents the minimal value of d such that two hosts (located at the grid centers) using the same channel will not hear each other. This is illustrated in Fig. 2(c). By simple calculus, we can find that each receiver of these two hosts will have a probability of 0.396 being interfered by the signals from the other sender. The value is the ratio of the intersection area that is covered by both hosts A and B to the area that is covered by either host A or host B.

- $d \approx 0$: This means that the grid size is infinitely small. This degenerates to the case that a mobile host will randomly choose a channel to transmit its packets, and thus little channel reuse can be exploited.

The above analysis has indicated some tradeoffs. This concept will be captured by the ratio r/d. If the ratio is too large, then the chance of co-channel interference will be high. On the other hand, if the ratio is too small, although co-channel interference can be reduced, the channel reuse will be reduced too since a channel will be unavailable in



(a) (b)

Figure 3: Tests of blocked sender-receiver pairs at different r/d ratios: (a) $n = 36$ and (b) $n = 81$.

many locations. Thus, we need to carefully adjust the r/d ratio for the best network performance. This will be further investigated through simulations in Section 4.2.

2.2.2 Some Experiments on the r/d Ratio

At this point, it deserves to predict, under ideal situations, how much benefit our location-aware channel assignment can offer over a non-location-aware one. We developed a simple simulation without concerning the details of medium access, such as collision, timing, etc. (this will be explored in Section 4). We simulated an area of size 1000×1000. On this area, we randomly generated a sender A and then randomly generated a receiver B in the circle of radius $r = 100$ centered at A. A transmitted using a channel selected by two methods: (i) a static one based on host ID (referred to as SCA, static channel assignment), and (ii) our GRID approach. We then repeated this process to generate more sender-receiver pairs. However, for each pair generated, we tested whether this transmission will interfere any earlier ongoing pairs. If so, the current pair will be deleted; otherwise, it will be granted.

Through this ideal experiment, we intend to observe how many more sender-receiver pairs can be generated in the physical area by GRID than SCA. This will verify whether

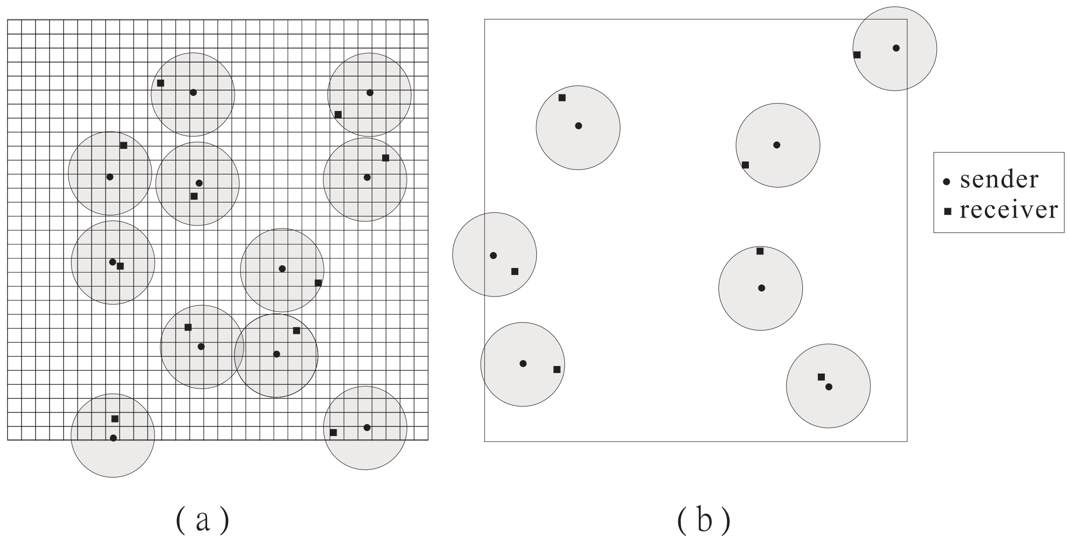


Figure 4: A snapshot of our experiment in Fig. 3 when $n = 36$ and $r/d = 3.0$: (a) GRID and (b) SCA. The snapshots are taken on a 1000×1000 area, and each circle means a sender-receiver pair.

GRID has a better channel reuse. Another important issue we would like to explore here is: what is best ratio r/d to maximize channel reuse?

Fig. 3 shows our first experimental results. The x-axis is the number of sender-receiver pairs generated. The y-axis shows the number of pairs that fail and thus are deleted. For our GRID, we tested different r/d ratios. Fig. 3(a) uses a total number of $n = 36$ channels, and Fig. 3 (b) uses $n = 81$. Indeed, some r/d ratios are better than SCA, while some are worse. In Fig. 3(a), we see that the r/d ratios $2.5, 3.0$, and 3.5 will outperform SCA, while in Fig. 3(b), the r/d ratios $4.0, 4.5$, and 5.0 will outperform SCA.

We conclude from the above experiments that when $r/d \approx \sqrt{n}/2$, our GRID will perform well. The reason is as follows. Let's consider any channel. At this ratio, it is more likely that we can place most circles (which represent transmission activities of this channel) in a physical area, while incurring the least overlapping among circles (which represents co-channel interference). This is how our GRID can offer better channel reuse. Fig. 4 shows a snapshot in our experiment when $n = 36$ and $r/d = 3.0$ on the use of channel 1. Clearly, the placement of circles by GRID is denser and more regular than that of SCA.

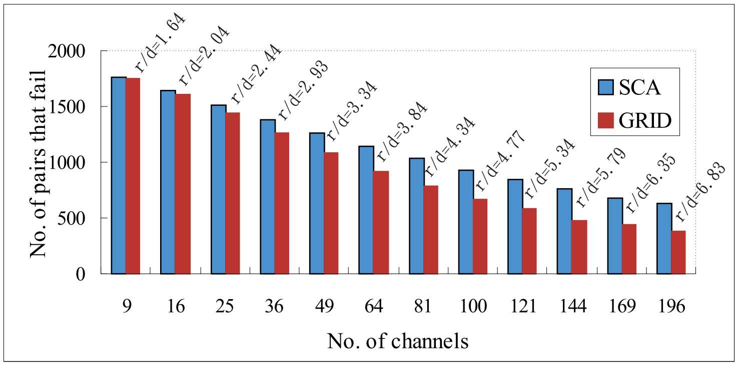


Figure 5: Tests of blocked sender-receiver pairs at various n's.

In Fig. 5, we further vary the value of n to observe the trend. In this figure, we have picked the best r/d ratio for each n. The number of sender-receiver pairs generate is 2000. As can be seen, the best ratios are all very close to $\sqrt{n}/2$, as we have predicted. Also, with more channels, there are less pairs being blocked by both GRID and SCA. But the gain of GRID over SCA will enlarge as a larger n is used.

3 The MAC Protocol

This section presents the medium access part of our protocol by integrating the channel assignment part in the previous section. The channel model is as follows. The overall bandwidth is divided into one control channel and n data channels $D_1, D_2, \ldots, D_n$. Each channel, including control and data ones, is of the same bandwidth. The purpose of data channels is to transmit data packets and acknowledgements. Control channel serves in many important management purposes: (i) to synchronize the use of data channels among hosts, (ii) to broadcast beacons periodically, and (iii) to search for routes. Note that beacons can help mobile hosts to discover which hosts are currently neighbors. Hosts can always communicate with others through the control channel, but they can only communicate with each other through data a channel if they switch to the same one. Route discovery and routing functions are beyond the scope of this paper and will not be elaborated, but can be supported by the control channel.

In our protocol, the channel assignment should be done in advanced. We think that the organization, e.g. city governments or corporations, should take the responsibility of channel allocation if it wants to use GRID in its district such that the best performance can be got. It is something like that FCC regulates the use of radio spectrum to satisfy the communications needs without interference.

Each mobile host is equipped with two half-duplex transceivers:

- *control transceiver:* This transceiver will operate on the control channel to exchange control packets with other mobile hosts and to obtain rights to access data channels.

- *data transceiver:* This transceiver will dynamically operate on one of the data channels, according to our channel assignment, to transmit data packets and acknowledgements.

Each mobile host X maintains the following data structure.

- $CUL[\ ]$: This is called the *channel usage list*. Each list entry $CUL[i]$ keeps records of how and when a host neighboring to X uses a channel. $CUL[i]$ has three fields:
  - $CUL[i].host$: a neighbor host of X.
  - $CUL[i].ch$: a data channel used by $CUL[i].host$.
  - $CUL[i].rel\_time$: when channel $CUL[i].ch$ will be released by $CUL[i].host$.

Note that this CUL is distributedly maintained by each mobile host and thus may not contain the precise information.

The main idea of our protocol is as follows. For a mobile host A to communicate with host B, A will send a RTS (request-to-send) to B. This RTS will also carry the channel number that A intends to use in its subsequent transmission. Then B will match this request with its in $CUL[\ ]$ and, if granted, reply a CTS (clear-to-send) to A. All these will happen on the control channel. Similar to the IEEE 802.11 [1], the purpose of the RTS/CTS dialogue is to warn the neighborhood of A and B not to interfere their subsequent transmission, except that a host is still allowed to use the channels different from that indicated in the RTS and CTS packets. Finally, transmission of a data packet will occur on the data channel.

Table 2: Meanings of variables and constants used in our protocol.

T_{SIFS}	length of short inter-frame spacing
T_{DIFS}	length of distributed inter-frame spacing
T_{RTS}	time to transmit a RTS
T_{CTS}	time to transmit a CTS
T_{curr}	the current clock of a mobile host
T_{ACK}	time to transmit an ACK
NAV_{RTS}	network allocation vector on receiving a RTS
NAV_{CTS}	network allocation vector on receiving a CTS
L_d	length of a data packet
L_c	length of a control packet (RTS/CTS)
B_d	bandwidth of a data channel
B_c	bandwidth of a control channel
τ	maximal propagation delay

The complete protocol is shown below. Table 2 lists the variables/constants used in our presentaiton.

1. On a mobile host A having a data packet to send to host B, it first checks whether the following two conditions are true:

   a) B is not equal to any $CUL[i].host$ such that

   $$CUL[i].rel\_time > T_{curr} + (T_{DIFS} + T_{RTS} + T_{SIFS} + T_{CTS}).$$

   If so, this means B will still be busy (in using data channel $CUL[i].ch$) after a successful exchange of RTS and CTS packets.

   b) Suppose A determines that its current data channel is D_A. Then for each $i = 1..n$,

   $$(D_A = CUL[i].ch) \Longrightarrow (CUL[i].rel\_time \leq T_{curr} + (T_{DIFS} + T_{RTS} + T_{SIFS} + T_{CTS})).$$

   If so, this means A's data channel is either not currently being used by any of its neighbors, or currently being occupied by some neighbor(s) but will be released after a successful exchange of RTS and CTS packets. (Fig. 6 shows how the above timing is calculated.)

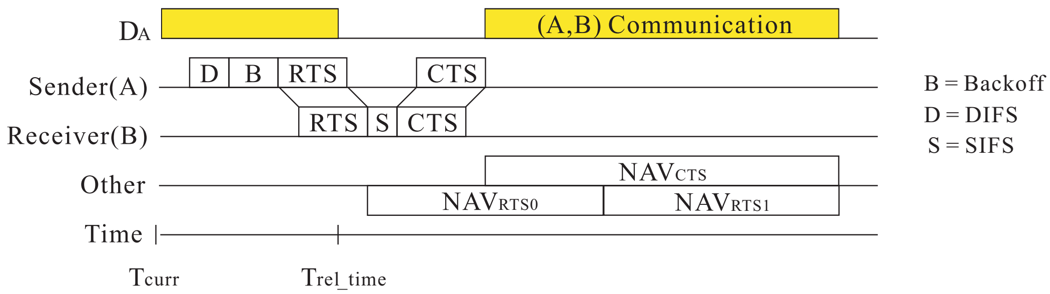


Figure 6: Timing to determine whether a channel will be free after a successful exchange of RTS and CTS packets.

If the above two conditions are true, proceed to step 2; otherwise, A must wait at step 1 until these conditions become true.

2. Then A can send a $RTS(D_A, L_d)$ to B, where L_d is the length of the yet-to-be-sent data packet. Also, following the IEEE 802.11 style, A can send this RTS only if there is no carrier on the control channel in a T_{DIFS} plus a random backoff time period. Otherwise, it has to go back to step 1.

3. On a host B receiving the $RTS(D_A, L_d)$ from A, it has to check whether the following condition is true for each $i = 1..n$:

$$(D_A = CUL[i].ch) \Longrightarrow (CUL[i].rel\_time \leq T_{curr} + (T_{SIFS} + T_{CTS})).$$

If so, D_A is either not currently being used by any of its neighbors, or currently being used by some neighbor(s) but will be released after a successful transmission of a CTS packet. Then B replies a $CTS(D_A, NAV_{CTS})$ to A, where

$$NAV_{CTS} = L_d/B_d + T_{ACK} + 2\tau.$$

Then B tunes its data transceiver to D_A. Otherwise, B replies a $CTS(T_{est})$ to A, where T_{est} is the estimated time that B's data channel D_A will change minus the time for an exchange of a CTS packet:

$$T_{est} = \max\{\forall i \ni CUL[i].ch = D_A, CUL[i].rel\_time\} - T_{curr} - T_{SIFS} - T_{CTS}.$$

4. On an irrelevant host $C \neq B$ receiving A's $RTS(D_A, L_d)$, it has to inhibit itself from using the control channel for a period

$$NAV_{RTS0} = T_{SIFS} + T_{CTS} + \tau.$$

This is to avoid C from interrupting the RTS $\rightarrow$ CTS dialogue between A and B. Then, C senses channel D_A for a period of τ to determine whether this communication is success or not. If so, it appends an entry $CUL[k]$ to its CUL such that:

$$\begin{aligned} CUL[k].host &= A \\ CUL[k].ch &= D_A \\ CUL[k].rel\_time &= T_{curr} + NAV_{RTS1} \end{aligned}$$

where

$$NAV_{RTS1} = T_{curr} + L_d/B_d + T_{ACK} + \tau.$$

5. Host A, after sending its RTS, will wait for B's CTS with a timeout period of $T_{SIFS}+T_{CTS}+2\tau$. If no CTS is received, A will retry until the maximum number of retries is reached.

6. On host A receiving B's $CTS(D_A, NAV_{CTS})$, it performs the following steps:

   a) Append an entry $CUL[k]$ to its CUL such that

   $$\begin{aligned} CUL[k].host &= B \\ CUL[k].ch &= D_A \\ CUL[k].rel\_time &= T_{curr} + NAV_{CTS} \end{aligned}$$

   b) Send its DATA packet to B on the data channel D_A.

   On the other hand, if A receives B's $CTS(T_{est})$, it has to wait for a time period T_{est} and go back to step 1.

7. On an irrelevant host $C \neq A$ receiving B's $CTS(D_A, NAV_{CTS})$, C updates its CUL. This is the same as step 6a) except that

   $$CUL[k].rel\_time = T_{curr} + NAV_{CTS} + \tau.$$

   On the other hand, if C receives B's $CTS(T_{est})$, it ignores this packet.

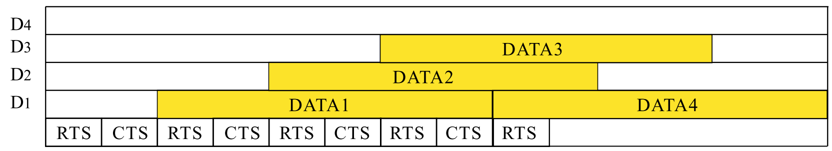


Figure 7: An example that the control channel is fully loaded and the data channel D_4 is not utilized.

8. On B completely receiving A's data packet, B replies an ACK on D_A.

To summarize, our protocol relies on the control channel to negotiate the transmissions among hosts using the same data channel. Also, note that although our protocol will send timing information in packets, these are only relative time intervals. No absolute time is sent. So there is no need of clock synchronization in our protocol.

4 Analysis and Simulation Results

4.1 Arrangement of Control and Data Channels

One concern in our protocol is: Can the control channel efficiently distribute the communication jobs to data channels? For example, in Fig. 7, we show an example with 5 channels, one for control and four for data. For simplicity, let's assume that the lengths of all control packets (RTS, and CTS) are L_c, and lengths of all data packets $L_d = 6L_c$. Then Fig. 7 shows a scenario that the control channel can only utilize three data channels D_1, D_2, and D_3. Channel D_4 may never be used because the control channel can serve at most three data channels. Although L_d is typically larger than L_c by an order of at least tens or hundreds, it still deserves to analyze this issue to understand the limitation.

The above example shows that how to arrange the control and data channels is a critical issue. In the following, we consider two bandwidth models.

- *fixed-channel-bandwidth:* Each channel (data and control) has a fixed bandwidth. Thus, with more channels, the network can potentially use more bandwidth.

- *fixed-total-bandwidth:* The total bandwidth offered to the network is fixed. Thus, with more channels, each channel will have less bandwidth.

We comment that the first model may reflect the situation in CDMA, where each code has the same bandwidth, and we may utilize multiple codes to increase the actual bandwidth of the network. On the other hand, the second model may reflect the situation in FDMA, where the total bandwidth is fixed, and our job is to determine an appropriate number of channels to best utilize the given bandwidth.

We will show how to arrange the control and data channels under these models so as to well utilize a given bandwidth. Let's consider the fixed-channel-bandwidth model first. Apparently, since the control channel can arrange a data packet by sending 2 control packets of total length $2L_c$, the maximum number of data channels should be limited by

$$n \leq \frac{L_d}{2 \times L_c}. \tag{1}$$

Also, consider the utilization U of the total given bandwidth. Since the control channel is actually not used for transmitting data packets, we have

$$U \leq \frac{n}{n+1}. \tag{2}$$

From Eq. (1) and Eq. (2), we derive that

$$\frac{U}{1-U} \leq n \leq \frac{L_d}{2 \times L_c} \Longrightarrow U \leq \frac{L_d}{2 \times L_c + L_d}. \tag{3}$$

The above inequality implies that the maximum utilization is a function of the lengths of control and data packets. Thus, decreasing the length of control packets or increasing the length of data packets will improve the utilization. Since the maximum utilization is only dependent of L_d and L_c, it will be unwise to unlimitedly increase the number of data channels.

Next, we consider the fixed-total-bandwidth model. Suppose that we are given a fixed bandwidth. The problem is: how to assign the bandwidth to the control and data channels to achieve the best utilization. Also, how many data channels (n) will be most efficient? Let the bandwidth of the control channel be B_c, and that of each data channel B_d. Again, the number of data channels should be limited by the assignment capability of the control channel:

$$n \leq \frac{L_d/B_d}{2 \times L_c/B_c}. \tag{4}$$

Similarly, the utilization U must satisfy

$$U \leq \frac{n \times B_d}{n \times B_d + B_c}. \tag{5}$$

Combining Eq. (4) and Eq. (5) gives

$$\frac{UB_c}{B_d - UB_d} \leq n \leq \frac{L_d B_c}{2 \times L_c B_d} \Longrightarrow U \leq \frac{L_d}{2 \times L_c + L_d}. \tag{6}$$

Interestingly, this gives the same conclusion as that in the fixed-channel-bandwidth model. The bandwidths B_c and B_d have disappeared in the above inequality, and the maximum utilization is still only a function of the lengths of control and data packets. Thus, decreasing the length of control packets or increasing the length of data packets may improve the utilization. To understand how to arrange the bandwidth, we replace the maximum utilization into Eq. (5), which gives

$$\frac{L_d}{2 \times L_c + L_d} = \frac{n \times B_d}{n \times B_d + B_c} \Longrightarrow \frac{B_c}{nB_d} = \frac{2L_c}{L_d}. \tag{7}$$

Thus, to achieve the best utilization, the ratio of the control bandwidth to the data bandwidth should be $2L_c/L_d$. Furthermore, since the maximum utilization is independent of the value of n, theoretically once the above ratio ($2L_c/L_d$) is used, it does not matter how many data channels that we divide the data bandwidth into. (Thus, one can even adjust the value of n according to the number of mobile hosts or host density.)

Finally, we comment on several minor things in the above analysis. First, if the control packets are of different lengths, the $2L_c$ can simply be replaced by the total length of RTS, and CTS. Second, the L_d has included the length of ACK packets. So the real data packet length should be L_d minus the length of an ACK packet. Last, we did not consider protocol factors (such as propagation delay, SIFS, DIFS, collisions of control and data packets, backoffs, etc.) in the analysis and hence the bandwidth considered above is not “effective” bandwidth. In reality, these factors will certainly affect the performance. In the next section, we will explore this through simulations.

4.2 Experimental Results

We have implemented a simulator to evaluate the performance of our GRID protocol. We mainly used the SCA protocol as a reference for comparison. SCA only differs from our GRID in its channel assignment strategy. Specifically, in SCA, the overall bandwidth is still divided into one control channel and n data channels. But each host is statically assigned to only one data channel. To use its data channel, a host must go through a RTS/CTS exchange with its intending receiver before using the data channel. Since both SCA and GRID use the same channel model and medium access approach, we believe that the experiment can give a clear indication how much more channel reuse that GRID can offer. Also, whenever appropriate, we will include the performance of IEEE 802.11, which is based on a single-channel model, to demonstrate the benefit of using multiple channels.

The parameters used in our experiments are: physical area = 1000×1000, transmission range $r = 200$, hosts = 400, $DIFS = 50\mu sec$, $SIFS = 10\mu sec$, backoff slot time = $20\mu sec$, control packet length $L_c = 100$ bits. A data packet length L_d is a multiple of L_c. Packets arrived at each mobile host in an Poisson distribution with arrival rate λ packet/sec. For each packet arrived at a host, we randomly chose a host at the former's neighborhood as its receiver. Both of the earlier bandwidth models are used. If the fixed-channel-bandwidth model is assumed, each channel's bandwidth is 1 Mbps/sec. If the fixed-total-bandwidth model is assumed, the total bandwidth is 1 Mbps/sec. In the following, we make observations from four aspects.

A) Effect of the r/d Ratios: In this experiment, we change the r/d ratio to observe the effect. We use $n = 16$ data channels and $L_d/L_c = 200$. Fig. 8 shows the network throughput under different loads under the fixed-channel-bandwidth model. We can see that both SCA and GRID have similar throughput curves. When $r/d = 0.5$, 1.0, and 1.5, our GRID protocol is worse than the SCA protocol. When $r/d \geq 2.0$, our GRID will outperform SCA. At $r/d = 3.5$, GRID will deliver the highest throughput, which is about 25% more than the highest throughput of SCA. After $r/d > 3.5$, GRID will saturate and degrade slightly, but still outperform SCA. It is worth to mention that according to our earlier ideal analysis in Section 2, the best performance of GRID

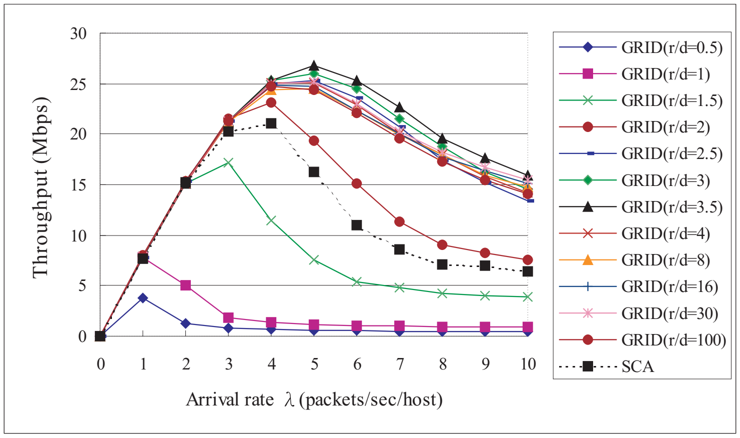


Figure 8: Arrival rate vs. throughput under the fixed-channel-bandwidth model at different r/d ratios with $n = 16$.

will appear when $r/d = \sqrt{n}/2 = 2$. This ratio is somewhat smaller than the ratio 3.5 that we obtain here. We believe that this is because in this experiment we have taken timing factors (such as different packet arrival time and different backoff intervals) into consideration, while in Section 2 we have disregarded this factor. Thus, different sender-receiver pairs may be time-differentiated, and thus more pairs may coexist. In fact, this is a favorable result to GRID because a higher r/d ratio means more signal overlapping, and thus higher channel reuse.

Fig. 9 shows the similar experiment under the fixed-total-bandwidth model. Again, the best r/d ratio appears at around 2.5 to 4. The trend is similar to that of the fixed-channel bandwidth model. Also, as a reference point, this figure contains the performance of IEEE 802.11.

B) Effect of the Number of Channels: In this experiment, we still use $L_d/L_c = 200$, but vary the number of channels n, to observe its effect. Fig. 10 shows the result under the fixed-channel-bandwidth model. Note that in this figure we have picked the best r/d ratio (through experiments) for each given n for our GRID protocol. We see that both SCA's and GRID's throughputs will increase as more data channels are used. This is quite reasonable because under the fixed-channel-bandwidth model, a larger n means more total bandwidth being provided. As n enlarges, the gap between GRID and SCA will increase slightly.

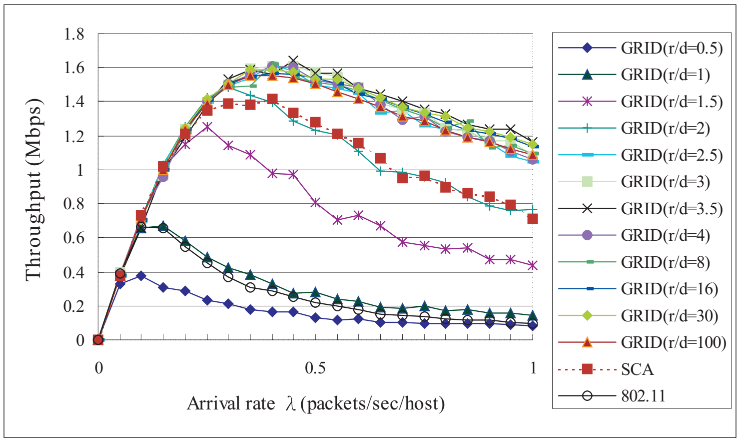


Figure 9: Arrival rate vs. throughput under the fixed-total-bandwidth model at different r/d ratios with $n = 16$.

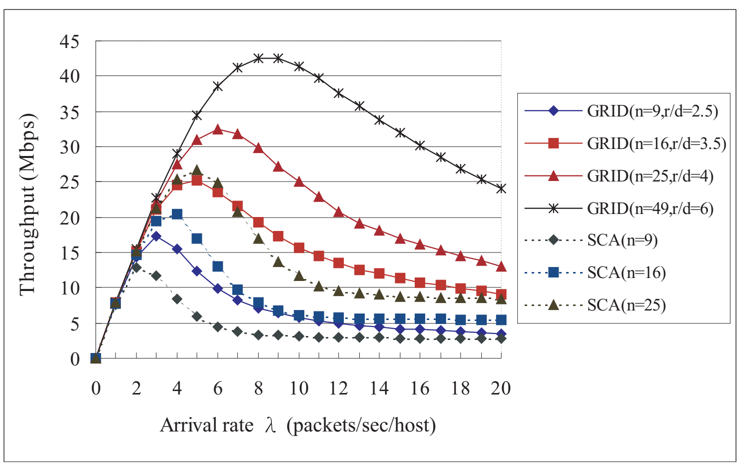


Figure 10: Arrival rate vs. throughput under the fixed-channel-bandwidth model with different numbers of data channels.

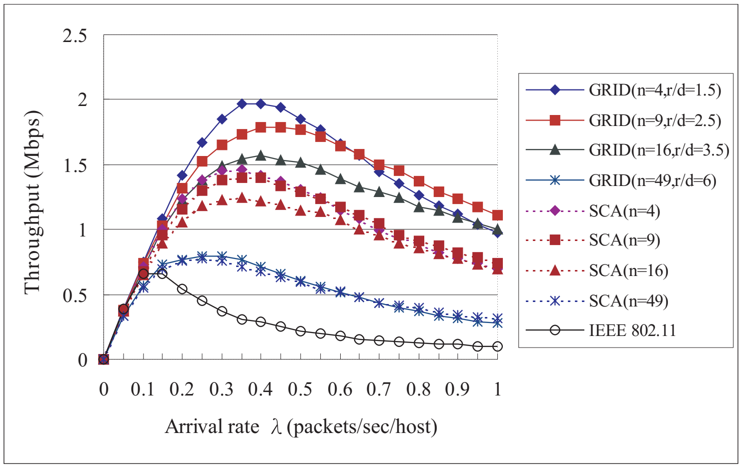
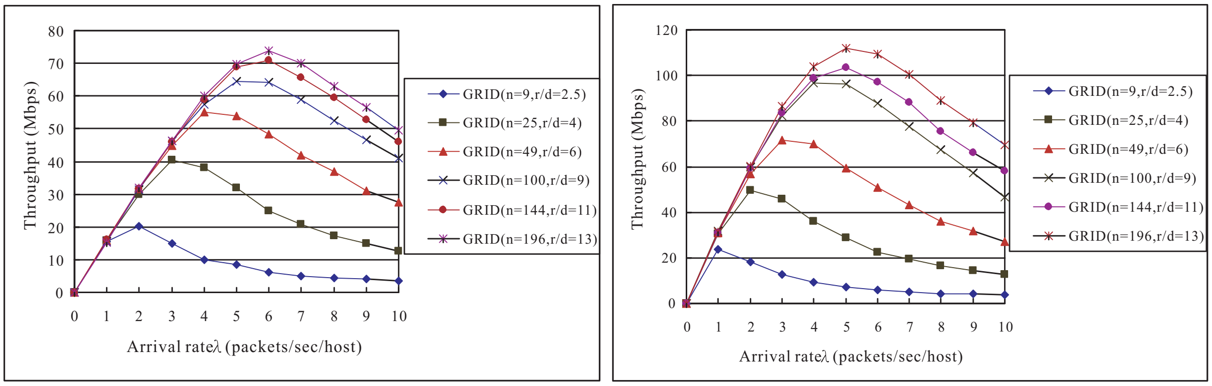


Figure 11: Arrival rate vs. throughput under the fixed-total-bandwidth model with different numbers of data channels.

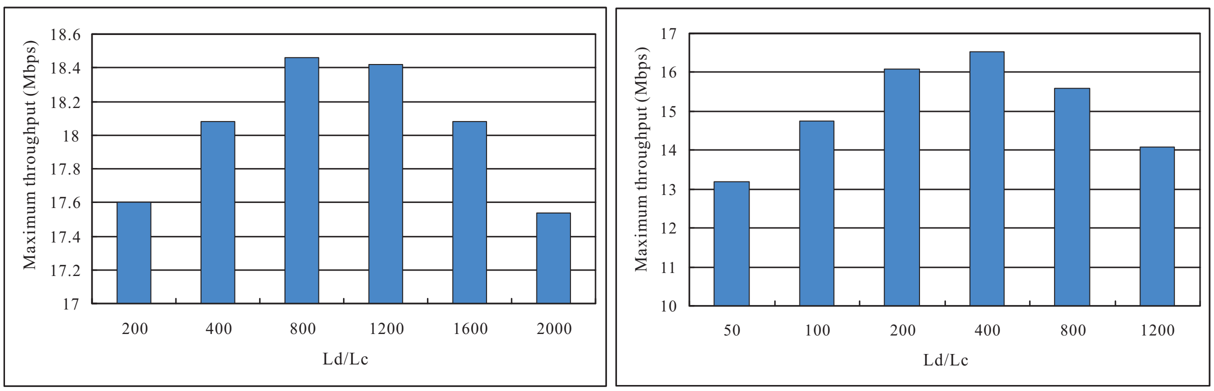
Fig. 11 shows the same simulation under fixed-total-bandwidth model. The trend is similar. One important observation is that the best performance for both SCA and GRID will appear at around $n = 4$ data channels. With more channels, the throughput will degrade significantly. Also, as comparing GRID and SCA, we see that when n is too large (e.g., $n = 49$), The gap between GRID and SCA will decrease significantly. This may due to two reasons: either the control channel is overloaded, or the control channel has not been fully loaded but there are too few mobile hosts to fully utilize these data channels.

C) Effect of the L_d/L_c ratios: As discussed earlier, the performance of GRID will be limited by the use of the control channel. One way to increase performance is to increase the data packet length in order to reduce the load on the control channel. To understand this issue, observe Fig. 12(a), which assumes $L_d/L_c = 50$ and the number of hosts = 1600 under the fixed-channel-bandwidth model. Comparing the curves in this figure, we see that there is a large performance improvement between using $n = 9$ channels and $n = 25$ channels. However, the improvement reduces significantly from using $n = 25$ to using $n = 49$ channels. When using $n = 100$ channels, the gain relative to using $n = 49$ is very limited (note that under the fixed-channel-bandwidth model, this means much bandwidth being wasted). To resolve this problem, in Fig. 12(b), we increase L_d/L_c to 200. Now the improvements all enlarge. This has justified our



(a) (b)

Figure 12: Arrival rate vs. throughput under the fixed-channel-bandwidth model at different numbers of data channels: (a) $L_d/L_c = 50$ and (b) $L_d/L_c = 200$.



(a) (b)

Figure 13: Ratio L_d/L_c vs. maximum throughput under the fixed-channel-bandwidth model with $n = 9$: (a) bit error rate $= 10^{-6}$ and (b) bit error rate $= 5 \times 10^{-6}$.

argument. As a result, given an n, one has to wisely adjust the ratio L_d/L_c so as to get the best throughput.

D) Effect of Transmission Error Rates: In the previous experiment, we have made a strong assumption: the transmission is error-free. To take this into consideration, we further assume a bit error rate during transmission. Under the fixed-channel-bandwidth model with $n = 9$ channels, Fig. 13(a) and (b) show our simulation results under the transmission bit error rates of 10^{-6} and 5×10^{-6}, respectively. Under an error rate of 10^{-6}, $L_d/L_c = 800$ has the best maximum throughput. With a larger error rate of 5×10^{-6}, the best maximum throughput will appear at the smaller ratio $L_d/L_c = 400$.

5 Conclusions

We have developed a new MAC protocol for a multi-channel MANET. Our channel assignment is characterized by location awareness capability and it incurs no communication cost to conduct the assignment. This is a significant breakthrough compared to existing protocols which require clock synchronization and/or which dictate a number of channels which is a function of the network degree. Our simulation results have also indicated that it is worthwhile to consider using multiple channels under both the fixed-channel-bandwidth model and the fixed-total-bandwidth model.

In this paper, we focus on the scenario where hosts are randomly deployed. In such an environment, GRID is a simple yet efficient solution. For larger areas where users have geographical locality, the GRID-B proposed in [29] tries to explore channel borrowing to make an efficient use of channels. However, due to its channel relocation behavior, GRID-B involves higher complexity. The purpose of this paper is to develop a light-weight MAC protocol that is suitable for an ad hoc environment.

We believe that there are many open research problems from this work. In our simulations, we have used a number of data channels (n) which is a square of some integer. Other values of n deserve investigation. In practice, the best r/d ratio may change due to many factors, such as system load, which also deserves studies. While GPS is widely available, indoor positioning is still an open issue. Since our work relies on physical locations to assign channels, for indoor environment pre-assignment of channels to each location may be necessary.

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附錄二：

“IEEE 802.11 無線網狀網路之分散式時槽分割式多重頻道協定”

碩士論文

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民國九十五年六月

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碩 士 論 文

IEEE 802.11 無線網狀網路之分散式時槽分割式多重頻道協定

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IEEE 802.11 無線網狀網路之分散式
時槽分割式多重頻道協定

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摘要

IEEE 802.11 協定允許使用不同頻道來提升網路效能，儘管近年來有不少這方面的研究，但分散式的分配頻道仍然是個複雜而具有挑戰性的問題。無線網路網路(Wireless Mesh Network)近年來倍受國際矚目，它提供有線區網的另一種更方便、更便宜的選擇，而在無線網狀網路上使用多重頻道(Multiple Channel)是非常吸引人的一個議題，因為無線網狀網路必需提供很大的頻寬給使用者。我們提出一個適合無線網路網路的鏈結層的多重頻道管理協定來增進整個網路的效能，這個協定使用已普及的 IEEE 802.11 相容的網路卡介面，每個存取點只需一個介面便能順利運作，並且很容易能擴充至裝備多張介面卡的存取點。我們設計此協定使用接收端為主(Receiver-Based)的頻道分配演算法，並使用時槽來控制什麼時候該送、什麼時候該收。我們在真實的網路環境下實作這個協定來驗證我們方法，發現確實的提升了網路的吞吐量(Throughput)。

關鍵字：無線網狀網路、多重頻道、媒體存取控制、頻道分配、排程分配

A Distributed Time-Slotted Multi-Channel Protocol for IEEE 802.11 Wireless Networks

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ABSTRACT

Multiple channels which are available for use in IEEE 802.11 can increase network capacity. Despite being the subject of many years of research, distributed channel assignment remains a challenging problem. The idea of exploiting multiple channels is particularly appealing in wireless mesh networks because of their high capacity requirements to support backbone traffic. We propose a link-layer protocol for wireless mesh network that utilizes multiple channels dynamically to improve performance. The protocol can be implemented in software over an IEEE 802.11-compliant wireless card. We only need one interface and easily extend our protocol to multiple interfaces. We are based on receiver-based channel assignment algorithm to design our protocol and use time slot to control when to send and when to receive. Our protocol is implemented in real environment and indeed improves the network performance.

Keywords: Wireless Mesh Network, Multi-Channel, Medium Access Control, Channel Assignment, Scheduling

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中華民國九十五年六月

論文目次

第壹章、緒論

近年來國際間「無線網狀網路」(Wireless Mesh Networks)的相關研究備受矚目，更引起了學界與業界的廣泛討論。無線網狀網路系統是基於IEEE 標準技能實現的新型網路系統，可提升無線域區的覆蓋能力，並可為有線網路提供另一種便宜、方便的選擇。

第一節、研究背景

我們的研究背景及立基於目前漸趨成熟的無線網狀網路，由於無線網狀網路的高頻寬需求，我們引進使用複數頻道的優點來改善其效能。

在多躍式隨意無線網路(multi-hop ad-hoc wireless network)裡，傳輸速率會因為鄰近存取點的同時傳輸而干擾降低速率，利用複數頻道(multiple channel)可以避免這種干擾進而有效提升無線網路的效能。在IEEE 802.11 網路中有存在著數個不與其他頻道重疊(non-overlap)的頻道，例如在 IEEE 802.11b 中有三個不相互重疊的頻道可使用，而在 IEEE 802.11a 中更多達十二個不相互重疊的頻道可使用，雖然在 IEEE 802.11 Infrastructure mode 中，相鄰的基地台使用不同的頻道來降低干擾的方法已經被提出（例如交大研究團隊在去年與資策會所合作的計劃「新世代無線區域網路架構與技術」[14]中提出了動態調整無線網路基地台的頻道的方法)，然而在 IEEE 802.11 Ad-hoc mode 中如何有效的利用複數頻道增進網路效能卻還是一個值得研究的領域。

我們可以從下圖 1-1 看出使用多重頻道的好處，在使用同一個頻道時，兩個連線(link)會互相競爭干擾這個頻道，雙方理想值只能得到這個頻道頻寬的一半，或許還會更低，但如果這兩個連線使用不同的頻道，便能獲得這個頻道的整個頻寬，進而提升整個網路的效能。

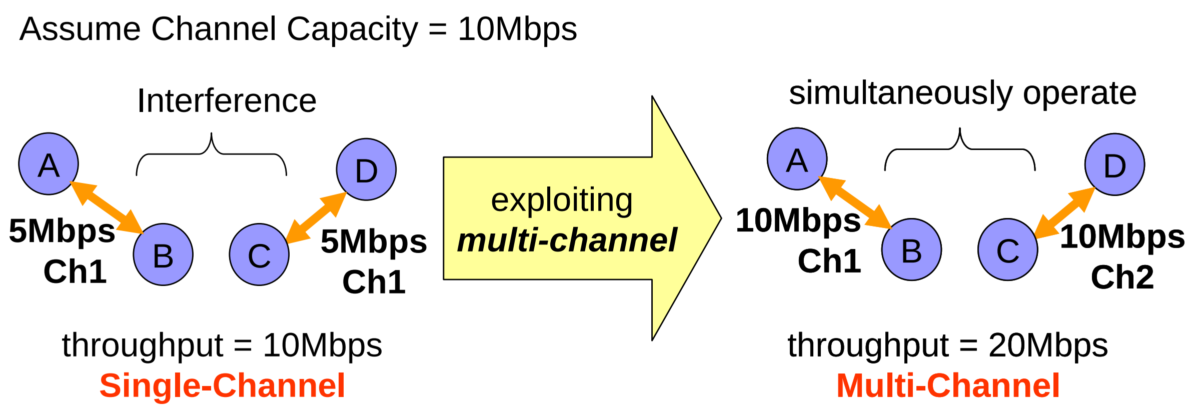

圖 1-1、頻道干擾範例

無線網狀網路(Wireless Mesh Network)就像一個由一群路由器(routers)組成的網路，如圖 1-2，差別在於網狀網路之間是用無線網路的方式來連線，最近這幾年無線網狀網路也獲得相當大的重視，無線網狀網路可提供無線寬頻服務的網路，它融合了無線區域網路(Wireless LAN)和隨建即連(Ad Hoc)網路的優勢，無線網狀網路採用無線傳輸的方式連結了許多的存取點(Access Point)，有了無線網狀網路的架設可讓網路服務提供者透過少量的有線網路達成大範圍的服務區域，且因為少了使用有線網路時纜線的建置過程，時間與成本都可大量的減少。

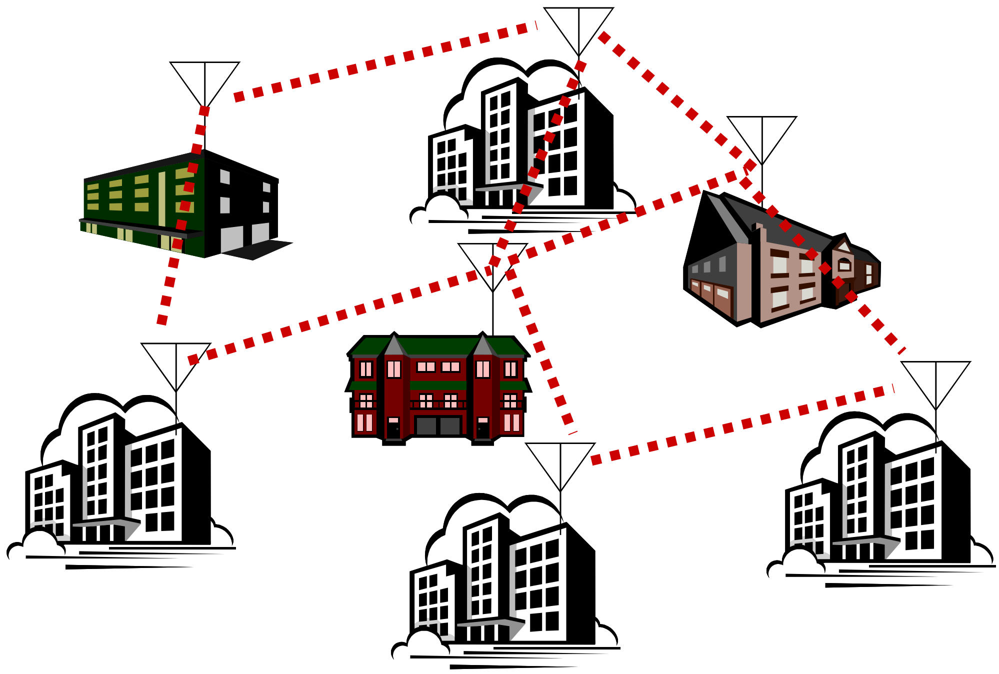

圖 1-2、無線網狀網路

第二節、研究動機與目的

在 IEEE 802.11 網路中有存在著數個不與其他頻道重疊(non-overlap)的頻道，雖然在 IEEE 802.11 Infrastructure mode 中，相鄰的基地台使用不同的頻道來降低干擾的方法已經被提出，然而在 IEEE 802.11 Ad-hoc mode 中如何有效的利用複數頻道增進網路效能卻仍是一個值得研究的領域，例如 IEEE 802.11 的媒介存取協定(Medium Access Control Protocol)的設計只針對單一頻道的使用，因此造成效能無法進一步地提升。當複數個頻道被利用時，可預期的是網路的吞吐量(Throughput)會增加，干擾可降低，空間中頻道的再使用率(Spatial Reuse)能提高。

我們最終的目的是要能夠利用複數頻來提升無線網狀網路的吞吐量。由於在無線網狀網路中，有一些特性是和傳統無線區域網路或無線隨建即連網路不同的，例如存取點是不具移動能力的，此外在無線網狀網路中，通常會有一個連接至有線網路的存取點(Gateway)，因此在網路協定的設計上會有別於傳統的無線區域網路或隨建即連網路。例如由於網狀網路是以效能的提升(Throughput)為考量，而非以省電或硬體成本為考量，因此通訊協定上的設計可以較為複雜，而多頻道的使用也更為合理，因此針對無線網狀網路的環境，在頻道的切換與管理方面我們預期提出一個可在鏈結層(Link Layer)實作的解決方法，以便提供整個無線網狀網路最大的效能。

第三節、研究範圍

主要的研究範圍為鏈結層的頻道管理協定：

我們將針對無線網狀網路提出一個頻道管理協定，並於鏈結層上實作出來，我們預期此頻道管理能夠在只有一個網路介面卡的環境下運作，如此

能減少硬體的成本與硬體的相容性。

在頻道管理上要考量的主要問題是，要分配哪一個頻道給哪一個網路介面卡使用，分配好頻道後使用多久要再度切換頻道等問題，在決定如何分配頻道的問題上，我們必須考量每個鏈結(Link)的負載量(Load)，以及鄰近存取點所使用的頻道狀況，如此才能降低干擾進而提高整體網路效能。

而頻道的使用問題上，我們必須考量每個存取點都可隨著時間的變化而動態的調整所使用的頻道，在這過程中要考慮如何跟鄰居交換頻道的使用資訊，另外在複數頻道環境下廣播(Broadcast)封包的傳送也是我們必須考量的問題。

第四節、論文結構

本篇研究論文之架構區分為五個章節：第壹章為緒論，闡明本研究之背景、動機與目的，第貳章為相關研究，介紹目前有關使用複數頻道應用於的隨建即連(Ad Hoc)或無線網狀(Wireless Mesh)網路的文獻，在第參章中，會詳盡說明我們提出的方法和管理協定，以及衍生出來的議題，第肆章利用真實的網路環境形成無線網狀網路，並修改網路卡的驅動程式，把我們的協定加進裡面，並測試成果，於第伍章中則是本研究論文的結論與未來發展方向，以供為後續研究之參考。

第貳章、相關研究

下面我們將透過本章相關文獻的整理，進一步了解應用複數頻道在無線網路方面的發展，並分析它們的優缺點。

協定	[1]	[2]	[3]	[4][5]	[12]	[13]
頻道數	有限	有限	有限	有限	無限	有限
MAC 協定	需新 MAC	802.11 相容	802.11 相容	802.11 相容	需新 MAC	需新 MAC
網路卡數	兩張	均可	單張	多張	單張	多張
時間同步	不需要	不需要	需要	不需要	不需要	不需要
演算法	分散式	集中式	分散式	分散式	分散式	分散式
應用網路	Ad-Hoc	Mesh	Mesh	Mesh	Ad Hoc	Ad-Hoc

表(一)、複數頻道範疇

上表列出了相關複數頻道應用在無線網路的範疇，在[12]中假設頻道數無限多，在[1]中需要新的媒介存取層協定(Medium Access Protocol, MAC)，在[4][5]中網路卡數必須多於一張才能實行，[3]為單張網路卡，利用不停的切換頻道來提高空間的再使用率，但必需時間同步，[2]中的演算法是集中式演算法，必須知道整個網路的拓撲，[13]把複數頻道建立在繞徑(routing)上，應用的網路類型屬於隨建即連(Ad-Hoc)式網路。

我們的問題鎖定在無線網狀網路和不改變 IEEE 802.11 協定的條件下，所以以下僅簡介在此條件下的研究，頻道數無限多不是一個合理的假設，我們討論的是有限的頻道數。下面列出幾個相關的研究，並分析其缺優點，以便帶入我們的研究主題。

第一節、複數頻道多網卡無線網路（MCR）

[5]為美國伊利諾科技學院(University of Illinois at Urbana-Champaingn) Pradeep Kyasanur 及 Nitin H. Vaidya 提出的分散式方法，他們稱作 MCR(Multi-Channel Routing)，不需要時間同步，使用兩張以上網路卡，一為固定式網路卡，另為可切換式網路卡，每個存取點都需為固定式網路卡設置一個頻道，這個頻道稱之為接收頻道，在鄰近區(neighborhood)的每個存取點的接收頻道需盡量不同，為了達到這個要求，每個存取點會廣播自己的接收頻道給鄰居知道，當鄰居收集到這些資訊，便可設定一個較適當的頻道給固定式網卡。這是一個接收頻道為主(receiver-based)的演算法，利用固定式網卡來接收資料，用可切換式網

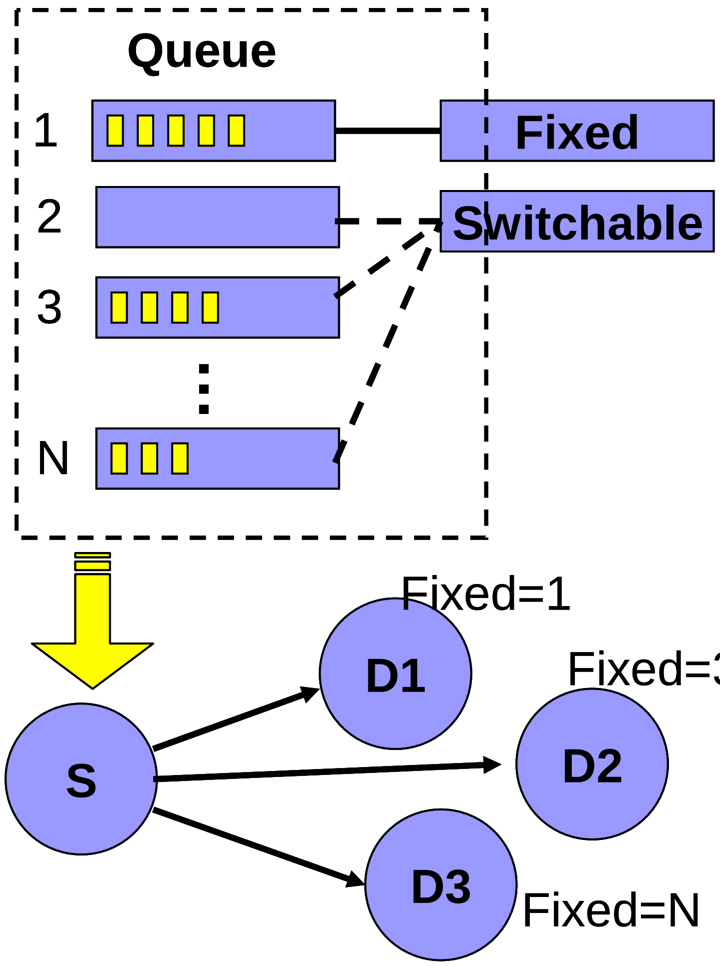

圖 2-1 複數頻道多網卡無線網路（MCR）

卡來送資料，當要送資料時，可切換式網卡必需切換到接收端的接收頻道上，這樣才能使用相同的頻道進行傳送。

每個存取點會維護每個頻道佇列，如上圖 2-1，當上層有封包要求傳送時，封包會依照接收端的接收頻道來放進頻道佇列中，固定式網卡只需管理接收頻道的那個頻道佇列，而可切換式網卡必需適時的切換到其它的每個頻道佇列去把未送出的資料送出。另一個需要解決的問題是廣播問題，因為每個存取點接收資料的頻道不同，所以當需要廣播的時候，每個頻道都要發送，所以廣播的封包必需在每個頻道佇列上都複製一份，才能確保接收端能收到資訊。

MCR 的缺點在於網路上每個存取點的接收和傳送的比例不盡相同，某些存取點可能時常在做傳送的動作或接收的動作，便會有一張網路卡是浪費的，而且在這個方法的運作下，廣播需要很大的花費，因為每個頻道都要廣播一次。

第二節、時槽式跳頻法（SSCH）

[3]為微軟(Microsoft) Raramvir Bahl、Ranveer Chandra 及 John Dunagan 的研究，簡稱 SSCH(Slotted Seeded Channel Hopping)，是一個分散式的演算法，只需用單張網卡，不過需要時間的同步。

SSCH 利用跳頻的方式來達成提升空間中頻道的再使用率(Spatial Reuse)，SSCH 定義一個時槽(slot)為在單一頻道上所花費的時間，他們選擇一個時槽為 10 毫秒，並定義頻道排程(channel schedule)為一連串的頻道作為跳頻順序，頻道排程包含目前的頻道和一個跳頻的規則，可利用規則計算出下一個時槽該跳至那一個頻道，利用規則來跳頻可以節省使用龐

大的空間來儲存跳頻的順序，每個存取點必須儲存並維護其它鄰居區內所有存取點的頻道排程。

SSCH 用四組(channel,seed)配對來組成頻道排程，他們實驗結果顯示四組已經有很好的效能增進，用(xi,ai)來表示(channel,seed)，xi 表示[0,12]共 13 個可能的頻道，而 seed ai 為[1,12]個整數，每個存取點會一直根據頻道排程的資訊來跳頻，下個時槽的跳頻頻道公式為：

xi<-(xi+ai)mod 13 ---公式(一)

13 為可能的頻道，此篇假設可用頻道為 13 個，這個可用頻道數必需為質數個，如此一來，利用公式(一)的跳頻方法，任意兩個存取點的 ai 不同，便能保證在跳躍過程中，每 13 次的跳躍，一定會有某個時槽，而且只有一個，兩個存取點的頻道會跳到同一個頻道，如此即使頻道排程互不相同的兩個存取點也能經由這種偶發性的頻道重疊而進行溝通，主要是用在廣播上。但如果兩個存取點頻道排程的頻道不同，但種子(seed)相同時，使可能發生兩個頻道永遠沒有重疊的情況，所以走訪完每個配對的所有頻道必須加上一個配類時槽(parity slot) ，這個配類時槽所需跳躍到的頻道為第一個配對的種子(seed)的數值，加了這個配類時槽就能避免兩個存取點的頻道不一樣但種子(seed)相同帶來頻道永遠無法重疊的情況，因為至少能在配類時槽能重疊。

下圖 2-2 為兩個存取點的頻道排程，此例共有 3 個頻道，2 組(channel,seed)配對，每一個週期(cycle)必須走訪完每個配對的所有頻道加上一個配類時槽(parity slot)，接著一直重覆著這個週期。如圖可以看到存取點 A 的第一組配對(x1,a1)和存取點 B 的第一組配對(x1,a1)均為(1,2)，表示兩個存取點的第一組配對頻道排程相同，跳頻的順序均會相同，在此配對的時槽中，兩個存取點便可進行溝通，因為種子(seed)也相

同，所以在配類時槽(parity slot)也會重疊。在圖中時槽裡的數字表示在這個時槽要跳躍到的頻道，存取點 A 的第一個配對(1,1)表示目前這個時槽使用頻道是頻道 1，而更新的規則是加上種子(seed)1，所以下一次跳躍的頻道是(1+1)mod 3=2。

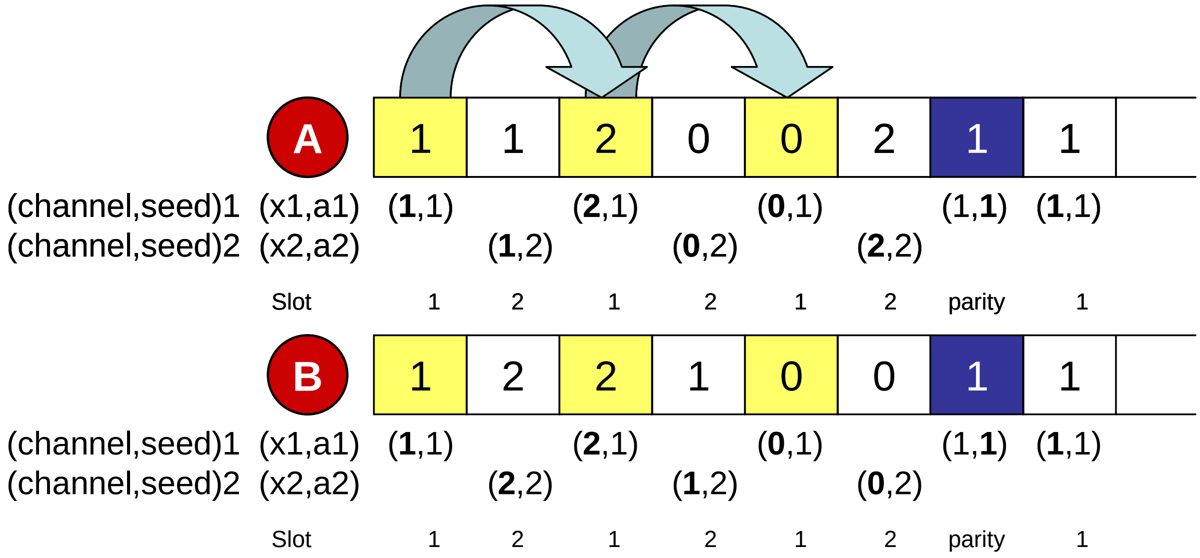

圖 2-2 時槽式跳頻法 (SSCH)

當某存取點想要傳輸資料給某鄰居存取點時，可以改變自己的某個(channel,seed)配對跟鄰居相對應的配對相同，如此，這個配對所屬的時槽便能對行溝通，而另外的配對可以再跟其它的鄰居配合傳輸，在他們的模擬中是以四組配對作為實驗。在如此的架構下，除非(channel,seed)配對完全相同，否則最多頻道只會重疊一次，可降低干擾，至於廣播的問題，他們在每個時槽均廣播自己的頻道排程，因為每個週期(cycle)會至少頻道重疊一次，所以利用這個性質，便可確保在一個週期內可把廣播訊息廣播出去。

SSCH 的缺點在於廣播所需要的時間很長，假設每個時槽為 10 毫秒、13 個可用頻道，需 530 毫秒才能確保廣播至所有的鄰居存取點，對於繞徑(routing)尤其不利，找尋路徑需花費很長的時間。而且每個時槽都必需廣

播自己的排程，花費高。另外，SSCH 並沒有規定那些需固定用那些頻道排程，不小心就一群人都使用相同的排程，造成頻道擁塞(channel congestion)，需要做反同步(de-synchronization)來把頻道使用再度打散。

第參章、時槽式複數頻道管理協定

整理了許多複數頻道相關的研究，我們想出了一個方法適用在有限頻道數、單張網路卡、IEEE 802.11 相容、分散式(distributed)、網狀網路(Mesh Network)的複數頻道管理協定，不過需要時間的同步，並可容易擴充至多張網路卡的環境。

第一節、基本原理

我們發展了一套適用於無線網狀網路(Wireless Mesh Network)，而可實作在鏈結層(Link layer)上使用複數頻道的頻道管理協定(Channel Management Protocol)，這是一個以接收端的頻道(Receiver-based)做為傳輸頻道的方法，主要想法是假設當有一個存取點(Access Point)A 要傳送資料給另一個存取點 B 時，A 就要切換到 B 所使用的頻道上進行通訊。我們假設每個存取點都會分配到一個接收頻道(receiving channel)，此頻道應該與存取點的鄰居(Neighbors)所使用的接收頻道(receiving channel)要盡量的不同。如圖 3-1，存取點 A,B,C,D,E,F 都盡量的使用不同的頻道做為接收頻道(receiving channel)，例如 A 使用 Channel 3，B 使用 Channel 5，C 使用 Channel 4 等，假如 B 要傳輸資料給 A，會用 Channel 3 去傳輸資料，同時 C 要傳資料給 D，會用 Channel 1 去傳輸，所以在同一個鄰居區內，CD 和 AB 的傳輸會用不同的頻道，而降低干擾使網路吞吐量(Throughput)增加。至於一個存取點如何選取適當的頻道，稍後會做詳細的說明，下面先將設計頻道分配演算法所要用到的其它概念做敘述。

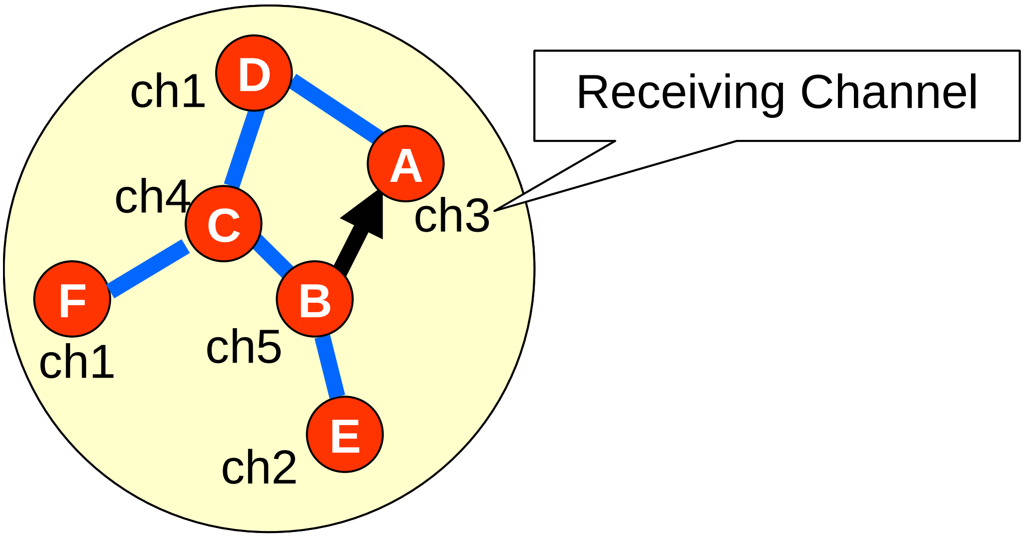

圖 3-1 接收頻道示意圖

接收端為主(receiver based)的設計會產生一個問題，如果存取點 A 要傳送資料給存取點 B，A 會切換到 B 的接收頻道(receiving channel)上，但假設此時 B 也正要傳送資料給 C，則 B 會切換到 C 的接收頻道(receiving channel)上，如此有可能形成死結(deadlock)的情形，為此我們加進“分時”的概念，我們把每個存取點的時間軸切成一個一個的時槽(time slot)，我們假設每個存取點的時槽的開始是同步的，並設定每 k 個時槽為一個週期(cycle)，然後重覆這 k 個時槽。在這 k 個時槽裡，每個存取點（假設為 A）需指定好那些時槽是用來傳送資料給其它存取點，那些時槽是用來接收其它存取點所送過來的資料，然後把這個資訊廣播給其鄰居（假設為 B），當 B 收到這個資訊時，B 就能知道 A 的時槽使用情況，如此 B 有資料要傳送給 A 時，B 能夠知道 A 何時可接收資料，何時不能接收資料，當然 B 會選擇 A 可接收資料的時槽傳送資料給 A，除了傳送時槽、接收時候和廣播時槽外，我們還選了一個接收時槽當固定式接收時槽(Fixed Receiving Slot)，因為我們可以動態改變排程，所以接收時槽可能轉變為傳送時槽，為了不讓所有的時槽都變成傳送時槽，固定式的接收時槽是不能變成傳送時槽的，至於固定式接收時槽的選取，也是在鄰居區內要盡量不同。

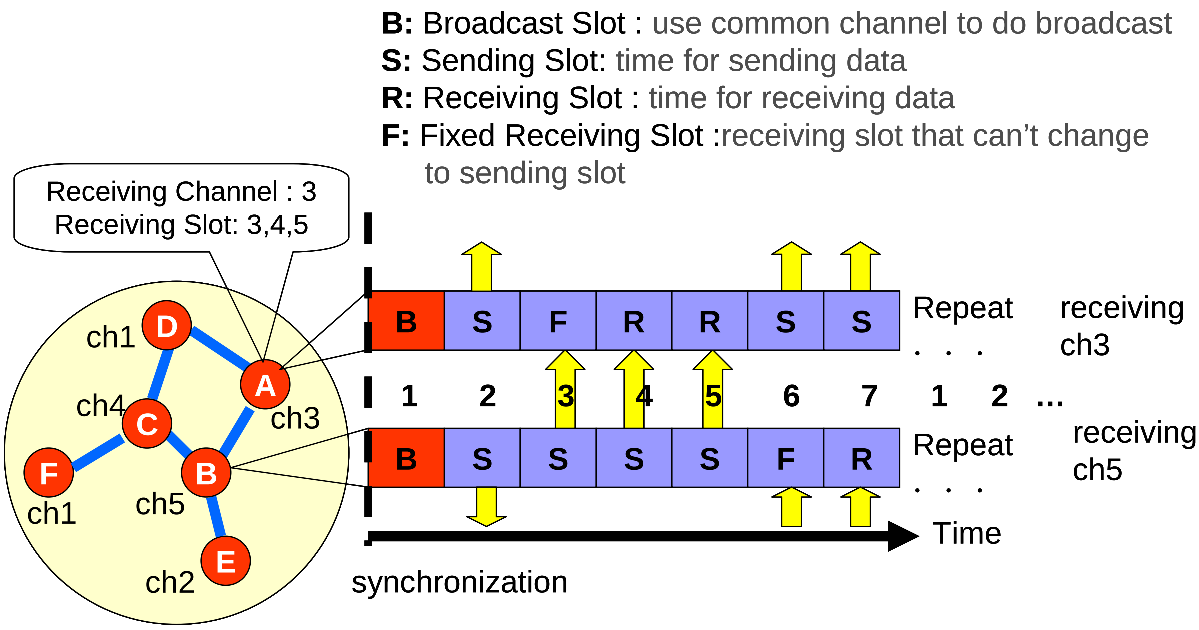

圖 3-2 channel model

如圖 3-2：在這個例子中 k 值為 7，當存取點 B 要傳送資料給 A，因為 B 有收到 A 的廣播說 A 的時槽 3,4,5 是用來接收資料的時槽，因此 B 要送資料給 A 時，B 會利用時槽 3,4,5 將頻道切換到 A 的接收頻道(receiving channel)，在此例中為 Channel 3，來進行資料的傳輸。

另外要解決的問題是廣播(Broadcast)的問題，在複數頻道(Multi-channel)的網路環境下，因為每個存取點可能正使用不同的頻道，如何做有效率的廣播就是一個問題，相關研究中廣播的方法大多是複製多個廣播封包(Broadcast Packet)在每個頻道都廣播出去，以便確保鄰居都能接收到此廣播封包，我們所採取的做法是選擇第一個時槽裡當成廣播時槽(broadcast slot)，在這個廣播時槽裡，頻道會切換到一個大家共同的頻道，其目的就是要把所有的存取點在這個時候同時切換至此共同頻道上，如此一來，所有的存取點便能同時接收或傳送廣播封包，因為我們並沒有改變 IEEE 802.11 的 MAC 協定，所以這時的接收和傳送是經由 IEEE 802.11 的競爭機制在傳送。這樣做的好處在於每一次的廣播只需廣播一次便所有的存取點都接收的到，並不需要在每個頻道上做廣播的動作，也不

需要複製多個廣播封包。此外，我們必須考量一個問題：這樣是不是會造成頻道擁塞(channel congestion)？因為這個時候所有的存取點都切換到這個頻道上，造成封包過多超過這個頻道所能負荷的量。我們想這是可以避免的，因為我們定義的一個週期的時槽數可經由廣播封包和一般封包的比例來設定，廣播時槽可以在一個週期不一定只有一個，可以有兩個或三個，可視這個網路的特性去調整這個參數。廣播時槽帶來的好處還不只這些，在複數頻道上同步是有困難的，因為所有的人不在相同的頻道上，有了這個廣播時槽，順使可以在這個時槽發送信號彈(beacon)來達成時間同步的效果。

第二節、衍生議題

發展這個管理協定，衍生出一些待解決的議題，第一，如何決定每個存取點的接收頻道(receiving channel)？第二，如何分配每個存取點傳送時槽(sending slot)和接收時槽(receiving slot)的比例？第三，如何分配傳送時槽和接收時槽的順序？第四，進入某個傳送時槽時，要選擇傳送給那一個鄰居才不會造成不公平？

第一項、接收頻道分配

(Receiving Channel Assignment)

為什麼要選擇接收頻道？因為接收頻道的選擇關係到整個網路的效能，每個連線(wireless link)使用愈不同的頻道，干擾的情況就愈小，整體網路效能便能上升。現在我們已經可以來敘述一個存取點如何選擇接收頻道，其方法如下：

1. 加入網路後，先聽數個週期，但不發送訊息，得知鄰居(Neighbor)使用頻道的資訊。
2. 選一個較沒有其它存取點在用的頻道當接收頻道(receiving channel)。
3. 廣播自己的接收頻道(receiving channel)。
4. 收到其它存取點的資訊，更新自己的資料表。
5. 每隔一段時間檢查資料表，發現有太多人用同一個頻道當接收頻道(receiving channel)時，重覆步驟 2~4。

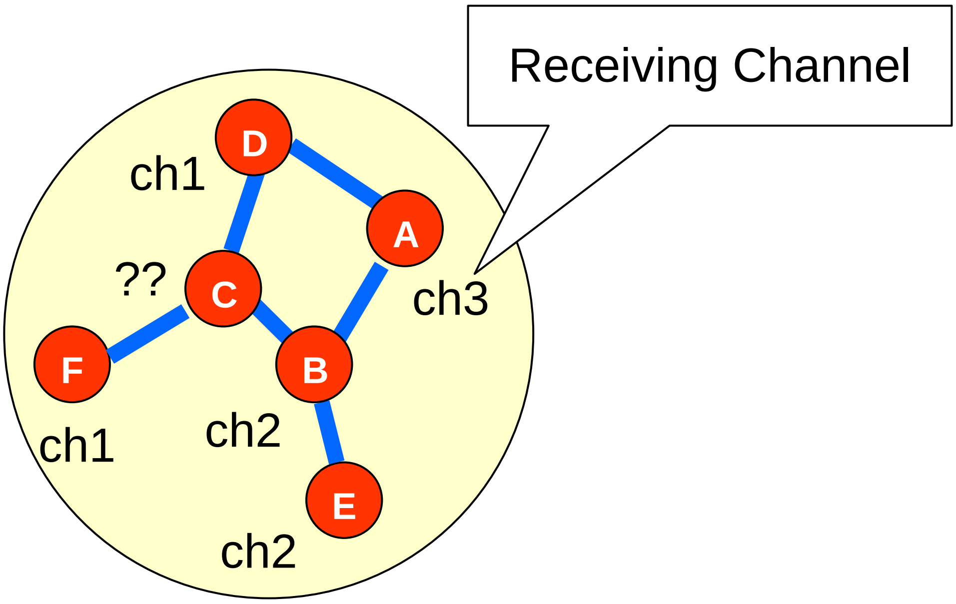

圖 3-2 選擇接收頻道

以上圖 3-2 為例，A、B、C、D、E、F 節點代表的是每個存取點，旁邊的數字代表接收頻道為幾號頻道，總頻道數為 3，以 ch1、ch2、ch3 表示。點 C 是剛加入網路的存取點，因為鄰居存取點會在廣播時槽廣播自己的接收頻道是那一個頻道，當 C 收集一段時間後便能得知存取點 B、D、F 的接收頻道為 ch1、ch1、ch2，所以選一個較沒有其它存取點在用的頻道 ch3 當接收頻道，接著廣播自己的接收頻道讓其它存取點知道。因為我們的演算法是一個以接收端的頻道(Receiver based)做為傳輸頻道的方法，要傳輸

就必需切換到接收端的接收頻道去傳輸，如果每個接收端的接收頻道均不同，則每條連線所使用的頻道就會不同，干擾便會大幅降低，這就是為何我們需要選擇盡量不同接收頻道的原因。

第二項、傳送接收時槽比例分配

(The Ratio of Sending and Receiving Slot)

我們觀察到並不是每個存取點傳送量和接收量總是一半一半，每個存取點的網路流量特性不同，某些存取點比較傾向接收資料，或總是傳送資料，所以傳送時間和接收時間的比例我們設計成可變的，因此傳送時槽數(sending slot)和接收時槽數(receiving slot)變成可動態依照情況而改變，至於如何分配每個存取點傳送時槽和接收時槽的比例，我們使用單位時間內需傳送量和需接收量的比例來當傳送時槽和接收時槽的比例。公式為：

$\alpha * S/R < O/I \rightarrow S+1, R-1$

$\beta * S/R > O/I \rightarrow S-1, R+1$

α、β為穩定度的參數，S 為目前傳送時槽的數目，R 為目前接收時槽的數目(包含固定接收時槽 Fixed Receiving Slot)，O 為單位時間需傳送量，I 為單位時間需接收量。雖然說無線網狀網路(mesh network)的網路流量穩定，但如果單位時間需傳送量和單位時間需接收量的比例剛好在某個臨界數值之徘徊，會造成一下增開傳送時槽，一下子又減少傳送時槽，造成排程不穩定，這不是我們希望的，我們取α為 1.2、β為 0.8，表示如果比例沒有超過α倍的話，不會增開傳送時槽，沒有低於β倍的話，不會減少傳送時槽，避免時槽比例經常性更換。另外有一條規定是，接收時槽最小個

數需保持一個來接收資料，我們使用固式定接收時槽來達成這個規定，因為這個時槽是不能改變成傳送時槽的，以免其它人永遠無法傳送資料給它。如此一來，每個存取點便依照著自己的流量特性來分配傳送和接收比例，達到較好的傳輸效果。

接下來我們探討 I（單位時間內所需接收的量)要怎麼衡量？O（單位時間內要傳送的量）可以用上層(IP 層)要求傳送封包的量來衡量，但是 I 並不能用收到的流量來衡量，因為只能由接收時槽上獲得，無線網狀網路是高負載的網路，IP 層要求傳輸的量遠大於從接收時槽上接收到的流量，所以依照我們的演算法，傳送時槽會一直增開，所以我們衡量 I 需用別人想要送給我們的流量來計算，但是每個存取點並不知道別人有多少流量要傳給它，所以每個存取點在廣播頻道排程時，要順便將要傳送給其它存取點的流量含進排程，其它存取點收到時，便能清楚知道別人要傳給自己有多少流量，用這個一數值來衡量 I，這個演算法便可以讓傳送時槽和接收時槽的比例正確的分配。

第三項、傳送接收時槽順序分配

(The Order of Sending and Receiving Slot)

當決定了傳送時槽(sending slot)和接收時槽(receiving slot)的比例，還必需決定傳送時槽和接收時槽要放在週期內的那一個時槽裡，如果排序的不好，如下圖 3-3，兩個存取點的排程幾乎都一樣時，會造成無法連線的情況，所以排程也必需詳細考慮。

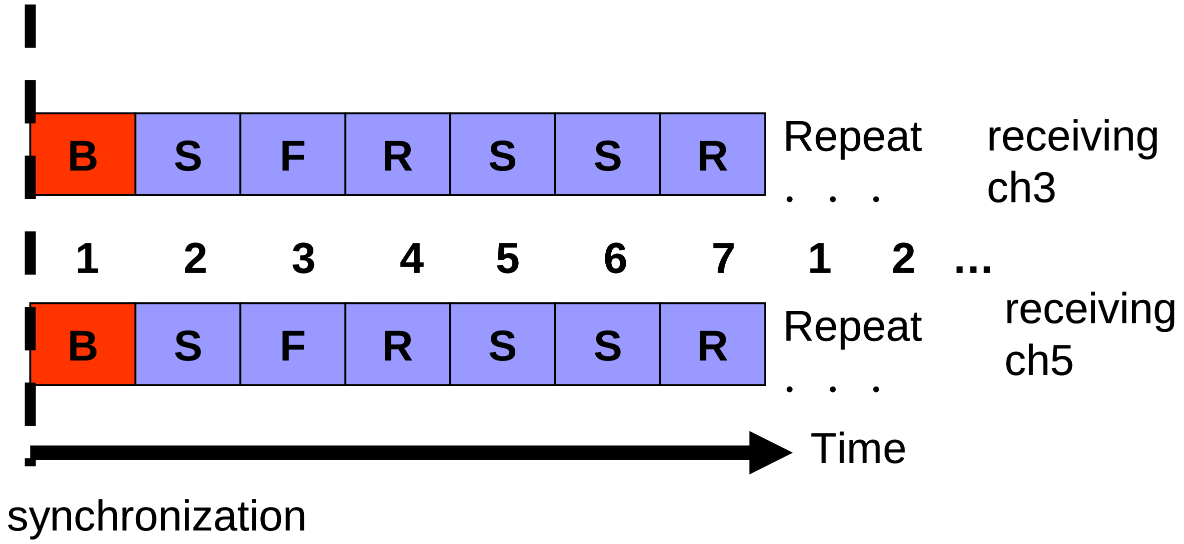

圖 3-3、排程相同造成無法溝通

初始時，除了廣播時槽(broadcast time slot)外，其它時槽(time slot)預設均為接收時槽(receiving time slot)包含固定接收時槽(fixed receiving slot)，隨著傳送流量的變化，會有三種情況，第一種情況，傳送時槽和接收時槽比例 S/R 沒有變，第二種情況，需要增開傳送時槽，第三種情況，需要減少傳送時槽。

第一種情況是最簡單的，排程根據之前傳送時槽和接收時槽的順序便可。第二種情況，當需增加傳送時槽(sending slot)時，定義 Lj 為第 j 個時槽衡量值，這個值愈高代表這個時槽改成傳送時槽會有比較好的效果，下面會有 Lj 的正式定義，當要增開傳送時槽時，我們會選擇所有接收時槽(不含固定接收時槽)中 Lj 值最大的那個時槽。第三種情況，跟第二種情況反其道而行，選擇所有傳送時槽中 Lj 最小的時槽變成接收時槽。

接下來，我們正式的定義 Lj：

$$Lj = \sum_{Sij=\text{'R' or 'F'}} Wi/Gi \; - \; \sum_{Sij=\text{'S'}} Oi/Ti$$

Sij 為鄰居 i 的第 j 個時槽的類型(接收時槽、傳送時槽或固定接收時槽)，Wi 為對鄰居 i 要傳送的量的比重(weight)，這個比重通常用封包到達速率(packet arrival rate)來代表，也可以用其它方式來代表，定義 Oi 為其它鄰居有多少流量要傳給我的比重，從鄰居發送排程時內含的資料獲得，Gi 為自已的傳送時槽配合到鄰居 i 的接收時槽的總個數，意義上是說自己共有 Gi 個時槽可以跟 Ni 進行傳輸，如下圖 3-4，自己和鄰居 1 的 Gi 值為自己的傳送時槽數配合到鄰居相對應的接收時槽，此例，G_1是 2，因為共有 2 個時槽有配合到。反之，Ti 為鄰居 i 的傳送時槽配合到自己的接收時槽的總個數，根據 Lj，我們便可以選出效果較好的時槽來改變成接收時槽或傳送時槽。

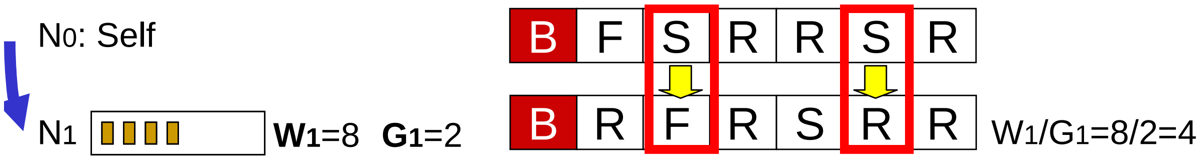

圖 3-4、Gi 值的計算

下圖 3-5 為某存取點增開新傳送時槽的範例，N_4有資料要傳給 N_0，N_0有資料要傳給 N_1、N_2、N_3，我們先對每個鄰居計算 W_i/G_i和 O_i/T_i，現在是 N_0要增開傳送時槽，所以我們考慮第 4、5、7 個接收時槽，L_4為衡量第 4 個時槽的值，改變 L_4變成傳送時槽會影響到 N_1、N_3、N_4，這個改變對 N_1和 N_3是好的，因為 N_1和 N_4便可接收到來自 N_0的資料，但對 N_4是不好的，因為 N_4的資料便無法從這個時槽傳送到 N_0，所以 L_4的值是 W_1/G_1加 W_3/G_3再減 O_4/T_4，算出來是 9，我們再計算其它時槽的 Lj，L_5是 5，L_7為無限大，因為 N_0有資料要傳給 N_2，但 N_0跟 N_2並沒有可溝通的時槽，G_2為 0，W_2/G_2變成無限大，所以我們目前最迫切的是要能跟 N_2溝通，所以改變第 7 個時槽成為傳送時槽

是最好的。我們可以發現並不是只考慮比重(Weight)最重的那一個鄰居，因為有可能這個鄰居已經可以用其它的時槽去溝通了，最迫切的是那些有資料要傳，但可溝通的時槽很少的那些鄰居。

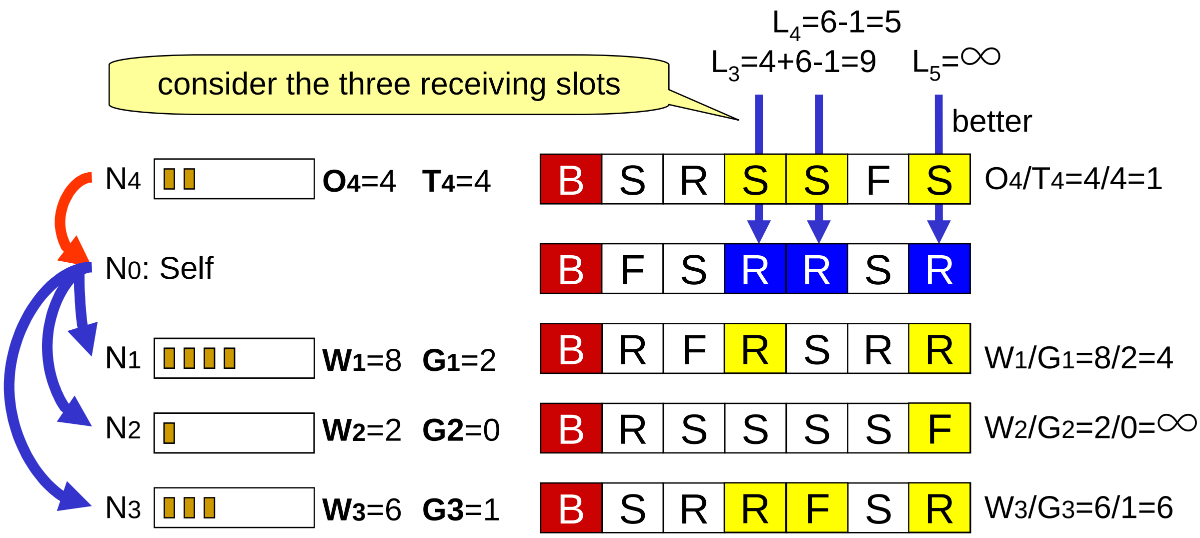

圖 3-5、增開傳送時槽範例

第四項、傳送端選擇

(The Destination of Sending Slot Selection)

在接收時槽，只需靜待在自己的接收頻道上等待別人傳送資料，但在傳送時槽卻還需判斷要傳送給誰資料。在解決這個問題前，會先碰到我們用什麼資料結構來儲存將要送出去的封包，不同的資料結構會造成能判斷的能力不同。

我們用時槽來分割時間，但是封包並不是盲目的傳送出去，因為在某個傳送時槽，頻道是換到某個鄰居的接收頻道上，只有當鄰居是處在這個接收頻道上時才能把封包傳送出去，其它的封包必需用其它的緩衝區(佇列)暫存起來，這衍生出一個問題，用頻道佇列(per channel queue)還是鄰居佇列(per neighbor queue)來實作緩衝區會比較好？頻道佇列和鄰居佇列

範例如下圖 3-5 所示，存取點 A 需暫存鄰居的資料，圖左為鄰居佇居，就是為每一個鄰居都開一個佇列來當緩衝區，圖右為頻道佇列，總頻道數固定時，佇列數也就固定，存取點 B 和存取點 E 使用相同的接收頻道 ch1，所以 B 和 E 必需共同一個緩衝區。使用頻道佇列和使用鄰居佇列各有一些優缺點，使用頻道佇列的優點是總頻道的數道固定，將來網路卡硬體可提供這類型的佇列，效能上會比較好。而使用鄰居佇列，當在自己的接收時槽時，可以視接收到的封包的傳送端為可溝通對象，把一些要回傳的封包從鄰居佇列中拿出傳回去，但頻道佇列沒有辦法做到這點，因為他是以頻道為佇列，一定要等到傳送時槽到來換到該頻道，才會從該頻道佇列取出封包來傳送。

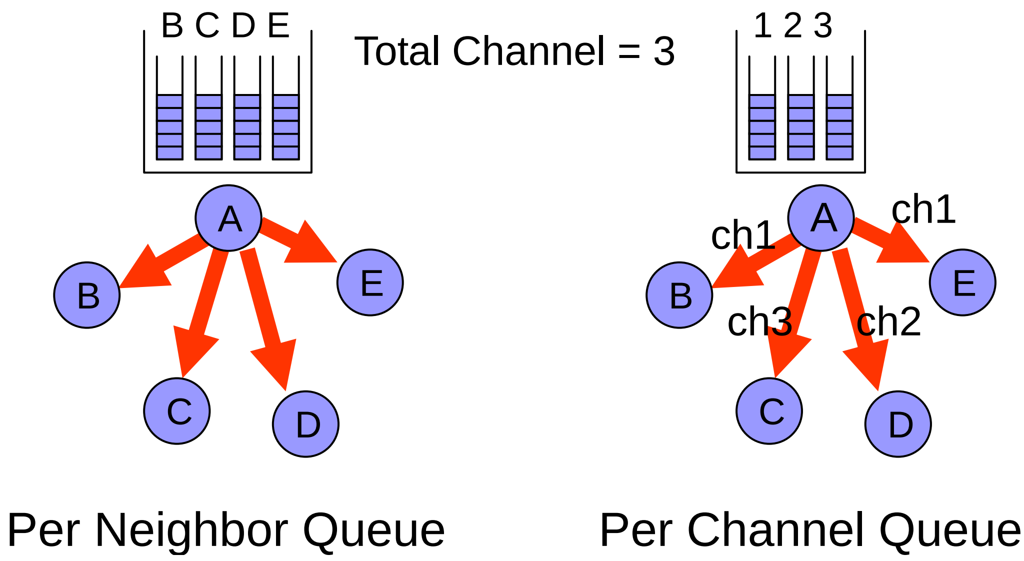

圖 3-5、鄰居佇列與頻道佇列

　　評估衡量的結果，我們決定以頻道佇列來實作，這關係到進入每一個傳送時槽選擇接收端(destination)的演算法，我們使用傳統的比重分配法(Weighted Round Robin Algorithm)，但這並不全適用於我們的架構之下，因為每一個傳送時槽，我們只能選擇正處理接收時槽的存取點來傳輸。我們為每一個頻道佇列設定一個比重(weight) Wi 和優先權(priority) Pi，

這個比重和傳送接收時槽順序分配時用到的那個比重是一樣的。一開始每個頻道佇列的 Pi 相同，Wi 隨著網路流量的變化而改變，在無線網狀網路(mesh network)裡，這個變化量比較小。

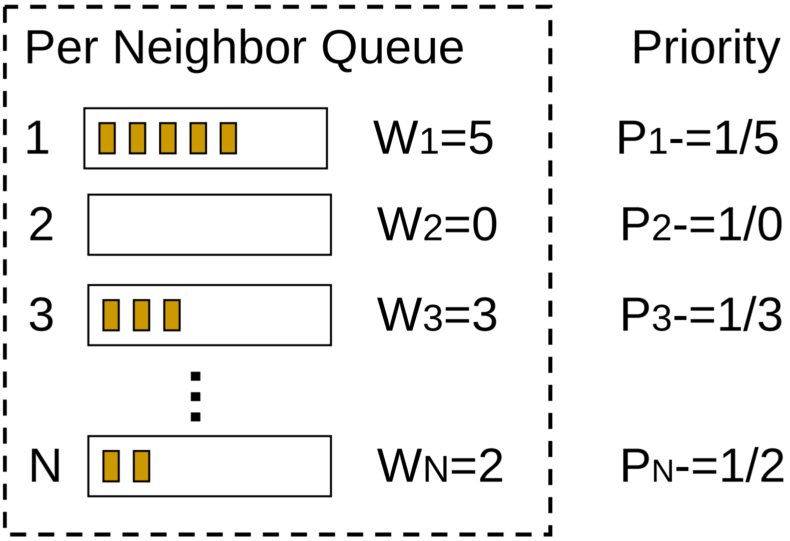

圖 3-6、傳送端選擇及優先權

當進入每個傳送時槽時，先去查看排程資料表，得知那些鄰居正處於接收時槽，再比較這些鄰居誰的 Pi 最高，選擇這個鄰居當成接收端傳輸資料，當結束這個傳送時槽時，降低 Pi 的優先權 1/Wi，如此一來，Wi 重的鄰居優先權降的慢，能得到傳輸的機會比較多，而且一直沒有輪到可傳輸時槽的那些鄰居，因為沒有輪到機會傳輸，一輪到機會時也會因為 Pi 值比其它存取點高而競爭到傳輸的優先權，如圖 3-6，N_1一次降 1/5 而 N_3一次降 1/3，N_1 比較容易取得傳送權，但 N_3 也不會一直競爭不到傳送權。我們會週期性的把 Pi 重置(reset)，不會造成不公平的狀況發生。

第肆章、實作

第一節、硬體和系統環境

為了在真的環境上面實作，查閱了相關的研究，去找尋有公開原始碼的驅動程式(Open Source Driver)，Atheros 有公開晶片 Linux 驅動程式的原始碼，使用 Atheros 晶片的網卡都可以使用這個驅動程式來驅動。所以我們選擇了幾台筆記型電腦，每台上面均裝有 D-link DWL-AG650 的網路卡，它是使用 Atheros 的晶片，這讓我們可以修改公開的原始碼來把我們的管理協定實作在上面。

圖 4-1、D-link 網路卡

圖 4-2、實作環境

我們利用這些筆記型電腦當成是一個個的存取點，讓它們形成隨建即連網路(Ad-Hoc Network)，並固定其位置模擬無線網狀網路(Mesh Network)，無線網狀網路和隨建即連網路有很大的共通性，差別在於無線網狀網路有閘道可以連上網際網路(Internet)，且無線網狀網路沒有行動性(Mobility)，我們讓這些筆記型電腦模擬一個無線網狀網路的雛形(Prototype)來當成我們要的環境。

第二節、程式架構

我們的管理協定是屬於 OSI 七層網路模型的鏈結層(Data Link Layer)，而在 Linux 的網路架構下是簡易的分成四層，我們所要修改的是最底層鏈結層。

下圖 4-3 為 Linux 使用者空間(User Space)、核心空間(Kernel Space)和網路驅動程式(Network Driver)之間的關係，上層使用者空間的行程(User Space Processes)透過系統呼叫(System Call)跟核心來通溝，而核心內行程的溝通都是使用函式呼叫(Function Call)的方式，在 Linux 的網路分層架構下，每一層之間都有佇列(Queue)把出境封包暫存起來，等到有時間可以處理時，才使用函式呼叫去處理送出的動作，而外來的封包則是使用中斷(Interrupt)的方式來通知驅動程式，驅動程式需處理鏈結層的一些必要動作，例如拿掉鍵結層的標頭(Header)、過濾封包…，處理完後再使用軟體中斷(Software Interrupt)通知核心處理更上層的工作。

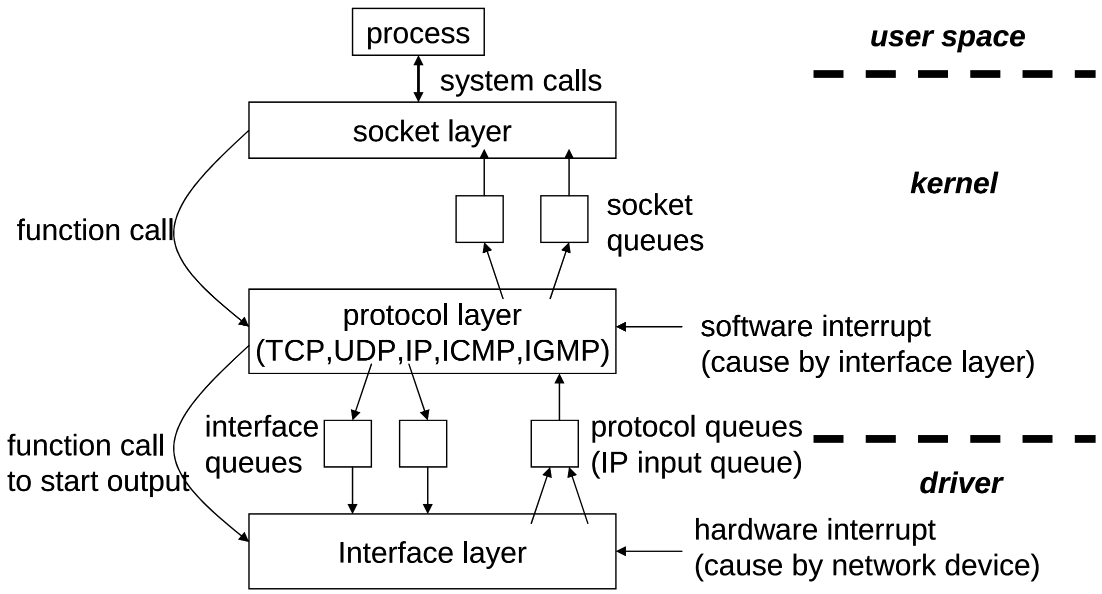

圖 4-3、 Linux User Space、Kernel 和 Network Driver 之間的關係

圖 4-4 是我們使用的 Madwifi 驅動程式和 Linux 核心之間的架構，驅動程式利用模組(Module)方式掛載(Mount)進入核心時，會跟核心註冊一些函式。當核心需要送出資料封包時，會先將資料排入「出境佇列」(Outgoing Queue)，然後呼叫網路介面的 hard_start_transmit()作業方法。在 Madwifi 驅動程式裡是註冊 ath_start()這個函式作為 hard_start_transmit()的函式指標。利用 request_irg()函式註冊 Madwifi 驅動程式的 ath_intr()為中斷函式，而 netif_rx()是核心給驅動程式呼叫用來通知上層處理接收到的封包。

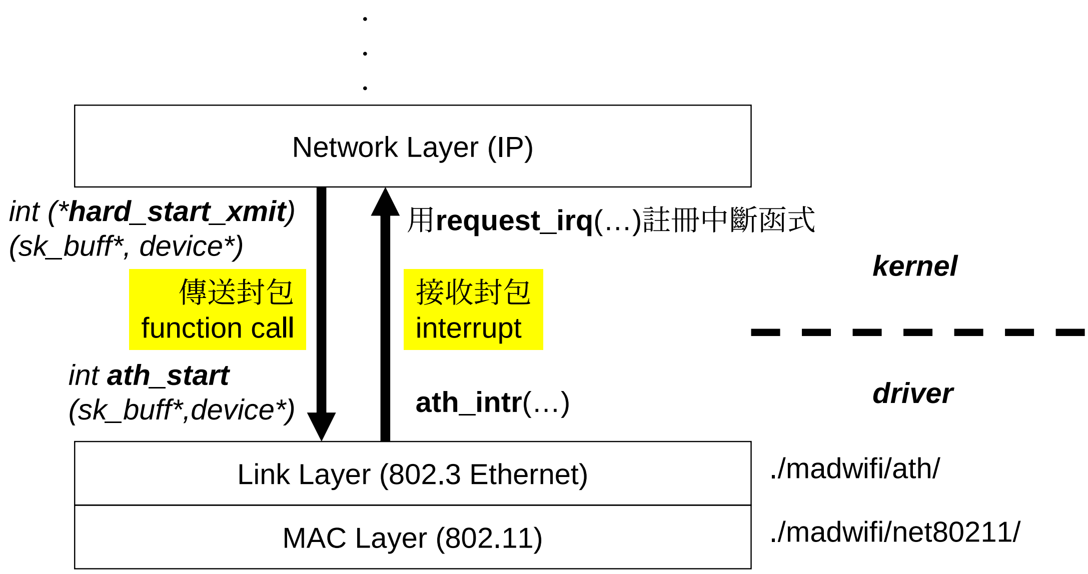

圖 4-4、Madwifi Driver 和 Linux Kernel 之間的架構

下圖 4-5 為對 Madwifi 程式我們所要修改新增的部分。當有封包要送出的時候，我們必須多加鄰居佇列(Per Neighbor Queue)來暫存封包，分類的依據是檢查 MAC 位址，放進適當的佇列中，然後選擇一個此時該傳送的封包傳送出去，接著再觸發傳送下一個封包。當接收到封包時，檢查是否為時間同步或頻道排程的封包，如果是時間同步的封包則更新計時器，若是頻道排程的封包，則更新排程資料表，若都不是則交於上層去處理。新增切換頻道計時器，每 3 秒中斷一次，當切換頻道計時器中斷時，需去查

詢排程資料表，如果進入的是廣播時槽(Broadcast Slot)，則切換至共同頻道上，然後檢查頻道使用情況是否該換接收頻道，也檢查網路流量是否需換排程，一定的週期需重設所有佇列的優先權，如果進入的是接收時槽，最簡單，就切換到接收頻道上去等待資料就好了，如果進入的是傳送時槽，必須依照佇列優先權和頻道排程選一個傳送端，並切換到此傳送端的接收頻道上。

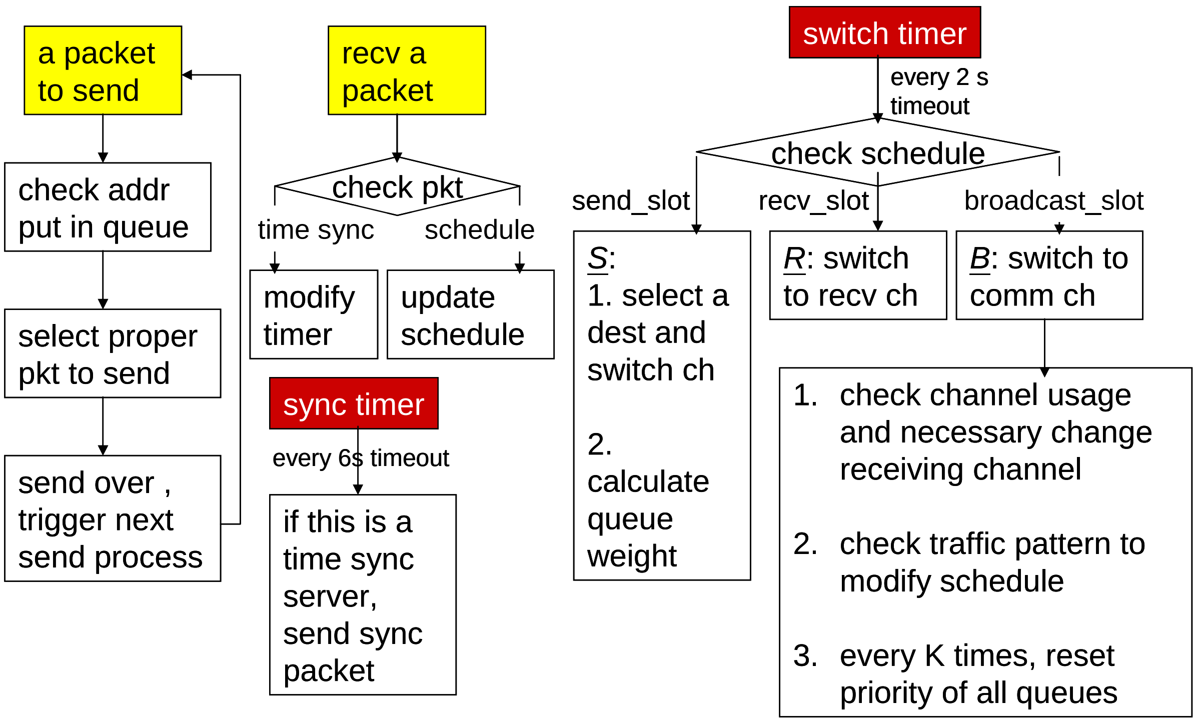

圖 4-5、程式架構圖

第三節、實作的過程與碰到的難題

萬事起頭難，驅動程式是屬於 Linux 核心的部分，跟一般我們碰到的程式限制多了不少，使用不當就會死當，挑戰變成很高，龐大的程式碼也讓我們不知從何下手，經過幾個月的努力查詢程式碼，並閱讀有關 Linux 網路驅動程式相關的書藉來封包了解，才漸漸了解 Madwifi 驅動程式的架構。

首先碰到的問題便是每 3 秒切換一次頻道的問題，因為 Linux 核心不會

做置換(context switch)的動作，所以不用能 while 迴圈或 sleep 去實作，會耗盡 CPU 的時間，其它的工作都停滯不動，其它的行程也都受到牽涉。我們研究 Linux 核心可用的函式庫及語法，利用 timer_list 這個資料結構來實作，向核心註冊中斷時間和中斷服務函式，時間一到，核心會自動去呼叫註冊的中斷服務函式，就可以達成不吃 CPU 時間而且 5 秒中斷一次。我們最後成功的在驅動程式裡加入計時器中斷，完成分時的概念。

接下在我們在 Linux 核心和驅動程式間之間加新一層暫存區，把上層的封包用佇列(queue)暫存，等到適當的傳送時槽(sending slot)再傳送出去，佇列怎麼做？經過一番的努力，發現核心經手的每一個封包，都是包裝成一個 struct sk_buff 結構，要把封包用佇列暫存，需要加上一個 sk_buff_head 把上層下來的封包串起來，便可行成一個佇列。分析封包要傳送的位置，便可利用多個 sk_buff_head 來實作成頻道佇列(per channel queue)或鄰居佇列(per neighbor queue)， 我們是以鄰居佇列為實作佇列的方式。

困擾著我們最久的是切換頻道的問題，我們試了諸多的切換頻道的方法，第一種，直接下 iwconfig 系統呼叫(System Call)去切換頻道，是屬於使用者空間的方法，第二種，使用 Madwifi 驅動程式被 iwconfig 下命令著第一個呼叫的函式，並傳入相同的參數，第三種，在 Madwifi 驅動程式與 IEEE 802.11 無關的程式碼裡選一個最直接切換 channel 的函式去切換，發現一件事，切換頻道均沒有問題，但是同時兩個存取點同時換到同一個頻道時，有時連得起來，有時連不起來，有時連的很快，有時連的很慢，這對我們的管理協定有非常嚴重的效能折扣。第四種，我們直接去呼叫燒在網路卡的韌體函式去切換頻道，發現沒有效果，根本不能切換頻道。正當一切均失敗的時候，我去查閱了 IEEE 802.11 的書籍，找到 802.11 在

Ad-Hoc 模式下為了要同步的關係，需要產生信號彈(Beacon)，我們發現，當一起切換到相同的頻道，當信號彈 BSSID 有合併的時候，網路才有辦法連線，所以我們去追蹤查詢，換頻道的時候會重新產生 BSSID，就是即使隨建即連(Ad-Hoc)網路的名稱相同, BSSID 也會不同, 推測因為 BSSID 的不同, 所以封包在過濾的時候被過濾掉了，所以連線連不起來，當 BSSID 合併才連得起來，我們查到網路卡支援一種 Ad-Hoc Demo 的模式可以把 BSSID 一直維持在 00:00:00:00:00:00，所有封包都會收進來，如此才成功的解決問題。

第四節、測試程式

我們的程式碼架構在驅動程式裡, 為了監控驅動程式的狀態以及測試效能，我們在使用者空間透過共享記憶體的方式去監控驅動程式的狀態，並在上面使用 UDP 的封包進行頻寬的測試，測試畫面如下圖。

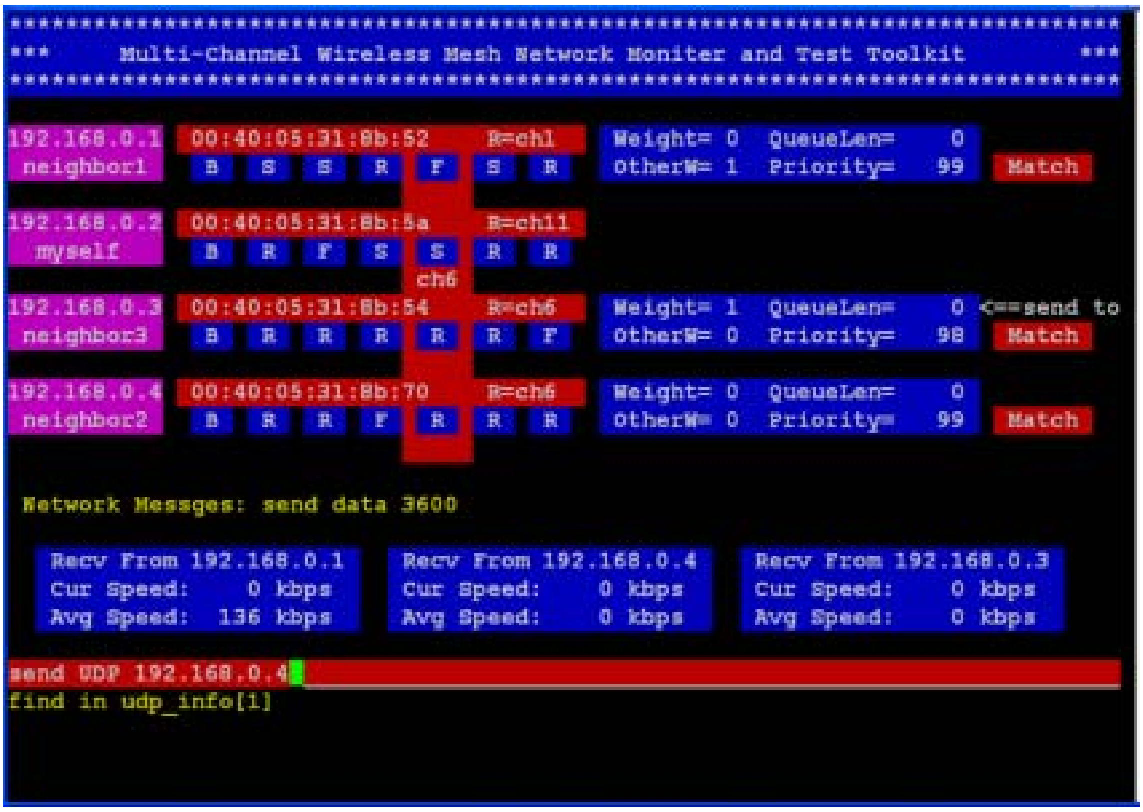

圖 4-6、測試監控程式

看圖 4-6，這是 192.168.0.2 筆記型電腦的測試監控程式的截取畫面，此存取會對其它鄰居紀錄一些資訊，驅動程式儲的資訊放在畫面中 Network Messages 的上方，例如排程、MAC 位址、比重(Weight)、對方對我的比重(OtherW)、目前佇列的長度(QueueLen)以及優先權(Priority)，為了實作上的方便，比重有就是 1，沒有就是 0，因為這個不是實作上的重點，所以用最簡單的方式。畫面中 R=ch1 表示存取點的接收頻道是頻道 1，一個個藍色小方框裡標著 B、R、F、S 是鄰居和自己存取點的排程，會有紅色的長條顯示目前在那一個時槽，而長條中標示的 ch6 表示自已目前這個時槽是用頻道 6，當自己在傳送時槽時而鄰居又剛好在接收時槽時，最右方會有”Match”的紅色標示，這個傳送時槽目前傳輸的對象會在最右邊標示”send to ”的字眼，以上是驅動程式的資訊，用共享記憶體的方式讀取出來顯示在畫面上。

在 Network Messages 的下方在網路層的監控和測試工具，在 Network Messages 可以看目前產生的封包序號或接收到的封包序號，而下方藍色大方框可以看目前網路層有接收到從那來的流量，Cur Speed 是目前這一秒鐘的傳輸速度，而 Avg Speed 是過去一段時間傳輸速度的平均，因為我們有分時槽的關係，用 Cur Speed 來看傳輸速度是不準確的，有時是 700 KBps 而有時是 0 KBps，所以要加 Avg Speed 來測試頻寬。最下方的命令列可以用來下命令測試頻寬，例如”send UPD 192.168.0.4”可以起始一條新的流量(最佳傳輸的方法)給 192.168.0.4 這個存取點。

第五節、測試結果

我們利用測試監控程式來測試所設計的協定，下圖 4-7 為我們測試的第

一個網路環境，四個存取點都在互相的干擾範圍內，A 使用最佳傳送的方式(Best Effort)傳送 UDP 封包給 B，而 C 也使用最佳傳送的方式傳送 UDP 封包給 D，由表 4-1 可看出，在相同未經修改的驅動程式下，使用不同頻道時的效能是使用同頻道的效能的 190%而已，但是 2 個連線此時使用調頻的方式是手動去改變頻道的方式，接著可以看到經過我們經改過的驅動程式(隨著不同環境自動調整頻道)的效能是單一頻道的 135%的效能。

圖 4-7、第一個測試環境

A-B 傳輸速度	C-D 傳輸速度	頻道使用	驅動程式
374KBps	358KBps	同頻道	未經修改的驅動程式
700KBps	698KBps	不同頻道	未經修改的驅動程式(手動調頻)
503KBps	486KBps	不同頻道	我們修改過的驅動程式

表 4-1、測試結果一

接著我們測試鏈狀的網路，如下圖 4-8。

圖 4-8、第二個測試環境

在第二個測試環境下，因為我們無法把四台筆記型電腦拉開至幾百公尺之長來造成鏈型網路，所以我們透過寫死繞送路徑(Routing Path)的方式來

達成這個效果，此時 A 使用最佳傳送的方法傳送 UDP 封包至 D，會經過 B 和 C 最後才到達 D。下圖 4-9 為四個存取點在此環境下排程變化的最終結果，可以看到 A 有三個時槽使用頻道 11 傳送到 B，同一時刻，C 也利用這些時槽使用頻道 1 傳送到 D，此時便有使用到多重頻道帶來的好處，增進鏈狀傳輸的效能。由表 4-2 可看出在我們的多重頻道協定是單一頻道效能的 122%。

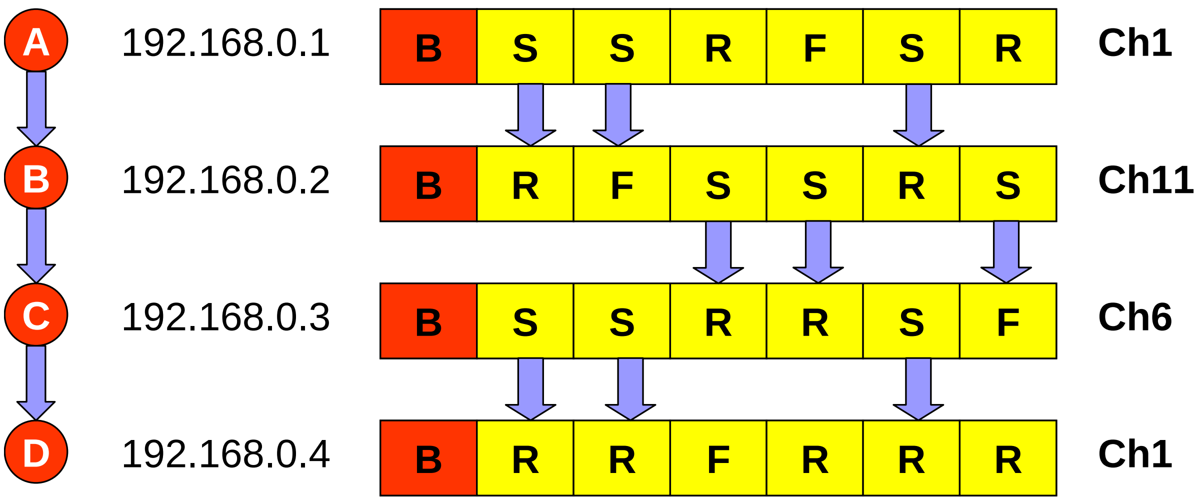

圖 4-9、在第二個測試環境下的排程結果

A-B-C-D 傳輸速度	頻道使用	驅動程式
245KBps	單一頻道	未經修改的驅動程式
300KBps	多重頻道	我們修改過的驅動程式

表 4-2、測試結果二

第伍章、未來展望與結論

第一節、本文研究貢獻

我們提出了一個可套用在 IEEE 802.11 MAC 層上的複數頻道管理協定，透過不斷的切換頻道提高空間中頻道的再使用率(Spatial Reuse)，只需單張網路卡便能達成這個效果，並且容易擴充至多張網路卡。

我們的研究並不是紙上談兵，因為我們模擬真正的環境去修改網路卡的驅動程式，把我們的管理協定實作在上面，並寫了測試程式去監控是否運作的正常，有成功的在實機上面套用我們的協定。

第二節、未來的研究方向

在我們提出的架構之下，衍生出了很多議題，像是傳送接收時槽比例的分配、傳送接收時槽順序的分配、傳送端的選擇，每個議題都是可以再繼續深入研究最佳化的方法，例如傳送接收時槽比例在分配時，α、β穩定的參數如果調整才能達到最好的效果，既穩定又能配合流量來分例。另外，廣播時槽和其它時槽的比例和時槽的總數也是另一個研究的議題，怎麼樣分配才不會造成時間的浪費，又不會造成廣播時槽內頻道擁塞(channel congestion)的問題。

在實作的方面可以擴大其規模，用更多的存取點或用多的頻道來測試我們的管理協定，適當的修改，讓它能夠更成熟更穩定。

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附錄三：

Joint Multi-Channel Link Layer and Multi-Path Routing Design for Wireless Mesh Networks

Y W.-H. Tam and Y.-C. Tseng, “Joint Multi-Channel Link Layer and Multi-Path Routing Design for Wireless Mesh Networks”, IEEE INFOCOM, 2007

Joint Multi-Channel Link Layer and Multi-Path Routing Design for Wireless Mesh Networks

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***Abstract*— In recent years, the wireless mesh network (WMN) attracts the interest of many people as a new broadband Internet access technology. However, increasing throughput is still an open and challenging research issue. One potential solution is to enable transceivers to utilize multiple channels dynamically. However, most of existing works do not consider the routing issue, and trivially use some popular single-path routing protocols like AODV and DSR. In this paper, we exploit the benefit of multi-path routing in multi-channel WMNs from the aspect of end-to-end throughput. Between medium access control and network layers, we propose a novel protocol named *Joint Multi-channel and Multi-path control* (JMM) which combines multi-channel link layer with multi-path routing. Dividing time into slots, JMM coordinates channel usage among slots and schedules traffic flows on dual paths. Our scheme efficiently and intelligently decomposes contending traffics over different channels, different time, and different paths, and hence leads to significant throughput improvement. To the best of our knowledge, this is the first work discussing the joint design of multi-channel control and multi-path routing for WMNs.**

I. Introduction

Wireless Mesh Networks (WMNs) are believed to be a promising technology to offer broadband wireless access to the Internet and to build self-organized networks in places where wired infrastructure is not available or not worthy to deploy [3]. A WMN consists of a collection of wireless *mesh routers*, which are able to self-configure themselves as a backbone and also serve as an access network to offer connectivity to end-users by standard radio interfaces like 802.11 [1]. A WMN typically has a *two-tier architecture* as shown in Fig. 1. On one hand, mesh routers self-organize themselves to form a wireless backbone, providing large coverage, connectivity, and robustness in the wireless domain. On the other hand, each mesh router is responsible of forwarding traffic on behalf of end-users in its coverage area. A logical separation is maintained between links connecting end-users and links forming the wireless backbone. One or more mesh routers with wired connections will serve as *gateways* to provide Internet access.

While benefiting from large coverage of multihop wireless connections, WMNs also inherit some scalability problems in terms of throughput, delay, and packet delivery ratio faced by all multihop wireless networks [9]. Previous studies have shown that end-to-end throughput of a flow may decrease rapidly as the number of hops increases [14], [25]. The main reasons are as follows:

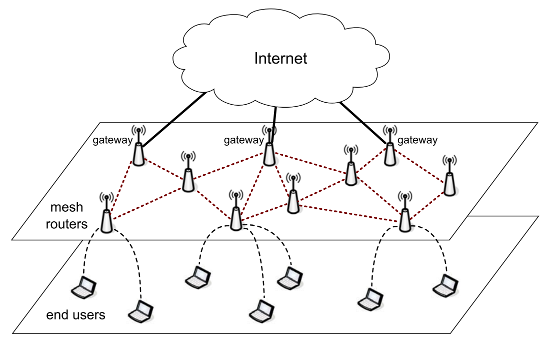


Fig. 1. The two-tier architecture of wireless mesh networks.

- Half-duplex property of the radios: Radios cannot transmit and receive at the same time. As a result, the capacity of relay nodes is halved.
- Broadcast nature of the wireless medium: When all nodes operate at a common communication channel, each node has to compete with neighboring nodes within extended hops, leading to a high collision probability as the traffic load increases.
- Difficulty of collision avoidance: In a multihop environment, the common phenomena of hidden and exposed terminals cause collision and unfairness, resulting in reduction of throughput.

There are several approaches to relieving the contention and collision problem, such as using directional antennas, implementing transmission power control [24], assisting by location information [20], and employing multiple channels. In this paper, we look for a more cost-effective solution by exploiting multiple non-overlapping channels using only one transceiver per host. While our goal is to improve network performance, we observe that using multiple channels alone is not very effective. Frequency diversity has to be exploited in concert with spatial and temporal reuse. We propose a protocol named *Joint Multi-channel and Multi-path control* (JMM), which can yield a significant performance improvement by decomposing the contending traffic over different channels, different time, and different paths.

The primary contributions of this paper can be summarized as follows:

- We first point out that multi-path routing has to be used in concert with multi-channel design to improve end-to-end throughput. However, using single-path routing cannot achieve this goal.
- We introduce a novel protocol which combines multi-channel link layer with multi-path routing. This protocol is able to increase end-to-end throughput by decomposing the traffic over different channels, time, and space.
- In the route discovery phase of our multi-path routing protocol, we propose a GREQ forwarding strategy to reduce the number of broadcast messages. A new routing metric which explicitly accounts for the disjointness between paths and interference among links is proposed. According to this metric, it is easy to select two maximally disjoint paths with less interference.

In Sec. II, we compare single-path routing with multi-path routing in both single-channel and multi-channel environments to motivate our work. Sec. III reviews related work. The proposed JMM protocol is introduced in Sec. IV. Sec. V presents our simulation results. Finally, Sec. VI concludes the paper.

II. Motivation

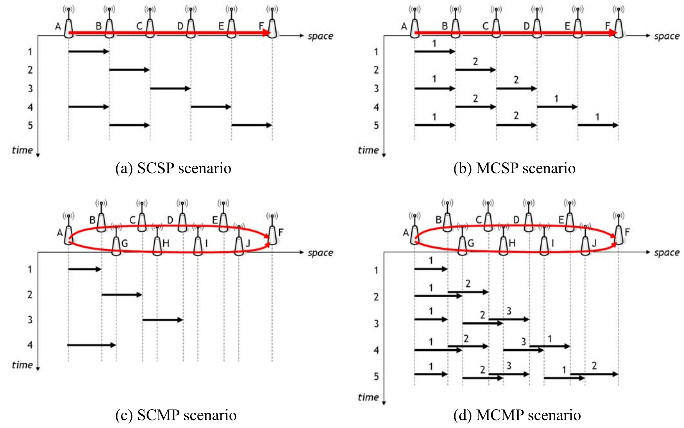
To motivate the problem, we first observe the upper bounds of end-to-end throughputs under (i) single-channel, single-path, (ii) multi-channel, single-path, (iii) single-channel, multi-path, and (iv) multi-channel, multi-path scenarios. We then show that case (iv) can achieve better performance.

A. Single-Channel, Single-Path (SCSP) Scenario

The most common combination is to use a single-channel MAC protocol like IEEE 802.11 with a single-path routing protocol like AODV (Ad-hoc On demand Distance Vector) [17]. In this case, packets travel along a chain of nodes toward their destinations. Successive packets on a single chain may interfere with each other as they move along, thus causing contention in the MAC layer.

In the SCSP scenario, we show that an ideal protocol could only achieve an end-to-end throughput at most $\frac{1}{3}$ of the effective MAC layer data rate. Consider the network in Fig. 2(a), where node A is the source and F is the sink. Assume for the moment that radios of nodes that are not neighbors do not interfere with each other. At time 1, A transmits the first packet to B. At time 2, A and B cannot transmit at the same time because B cannot receive and transmit simultaneously. At time 3, A and C cannot transmit at the same time because B cannot correctly hear A while C is sending. At time 4, A and D can send at the same time with the above assumption. Thus, a node can only send $\frac{1}{3}$ of the time.

However, if one assumes that radios can interfere with each other beyond the range at which they can communicate successfully, the situation is even worse. For example, in 802.11b, the interference range is about twice that of transmission range. Hence, in Fig. 2(a), node D's transmission will interfere with that from A to B. This may reduce a node's transmission



(a) SCSP scenario (b) MCSP scenario (c) SCMP scenario (d) MCMP scenario

Fig. 2. Ideal packet scheduling in (a) SCSP, (b) MCSP, (c) SCMP, and (d) MCMP scenarios.

opportunity to $\frac{1}{4}$ of the time. As can be seen, the throughput is even more pessimistic.

B. Multi-Channel, Single-Path (MCSP) Scenario

The above analysis shows the impact due to the broadcast nature of wireless medium. To improve the end-to-end throughput, a lot of researchers have proposed multi-channel solutions. Allowing each transceiver to switch among different channels, instead of waiting in the same channel, the MAC protocol has to deal with channel selection and the multi-channel hidden terminal problems [19].

In the MCSP scenario, we show that an ideal multi-channel MAC protocol could achieve end-to-end throughput as high as $\frac{1}{2}$ of the effective MAC data rate. Consider the scenario in Fig. 2(b). Assume that the MAC protocol can always select an appropriate channel and schedule packets perfectly. At time 1, node A transmits the first packet to B on channel 1. At time 2, A and B cannot transmit at the same time because B cannot receive and transmit simultaneously. At time 3, A and C can send at the same time since they use different channels. We can see that if the MAC protocol can switch channels perfectly, A can continuously inject one packet every other slot. This leads to the factor of $\frac{1}{2}$. Because of the half-duplex property of radios, the bottleneck appears in the intermediate nodes.

C. Single-Channel, Multi-Path (SCMP) Scenario

In this SCMP scenario, packets are split along two disjoint paths leading toward destinations. We will show that the broadcast nature of wireless medium may degrade throughput significantly.

In fact, the SCMP scenario can only achieve an end-to-end throughput slightly higher than the SCSP scenario. Consider the network in Fig. 2(c), where there are two disjoint paths from source A to destination F. At time 1, node A transmits the first packet along the upper path to B. At time 2, only one of nodes A and B can transmit because they are competitors. We suppose that B wins in the contention. At time 3, A can not transmit on the lower path because C will interfere the reception of G. So A can only transmit on the lower path at time 4. So A can only inject a packet every three slots.

D. Multi-Channel, Multi-Path (MCMP) Scenario

Some may believe that the factor of $\frac{1}{2}$ is the best case. Below, we show that using a multi-channel MAC protocol combined with a multi-path routing protocol can overcome the bottleneck at intermediate nodes. In the MCMP scenario, we show that the ideal MAC end-to-end throughput can be as high as the effective MAC data rate. Consider the network in Fig. 2(d). Assume that the routing protocol can split packets properly and the MAC protocol can perform ideal channel switching and scheduling. At time 1, node A transmits a packet along the upper path to B on channel 1. At time 2, A transmits a packet along the lower path to G. At the same time B can transmit along the upper path because they use different channels. Afterward, A can alternate between these two paths in every slot. This concludes our derivation.

III. Related Work

In the literature, a lot of efforts have been dedicated to multi-channel link protocols and multi-path routing protocols. However, these link layer protocols and routing protocols are investigated separately. This motivates us to design a joint protocol which combines these two approaches. Below, we review the related work in this field.

A. Multi-Channel MAC and Link Protocols

A lot of multi-channel link/MAC protocols focus on how to utilize multiple channels to reduce the contention and collision among stations. Depending on the number of radio interfaces per node, such protocols can be classified as *single-transceiver* schemes [19], [4], [26], [5] and *multi-transceiver* schemes [23], [28], [2].

For a single-transceiver system, the radio interface in each node needs to switch among channels. It may result in the *multi-channel hidden-terminal problem* [19]. The Multi-channel MAC (MMAC) protocol [19] proposes to embed a negotiation phase in the ATIM (Ad Hoc Traffic Indication Map) window that is periodically sent under the Power Save Mode (PSM). After the ATIM window, nodes may select different channels to transmit and receive packets. The Slotted Seeded Channel Hopping (SSCH) mechanism [4] divides the time axis into virtual channels. The hopping sequence of each virtual channel is determined by a (channel, seed) pair. SSCH requires a looser time synchronization than [19], but it has a higher channel switching overhead. The Multi-channel coordinated Temporal Topology control (MOTTO) [26] also divides the time axis into epochs. The active channel of an epoch is determined statically by the node's hop-count to a gateway and its direction (uplink or downlink).

B. Multi-Channel Routing Protocols

Several works consider utilizing multiple channels at the network layer [18], [10], [8], [12]. These works focus on how to assign channels to a flow and how to find the best path in a multi-channel environment.

The Hyacinth architecture [18] proposes a tree-based routing protocol for a multi-transceiver multi-channel WMN. From each gateway, a tree is constructed, along which packets are forwarded. Reference [10] proposes a CA-AODV protocol that combines channel assignment with AODV [17]. It assumes a system with one control channel and several data channels like DCA [23]. From the exchange of RREQ and RREP packets on the control channel, a source can achieve both route discovery and channel assignment of the flow.

A general multi-channel routing protocol can be designed by combining an existing single-channel routing protocol with a new routing metric by taking multi-channel effects into consideration. The WCETT (Weighted Cumulative Expected Transmission Time) metric in [8] is such a metric for routing in multi-radio multi-hop WMNs.

C. Multi-Path Routing Protocols

Recently, multi-path routing in WMNs has received some attention [16], [21], [13], [27], [15].

Based on Directed Acyclic Graph (DAG), TORA [16] can support multiple-path routing. However, it does not guarantee disjointness of paths. DSR [11] can also find multiple paths, naturally by its flooding behavior. But sometimes only small portions of the found paths are disjoint. The Split Multipath Routing (SMR) [13] can solve this problem because duplicate RREQs are not dropped, but this is at the cost or more RREQs.

AODVM [27] is an extension to AODV for finding reliable routing paths. Duplicate RREQs are not discarded by intermediate nodes. Again, the routing overhead is high. AOMDV [15] is also an extension to AODV for computing multiple loop-free and link-disjoint paths. It uses the notion of "advertised hop count" to guarantee loop-freedom and uses a particular property of flooding to achieve link-disjointness.

IV. JMM Protocol

A. Protocol Architecture

We assume that each node is equipped with an off-the-shelf 802.11 wireless adapter with a half-duplex radio which is allowed to switch among different channels and runs the 802.11 MAC protocol. The proposed JMM protocol is a cross-layer design on top of the 802.11 MAC layer and does not require any change to the 802.11 MAC and hardware. It is composed of a *multi-channel link layer part* and a *multi-path routing part*. These two parts cooperate with each other tightly. JMM has the following functionalities.

1) It decides the receiving channel of each node based on neighborhood information (see Sec. IV-C.1).
2) It constructs a dual path from each node to its gateway (see Sec. IV-C.2).
3) It conducts slot assignment for each node's superframes (see Sec. IV-B.2 and IV-C.4).
4) It schedules and forwards packets and adjusts the ratio of transmitting slots to receiving slots for each node (see Sec. IV-B.3).

Our presentation is bottom-up, from the link layer part (items 3 and 4) toward the routing part (items 1 and 2).

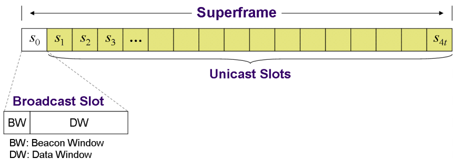


Fig. 3. The superframe structure.

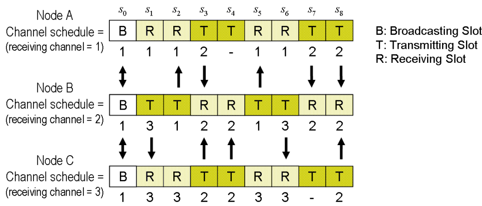


Fig. 4. An example of channel schedule.

B. Multi-Channel Link Layer Part

The link layer has two functionalities: *channel scheduling* and *packet scheduling*. The former is to control which channel the transceiver should stay on, and the latter is to schedule when a packet can be sent. Our design can avoid the *multi-channel hidden-terminal problem* [19].

1) Superframe Structure: The time axis is divided into slots of a fixed length l. Slots are organized into *superframes*. A slot may be designated as a transmitting slot or a receiving slot. We will determine the channels to be used in slots of a superframe. Our channel assignment strategy is *receiver-based*. The structure of a superframe is shown in Fig. 3. Superframes are loosely synchronized in time. Each superframe comprises $4t + 1$ slots, marked as s_0, s_1, ..., s_{4t}, where t is an integer. Slot s_0 is a *broadcast slot* in which only beacons and broadcast messages can be sent. Each broadcast slot is led by a *beacon window*, followed by a *data window*. Beacons also serve to synchronize stations' clocks. To ensure network connectivity, all nodes should stay on a pre-defined common channel in slot s_0. The remaining $4t$ slots are *unicast slots*, whose channels will be decided dynamically.

The receiver-based channel assignment helps two nodes to switch to the same channel for communication. Unicast slots are designnated as transmitting/receiving slots (refer to Sec. IV-B.2). A node will select a *receiving channel* for its receiving slots (refer to Sec. IV-C.1). Nearby nodes will try to avoid using the same receiving channel. During a receiving slot, a node will stay on its receiving channel. During a transmitting slot, a node can switch to its receiver's receiving channel and stay on that channel until the end of the slot. Hence, two nodes can communicate only if one is in a transmitting slot and the other is in a receiving slot. After switching to a new channel, a node first remains silent for a duration equals to the maximum packet transmission time so as to avoid the multi-channel hidden terminal problem which is resulted by loose time synchronization. Therefore, JMM does not require very precise clock synchronization.

An example is in Fig. 4. In s_0, all nodes stay on the common channel 1. In s_1, node B wants to send packets to C, so B switches to C's receiving channel 3. Suppose that A also wants to send packets to B. Since s_1 and s_2 of A are receiving slots, it has to wait until s_3 to transmit. Note that since both A and C want to send packets to B in s_3, they will use 802.11's CSMA/CA mechanism to contend for the medium.

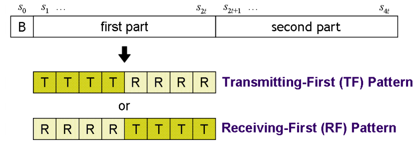


Fig. 5. The TF and RF patterns.

2) Transmitting and Receiving Patterns: Unicast slots of a superframe are designated as transmitting/receiving slots. However, since traffics on mesh networks are quite stable, slot assignment will not be changed too frequently. In each superframe, unicast slots s_1 to s_{4t} are evenly divided into two parts, with the fist part from s_1 to s_{2t} and the second part from s_{2t+1} to s_{4t}. One part is designated as the *upstream part* for communication with the node's upstream nodes (with respect to the node's gateway), and the other part is the *downstream part* for communication with its downstream nodes. These two parts are of the same length because for a relay node, the amount of traffics to and from upstream nodes is likely to be equal to that to and from downstream nodes.

Each part can follow a *Transmitting-First (TF) pattern* or a *Receiving-First (RF) pattern* as shown in Fig. 5. In a TF pattern, the first half is all transmitting slots, and the second half is all receiving slots. Contrarily, in a RF pattern, the first half is all receiving slots, and the second half is all transmitting slots. Considering the patterns of the first and the second parts, there are four types of superframe patterns, namely TF-TF, RF-RF, TF-RF, and RF-TF types. The ratio of the number of transmitting slots to the number of receiving slots can be adjusted dynamically (refer to Sec. IV-B.3).

3) Dynamic Adjustment of the T/R Ratio: Recall that each superframe has an upstream part and a downstream part. The ratio of the number T of transmitting slots to the number R of receiving slots in each upstream part, call T/R ratio, can be dynamically adjusted in a per node basis. Since in a relay node the amount of traffics from upstream nodes is likely to be equal to that to downstream nodes, the number of receiving slots in the upstream part should equal the number of transmitting slots in the downstream part. Similarly, the transmitting slots in the upstream part should equal the number of receiving slots in the downstream part. Therefore, in the downstream part, we can let T be the number of receiving slots and R be the number of transmitting slots.

The T/R ratio of each node is adjusted dynamically during runtime. Initially, we set $T = R = t$. A node should monitor the actual traffic through itself. Assume that the actual transmitting and receiving traffics on the upstream part are T_{actual} and R_{actual}, respectively. We then compute new weighted averages T_{smooth} and R_{smooth} as follows:

$$T_{smooth} \leftarrow \alpha * T_{actual} + (1 - \alpha) * T_{smooth}; \qquad (1)$$

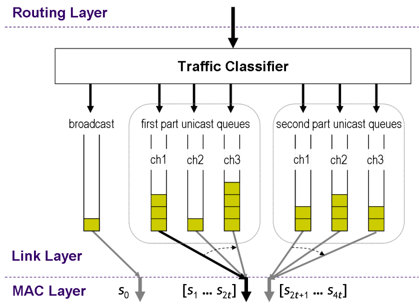


Fig. 6. The broadcast and unicast queues at the link layer.

$$R_{smooth} \leftarrow \alpha * R_{actual} + (1 - \alpha) * R_{smooth}. \tag{2}$$

The values of T and R will be changed slowly by the following rules:

```
if (T_smooth/T)/(R_smooth/R) > Threshold_h and R > 1 then
    T ← T+1;
    R ← R−1;
endif

if (T_smooth/T)/(R_smooth/R) < Threshold_l and T > 1 then
    T ← T−1;
    R ← R+1;
endif
```

T_{smooth}/T and R_{smooth}/R are the utilizations of transmitting and receiving slots, respectively. If the utilization ratio of transmitting to receiving slots is higher than a threshold $Threshold_h$, we increase T and decrease R by one. If the utilization ratio is lower than a threshold $Threshold_l$, a reverse process is preformed.

4) Packet Queues: When packets arrive, we need to allocate them to transmitting slots for transmission. JMM dispatches packets into a broadcast queue and two groups of unicast queues as shown in Fig. 6, where we assume that there are three non-overlap channels. Broadcast packets are enqueued in the *broadcast queue*, while unicast packets are classified as the *first part* or the *second part* and then are enqueued in the corresponding queues based on the receiving channels of receivers (refer to Sec. IV-C.5). The number of queues in each part is equal to the number of channels in the system.

The broadcast queue is served in broadcast slots. The first part unicast queues are served by transmitting slots of slots s_1 to s_{2t} in a round-robin manner. Each transmitting slot will serve one queue by switching to the channel of that queue, until the queue is empty or the slot expires. The second part unicast queues are served by transmitting slots of slots s_{2t+1} to s_{4t} in a similar way.

5) Permutation of Slots: In the above discussion, transmitting and receiving slots are clustered together. In practice, we can permute the slot sequence of a superframe to obtain some degree of randomness among these slots. The same permutation of $4t$ elements should be applied to all nodes.

C. Multi-Path Routing Part

The goal of the routing part is to construct two paths to the gateway. Since finding the best two paths requires the channel

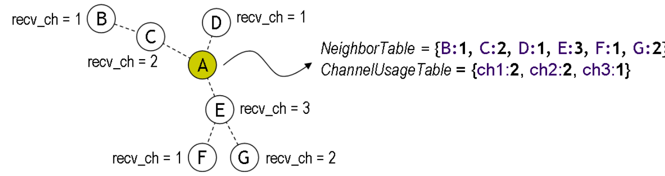


Fig. 7. A channel selection example.

information provided by the link layer part, we first describe how a node selects its receiving channel. We then present the multi-path route discovery phase and our path selection metric. Finally, we describe our packet scheduling scheme to exploit the benefit of multi-path routing. In JMM, route discovery is done in an on-demand manner. However, the selection of receiving channel will be changed less frequently.

1) Selection of Receiving Channels: When a node is first turned on, it can choose any channel as its receiving channel. Periodically, each node broadcasts its receiving channel to its 2-hop neighbors. This can be achieved by broadcasting a HELLO message carrying a node's direct neighbors' receiving channels. Each node maintains a *NeighborTable* containing the receiving channels of its 2-hop neighbors and a *ChannelUsageTable* to count the number of nodes using each channel. For example, Fig. 7 shows these tables of node A.

A node will choose the least used channel as its receiving channel. To prevent unnecessary fluctuation, when a node finds a better channel than its current receiving channel, it will only switch to that channel with a probability p.

2) Dual-Path Route Discovery: Our goal is to find from each node two paths to its gateway that are as disjoint as possible. However, a dilemma is: on one hand, we would like to avoid network-wide flooding of route search packets, while on the other hand, we do not expect too many duplicate route search packets being discarded by intermediate nodes.

Below, we propose an efficient discovery strategy to find a dual-path to each gateway in the network. A Gateway REQuest (GREQ) packet is used for this purpose. Instead of blindly flooding, limited rebroadcasts of GREQs are invoked. The format of GREQ is shown in Table I. The route discovery is performed in an incremental way. So when a node issues a GREQ, we can assume that each existing node has already established two paths to its gateway. For each node, let $gwAddr$ be its selected gateway and $hopCount$ be the length of the shorter path of its dual-path. When an intermediate node R receives a GREQ, the procedure in Fig. 8 is executed. It first checks whether the sequence number is up-to-date (lines 2-6). Then it verifies if its slot schedule mismatches with that of the transmitter (lines 7-9). Note that a "mismatch" happens when the superframe patterns of two neighboring nodes are the same (i.e., they choose the same type from TF-TF, RF-RF, TF-RF, and RF-TF), in which case these two nodes cannot communicate with each other. The $gwAddr$ and $hopCount$ fields guarantee that the GREQ packet is forwarded toward the gateway indicated in the $gwAddr$ flied and the $hopCount$ value progressively decreases on its way to the gateway (lines

TABLE I

STRUCTURE OF THE GREQ MESSAGE (S IS THE SOURCE NODE).

Field	Initial value	Meanings
$seqNum$	seqNum at S	the sequence number
$srcAddr$	S	the source address
$gwAddr$	unknown	the gateway address of the mesh network
$hopCount$	∞	the smallest number of hops to the gateway
$pathRecord$	{S}	the list of node records on the path

```
/*Executed when a non-gateway node R receives a GREQ from a node T */
01. begin
02.     if GREQ.seqNum < R.seqNum[srcAddr] then
03.         discard and exit;
04.     else
05.         R.seqNum[srcAddr] ← GREQ.seqNum;
06.     endif
07.     if the slot schedules of R and T mismatch then
08.         discard and exit;
09.     endif
10.     if GREQ.gwAddr ≠ unknown and
          GREQ.gwAddr ≠ R.gwAddr then
11.         discard and exit;
12.     endif
        /* Ensure that hopCount progressively decreases */
13.     if GREQ.hopCount < R.hopCount then
14.         discard and exit;
15.     elseif GREQ.hopCount = R.hopCount then
16.         if R ∈ GREQ.pathRecord then
17.             discard and exit;
18.         endif
19.     endif
20.     send GREQ(GREQ.seqNum, GREQ.srcAddr, R.gwAddr,
                  R.hopCount, GREQ.pathRecord ∪ {R});
21. end
```

Fig. 8. The GREQ propagation procedure of a non-gateway node.

10-19). This forwarding strategy can significantly reduce the rebroadcast overhead while traversing most wireless links. Finally, the node rebroadcasts the GREQ packets (line 20).

An example of the GREQ propagation procedure is shown in Fig. 9. The links indicated by dashed lines mean that the corresponding GREQs are discarded.

3) Path Selection Metric: After the above procedure, each gateway will collect a number of GREQs each carrying a path. Since our goal is to find a dual-path, the gateway will use a metric function to evaluate each pair of paths. For example, the gateway X in Fig. 9 will collect $n = 4$ paths, S-C-A-X, S-D-B-X, S-C-B-X, and S-D-A-X, from the route discovery initiated by S. So there are totally $\binom{n+2-1}{2} = 10$ path pairs

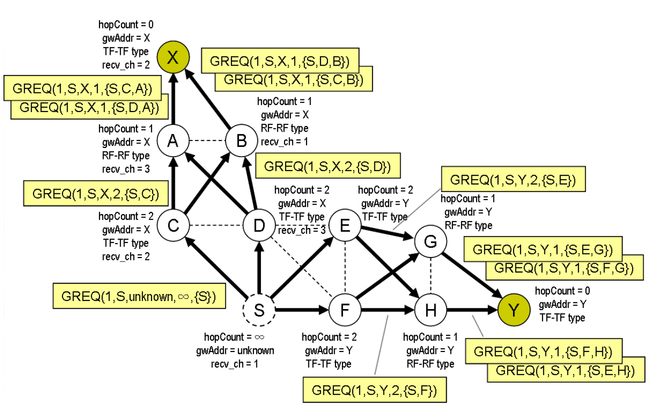


Fig. 9. A GREQ propagation example.

to be evaluated. Note that the combination with repetition is used because a path may serve as both paths of a dual-path in case that there is no good choice. The path pair with the lowest metric will be selected and two Gateway REPly (GREP) packets are unicast along the reverse directions to the source node. Then the source node will collect all GREP packets from different gateways and select the dual-path with the best path metric by sending two GREP ACKnowledgement (GREP_ACK) packets to the selected gateway along the dual-path.

The input of the path metric function is a path pair (P_1, P_2) and the output is affected by 3 factors V_{node}, V_{chl}, and V_{qlty}. V_{node} is the number of common nodes between P_1 and P_2 excluding the source node and the gateway. V_{chl} is defined as

$$V_{chl} = CN(P_1) + CN(P_2) + \delta(P_1, P_2), \tag{3}$$

where $CN(P_i)$ is the number of *channel contending pairs* along P_i, where two nodes on P_i are called a channel contending pair if they are within 2 hops and use the same receiving channel. For example, CN(S-C-B-X) = 2 because (S, A) and (C, X) are channel-contending pairs. Function $\delta(P_1, P_2) = 1$ if the difference of the lengths $|P_1|$ and $|P_2|$ is an odd number; otherwise, $\delta(P_1, P_2) = 0$. The value is so assigned because our algorithm prefers paths differ in lengths by an even number (refer to the discussion in Sec. IV-C.4). To reflect the signal quality perceived by nodes on P_1 and P_2, V_{qlty} is defined as $ETX(P_1) + ETX(P_2)$, where $ETX(P_i)$ is the expected transmission count of a packet along P_i [6]. Alternatively, other metrics for evaluating path quality [7], [8] can be used instead. We combine the three factors by taking their weighted average:

$$metric = w_{node}V_{node} + w_{chl}V_{chl} + w_{qlty}V_{qlty}, \tag{4}$$

where $w_{node} + w_{chl} + w_{qlty} = 1$. The one with a lower metric is preferred.

4) Determining Superframe Patterns: Next, we need to determine the superframe pattern (TF-TF, RF-RF, TF-RF, or RF-TF) of each node. The selection will be based on the result of the route discovery. We assume that all nodes on the dual-path except the source have already determined their superframe patterns. Without loss of generality, let the gateway choose the TF-TF type. Given any dual-path, the gateway will designates one path as the *master path*, and the other as the *slave path*. The requirement to be a master path is that the superframe patterns of the gateway and the first child must match in the first part (slots s_1 to s_{2t}), and the requirement to be a slave path is that they must match in the second part (slots s_{2t+1} to s_{4t}). A "match" happens if one side uses TF and the other side uses RF. Let S be the source, G be the gateway, and (P_1, P_2) be the dual-path, such that P_1 is the master path and P_2 is the slave path. The superframe pattern of S will be selected by the following rules:

1) $|P_1| - |P_2|$ is even: We refer to Fig. 10 for ease of presentation. If $|P_1|$ is odd, the pattern of S's first part should match with that of its parent on P_1 and the pattern

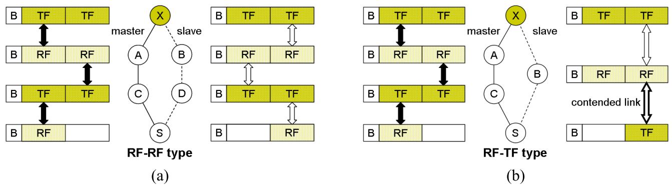


Fig. 10. The pattern selection of S when (a) $|P_1| - |P_2|$ is even and (b) $|P_1| - |P_2|$ is odd.

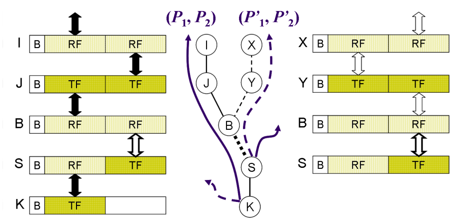


Fig. 11. An example where the contended link B-S on the slave path (..., X, Y, B, S) serves as a link on the master path (..., I, J, B, S, K).

of S's second part should match with that of its parent on P_2. If $|P_1|$ is even, the pattern of S's first part should match with that of its parent on P_2 and the pattern of S's second part should match with that of its parent on P_1. Hence, S chooses the RF-RF type in Fig. 10(a). This pattern selection can achieve high channel utilization. Packets on the dual paths are unlikely to interference with each other because they are separated in both the time domain and the space domain when they happen to use the same channels.

2) $|P_1| - |P_2|$ is odd: In this case, one of $|P_1|$ and $|P_2|$ is odd and the other is even. Let P be the longer path. If $|P|$ is odd, the pattern of S's first part should match with that of its parent on P_1 and the pattern of S's second part should match with that of its parent on P_2. If $|P|$ is even, the pattern of S's first part should match with that of its parent on P_2 and the pattern of S's second part should match with that of its parent on P_1. Hence, S chooses the RF-TF type in Fig. 10(b).

In case 2, packet transmission on the dual paths are also quite interference-free, except the link between S and its parent on the shorter path of P_1 and P_2, which is called the *contended link*. Because S matches with its parent on the same part as where S's parent matches with S's grandparent on that path. For example, B-S in Fig. 10(b) is a contended link. This competition may affect the end-to-end throughput of that path. So we let this happen on the shorter path. Also, the penalty is reflected by the earlier function $\delta(P_1, P_2)$ in the path metric V_{chl}.

Note that a contended link may play parts in both a master path of a dual path (P_1, P_2) and a slave path of another dual path (P'_1, P'_2). For example, in Fig. 11, B-S is a contended link in (P'_1, P'_2). If later on node S accepts a child K, which chooses the path along S as its master path, then B-S will be part of a master path in (P_1, P_2). However, the patterns of superframes of K's master path are not affected by the appearance of this contended link.

5) Packet Forwarding Rule: With dual-path routing, our system needs to inject packets to both paths to exploit communication parallelism. Below, we summarize our packet forwarding rule. When a source node or a gateway generates a sequence of packets, we will alternately mark them as to be sent along the master path or along the slave path. For each packet, we will compute a value $P = M \oplus E \oplus D \oplus C$, where

$$M = \begin{cases} 0 & \text{if the pkt is to be sent along the master path;} \\ 1 & \text{if the pkt is to be sent along the slave path;} \end{cases} \tag{5}$$

$$E = \begin{cases} 0 & \text{if the hop count to gw along the intended path is even;} \\ 1 & \text{if the hop count to gw along the intended path is odd;} \end{cases} \tag{6}$$

$$D = \begin{cases} 0 & \text{if the pkt is issued by a gateway;} \\ 1 & \text{if the pkt is issued by a source;} \end{cases} \tag{7}$$

$$C = \begin{cases} 0 & \text{if the pkt is to be transmitted to a non-contended link;} \\ 1 & \text{if the pkt is to be transmitted to a contended link.} \end{cases} \tag{8}$$

If $P = 0$, the packet will be forwarded to the first part unicast queues; otherwise, the packet will be sent to the second part unicast queues (refer to Fig. 6).

For a relayed packet, it is alternated between the first and the second parts except when it passes through a contended link. Specifically, if a packet is received from a contended link, it is enqueued to the same part of unicast queues; otherwise, it is enqueued to a different part of unicast queues from its original one. For example, in Fig. 10(b), when node B receives a packet from S in the second part, it enqueues the packet to the same second part, but when C receives a packet from S, it enqueues the packet to the different part.

6) Route Maintenance: Faulty links are detected by nodes' periodical HELLO messages. Losing a predefined number of HELLOs is an indication of a fault of link. When a node discovers a faulty link, it will propagate a Gateway ERRor (GERR) message to all its successors which use this link. Each successor will initiate a new gateway discovery procedure to find a new dual-path. Before new paths are found, the other (non-broken) paths can still be used for communication. Therefore, JMM is also quite resilient to failure.

V. Performance Evaluation

To evaluate the performance of the proposed protocol, we have implemented a JMM module in the NCTUns network simulator 2.0 [22]. The JMM module has a link layer, a routing layer, and some FIFO queues. The MAC layer is the IEEE 802.11a without using RTS/CTS. Data rate is 54 Mbps. Each node has a transmission range of 250 meters and an interference range of 550 meters. The default parameters used are shown in Table II.

A. Comparison of SCSP, SCMP, MCSP, and MCMP Routing

We first compare the performance of SCSP, SCMP, MCSP and MCMP routing. Network topologies as shown in Fig. 12 are tested. Assuming H = 200 meters, V = 300 meters, and five available channels, we vary the number of hops from the gateway to the destination and observe the end-to-end throughput. Continuous 512-byte packets are injected from the

TABLE II

THE DEFAULT PARAMETERS IN OUR SIMULATIONS.

Parameter	Default	Meanings
l	20 ms	the slot size
t	4	the number of slots in a quarter of a superfame
α	0.2	the weight between actual and smooth traffic
$Threshold_h$	2	the high threshold of adjusting the T/R ratio
$Threshold_l$	0.5	the low threshold of adjusting the T/R ratio
w_{node}	0.74	the weight of parameter V_{node}
w_{chl}	0.18	the weight of parameter V_{chl}
w_{qlty}	0.08	the weight of parameter V_{qlty}

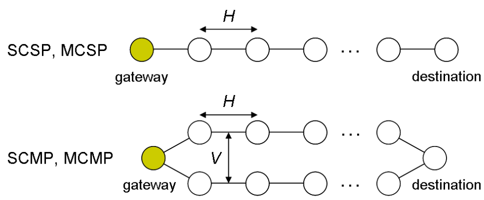
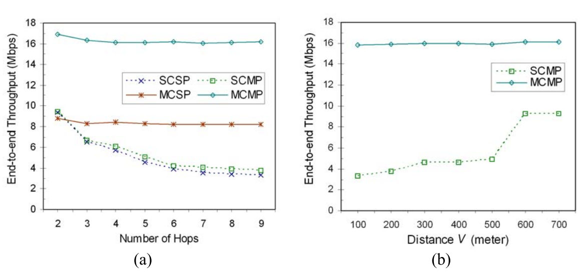


Fig. 12. Single-path and dual-path topologies used in our simulation.

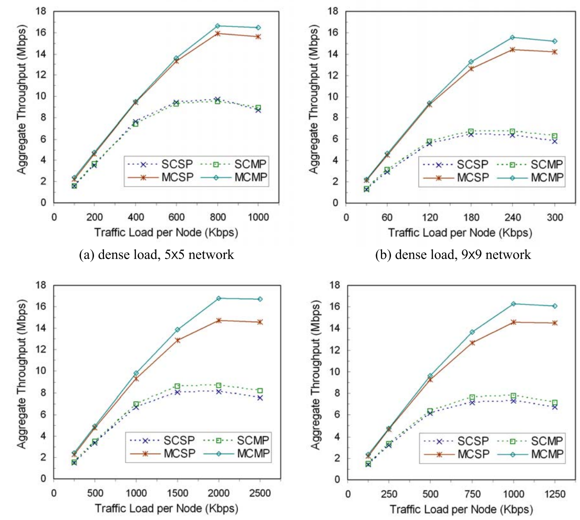
gateway to the destination. SCSP routing uses IEEE 802.11 MAC and AODV. SCMP routing uses the multi-path routing protocol AODVM. MCMP routing uses our JMM protocol. MCSP routing also uses our JMM protocol but it only employs a single path routing. The results are shown in Fig. 13(a). As can be observed, in SCSP and SCMP routing the end-to-end throughputs decrease dramatically as the number of hops increases. The SCMP routing is only slightly better than the SCSP routing since the two parallel paths still seriously interfere with each other. On the other hand, the throughputs of MCSP and MCMP remain relatively constant since newly added nodes will not interfere with existing nodes. The throughput of MCMP is about twice the throughput of MCSP. This demonstrates the advantage of our superframe structure in avoiding temporal and spatial interferences.

For SCMP and MCMP routing, we further vary the distance V between the two parallel paths. As shown in Fig. 13(b), as V decreases, the average end-to-end throughput of SCMP drops significantly due to higher and higher contention between the two paths. JMM achieves more than three time the throughput of SCMP routing as V reduces to below 400 meters. The throughput of JMM is quite insensitive to the value of V, which demonstrates the advantage of our JMM protocol in distributing packets to two parallel paths on which the transmissions are well interleaved.



(a) (b)

Fig. 13. (a) Average end-to-end throughput vs. number of hops, and (b) average end-to-end throughput vs. distance V (path length = 6)



(a) dense load, 5x5 network (b) dense load, 9x9 network

(c) 10 traffic-generating nodes, 9x9 network (d) 20 traffic-generating nodes, 9x9 network

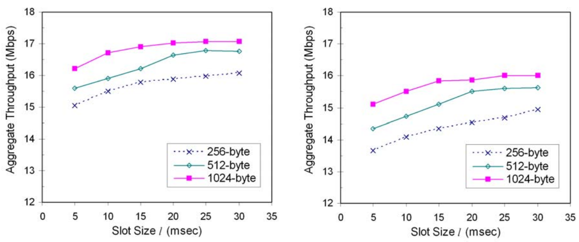
Fig. 14. Aggregate gateway throughput vs. traffic load under different numbers of traffic sources.

B. Impact of Traffic Load

To study JMM's performance for different traffic loads, we simulate a stationary 5×5 and 9×9 grid networks with only one gateway located in the center of the grid. Neighboring nodes are uniformly separated by 200 meters. Two different traffic loads are simulated: a dense load where each node in the grid generates even CBR (Constant Bit Rate) traffics towards the gateway, and a sparse load where only a few random-chosen nodes generate traffic. In both simulations, we gradually increase the traffic load of each flow and measure the gateway's throughputs, as shown in Figure 14. JMM outperforms SCSP and SCMP by over 100%, and outperforms MCSP by 10-20% depending on the traffic load. The amount of improvement is less significant in the dense load case. Because every node is transmitting and thus it is hard to see the advantage of multi-path routing. Our saturated throughput is close to the upper bound 19.5 Mbps (the maximum throughput between only two nodes after considering all MAC and PHY overheads). Note that this also includes JMM's overheads of broadcast slots and channel switching latency.

C. Impact of Slot Size l on JMM Protocol

Above simulations have fixed the slot size l to 20 msec. The length of l can influence the performance of JMM. Longer l may result in increased end-to-end delay as well as the buffer requirement at each node. On the other hand, if the length of l is too short, the channel switching overhead becomes considerable and degrades the system performance. To study this impact, aggregate throughput is measured using different l under 5×5 and 9×9 grid networks as shown in Fig. 15. In packet sizes of 256, 512, and 1024 bytes, we see consistent higher network throughputs as l increases from 5 to 30 msec, due to less channel switching overhead. However, this is at the cost of higher end-to-end delays. We recommend l = 20 msec from our experience.



(a) 5x5 grid network (b) 9x9 grid network

Fig. 15. Aggregate network throughput vs. slot size l.

VI. Conclusions

We have shown that multi-path routing, when being harmonized with multi-channel capability, has great potential to achieve good performance for WMNs. We then design the JMM protocol which combines multi-channel link layer and multi-path routing to offer this benefit. Dividing the time into slots, JMM coordinates channel usage among slots using a receiver-based channel assignment and schedules transmissions along dual paths. In the route discovery phase of JMM, we propose a GREQ forwarding strategy to reduce broadcast overhead. In addition, we define a new routing metric which explicitly accounts for the disjointness between paths and interference among links. According to this metric, it is easy to select two maximally disjoint paths with less interference. Our simulation results show that JMM yields a significant end-to-end throughput improvement in WMNs as compared to single-channel scenarios. In summary, JMM efficiently increases the performance by decomposing contending traffic over different channels, different time, and different paths.

Acknowledgment

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附錄四：

Exploiting Spectral Reuse in Resource Allocation, Scheduling, and Routing for IEEE 802.16 Mesh Networks

L.-W. Chen, Y.-C. Tseng, D.-W. Wang, and J.-J. Wu, "Exploiting Spectral Reuse in Resource Allocation, Scheduling, and Routing for IEEE 802.16 Mesh Networks", IEEE VTC, 2007-Fall

Exploiting Spectral Reuse in Resource Allocation, Scheduling, and Routing for IEEE 802.16 Mesh Networks

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***Abstract*—The IEEE 802.16 standard for wireless metropolitan area networks (WMAN) has been created to meet the need of wide-range broadband wireless access at low cost. The objective of this paper is to study how to exploit spectral reuse in an IEEE 802.16 mesh network through timeslot allocation, bandwidth adaptation, hierarchical scheduling, and routing. To the best of our knowledge, this is the first work which formally quantifies spectral reuse in IEEE 802.16 mesh networks and which exploits spectral efficiency under an integrated framework. Simulation results show that the proposed spectral reuse scheduling and load-aware routing significantly enhance the network throughput performance in IEEE 802.16 mesh networks.**

Keywords: IEEE 802.16, WiMax, Mesh Network, Resource Allocation, Routing, Wireless Network.

I. Introduction

The IEEE 802.16 standard for wireless metropolitan area networks (WMAN) is designed for wide-range broadband wireless access at low cost. It is based on a common medium access control (MAC) protocol with different physical layer specifications. The PHY layer can employ the orthogonal frequency division multiplexing (OFDM) below 11GHz or the single carrier (SC) scheme between 10GHz and 66GHz.

The MAC layer of IEEE 802.16 [4] can support the point-to-multipoint (PMP) mode and the mesh mode. In the PMP mode, subscriber stations (SSs) are directly connected to base stations (BSs). So all SSs associated to a BS must be within the transmission range of the BS. On the other hand, in the mesh mode, each SS can act as an end point or a router to relay traffics for its neighbors. So there is no need to have a direct link from each SS to its associated BS, and SSs may transmit at higher rates to their parent SSs/BS. Also, a BS can serve wider network coverage with lower deployment cost and higher robustness and flexibility [3]. However, intelligent routing and scheduling protocols are needed to fully exploit such benefits. For IEEE 802.16 mesh networks, efforts have been dedicated to topology design [10], packet scheduling [8], and QoS support [1].

This paper studies the spectral reuse issue in an IEEE 802.16 mesh network through multi-hop routing and scheduling. The proposed framework includes a load-aware routing algorithm and a centralized two-level scheduling scheme, which consider both traffic demands and interference among SSs. Given traffic patterns of SSs, we show how to achieve better spatial reuse and thus higher spectral efficiency. Table I compares our work against previous works. Reference [2] proposes an interference-aware route construction and a scheduling algorithms. However, the algorithm does not fully exploit spectral reuse and it is not load-aware (in the sense that the routing tree is a fixed one). How to attach a new SS to a mesh tree is discussed in [6], but scheduling is not addressed in that work. As pointed out in [5], the network performance highly depends on the order that SSs join the routing tree. Although [5] has taken routing tree reconstruction into account, the traffic demands of SSs are still not considered. Thus, the real traffic bottleneck of the network is not reflected. Compared to existing works, our work is most complete in exploiting spectral reuse in IEEE 802.16 mesh networks in the sense that it takes dynamic traffic loads of SSs into account and integrates not only a hierarchical bandwidth scheduling scheme for bandwidth adaptation and timeslot allocation, but also a routing algorithm with a tree optimization scheme.

TABLE I
Comparison of existing schemes and our results

	Scheduling		Routing	
Features	Reuse Quantification	Slot Assignment	Route Reconstruction	Load Awareness
Wei et al. [2]	N/A	Yes	N/A	N/A
Tao et al. [5]	N/A	Yes	Yes	N/A
Fu et al. [6]	N/A	N/A	N/A	N/A
Our work	Yes	Yes	Yes	Yes

The rest of the paper is organized as follows. Section II briefly reviews the IEEE 802.16 mesh mode and then formally defines our problem. Section III develops our resource allocation and scheduling framework, followed by our routing and tree construction algorithms. Performance evaluation is given in Section IV. Finally, Section V concludes this paper.

II. Backgrounds and Problem Definition

In an IEEE 802.16 mesh network, transmission schedules of SSs can be determined in a distributed manner by individual SSs, or in a centralized manner by the BS. In this work,

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to better exploit spectral reuse, we will focus on centralized scheduling, which is also most commonly used in the standard for Internet access.

In centralized scheduling, there are two control messages, MSH-CSCF (Mesh Centralized Scheduling Configuration) and MSH-CSCH (Mesh Centralized Scheduling). The BS can specify the current routing tree by using the last MSH-CSCF message and modify the tree by the last MSH-CSCH update. The BS will broadcast MSH-CSCF to all its neighbors, and all the BS neighbors rebroadcast this message to all their neighbors until all SSs have received the MSH-CSCF message. As a result, all SSs maintain a routing tree whose root is the BS and child nodes are SSs. On the other hand, SSs can transmit MSH-CSCH:Request messages to the BS for their traffic demands, which the transmission order is that the SS with the largest hop count transmits first, and retain the order to join the network for SSs with the same hop count. After collecting requests from all SS, the BS can broadcast its flow assignment for all SSs by the MSH-CSCH:Grant message. Since all SS know the current routing tree, they can determine the actual schedule from these flow assignments by dividing the frame proportionally.

In this work, we consider a mesh network with a gateway BS and a number of SSs for Internet access. For centralized scheduling, given the routing tree, the bandwidth demand requested by each SS, and the uplink and downlink data rates of each SS, a two-level scheduling scheme is designed for the following purposes: (1) dynamically adapt the bandwidths between uplink and downlink subchannels; (2) proportionally allocate frame timeslots among SSs; (3) obtain higher gateway throughput based on the above two manners. On the other hand, for routing tree construction, given the traffic demand generated by each SS and the data rate of each link between SSs, a load-aware routing algorithm is developed for constructing a load-balancing routing tree that can distribute evenly the forwarding data of all SSs and increase concurrent transmissions among SSs so as to get higher timeslot reuse ratio.

III. The Proposed Spectral Reuse Framework

A. System Model

We propose an integrated spectral reuse framework for IEEE 802.16 mesh networks, as illustrated in Fig. 1. There are a routing and a scheduling modules. The routing module collects the channel conditions and bandwidth requests of all SSs from MSH-CSCH:Request messages and computes a routing tree T for the mesh network. Next, the scheduling module conducts resource allocation, which contains *channel-level scheduling* (for bandwidth adaptation between uplink and downlink subchannels) and *link-level scheduling* (for timeslot allocation among SSs). Finally, the BS broadcasts the scheduling information to all SSs via MSH-CSCH:Grant messages. Below, we will focus on uplink traffic scheduling, since downlink traffic scheduling can be obtained similarly.

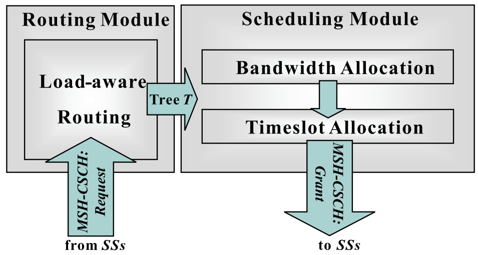

Fig. 1. The system model at BS

B. Resource Allocation and Scheduling Schemes

Below, we assume that the routing tree T is known (refer to Sec. 3-3 for the construction of T). We will derive our resource allocation schemes. Let the uplink data rate and uplink traffic demands of SS i be r_i^u and b_i^u, respectively. From T, we can calculate the aggregated uplink traffic demand $d_i^u = b_i^u + \sum_{j \in child(i)} b_j^u$ for SS i, where $child(i)$ is the set of children of i in T. Thus the demand of transmission time for the uplink of SS i is $T_i^u = d_i^u / r_i^u$. Let $C_{total}^u = \sum_{\forall i} T_i^u$ be the total uplink transmission time of the network, and $C_i^u = \sum_{j \in E_i} T_j^u$ be the total uplink transmission time of extended neighborhood of SS i, which contains SS i and its one-hop and two-hop neighbors. In the IEEE 802.16 standard, only a portion of T_i^u / C_{total}^u is allocated to the uplink transmission time of SS i. Clearly, SS i can detect busy carriers only in C_i^u / C_{total}^u portion of time. In the remaining $(1 - C_i^u / C_{total}^u)$ portion of time, SS i sees idle carriers. Our scheme is designed to exploit this portion of idle time for additional transmissions by raising the same ratio of allocated transmission time for all SSs.

For the fairness of all SSs in E_i, the portion of idle time should be divided proportionally by their transmission time demands. Thus the additional transmission time SS i can obtain is $(1 - C_i^u / C_{total}^u) \times T_i^u / C_i^u$. So the maximal transmission time with spatial reuse for SS i in the mesh network is $T_i^u / C_{total}^u + (1 - C_i^u / C_{total}^u) \times T_i^u / C_i^u = T_i^u / C_i^u$. Let $C_{max}^u = max\{C_i^u, \forall i\}$. For any SS i such that $C_i^u = C_{max}^u$, the SS could be the bottleneck of the network. Therefore, we propose to assign T_i^u / C_{max}^u portion of uplink transmission time to each SS i. It is clear that after assigning T_i^u / C_{max}^u portion of time to each SS i, the bottleneck SS will see 100% busy carriers, whereas other SSs such that $C_i^u < C_{max}^u$ can see some idle carriers. On the other word, we raise the same ratio of uplink transmission time for each SS i from T_i^u / C_{total}^u to T_i^u / C_{max}^u until the bottleneck SS sees 100% busy carriers.

As a result, the smaller C_{max}^u the mesh network can route, the larger transmission time each SS can get. Note that although the maximum of C_i^u among all SS i is used in the mesh network so that T_i^u / C_{max}^u is the lower bound of spectral reuse, actually the lower bound is also an upper bound when C_{max}^u is occurred at the one-hop neighborhood of the BS in most regular mesh networks since all the BS neighbors can not transmit or relay more data for themselves or other child SSs. Continuously, our two-level scheduling scheme with spectral reuse quantified above will be described in the following

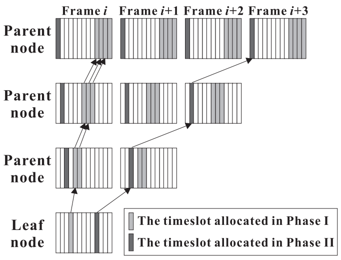

Fig. 2. The timeslots allocated in phase I and phase II

subsections.

1) Channel-Level Scheduling: The mesh mode supports only Time Division Duplex (TDD) to share the channel between downlink and uplink. The TDD framing is adaptive in that the bandwidth allocated to the downlink versus the uplink can vary. The split between uplink and downlink is a system parameter and is controlled at higher layers within the system. In our channel-level scheduling scheme, the ratio of downlink to uplink subchannel will be set to C^d_{max}/C^u_{max} that fits the traffic load distribution. Therefore, the first $F \times C^d_{max}/(C^d_{max} + C^u_{max})$ timeslots in each frame are assigned to downlink subchannel and the rest timeslots are assigned to uplink subchannel, where F is the number of timeslots in a frame. The well-arranged subchannel bandwidth for uplink and downlink could result in that the overall network throughput is increased significantly, which has been validated by simulation results in Section IV.

2) Link-Level Scheduling: In IEEE 802.16 mesh networks, SSs notify the BS their data transfer requirements and the quality of their links to their neighbors. The BS uses the topology information along with the requirements of each SS to decide the routing and the scheduling without spectral reuse. The frame fraction assigned to each SS i is T^u_i/C^u_{total} for uplink traffic in the IEEE 802.16 mesh mode specification, whereas the fraction is T^u_i/C^u_{max} in our scheduling with spectral reuse as mentioned at the beginning of Section III-B. Note that C^u_{max} is much smaller than C^u_{total} in a large IEEE 802.16 mesh network, which implies each SS can obtain much larger frame fraction from our scheduling algorithm.

For timeslot assignment, assume that there are N timeslots in a frame for uplink subchannel. We first allocate $N \times (T^u_i/C^u_{total})$ timeslots in phase I and then assign $N \times (T^u_i/C^u_{max} - T^u_i/C^u_{total})$ timeslots in phase II, which the total allocated timeslots to SS i is $N \times (T^u_i/C^u_{max})$. The allocated timeslots in phase I are assigned to each SS i in the mesh network according to its hop count from the BS, and retain the order to join the network for SSs with the same hop count. The allocated timeslots in phase II are inserted to the remaining space of frame allocation list for all SS j in E_i . As illustrated in Fig. 2, since the forwarding order for all SSs in the mesh network can be hold in phase I and thus the end-to-end delay between the BS and SSs can be minimized, SSs can utilize it by transmitting real-time traffic in order to reduce the packet delay. On the other hand, SSs can use the allocated timeslots in phase II without forwarding order to transmit non-real-time or best effort traffic since the packet delay is not crucial even though the end-to-end delay may be the duration of several frames. Note that the sum of the allocated timeslots for the SSs in the extended neighborhood with C^u_{max} equals to N exactly. Therefore, there are sufficient free timeslots in a frame to insert the allocated timeslots in phase I and phase II for those SSs in the extended neighborhood with C^u_i that is smaller than C^u_{max}. The link-level scheduling algorithm is described as follows.

Link-level scheduling algorithm

Phase I:

Allocate $N \times (T^u_i/C^u_{total})$ timeslots to each SS i according to the transmission order of MSH-CSCH:Request until all SSs have been allocated.

Phase II:

(1) Construct the frame allocation list L_i of E_i for each SS i in the network.

(2) According to the transmission order of MSH-CSCH:Request, assign the first $N \times (T^u_i/C^u_{max} - T^u_i/C^u_{total})$ free timeslots in L_i to SS i.

(3) Update all frame allocation lists L_j that E_j includes SS i.

(4) Repeat steps (2) and (3) until all SSs have been assigned.

C. Routing Tree Construction

The routing tree construction problem investigated in this section is to find a routing tree with the minimum C^u_{max} in a directed mesh network graph $G = (V, E)$ according to the traffic demand b_i requested by vertex $i \in V$ and the uplink data rate r^u_j of edge $j \in E$. We first prove that the routing tree construction problem is a NP-complete problem, and then propose a load-aware routing algorithm to reduce C^u_{max} for spectral efficiency. Below, we show the routing tree construction is NP-complete by proving that its decision problem is NP-complete.

The Problem

Given a directed mesh network graph $G = (V, E)$, the traffic demand b_i requested by vertex $i \in V$, the uplink data rate r^u_j of edge $j \in E$, and a real number R, determine whether G has a routing tree such that its $C^u_{max} \leq R$.

□ Theorem 1

The routing tree construction problem is NP-complete.

Proof: The routing tree construction belongs to NP, since we can guess a routing tree and check whether its $C^u_{max} \leq R$ easily in polynomial time. To prove that the routing tree construction problem is NP-complete, we have to reduce an NP-complete problem to it. We use the partition problem: the input is a set X such that each element $x \in X$ has an associated size $s(x)$. The problem is to determine whether it is possible to partition the set into two subsets with exactly the same total size. [7]

Consider a special case of mesh networks in Fig. 3. Assume that E_a and E_b are not overlapped, all uplink data rates in E_a and E_b are the same and low enough such that C^u_{max} is $max\{C^u_a, C^u_b\}$, and there are n SSs $(x_1, x_2, \ldots, x_n)$ be neighbors of SS c and SS d. Let the traffic demands of all SSs in the mesh network except x_1, x_2, ... , and x_n be zero.

Now we start to reduce the partition problem to the special case of the routing tree construction problem. Let $X = \{x_1, x_2, \ldots, x_n\}$, $s(x_k)$ be the traffic demand of x_k for $k = 1, 2, \ldots, n$, and $R = 5/2 \cdot \sum_{\forall k} s(x_k)/r_{slow}$, where r_{slow} is the data rate of slow link in Fig. 3. The parent node of x_k is either vertex c or vertex d. Thus, we can get the smallest C^u_{max} by partitioning $X(x_1, x_2, \ldots, x_n)$ into two subsets (SS c and SS d) with exactly the same total size. Therefore, if there is a routing tree such that $C^u_{max} = R$ in G, then there is a partition to divide X into two subsets with exactly the same total size. This reduction can obviously be performed in polynomial time. Since the special case of the routing tree construction problem is NP-complete, the general case is also NP-complete. □

To achieve efficient spectral reuse and high throughput in IEEE 802.16 mesh networks, we propose a load-aware routing algorithm to reduce C^u_{max} for uplink traffic. In our algorithm, we assume the initial value of C^u_i is $\sum_{j \in E_i} d^u_j / r^u_j(max)$ for each SS i in the mesh network, where $d^u_j = b^u_j$ and $r^u_j(max)$ is the highest data rate among links of SS j to its neighbors with less or equal hop count. The tree construction uses a bottom-up fashion that each SS i with the largest hop count to the BS will be first attached to its neighbors k which have less or equal hop count to estimate each new C^u_k, and then the SS which has minimum C^u_k will be chosen as the parent node of SS i. If there are several SSs with the same minimum C^u_k, the SS with smaller hop count has the higher priority. Once each SS with largest hop count has been attached to its parent node, the remaining SSs without a parent node repeat the above procedure until each SS in the mesh network has a parent node. Note that the step (2) in load-aware routing algorithm is to build the subtree with the minimum C^u_k first, which can balance the distribution of forwarding traffic and further reduce C^u_{max}.

Load-aware routing algorithm

(1) Let A be the set of SSs without a parent node that have the largest hop count, and B the empty set

(2) Estimate each C^u_k for all neighbors k with less or equal hop count when SS i in A becomes the child of SS k, and the SS with the smallest C^u_k will be chosen as the parent node of SS i

(3) Remove SS i from A, add SS i into B, and update C^u_l for all SS $l \in E_i \cup E_k$

(4) Repeat steps (2) and (3) until there is no SS in A

(5) Repeat steps (1) $\sim$ (4) until each SS has a parent node

The analysis of time complexity is as follows. Since each SS only has a parent node, steps (2) and (3) just repeat n times, where n is the number of SSs in the network. The dominant part of steps (2) and (3) is the step (2) that selects the smallest one from at most $m \times d$ estimated C^u_k values, where m is the maximum number of SSs with the same hop count, and d is the maximum degree of SSs. Therefore, the algorithm takes $O(nmd)$ time to build the routing tree.

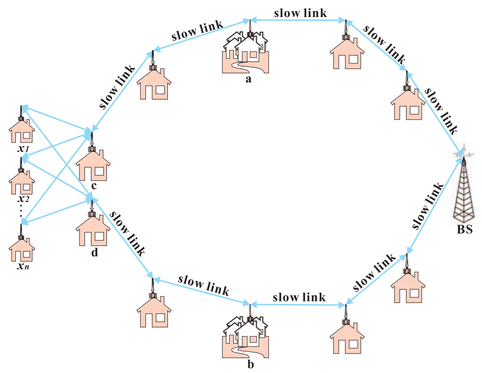

Fig. 3. The special case of the routing tree construction problem

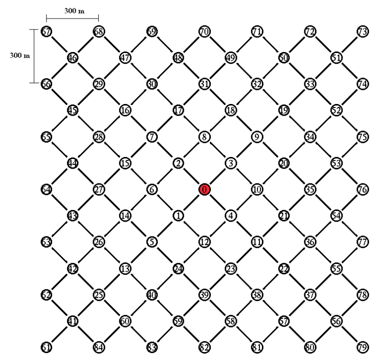

Fig. 4. The node placement in the regular mesh topology

IV. Performance Evaluation

In this section, we provide ns-2 [9] simulation results for the spectral reuse framework and compare it with the basic 802.16 mesh operation in [4] as well as the concurrent transmission with route adjustment in [5]. The typical TCP/IP/LL/MAC/PHY stack is used in our study. In addition, we adopt a single channel OFDM PHY and two-ray ground reflection model for radio propagation, and all the SSs are stationary and working in half duplex. In our work, we extend the TDMA MAC module in ns-2 for timeslot reuse in a multi-hop environment and use it to study the system performance.

In our simulation, the node placement in the regular mesh topology is shown in Fig. 4. There are totally at most 85 nodes which consist of a single BS (node 0) and 84 SSs (node 1 $\sim$ 84), and the one-hop neighbors are connected by lines. The channel bandwidth is set to 50 Mb/s and the data rates of all links are the same for simplicity. The extended neighborhood of each SS includes one-hop and two-hop neighbors. The random routing tree is used in the basic 802.16 mesh mode and our link-level scheduling except that the load-aware routing is marked on the figures. Note that the overall network throughput has been normalized by the performance of basic 802.16 mesh operation so that the scalability and improvement of our proposed framework are clearly demonstrated in the simulation results.

Fig. 5 shows the normalized gateway throughput with different scheduling and routing methods, respectively. The

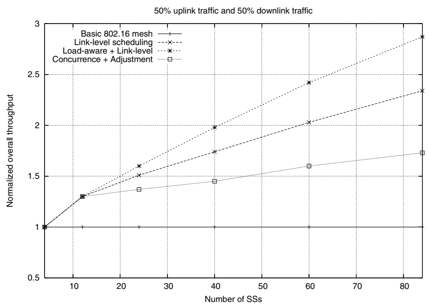

Fig. 5. The performance comparison for link-level scheduling

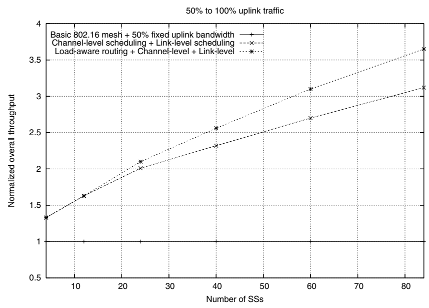

Fig. 6. The performance comparison for channel-level scheduling

number of SSs increases from 4, 12, 24, 40, 60, to 84 and all SSs request the same bandwidth for uplink and downlink. The throughput values are the average of simulation in 100 times with random load distribution among SSs. As shown in Fig. 5, the proposed link-level scheduling scheme outperforms the basic mesh mode significantly. Also, the routing tree generated by the load-aware routing algorithm further improves the throughput. It is because that in the basic 802.16 mesh scheme, the network throughput drops significantly as the number of SSs increases due to the fact that a packet needs to be forwarded several times since the length of relay route increases with the number of SSs in the network, whereas the proposed link-level scheduling is much more scalable than the basic scheme since the degree of spectral reuse increases with the network size. In addition, the load-aware routing algorithm produces better routing paths to distribute the traffic more evenly in the mesh network. Therefore, the scheme with both the load-aware routing and link-level scheduling achieves the highest network throughput. The scheme only using link-level scheduling still has the second best performance. On the other hand, since there is no scheduling algorithm provided in [6] and the concurrent transmission scheme in [5] outperforms that without route adjustment in [2], we also compare the performance of concurrent transmission with route adjustment in the simulation. The non-load-aware routing method constructs a routing tree according to the SS positions, which can not release the traffic bottleneck in the network efficiently. Thus, the benefit of route adjustment has been limited in the nature unless every SS generates the same traffic load under the same link data rate. In addition, the concurrent transmission algorithm forces SSs can not transmit data earlier than their child SSs so that the utilization of spectral reuse is reduced significantly. Therefore, its throughput improvement is much lower than our integrated spectral reuse framework.

Fig. 6 shows the normalized overall throughput with channel-level and link-level scheduling schemes. The configuration of simulation is as same as in Fig. 5. However, every SS requests 50% to 100% uplink bandwidth randomly, and thus the downlink bandwidth requested is 0% to 50% which depends on the uplink bandwidth requested. Note that the basic 802.16 mesh mode allocate the bandwidth equally for uplink and downlink subchannels. As shown in Fig. 6, the proposed channel-level and link-level scheduling scheme outperforms the basic mesh mode more significantly. Again, the combined routing and scheduling scheme gets the highest system throughput. This is because that channel-level can adapt dynamically the bandwidth between uplink and downlink subchannels based on the traffic load distribution in the mesh network. Using load-aware routing, the network throughput can be enhanced as the number of SSs increases. As a result, the combination of channel-level and link-level scheduling as well as load-aware routing can fit more traffic patterns so as to keep high network performance.

V. CONCLUSIONS

In this paper, we have formally quantified spectral reuse in IEEE 802.16 mesh networks. Also, an integrated spectral reuse framework for centralized scheduling scheme and routing tree construction is developed. Compared to existing works, our work is most complete in exploiting spectral reuse in IEEE 802.16 mesh networks in the sense that it takes dynamic traffic loads of SSs into account and integrates bandwidth adaptation, timeslot allocation, as well as routing tree construction under a framework. Simulation results indicate that the spectral reuse scheduling and load-aware routing significantly increase the overall throughput in IEEE 802.16 mesh networks.

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附錄五：

Route Throughput Analysis with Spectral Reuse for Multi-Rate Mobile Ad Hoc Networks

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Route Throughput Analysis with Spectral Reuse for Multi-Rate Mobile Ad Hoc Networks

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Abstract

The mobile ad hoc networks (MANETs) have received a lot of attention for its flexible network architecture. While many routing protocols have been proposed for MANETs based on different criteria, few have considered the impact of multi-rate communication capability that is supported by many current WLAN products. Given a routing path, this paper provides an analytic tool to evaluate the expected throughput of the route with spectral reuse, assuming that hosts move following the discrete-time, random-walk model. The derived result can be added as another metric for route selection. Simulation results show that the proposed formulation can be used to evaluate path throughput accurately.

Keywords: Ad hoc networks, mobile computing, mobile networks, routing, wireless communication, spectral reuse.

I. Introduction

The mobile ad hoc network (MANET) is a flexible and dynamic architecture that is attractive due to its ease in network deployment. Routing is perhaps one of the most intensively addressed issues in MANET. Many different criteria have been used in route selection, including hop count [5], signal strength [11], route lifetime [3], and energy constraint [12]. Among these metrics, hop count may be the most widely used metric in choosing routes. When a hop-count based routing protocol is given multiple paths, the shortest path is normally selected and a random path is selected when there is a tie. This metric has the advantage of simplicity, requiring no additional measurements and incurring the least number of transmissions. The primary disadvantage of this metric is that it does not take packet loss or available bandwidth into account, especially when network interfaces can transmit at multiple rates [10]. It has been shown in [4] that a route which minimizes the hop count does not necessarily maximize the throughput of a flow.

While it is true that there is no single route selection metric that is able to best fit all possible routing scenarios in MANET, few works have considered the impact of multi-rate communication capability that is widely supported by many current wireless LAN products. For example, IEEE 802.11b supports rates of 11, 5.5, 2, and 1 Mbps, while IEEE 802.11a supports rates of 6, 9, 12, 18, $\cdots$, and 54 Mbps. Route selection is more complicated in a multi-rate MANET than in a single-rate environment. Also, there exists an inherent tradeoff between transmission rates and their effective transmission ranges [2]. To support reliable data transmissions, longer-range communications must use lower rates, and vice versa. Auto-rate selection protocols [1], [6] do exist at the link level. Reference [13] proposes a multi-rate-aware topology control algorithm to enhance the network throughput in multi-hop ad hoc networks, and [14] uses fast links (with high nominal bit rate) to improve the system throughput in wireless mesh networks. However, they only focus on static network environment without taking mobility into account. Reference [17] proposes a multi-rate-aware sub-layer between the MAC and the network layers to improve resource utilization

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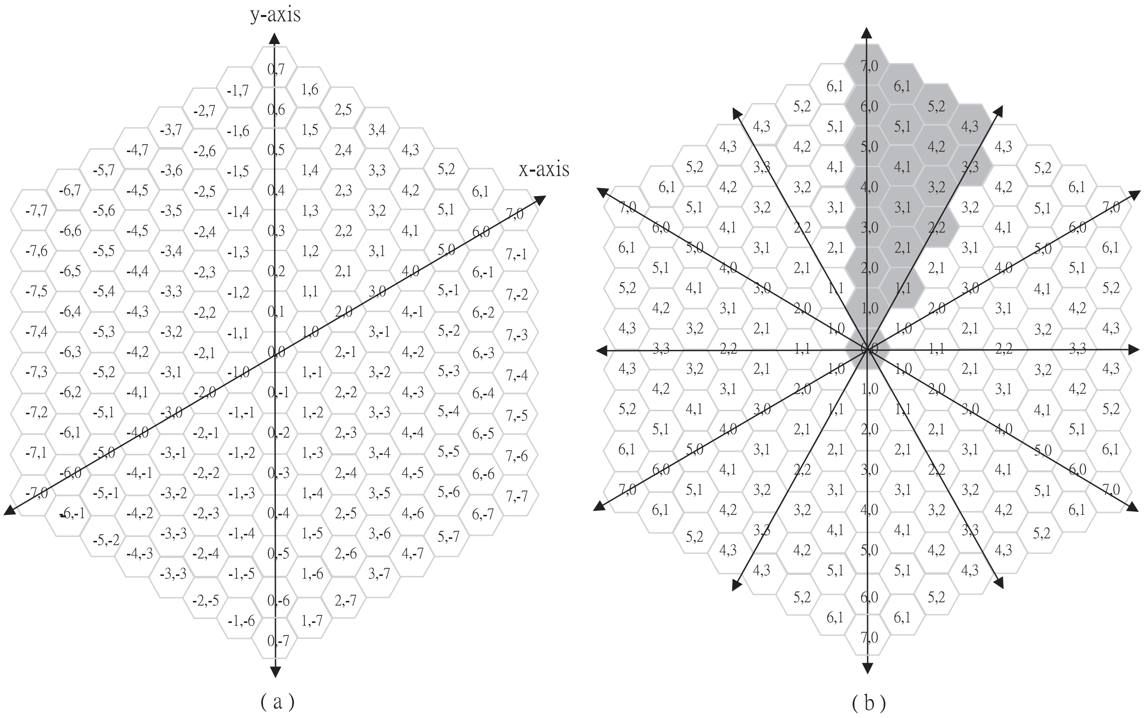

Fig. 1. (a) a cellular system to model station mobility, and (b) the “folding” of link states.

and to minimize power consumption, but the effect of multi-rate communications at the routing level is not yet fully addressed.

In this paper, we consider a MANET where each wireless link can support multiple rates and has the auto-rate selection capability. Given a routing path, this paper provides an analytic tool to evaluate the expected throughput of the route with spectral reuse, assuming that hosts move following the discrete-time, random-walk model. The result can be added as a new metric for route selection in MANET. (We comment that we do not intend to propose a new routing protocol here. But the proposed results may be used in many current protocols to compute a new route selection metric.)

The rest of this paper is organized as follows. Section 2 presents our system model. Section 3 shows our analysis results. Simulation results are presented in Section 4. Section 5 concludes this paper.

II. System Model

In this paper, we assume that each mobile host roams around in the network area following the discrete-time, random-walk mobility model, which has been widely used in several works [7], [8], [9]. In this model, the network area is partitioned into a number of hexagonal cells, each with radius r and each with a coordinate (x, y), as shown in Fig. 1(a). Cells on the x-axis are numbered $(x, 0)$, and those on the y-axis $(0, y)$. The coordinates of other cells are obtained by mapping them onto these two axes, as is normally done in the Cartesian coordinate system.

Although hosts actually roam around in continuous time domain, we will work in discrete time domain by dividing time into fixed-length units. We assume that mobile hosts roam around in a cell-to-cell basis following the random walk model. Given a mobile host at any cell, it will move into any one of its six neighboring cells in the next time unit with an equal probability of $1/6$.

Cells in the network are further divided into layers as follows. Cell $(0, 0)$ is on the layer-0. The six neighboring cells of cell $(0, 0)$ are the layer-1 cells, and the outer cells surrounding layer-i cells are said to be on layer $(i + 1)$. The number of cells included in an n-layer network is $3n^2 + 3n + 1$. In this paper, we will model the transmission range of a mobile host by a certain number of layers, by assuming its current location at layer 0.

Following [16], we use a vector to represent the state of a wireless link. Specifically, given a wireless link between two hosts located at cells (x, y) and (x', y'), we represent the link's state as a vector $\langle x' - x, y' - y\rangle$. A routing path thus may contain a sequence of vectors, each representing a wireless link. For example, a routing path containing hosts in cells $(0, 1)$, $(3, 1)$, and $(7, -3)$ in the order can be written as $[\langle 3, 0\rangle, \langle 4, -4\rangle]$.

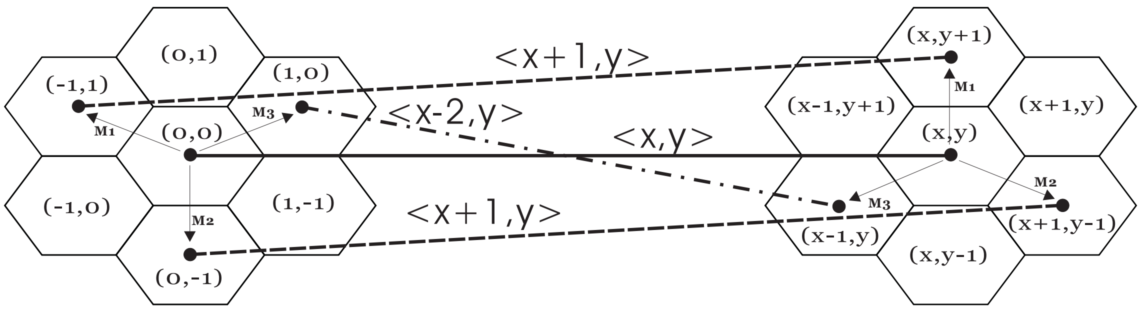

Fig. 2. Example of link state changes.

TABLE I

THE PROBABILITY DISTRIBUTION FOR A WIRELESS LINK TO SWITCH FROM STATE $\langle x, y\rangle$ TO STATE $\langle x', y'\rangle$ AFTER ONE TIME UNIT.

<x',y'>	<x,y>	<x-1,y>	<x-1,y-1>	<x,y-2>	<x+1,y-2>	<x+1,y-1>	<x+1,y>	<x,y-1>	<x+2,y-2>	<x+2,y-1>
Probability	6/36	2/36	2/36	1/36	2/36	2/36	2/36	2/36	1/36	2/36

<x',y'>	<x+1,y+1>	<x,y+1>	<x+2,y>	<x,y+2>	<x-1,y+2>	<x-1,y+1>	<x-2,y+2>	<x-2,y+1>	<x-2,y>
Probability	2/36	2/36	1/36	1/36	2/36	2/36	1/36	2/36	1/36

Based on the random walk model, we can derive a probability model for the state change of a wireless link. Let $\langle x, y\rangle$ be the state of a wireless link connecting two neighboring hosts at time t, At time $t + 1$, each of the hosts may move into one of its six neighboring cells with probability of $1/6$. This gives 36 combinations of the two hosts' next locations (as shown in Fig. 2), which can be reduced to 19 link states with different probabilities (as shown in Table I) [16].

Suppose that the transmission distance of a host is n layers. Then the number of states for a wireless link will be as large as $3n^2 + 3n + 1$. To prevent the problem of state explosion that so many states need to be taken into consideration, [15] proposes to merge equivalent cells by "folding" the 12 sectors in Fig. 1(b) into one (cells of the same indices are equivalent). This reduces the number of states by around $1/12$. Detailed derivations can be found in [15]. We will adopt the state reduction in this paper.

Most current wireless LAN cards support automatic rate selection depending on channel conditions. For example, IEEE 802.11b standard supports four transmission rates: 11, 5.5, 2, and 1 Mbps. When the MAC layer overheads are taken into account (control overheads, contention overheads, collision costs, etc.), the effective link rates may be reduced from 11, 5.5, 2, and 1 Mbps to 4.55, 3.17, 1.54, and 0.85 Mbps, respectively [2]. We assume that the rate of a wireless link will depend on the distance between the two hosts of the link. Reference [2] provides a general theoretical model of the attainable throughput in multi-rate ad hoc wireless networks.

III. ROUTE THROUGHPUT ANALYSIS

A route consists of a number of wireless links. Given a routing path, our goal is to determine the expected route throughput based on the random walk model. In the previous section, we have derived how a wireless link changes states. Suppose that each mobile host has a transmission range of n layers. Then we can model a wireless link by considering an $(n + 2)$-layer network. For example, Fig. 3 shows the state transition diagram of a wireless link when $n = 5$. Note that states $\langle 6, 0\rangle, \langle 5, 1\rangle, \langle 4, 2\rangle, \langle 3, 3\rangle, \langle 7, 0\rangle, \langle 6, 1\rangle, \langle 5, 2\rangle$, and $\langle 4, 3\rangle$ are "absorbing" states such that $x + y > n$ for state $\langle x, y\rangle$, which means the distance between mobile hosts is larger than the transmission range and once a wireless link changes to any of these states, the link is considered broken.

The state transition probability of a wireless link in Fig. 3 can be modeled by a matrix M in Fig. 4, where each element $M_{i,j}$ represents the probability for a link to transit from state i to state j. M^k is the k-th power of M, which represents the state transition probabilities after k time units. That is, $M^k_{i,j}$ is the probability that a link at state i transits to state j after k time units. Therefore, M is a $C(n + 2) \times C(n + 2)$ matrix. The formal derivation

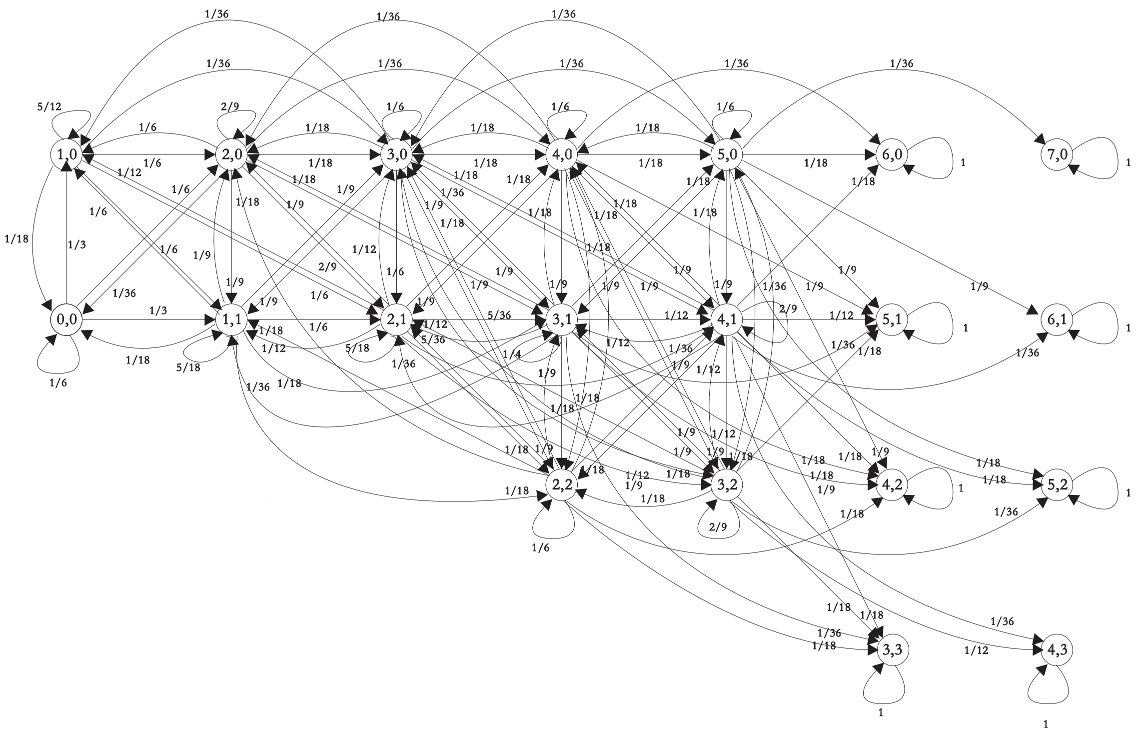

Fig. 3. State transition diagram of a wireless link when $n = 5$.

$$
M = \begin{array}{c} \\ \langle 0{,}0\rangle \\ \langle 1{,}0\rangle \\ \langle 2{,}0\rangle \\ \langle 1{,}1\rangle \\ \vdots \\ \langle 5{,}2\rangle \\ \langle 4{,}3\rangle \end{array}
\begin{array}{c}
\begin{array}{ccccccc} \langle 0{,}0\rangle & \langle 1{,}0\rangle & \langle 2{,}0\rangle & \langle 1{,}1\rangle & \cdots & \langle 5{,}2\rangle & \langle 4{,}3\rangle \end{array} \\
\begin{bmatrix}
\frac{6}{36} & \frac{12}{36} & \frac{6}{36} & \frac{12}{36} & \cdots & \frac{0}{36} & \frac{0}{36} \\
\frac{2}{36} & \frac{15}{36} & \frac{6}{36} & \frac{6}{36} & \cdots & \frac{0}{36} & \frac{0}{36} \\
\frac{1}{36} & \frac{6}{36} & \frac{8}{36} & \frac{4}{36} & \cdots & \frac{0}{36} & \frac{0}{36} \\
\frac{2}{36} & \frac{6}{36} & \frac{4}{36} & \frac{10}{36} & \cdots & \frac{0}{36} & \frac{0}{36} \\
\vdots & \vdots & \vdots & \vdots & \ddots & \vdots & \vdots \\
\frac{0}{36} & \frac{0}{36} & \frac{0}{36} & \frac{0}{36} & \cdots & \frac{36}{36} & \frac{0}{36} \\
\frac{0}{36} & \frac{0}{36} & \frac{0}{36} & \frac{0}{36} & \cdots & \frac{0}{36} & \frac{36}{36}
\end{bmatrix}
\end{array}
$$

Fig. 4. A state transition matrix of a wireless link when $n = 5$.

of $C(n)$ can be found in [15]:

$$
C(n) = \begin{cases} 1 & n = 0 \\ \frac{(n+1)(n+3)}{4} & n > 0 \text{ and n is odd} \\ \frac{n(n+4)}{4} + 1 & n > 0 \text{ and n is even} \end{cases} .
$$

Suppose that a wireless link is in state i at time 0. The probability that the link will become broken at time t is

$$
P_1(i,t) = \sum_{j \in \text{layer } n+1,\, n+2} M_{i,j}^t .
$$

The probability that the wireless link is alive at time $t-1$ but becomes broken at time t is

$$P_2(i,t) = \begin{cases} P_1(i,t) & \text{if } t = 1 \\ P_1(i,t) - P_1(i,t-1) & \text{if } t > 1 \end{cases}.$$

Now consider an α-hop route $R = [s_1, s_2, \cdots, s_\alpha]$, where $s_i, i = 1..\alpha$, is the state of the ith wireless link in R. The probability that R is still alive after t time units is

$$P_3(R,t) = \prod_{i=1}^{\alpha}(1 - P_1(s_i,t)).$$

A path breaks when one or more of its links break. So the probability that R becomes broken after t time units is

$$P_4(R,t) = 1 - P_3(R,t)$$

and the probability that R is alive at time $t-1$ but becomes broken at time t is

$$P_5(R,t) = \begin{cases} P_4(R,t) & \text{if } t = 1 \\ P_4(R,t) - P_4(R,t-1) & \text{if } t > 1 \end{cases}.$$

Let each wireless LAN card support m rates, $R_1, R_2, \ldots, R_m$, such that rate $R_i, i = 1..m$, will be used if the destination host falls between (including) layers $n_{i-1}+1$ and n_i from the source, where $n_0 = -1$ and $n_m = n$. For example, reference [2] models an IEEE 802.11b card by $R_1 = 11$, $R_2 = 5.5$, $R_3 = 2$, $R_4 = 1$, $n_0 = -1$, $n_1 = \lfloor \frac{n}{2} \rfloor$, $n_2 = \lfloor \frac{2n}{3} \rfloor$, $n_3 = \lfloor \frac{5n}{6} \rfloor$, and $n_4 = n$. Given the initial state of link $s_i, i = 1..\alpha$, the probability that the link's rate falling in R_j (i.e., the link's distance is between layers $n_{j-1}+1$ and n_j) at time t is

$$P_6(s_i, R_j, t) = \sum_{k \in \text{ layer } (n_{j-1}+1)..n_j} M_{i,k}^t.$$

Therefore, the bandwidth of R at time t can be modeled by summing the expected transmission rate of the route over all possible rate combination of links in R at time t as follows

$$B(R,t) = \sum_{i_1=1}^{m}\sum_{i_2=1}^{m}\cdots\sum_{i_\alpha=1}^{m} P_6(s_1, R_{i_1}, t)$$
$$\times P_6(s_2, R_{i_2}, t) \times \cdots \times P_6(s_\alpha, R_{i_\alpha}, t) \times f(R_{i_1}, R_{i_2}, \cdots, R_{i_\alpha}),$$

where the function f is the transmission rate of the route. It will be estimated in next subsection. Finally, the expected throughput of route R, denoted by $E(R)$, can be derived by summing the expected route throughput over all possible route lifetime of R as follows

$$E(R) = \sum_{t_1=1}^{\infty}\left(P_5(R,t_1) \times \sum_{t_2=1}^{t_1}\frac{B(R,t_2)}{t_1}\right). \tag{1}$$

A. *Estimation of the Function* $f(\cdot)$

In this subsection, we will propose a method to estimate the throughput of a given α-hop route $R = [s_1, s_2, \cdots, s_\alpha]$, where $s_i, i = 1, 2, .., \alpha$, is the state of the ith wireless link. Recall that we represent link state as a vector in a 2-dimensional space. So from each s_i, we can derive the distance between the two endpoints of the link and the most appropriate rate r_i that should be used by this link. Given such a route R, our goal is to derive its transmission rate $f(r_1, r_2, \cdots, r_\alpha)$. An ideal channel condition is assumed in the estimation such that a transmission fails only when collisions occur.

The hosts in routing path R are numbered from 0 to α such that host 0 is the traffic source and host α the sink of the path. Therefore, s_i is the state of the link between host $i-1$ and host i. Except the sink host, we can assign each host i in R an interference group G_i, which contains host i and each host j in front of i (i.e., $i > j$) that can sense the signal of i when i is transmitting. Intuitively, hosts in the same group will not transmit at the same time, but hosts in different groups may be allowed to transmit simultaneously. Note that the interference group G_i is defined to make hosts in G_i can not transmit at the same time. If hosts behind host i are included in G_i, hosts

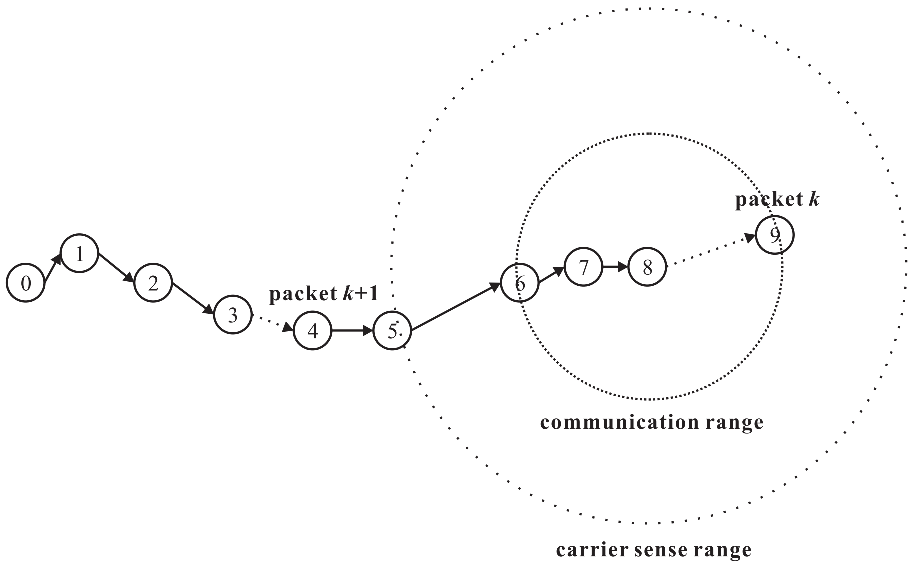

Fig. 5. The 9-hop route with its most interfered region including host $5 \sim 9$.

in front of host i and hosts behind host i may be allowed to transmit simultaneously. In our estimation, we model the function $f(\cdot)$ by

$$f(r_1, r_2, \cdots, r_\alpha) = \frac{1}{\max_{i=1\sim(\alpha-1)}\{\frac{1}{r_{(i+1)}} + \sum_{j \in G_i} \frac{1}{r_j}\}},$$

where $\max_{i=1\sim(\alpha-1)}\{\frac{1}{r_{(i+1)}} + \sum_{j \in G_i} \frac{1}{r_j}\}$ is the time required to transmit a bit along R in the most interfered region.

The basic concept of our modulation is that host a receiving packet $k+1$ can not be in the carrier sense range of host b sending packet k. In other words, these two packets can be transmitted simultaneously if host a is not in the carrier sense range of host b. From the view of pipelining, when packet k arrived at host b, packet $k+1$ has arrived at the host near but out of the carrier sense range of host a. Since the slowest stage of a pipeline dominates its throughput, we take the time T of travelling through the most interfered region as the transmission time for packets in R. So every packet except the first one in R only takes T to arrive at sink host α after its previous packet arrived at the sink host. Therefore, the expected transmission time for each packet of size S being transmitted in R is $T = \max_{i=1\sim(\alpha-1)}\{\frac{S}{r_{(i+1)}} + \sum_{j \in G_i} \frac{S}{r_j}\}$, and the transmission rate of R is $\frac{S}{T} = \frac{1}{\max_{i=1\sim(\alpha-1)}\{\frac{1}{r_{(i+1)}}+\sum_{j \in G_i} \frac{1}{r_j}\}}$. For example, Fig. 5 shows the 9-hop route with its most interfered region including host $5 \sim 9$. The links that can transmit concurrently are indicated by dashes. We can find that when packet k arrived at sink host 9, packet $k+1$ has arrived at host 4. It is because that when host 3 is sending packet $k+1$ to host 4, host 8 can send packet k to sink host 9 at the same time without interfering the receiving of host 4. Therefore, packet $k+1$ in the 9-hop route only takes the time of travelling through hosts $5 \sim 8$ to arrive at sink host 9 after packet k arrived at the sink host. Accordingly, the transmission rate of the 9-hop route is $\frac{1}{\frac{1}{r_9}+\sum_{j \in G_8} \frac{1}{r_j}} = \frac{1}{\frac{1}{r_5}+\frac{1}{r_6}+\frac{1}{r_7}+\frac{1}{r_8}+\frac{1}{r_9}}$.

IV. Simulation Results

In this section, we present our simulation results. Most current products of IEEE 802.11b have a transmission distance of $150 \sim 300$ meters. We set the radius of each hexagonal cell to 10 meters, so hosts' transmission range is around $n = 15 \sim 30$ layers. The carrier sense range is set to be the same with the transmission range, and the mobile host is set to randomly select its roaming direction per time unit. Each time unit is set to 10 seconds, so as to model the roaming speed of pedestrians (around 1 m/s). The saturated traffic and unlimited buffer are used in our simulation, and the roaming speed of each mobile host is set to 1 m/s.

First, we try to determine the level of accuracy. Observe that index t_1 in Eq. (1) ranges from 1 to infinity. This is computationally infeasible. So we need to determine an upper bound for t_1 (called t_1^{max} below). We randomly generate five routing paths with 1, 3, 6, 9, and 12 hops, respectively. We calculate their expected throughput by varying t_1^{max} from 100 to 1000. The results are in Fig. 6 for $n = 15$ and 25, respectively. Since $E(R)$ stablizes at $t_1^{max} \approx 300$, we will set $t_1^{max} = 1000$ in the rest of the simulations.

Our results can be used to help route selection in a MANET. Hop count is probably the most widely used route selection criterion. Our result may provide an alternative choice if throughput is the main concern, especially under a multi-rate environment. We pick a source cell and a destination cell, and place some relay hosts between them which are separated uniformly. We evaluate the expected route throughput by varying the number of relay hosts (and thus path length that is the number of links in the path). Fig. 7 shows our results for $n = 15$ and 25, respectively.

In both cases, we see that the throughput increases with the path length initially, but decreases afterwards after certain thresholds. In fact, there are two contradicting factors here. A very small path length implies a low transmission rate in each hop, thus leading to low path throughput. On the contrary, a longer path implies potentially higher rates and the higher degree of spectral reuse, but may risk a higher probability of existence of low-rate links in the path (thus becoming a bottleneck). Our result may be used here to make a smart choice.

Fig. 7 also contains comparisons of simulation and analytical results. In each simulation, we evaluate the throughput of the path every time unit until it is broken and then calculate the average throughput. Each simulation is repeated 20,000 times to capture the random roaming of mobile hosts, and then we take the average throughput. As can be seen, the simulated and analytical results are quite close, which justifies the correctness of our derivation.

V. CONCLUSIONS

In this paper, we have shown how to formulate the throughput given a path in which hosts roam around in a random walk model and the communication interfaces have the rate adaptive capability. As far as we know, this issue has not been carefully studied yet. Simulation results show that the proposed formulation can be used to evaluate path throughput accurately. We believe that the path throughput is a better metric than the traditional metrics, such as the hop count, for route selection in multi-rate ad hoc networks and that the proposed mechanism can be easily embedded into most of the current routing protocols for mobile ad hoc networks.

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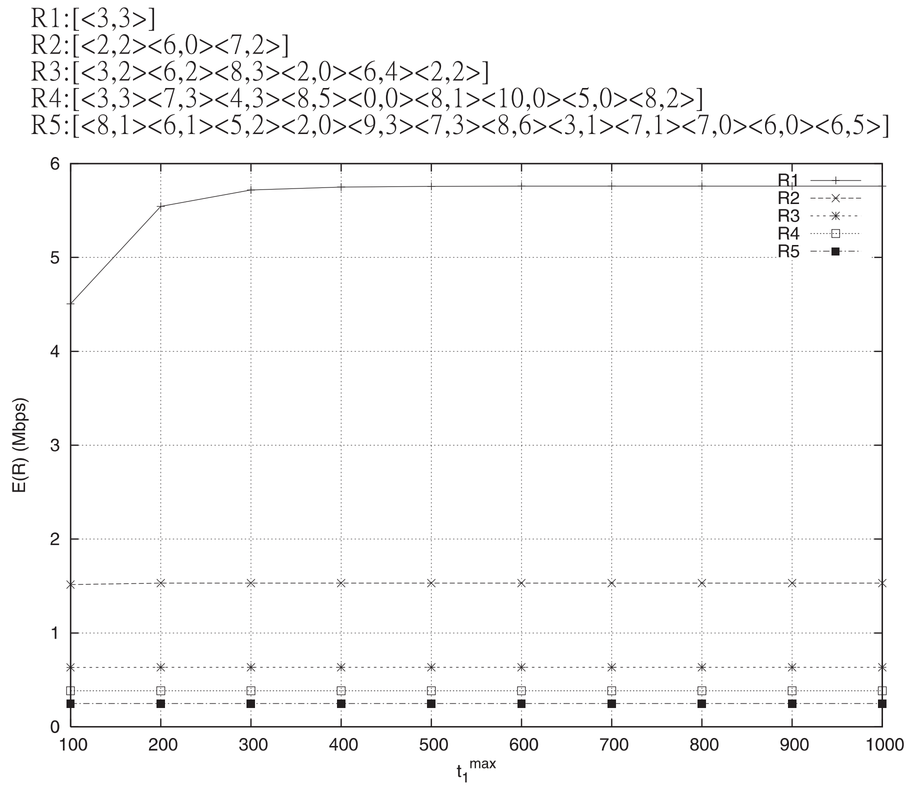
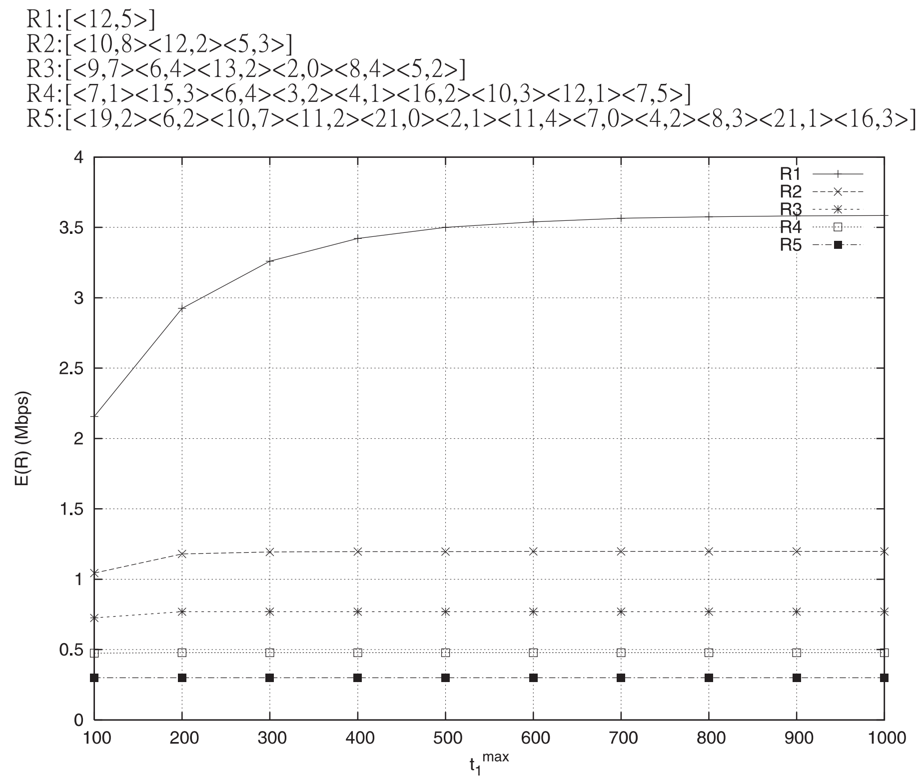
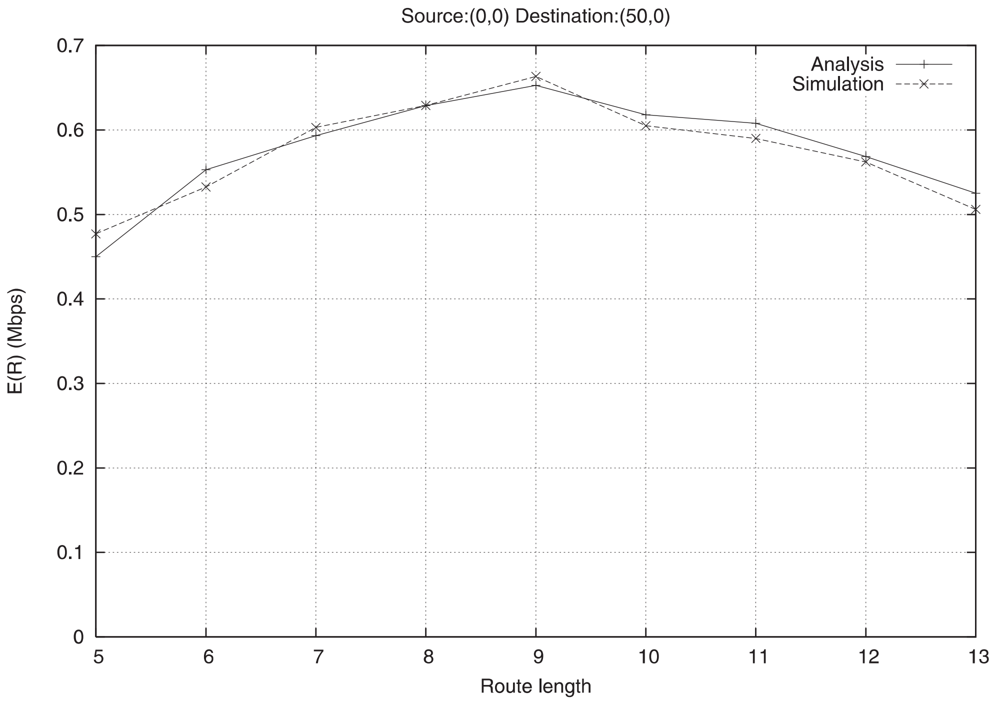


(a)

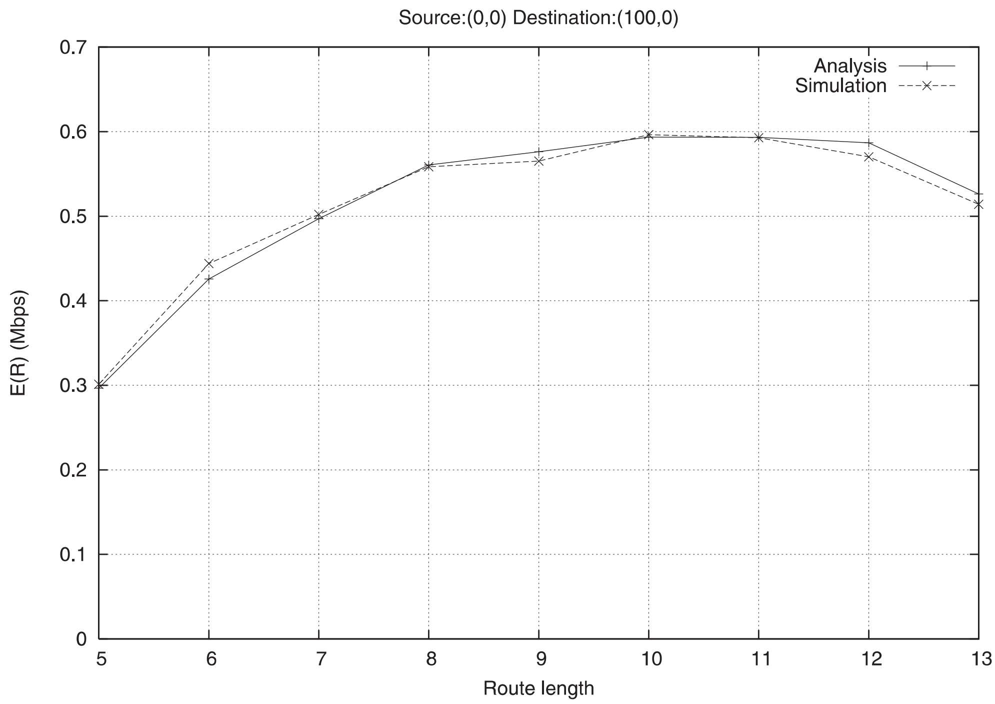


(b)

Fig. 6. Expected route throughput vs. t_1^{max}: (a) $n = 15$ and (b) $n = 25$.

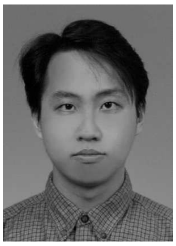


(a)



(b)

Fig. 7. Expected route throughput vs. path length: (a) $n = 15$ and (b) $n = 25$.



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附大陸出差或研習心得報告一份

會議/訪問時間地點	2007/8/8~2007/8/11 北京,中國
會議名稱	互聯網服務主題研討會
發表論文題目	Mobile GeoWeb Services

□ 赴國外出差或研習
■ 赴大陸地區出差或研習
□ 出席國際學術會議
□ 國際合作研究計畫出國

心得報告

計畫名稱	多天線多通道多模多速率無線網狀網路之設計與實作-子計畫一：~M4 無線網狀網路之網路規劃及資源分配問題(2/3)	計畫編號	**95-2219-E-009-007-**
報告人姓名	郭聖博	服務機構及職稱	國立交通大學資訊工程學系博士班學生
會議/訪問時間地點	2007/8/8~2007/8/11 北京,中國		
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一、主要任務摘要（五十字以內）

參與計畫海報展，展示目前計畫方向與進度，並且與其他與會學者交流。

二、對計畫之效益（一百字以內）

蒐集多媒體資料探勘在國際上最新之發展趨勢，並且於現場海報展與實機展示中獲取目前世界上各主要實驗室目前的最新研究方向與主題，並且將目前在執行中的計畫進度向亞洲微軟研究院報告。

三、經過

亞洲微軟研究院舉行本次會議的目的在於交流最新相關於Internet Service這個研究主題上的最新發展，特別是在各種search的技術。本次workshop所邀請的對象主要為獲得MSRA Internet Services Invitation for Proposal Program的學者，希望藉由本次的聚會展示最新的研究進展。舉行的方式包括Poster Session，Group Discussions，和各種有趣的Demo，此外還有多個研究主題的演講：

- Social Network for the Appreciation and Learning of Arts
- Overview of Web Search & Data Mining Group at Microsoft Research Asia
- Overview of Internet Services Theme Program at Microsoft Research Asia
- Automatic Search Engine Evaluation Using Click-through Data
- Supporting Data Retrieval over Programmable Web
- Dolphin: An Academic Web Search Engine
- From Multimedia Search, Streaming To Internet Infrastructure for Multimedia Service
- Privacy Preservation - Approximate Algorithms for k-Anonymity
- Improving the Quality of Responses to Health Search Queries by Evidence-based Context
- Enhancing Web Search by Web Knowledge and User Behaviors
- A Personalized Million-book Recommender System
- Bias and Controversy in Evaluation System

Group Discussion的部分是以平行方式進行，共計有五個topics，地點在亞洲微軟研究院，每個topic都有多位微軟亞洲研究院的研究員參與討論。主要topic大致如下：

（一） Building Infrastructure for Web Data Management
（二） Object Level Search
（三） Enterprise Search
（四） Learning to Rank
（五） Large scale Multimedia Search

四、心得

本會是亞洲微軟研究院所邀請的workshop，本實驗室因獲得MSRA Internet Services Invitation for Proposal Program而獲得邀請，並於Poster Session展示我們正在執行中的GeoWeb計畫，此計畫包含三位教授，藉由本次會議的交流互動，並且與其他學者交換意見，收獲甚多。出席國際性大型機構的會議對於做研究學者是一種莫大的鼓勵，並能增廣見聞同時吸收新知更能和研究同一領域之外國學者互相切磋討論。

五、建議與結語

筆者很感謝此次的補助。出席國際性學術會議對於做研究學者是一種莫大的鼓勵，並能增廣見聞同時吸收新知更能和研究同一領域之外國學者互相切磋討論。

六、攜回資料

（一）大會手冊
（二）微軟亞洲研究院各主要研究單位的宣傳資料

附錄七：

出席國際學術會議心得報告兩份

會議/訪問時間地點	第 5 屆一年一期網路計算與應用研討會 The 5th IEEE International Symposium on Network Computing and Applications (IEEE NCA06) 2006/07/24~2006/07/26, Boston, USA
會議名稱	第 5 屆一年一期網路計算與應用研討會(IEEE NCA 2006)
發表論文題目	Implementation of an Emergency Guiding System by Wireless Sensor Networks

會議/訪問時間地點	16th International Conference on Computer Communications and Networks, Aug. 13 - 16, 2007, Honolulu, Hawaii USA
會議名稱	ICCCN 2007
發表論文題目	Exploring Load-Balance to Dispatch Mobile Sensors in Wireless Sensor Networks

□ 赴國外出差或研習

□ 赴大陸地區出差或研習

■ 出席國際學術會議

□ 國際合作研究計畫出國

心得報告

計畫名稱	M4 無線網狀網路之網路規劃及資源分配問題	計畫編號	NSC 94 - 2219 - E - 009 - 004
報告人姓名	潘孟鉉	服務機構及職稱	國立交通大學資訊工程學系博士班學生
會議/訪問時間地點	第 5 屆一年一期網路計算與應用研討會 The 5th IEEE International Symposium on Network Computing and Applications (IEEE NCA06) 2006/07/24~2006/07/26, Boston, USA		
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發表論文題目	Implementation of an Emergency Guiding System by Wireless Sensor Networks		

一、參加會議經過：

這一次研討會的舉辦地點是位於美國波士頓(Boston)的 Boston Marriott Cambridge 飯店，會議中除了個人報告(Oral)外，還有邀請知名人士針對網路計算與應用的發展做專題演講。

我個人報告的時間是會議的第一天(當地時間 07/24)，這個 Session 是針對現在目前世界各國之研究機構目前正在發展之研究題目以及實做平台做討論的子會議，每個人約有 15 分鐘左右的時間可以報告自己所做的研究，並預留 5 分鐘的時間讓其他的聽眾得以詢問問題。在這個子會議中，大家談論的議題包括在無線感應網路中感應器的安全問題、在感應網路中資料查詢機制、分散式網路上資料存放擷取問題，參與此討論讓我對世界目前頂尖之研究題材有著更深入的了解。

二、與會心得：

這一次研討會主要探索的方向是集中在網路計算與應用(Network Computing and Application)的範疇，討論的議題包括有 Overlay 網路以及 P2P 網路的發展、無線網路(Wireless Networks)的發展及無線感應網路(Wireless Sensor Network)等等不同的領域，由於這些領域和我目前的研究習習相關，因此藉由聆聽別人對這些領域的研究報告，讓我了解到目前別人在這些領域中是朝哪些方向努力、並且發掘到哪些問題，相信這對我目前的研究有著極大的幫助。

此外，藉由這次出國的機會，讓我得以和不同國家的人們交談並交換心得，除了能夠加強自己的語文能力外，也增加了自己的國際觀。

三、建議與結語：

在這次的會議中，有幾個與會的人員於會後針對我所做的報告詢問相關的問題，而我也於會後去和別的報告者交換心得，我覺得這是一個很不錯的經驗，我會建議學校或相關研究單位除了鼓勵學生出國於國際會議報告之外，更應鼓勵學生積極與其它學者做交流，這樣應更能提升出國報告的收穫。

四、攜回資料：

NCA 2006會議手冊：記載本研討會會議流程、報告人員、地點、及報告題目等與研討會相關的會議手冊，其中並包含了會議中所有的論文集。

五、其他：無

出國報告書

撰寫時間: 96 年 8 月 20 日

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職　　別	教授/系主任		
出席國際會議名稱	ICCCN 2007		
到達國家及地點	Hawaii, 美國		
出國期間	自 96 年8 月14 日　迄　96 年 8 月 18 日		

內容	ICCCN 2007 於 2007 年 8 月 13 日至 8 月 16 日在 Hawaii, USA 召開，由 IEEE 贊助主辦，並出版之論文集。會議中有 keynote, Technical Session, 和多個 workshop 同時召開。Technical Sessions 以 parallel session 方式進行。有許多相關 wireless communication, mobile computing, location management, multicast, multiple access, energy-efficient protocols, power control, routing protocols, ad hoc network 的論文發表，會議的方向為: • Broadband Networking & Protocols • Networking Algorithms & Performance Evaluation • QoS Control & Traffic Modeling • Communications & Information Theory • Optical Networking • Peer-to-Peer & Grid Networking • Network Security • Signal Processing for Communications • Computer Architecture for Networking • Broadband Wireless Access • Multimedia Communications • Pervasive Computing & Mobile Networking • Sensor & Ad-Hoc Networking • Cross-Layer Design and Optimization • Wireless & Mobile Network Architecture • Emerging Technology & Standards • Parallel & Distributed Computing 本會可謂一個極大型的國際會議，會中有來自多個國家，約 300 位學者與會。本會議可稱計算機網路不錯的會議，論文接受率約為 33%。會中並附數幾個 workshop，如 WiMAN, IMAP, PMECT, DSS 等。 筆者發表的論文為，會中並與多位與會人員交換意見: Y.-C. Wang, W.-C. Peng, M.-H. Chang, and Y.-C. Tseng, "Exploring Load-Balance to Dispatch Mobile Sensors in Wireless Sensor Networks", ***Int'l Conf. on Computer Communication and Networks (ICCCN)***, 2007 本人在會中並與其它學者交換意見，收穫甚多。出席國際性學術會議對於做研究學者是一項重要活動，能增廣見聞、吸收新知、互相切磋討論。