Cross Layer Optimization in 4G Wireless Mesh Networks

Imam Mahmud Taifur Rahman AL-Wazedi

A Thesis

in

The Department

of

Electrical and Computer Engineering

Presented in Partial Fulfillment of the Requirements for

The Degree of Doctor of Philosophy at

Concordia University

Montreal, Quebec, Canada

2010

Your file Votre référence
ISBN: 978-0-494-67376-8
Our file Notre référence
ISBN: 978-0-494-67376-8

ABSTRACT

Cross Layer Optimization in 4G Wireless Mesh Networks

Imam Mahmud Taifur Rahman AL-wazedi

Concordia University, 2010

Wireless networks have been rapidly evolving over the past two decades. It is foreseen that Fourth generation (4G) wireless systems will involve the integration of wireless mesh networks and the 3G wireless systems such as WCDMA. Moreover their wireless mesh routers will provide service to wireless local networks (WLANs) and possibly incorporate MIMO system and smart admission control policies among others. This integration will not only help the service providers cost effectiveness and users connectivities but will also improve and guarantee the QoS criteria. On the other hand, cross layer design has emerged as a new and major thrust in improving the quality of service (QoS) of wireless networks. Cross layer design involves the interaction of various layers of the

network hierarchy which could further improve the QoS of the 4G integrated networks.

In this work we seek new techniques for improving the overall QoS of integrated 4G systems. Towards this objective we start with the local low tier WLAN access. We then investigate CDMA alternatives to the TDMA access for wireless mesh networks. Cross layer design in wireless mesh networks is then pursued.

In the first phase of this thesis a new access mechanism for WLANs is developed, in which users use an optimum transmission probability obtained by estimating the number of stations from the traffic conditions in a sliding window fashion, thereby increasing the throughput compared to the standard DCF and RTS/CTS mechanism while maintaining the same fairness and the delay performance.

In the second phase we introduce a code division multiple access/Time division duplex technique CDMA/TDD for wireless mesh networks, we outline the transmitter and receiver for the relay nodes and evaluate the efficiency, delay and delay jitter performances. This CDMA based technique is more amenable to integrating the two systems (Mesh networks and WCDMA or CDMA 2000 of 3G). We compare these results with the TDMA operation and through analysis we prove that the CDMA system outperforms the TDMA counterparts.

In the third phase we proceed to an instance of cross layer optimized networks, where we develop an overall optimization routine that finds simultaneously the best route and the best capacity allocation to various nodes. This optimization routine minimizes the average end to end packet delay over all calls subject to various constraints. In the process we use a new adaptive version of Spatial TDMA as a platform for comparison purposes of the MAC techniques involved in the cross layer design. In this phase we also combine CDMA/TDD and optimum routing for cross layer design in wireless mesh networks. We compare the results of the CDMA/TDD system with results obtained from the STDMA system.

In our analysis we consider the parallel transmissions of mesh nodes in a mesh topology. These parallel transmissions will increase the capacity resulting in a higher throughput with a lower delay. This will allow the service providers to accommodate more users in their system which will obviously reduce the cost and the end users will enjoy a better service paying a lower amount.

ACKNOWLEDGEMENTS

This academic endeavour would not have been possible without the help and support of many individuals. In my whole life, I met a lot of people of different cultures and often times of the same culture in different places. Many of them became good friends; while others became a part of my family. I wish to acknowledge those people who were really close to me and gave me full mental support during my research.

Among them, first, I want to give my heartiest thanks to my supervisor, Dr. Ahmed K. Elhakeem for his encouragement, guidance, constructive advice and evaluation of my work. Moreover, without his technical support I would not have had such an achievement for which my family and friends say "Bravo".

I also wish to acknowledge with many thanks Dr. A. Agarwal, Dr. Yousef R. Shayan and Dr. Adel M. Hanna for their suggestions which helped me a lot to gear up my research.

My thanks are also due to my friends Mazharul Hoque, Abul Hossain, Yousef Khan, Nasim Ali Khan, Shaha Babu, Syed Kamal, Enna, Ayman, Hassan, Atik, Anupom, Kaushik, Reza, Bappi and Beajon who provided practical and moral help.

DEDICATION

To my parents Dr. Wazed Ali and Khaleda Parveen Banu and my wife Moushumi Choudhury.

Table of Contents

List of Figures

List of Symbols

p	Transmission probability of each node
M	Number of active stations
P_s	Probability of successful transmission
P_o	Probability of an idle slot
P_{cc}	Probability of unsuccessful transmission
P_c	Probability of correct packets
W	Average length of each idle period
η	Throughput
I	Average number of cycles
C	Number of collisions
λ	Offered Traffic
μ	Service rate
C_s	Scrambling Code
w_i	ith Walsh funtion
N_p	Number of Walsh functions
d_w	Bits indicating number of Walsh function
d_i	Bits indicating initial shift of C_s code
T_b	Bit duration
T_c	Chip duration
C_d	Short data pilot code
WW	Bandwidth
Nb	Number of Bits/Packet
Rb	Bit rate
PG	Processing Gain
r	Turbo Code rate
α_l	Probability of Packet generation
β_l	Service probability

a_{ij}	State transition probabilities
B	Maximum number of packets simultaneously sent in parallel
U	Number of Users
K	Maximum Number of States
PP	Probability that the user's transmitting certain Number of Packets
φ_n	Probability of packet success
n	Total number of parallel packets transmitted on the channel
k	Number of packets transmitting in parallel
E_b/N	Thermal Noise
P_{b_n}	Probability of bit error given n parallel packets on the channel
ξ	Steady state distribution of the values of packets in the queue
l	Link number
m_i	Number of links in ith path
b	Probability of sum of packets in queue
n_p	The value of the number of paths available
$\alpha\alpha$	Probability that the most crowded path taken by the control packets have a certain number of packets
$\overline{\lambda}$	The average delay for the call establishment
σ_θ	The delay jitter for the control packets
γ	Probability that the least crowded path taken by the Data packets have a certain number of packets
$\overline{D}$	The average delay for the data Packets
σ_ε	The delay jitter for the data packets
σ_i	The delay jitter for the TDMA System
$\eta\eta$	Network Efficiency
II	Number of Calls
X	Control variable
CC'	Total number of calls generated by all nodes
c	Call Number

R_c	Total number of routes for a certain call
θ_c'	Call traffic intensity of call number c
θ	Total probability of a node making a call
P_{c_l}	Packet success probability of link l
T_F	Frame size
ζ	Slots occupied by links
F_{dl}	Synchronization Time for link l
F_{Dl}	Synchronization time plus packet time for link l
R	Route Number
ρ	Ratio of Traffic and the service rate
F	File size in packets
L_c	Set of consecutive links that are used by a certain call to establish a connection according to the route decision decided by the optimum routing policy
σ	Delay jitter for STDMA system
L_c'	Optimum set of links of classic shortest path route of call c

List of Abbreviations

MAC	Medium Access Control
WLAN	Wireless Local Area Networks
WMN	Wireless Mesh Network
WIMAX	Worldwide Interoperability for Microwave Access
DCF	Distributed Coordination function
CDMA	Code division Multiple Access
TDMA	Time division Multiple Access
FDMA	Frequency division Multiple Access
TDD	Time division duplex
DL	Downlink
UL	Uplink
RTS	Request to send
CTS	Clear to send
DIFS	DCF interframe space
SIFS	Short interframe space
MAN	Metropolitan Area network
FDD	Frequency division duplex
MSH-NCFG	Mesh Network Configuration
MSH-NENT	Mesh Network Entry
BS	Base Station
SS	Subscriber Station
NAV	Network Allocation Vector
CW	Contention Window
ACK	Acknowledgment
RREQ	Route Request
RREP	Route Reply
AODV	Ad hoc On Demand Distance Vector
CRC	Cyclic Redundancy Check
TX	Transmitter

CSMA-CA	Carrier Sense Multiple Access-Collision Avoidance
AWGN	Additive white Gaussian noise
PHY	Physical Layer
CID	Connection Identification Number
OFDM	Orthogonal frequency division multiple access
STDMA	Spatial Time division Multiple Access
AP	Access Point

List of Tables

CHAPTER 1

INTRODUCTION

1.1 4G Wireless System

For the past few decades wireless networks has attracted both academic and industrial interest. As an integral part of such wireless networks, WLANs such as IEEE 802.11 are offering high bandwidth radio communication to low tier users. The eventual need for convergence of these with high tier backbone like platforms has led to the outgrowth of broadband wireless access and to the standardization of a wireless MAN air interface. On the other hand, cellular networks such as WCDMA can achieve higher data rates comparable to these provided by 2G GSM system. It is foreseen that in fourth generation 4G, there will be integration among WLANs, Mesh Networks and WCDMA cellular system [1].

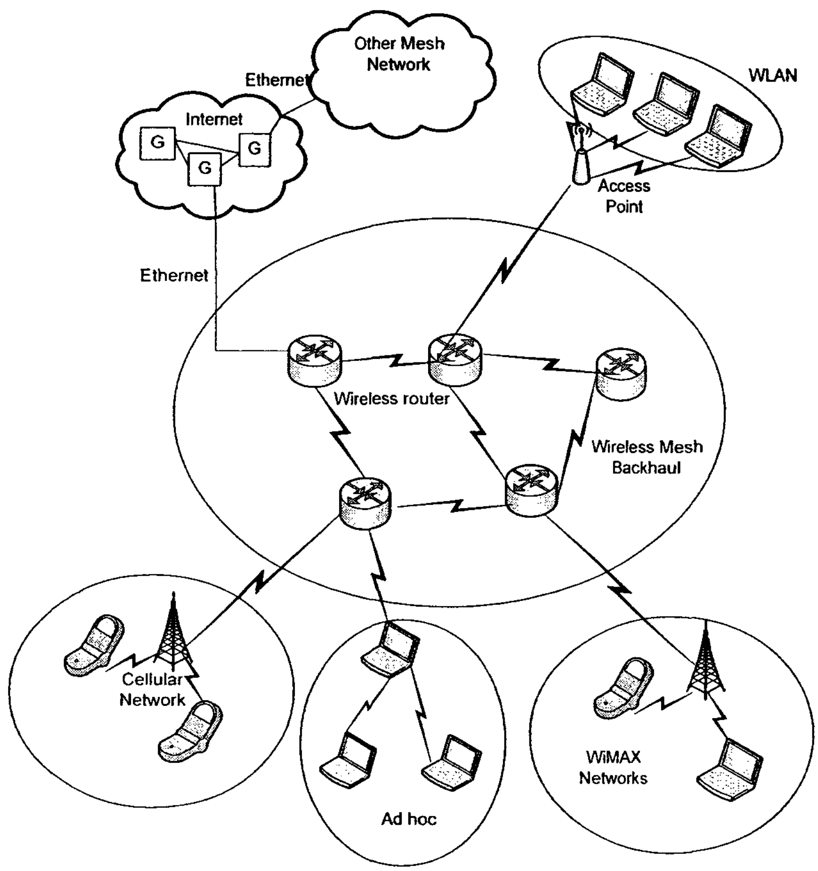


Figure 1.1-1 Future 4G Network Scenario

Figure 1.1-1 shows a typical 4G Network scenario where WLAN and cellular networks are interconnected with the wireless mesh backhaul.

Wireless mesh networking is relatively new and promising key technology for next generation wireless networking. Mesh networks are expected to replace the existing wired infrastructure by providing higher communication QoS, accommodation of new services such as video streaming, gaming etc [2, 3]. Mesh networks are composed of wireless routers [4]. These routers are able to relay packets from other nodes without direct access to their destinations. The destinations can either be an internet gateway or a mobile device served by another AP in the same or different mesh networks. The routers are designed in such a manner that they meet the QoS requirement for wireless networks.

Most of the current work on wireless and ad hoc network protocol analysis is based on layered approach. Layered architectures have traditionally culminated [5, 6] in easier integration of communication equipment from different vendors and cost effective design of interfaces while providing transparency between the layers. So they have become the de facto standard for wireless systems. However, in wireless mesh networks the spectral reuse and the unstable channel characteristics have made the layered approach inefficient for the overall system performance [4, 7]. This is why cross layer design for improving the network performance has been a focus of much recent work. In a cross-layer paradigm, the joint optimization of control over two or more layers can yield significantly improved performance compared to separately optimized layering.

1.1.1 WIMAX Networks (IEEE 802.16)

The IEEE 802.16 physical layer operates at both 10-66 GHz and 2-11 GHz with data rates that depend on bandwidth and modulation techniques. The use of OFDM (Orthogonal Frequency Division Multiplexing) makes the standard capable of high speed data connections for both fixed and mobile service stations. The IEEE 802.16 MAC protocol defines both frequency division duplex (FDD) and time division duplex (TDD) for its connections. The architecture is made of two components, a base station (BS) and a number of service stations (SS) with two directions of communication. The first one is the downlink (DL) transmission from the BS to the SSs, whereas the second one is the uplink (UL) direction. The UL channel is common to all nodes and is slotted via TDD method on a demand basis for multimedia data.

WiMAX technology has a high throughput which is capable of delivering backhaul for enterprise campuses, Wi-Fi, hotspots, and cellular networks. Based on the traffic characteristics of such a network, it is possible to cover the same area as the cellular base stations do today or even more.

1.1.2 UMTS/WIMAX Interworking

In [1], various interworking architectures has been proposed for the wireless networks. Different access networks (AN) such as 3G networks, WIMAX, and WLANs can be owned by different service providers or by the same provider. The wireless access gateways (WAGs) of WLAN, WIMAX, and 3G networks are connected to different proxy-call session control function (P-CSCF) servers in IP multimedia subsystems (IMS) via the internet. The WIMAX AN consists of WIMAX base stations, which are controlled by the WIMAX base station controller (WBSC). Several WBSCs are controlled by one WIMAX network controller (WNC). The WNC is connected to WAG to provide WIMAX users with 3G and IMS services. This is how the interworking is done between such networks.

1.2 Layered Architecture of OSI Model

The OSI [5] (open systems interconnection) divides the communication process into seven layers and provides the architecture to define the whole communication system. This was developed to generalize the standards for each layer. The seven layers are termed as physical layer, data link layer, network layer, transport layer, session layer, presentation layer and application layer. The organization of such layers has proved its cost effectiveness in wired networks [7].

1.3 Cross-Layer Architecture

It has been proved in recent literature [7], [8], [9] that the layered architecture does not perform well in wireless networks. For replacement of this layered architecture, researchers opted to the cross layer design for wireless networks [9],[10]. The layered architecture forces direct communication between the adjacent layers. As a result the protocols designed must respect the rules of the OSI reference model. This means that a higher layer protocol only makes use of the services at the lower layers and does not know the details of how the service is provided. In cross layer architectures [11], the protocol design encourage the exchange of few primitives between the layers so as to improve the overall performance of a group of layers. In cross layered architecture there are no hard and fast rules for the adjacent layers communications. Meaning, non adjacent layers can communicate and exchange information with each other. Cross layer does not violate the transparency etc. above to a great extent. It is basically an added benefit.

1.4 Literature Review

1.4.1 Performance Improvement of WLAN

Wireless local area networks (WLANs) have been widely deployed for the past decade. Their performance has been the subject of intensive research. In [12], [13] an improvement of throughput and fairness is shown by optimizing the backoff without estimating the number of active nodes in the network. In [14] a MAC layer based WLAN technique was introduced. Higher priority to access points was given so as to improve the throughput and the channel utilization. To improve the channel utilization backoff in [15] was tuned based on collision avoidance and fairness. In [16] a DCF model was proposed where the arrival and the service of the packets in the queue are controlled to improve the throughput and delay performance.

Cali in [17] pointed out that depending on the network configuration, DCF may deliver a much lower throughput compared to the theoretical limit. Cali derived a distributed algorithm that enables the stations to tune its backoff at run time where a considerable improvement in the throughput is shown. In [18] a contention based MAC protocol named fast collision resolution is presented where the backoff was also utilized. A model named DCF+ in [19] was proposed which uses the backoff to improve the fairness. In [20] a performance model for the IEEE 802.11 WLAN in ad hoc mode was proposed where a considerable improvement in throughput and delay performance is shown.

It is evident that the throughput, delay, fairness performances are improved by tuning the backoff in different scenarios considered in[12] - [20].

RTS/CTS mechanism with NAV is used to solve the hidden terminal problem. In [21] Khurana proposed the concept of *Hearing graph* to model the hidden terminals in static environment and analyzed the performance. Also in [22] Fullmer, proposed a three way handshaking technique to solve the hidden terminal problems of single channel WLANs. However, the first phase of our thesis does not emphasize on the hidden terminal problem but contributes on a modification of the standard DCF and the standard RTS/CTS mechanisms.

In this phase, table driven DCF and table driven RTS/CTS systems are proposed, which are similar to IEEE 802.11 (both DCF and RTS/CTS) standards without the use of the exponential backoff. In table driven DCF and table driven RTS/CTS the nodes estimate the number of active stations and transmit with an optimum probability measured from the traffic conditions (by sensing the channel) in a sliding window fashion, as will be detailed shortly. Simulation results show that our systems outperform the standard in terms of throughput while maintaining same delay and fairness.

1.4.2 Relaying Mechanism in Wireless Mesh Networks

As an example of mesh networks, WIMAX (IEEE 802.16) have been a subject of greater interest involving the classic access techniques such as TDMA/TDD and TDMA/FDD. [23] describes the general PHY and MAC layers and provides views on several cross layer issues related to WIMAX, especially the OFDMA/TDD system. In wireless metropolitan area networks (MAN), OFDMA PHY layer is based on OFDM modulation. TDD systems use the same frequency band for downlink and uplink and the frames are divided into the DL sub frames and UL sub frames in the time domain. WIMAX is capable of using both FDD and TDD operations. To provide highest transport efficiency in broad band networks, time division duplex (TDD) is preferred over FDD because it offers more flexibility in changing the UL/DL bandwidth ratio according to the dynamic traffic pattern [23]. The MAC protocol of IEEE 802.16 is connection oriented and each connection is identified by connection identification number (CID), which is given to each SS (subscriber stations) in the initialization process. In this protocol the SS use TDMA on the uplink and receives back from BS in a specific time slot [24].

In IEEE 802.16 Mesh mode, Mesh Network Configuration (*MSH-NCFG*) and Mesh Network Entry (*MSH-NENT*) messages are used for advertisement of the mesh network and for helping new nodes to synchronize and to join the mesh network. Active nodes within the mesh periodically advertise MSH-NCFG messages with Network Descriptor, which outlines the basic network configuration information such as BS ID number and the base channel currently used. A new node that plans to join an active mesh network scans for active networks and listens to MSH-NCFG message. The new node establishes coarse synchronization and starts the network entry process based on the information given by MSH-NCFG. Among all possible neighbors that advertise MSH-NCFG, the joining node (which is called *Candidate Node* in the 802.16 Mesh mode terminologies) selects a potential *Sponsoring Node* to connect to. A Mesh Network Entry message (MSH-NENT) with NetEntryRequest information is then sent by the Candidate Node to join the mesh [25].

IEEE 802.16 is a centrally controlled protocol but can also operate in Mesh mode. In the first case the BS controls the uplink bandwidth allocation and the SS's request transmission opportunities in the uplink channel. In the second case traffic can be routed through SS's utilizing a distributed scheduling algorithm. One node takes the role of the Mesh BS. Many researchers have concentrated their efforts towards the distributed scheduling of mesh networks. In [26] an efficient centralized scheduling algorithm in wireless mesh networks has been proposed where special attention to relay function of the mesh nodes in the transmission tree was given. In [27] a distributed scheduling algorithm is proposed where the links are assigned to slots in each frame and during each slot a number of non-conflicting links can transmit simultaneously using TDMA. In [28], an algorithm for routing and channel assignment for throughput maximization was proposed, which was also the basic objective of [27] and [28]. In [29] a joint routing and link scheduling was proposed to support high data rates for broadband wireless multi-hop networks to increase the throughput and reduce the delay. In [30], a joint routing and scheduling in TDMA based wireless mesh networks have been proposed for real time traffic. In [31] a shortest path routing protocol was proposed where interference free distributed TDMA is used to guarantee the bandwidth demand for nodes in a multi-channel wireless mesh network. In [32] a scheduling technique to access the control channels in a TDMA based mesh network was proposed where a single time slot can be reused by two nodes if they do not interfere each other. In [33] on demand TDD algorithm was proposed for achieving high channel utilization and fairness algorithm in a multi-hop wireless mesh network. These scheduling algorithms are based on TDMA/TDD system. The IEEE 802.16 MAC protocol defines both FDD and TDD for its connections. On the other hand, CDMA has been proposed as the access technique in wireless mesh networks. El-atty [34] proposed a cross layer packet scheduling technique utilizing the information delivered from the PHY layer in wireless CDMA based mesh networks. In [35], an extended link layer model to find the delay bound for a 3 hop mesh network is calculated in a probabilistic approach. In this phase of our work we introduce the transceiver and access techniques of a new CDMA/TDD based mesh network and analyze its performance. With the help of turbo coding, the new network uses parallel transmission from the SS's so as to improve the network QoS. Moreover we make comparisons with the TDMA based systems in terms of network efficiency, delay and delay jitter. Results show that CDMA system outperforms its TDMA counterparts. On top of this the delay performance of CDMA/TDD system outperforms the delay calculated in [35] and [36].

1.4.3 Cross layer Design in Wireless Mesh Networks

Mesh networks have been the subject of greater interest. Most of the efforts are towards the cross layer design where different layers such as MAC, PHY and network are simultaneously optimized. One of the conflict-free MAC protocol amenable to wireless mesh network is Spatial TDMA [37]. In STDMA the frame consists of a number of slots with each slot assigned to transmissions of a set of non conflicting links (e.g. situated far away from each other). STDMA has become popular due to its capability to guarantee Quality of Service (QoS). For example improving the delay performance [38], [39],[40]. Along with this, effective routing techniques are indispensible to satisfying QoS [11], [34],[41].

In [42], a resource assignment scheme to accommodate asymmetric traffic for connection-less services CDMA/ TDD cellular system is proposed. The proposed scheme controls the TDD boundary on the estimated uplink and downlink traffic. In [43], the authors consider a slot allocation to accommodate maximum traffic while satisfying the interference constraints.

In the past few years, combined routing, resource allocation and optimization in wireless networks were investigated [34], [32], [44],[30]. The authors in [38] investigated potential routing techniques for ad hoc networks, while separately assigning the resources at the MAC level.

In cross layer design it is necessary to design medium access control and the network layer routing protocols together, where the MAC determines when a node transmits and routing finds the efficient path for the destined packets [45], [46], [47]. This design will enable the nodes to have guaranteed QoS.

In our approach, the routing in the network layer is optimized utilizing information received from the MAC and PHY layers which is effectively a cross layer design. We use STDMA approach [37] for MAC allocation which is termed as uniform STDMA. Moreover, we have developed an adaptive STDMA technique which performs better than the uniform counterpart. This new adaptive STDMA allows the simultaneous operation of routing and MAC resource allocation based on the information received from the PHY layer. In addition, in this third phase of our work, we combine CDMA/TDD and optimum routing for cross layer design in wireless mesh networks and compare to STDMA based approach. In the second phase we have already proved by analysis that, CDMA/TDD outperforms the TDMA counterparts. Consequently, we started looking in the third phase for new MAC approaches which can compete with our CDMA based system.

1.5 Research Contributions

The goal of this thesis is cross layer optimization in 4G Wireless Mesh Networks.

The main contributions of this thesis are listed below:

- A new access mechanism for WLANs named table driven DCF and table driven RTS/CTS systems was proposed, which is similar to IEEE 802.11 (both DCF and RTS/CTS) standards, without the use of the exponential backoff.
- A transmitter, receiver and a queuing model for the analysis of CDMA/TDD system in wireless mesh networks was proposed.
- For comparisons an analytical model for a cross layer design using STDMA and optimum routing for wireless mesh networks was investigated.
- Finally, optimum cross layered routing design for wireless mesh networks using CDMA/TDD was developed.

1.6 Organization

The organization of this thesis is given as follows.

In Chapter 2, we discuss about some recent standards related to the wireless networks which will be useful for the subsequent development of this thesis.

In Chapter 3, we propose the new access mechanism for WLANs and compare the results with the current standards. This access mechanism can be extended for the communication among the mesh nodes which are to be investigated in the chapters latter on. The end to end delay and success probabilities will depend on the performance of the user access hop (WLANs as well as wireless mesh backbone networks). These make the investigation of both hops necessary for the overall 4G operation. For overall effectiveness of 4G networks, both local and backbone constituting networks should be optimized.

In Chapter 4, we propose a new relaying mechanism for wireless mesh networks. We also analyze its performance and compare it with the classic access technique.

In Chapter 5, we develop an adaptive load balancing mechanism for the network layer in cross layer architecture for wireless mesh networks using STDMA. This mechanism is used as a comparison platform. Moreover we propose a cross layered routing design for wireless mesh networks using CDMA/TDD approach and compare the results with the results obtained from the STDMA approach.

Finally, in Chapter 6 we bring up the conclusion and give proposals for future work.

CHAPTER 2

Preliminaries

In this chapter, some preliminaries in relation to the subsequent development of the thesis are introduced. These include the MAC protocols of WLAN, WIMAX networks and cross layer concepts. Moreover this chapter includes a description of star based STDMA system in multihop wireless networks.

2.1 IEEE 802.11 Wireless LAN

Figure 2.1-1 shows one of many transmission scenarios possible with the IEEE 802.11 DCF mode. In this mode a node with a packet to transmit initializes a backoff timer with a random value selected uniformly from the range [0, CW-1], where CW is the contention window in terms of time slots. After a node senses that the channel is idle for an interval called DIFS (DCF interframe space), it begins to decrease the backoff timer by one for each idle time slot observed on the channel. When the channel becomes busy due to other nodes transmission activity the node freezes its backoff timer until the channel is sensed idle for another DIFS. When the backoff timer reaches zero, the node begins to transmit. If the transmission is successful, the receiver sends back an Acknowledgement (ACK) after an interval called the SIFS. Then, the transmitter resets its CW to CW_{min}. In case of collisions the transmitter fails to receive the ACK from its intended receiver within the specified period, it doubles its CW subject to maximum value CW_{max}, chooses a new backoff timer, and starts the above processes again.

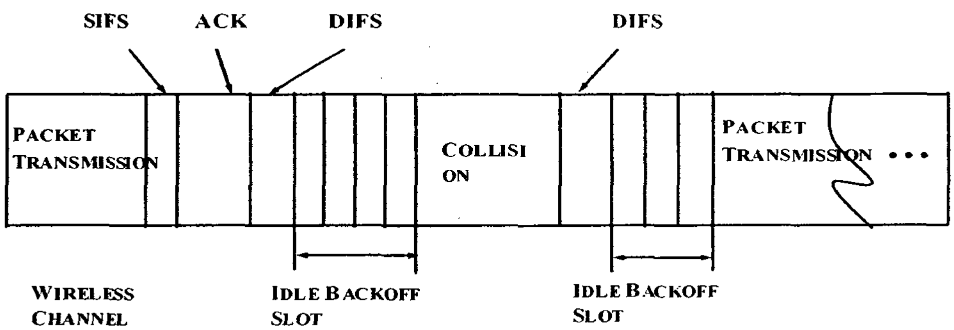


Figure 2.1-1 IEEE 802.11 MAC mechanism

In 802.11, DCF also provides a more efficient way of transmitting data frames that involves transmission of special short RTS and CTS frames prior to the transmission of actual data frame. As shown in Figure 2.1-2, an RTS frame is transmitted by a node, which needs to transmit a packet. When the destination receives the RTS frame, it will transmit a CTS frame after SIFS interval immediately following the reception of the RTS frame. The source station is allowed to transmit its packet only if it receives the CTS correctly. Note that all the other stations are capable of updating their knowledge about other nodes transmission duration by receiving a certain field in RTS, CTS, ACK, and packets transmission called network allocation vector (NAV). This helps to combat the *hidden terminal* problem. In fact, a node that is able to receive the CTS frames correctly, can avoid collisions even when it is unable to sense the data transmissions directly from the source station. If a collision occurs with two or more RTS frames, much less bandwidth is wasted when compared with the situations where larger data frames in collision, thus justifying the case for RTS, CTS mode of operation[13].

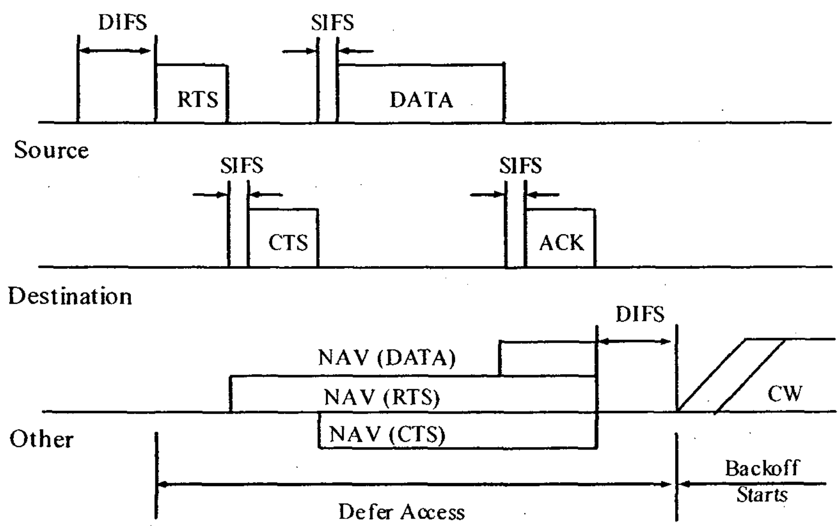


Figure 2.1-2 RTS/CTS access mechanism in DCF

2.2 Backbone Mesh Networks

The wireless mesh backbone can be built using various types of radio technologies, in addition to mostly used IEEE 802.11 technologies [48]. Figure 2.2-1 shows a general scenario of a wireless mesh backbone structure. With the help of gateways of internet, these routers can be connected to the internet which is often referred as *infrastructure meshing*. This provides the backbone for clients and enables the interworking of WMNs with other wireless technologies via gateway or bridges. Through Ethernet interface these clients can be connected to the mesh routers by Ethernet links. Clients with same radio technology can communicate directly with the mesh routers.

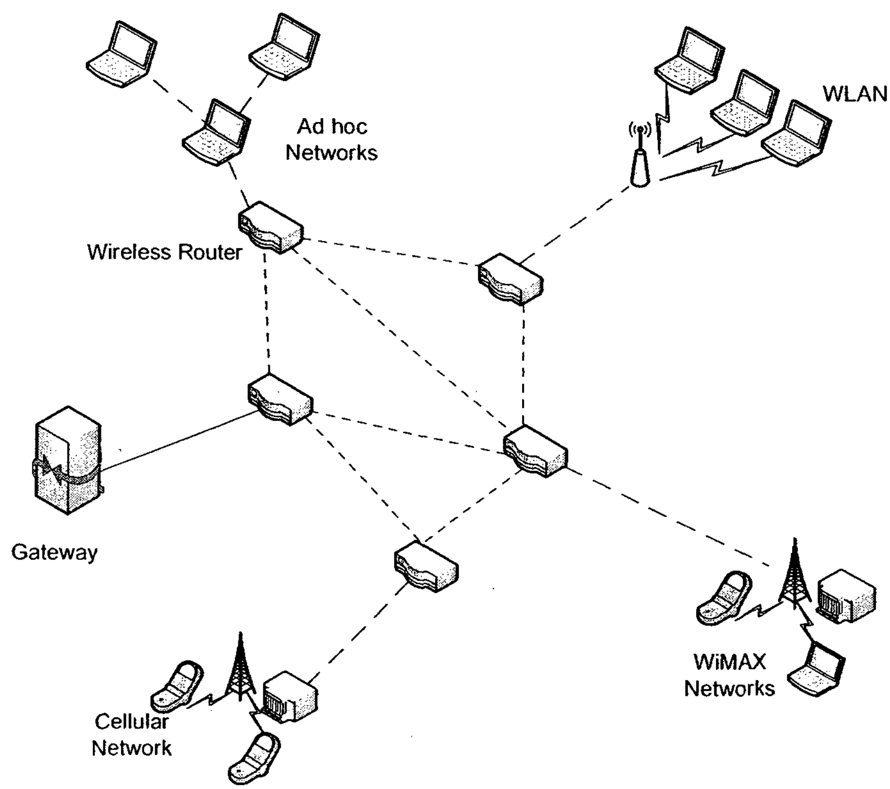


Figure 2.2-1 Wireless Mesh Backbone Scenario

However with different radio technology clients must communicate with their base stations that have Ethernet connections to mesh routers. The radio technology involves the design in higher layer protocols such as MAC and routing protocols.

The performance of WMNs depends on reliable mesh connectivity. Efficient MAC and routing protocols can significantly improve the performance of WMNs.

2.2.1 WIMAX MAC Protocol Overview

According to WIMAX standard, the MAC protocol is connection oriented and each Service station (SS) is provided with a Connection Identification number (CID) at the initialization process. The transmissions are divided by either TDD or FDD method. In the downlink direction (DL) the connection is multicast. In the uplink (UL) the SSs transmit packets using Time division multiple access (TDMA) which is always unicast.

WIMAX standard (IEEE802.16) is a centrally controlled protocol. However it can operate in mesh mode. In the mesh mode the traffic can be routed through SSs and use a distributed scheduling algorithm [49]. In this case one node takes role of the Mesh Base station (BS).

The above MAC standard can be applied to the wireless mesh backbone. However, other MAC protocols are proposed in the literature[50] for the wireless mesh backbone. In section 2.2.2 we describe elaborately one of such MAC protocol and its queue distribution.

2.2.2 An Alternative MAC Scheme and its Queuing Model

In [49], a collision free MAC scheme has been proposed for a wireless mesh back bone. According to this protocol a chain topology with six routers is considered [49] shown in Figure 2.2-2.

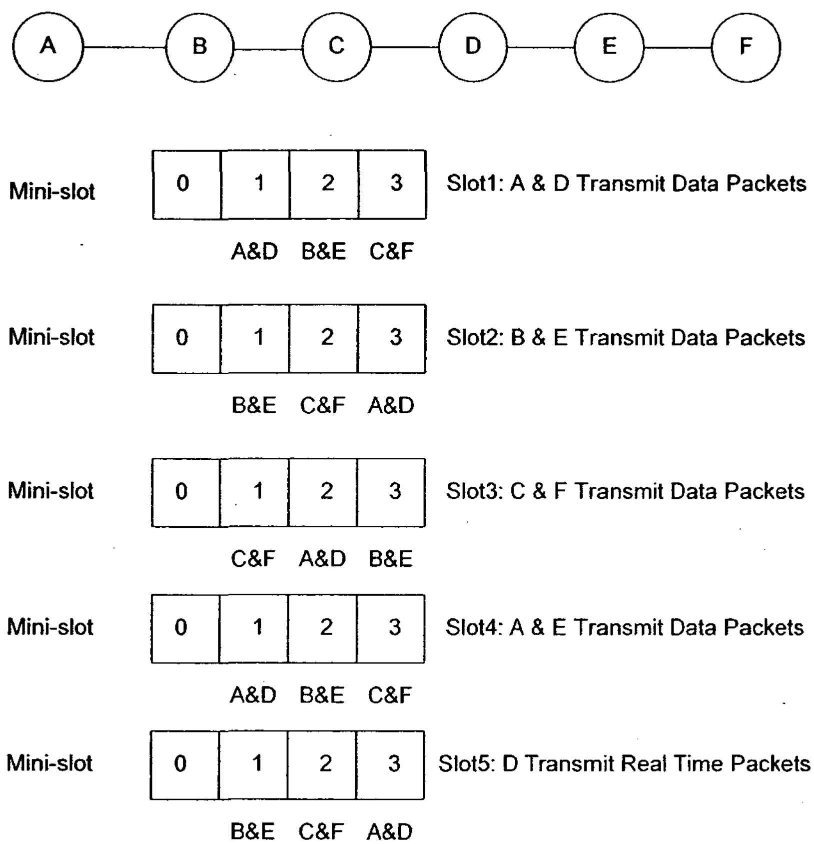


Figure 2.2-2 Operation of a MAC Scheme

According to [49], routers A and D are assigned to mini-slot 1, routers B and E are assigned to mini slot 2, routers C and F are assigned to mini-slot 3, and mini-slot 0 is reserved for real time traffic. At the beginning all routers have data packets to transmit. For slot 1, since no router have real-time packet, mini-slot 0 will remain idle. So routers A and D will sense idle channel and send a jamming signal at mini-slot1. Other routers B, C, E and F will sense a busy channel and differ their transmissions. As a result router A and D will transmit in mini-slot 1 without any collision. At the end of each slot, all the routers rotate their mini-slot indices corresponding to Figure 2.2-2. For slot 2 routers B and E are associated with mini-slot 1, and for slot 3, routers C and F are associated with mini-slot 1. Routers B and E will transmit their packets at slot 2 and C and F will transmit at slot 3. At the end of slot 3, the routers A, B and E has packets to transmit and router D does not. At slot 4, at mini slot 1, router A will send the jamming signal as it is assigned to this slot. Router D will remain silent in this period as it has no packet to transmit. As a result router E will sense an idle channel at mini-slot 0 and 1. So router E will transmit its packet in slot 4. At the end of slot 4, routers B and F have data packets and router D has a real time packet. For slot 5, router D will first send the jamming signal at mini-slot 0 which can be sensed by router B and F. For this reason router B and F will not transmit their jamming signal. After sensing an idle channel at mini-slot 1 and 2, router D will transmit the jamming signal again at mini-slot 3. As a result D transmits its real time packets at slot 5.

For the above spaced based TDMA protocol [49], a queuing model is proposed where a simple case is analyzed. In this model, the voice and video call arrivals at each node are assumed to be independent and follow Poisson's process and the call duration has an exponential distribution.

The average arrival rates of voice and video calls at each node in Figure 2.2-2 are denoted as λ_o and λ_g. The average call durations for the voice and video are denoted as $1/\mu_o$ and $1/\mu_g$. The maximum number of allowable voice and video calls are denoted as No and Nv. It is considered that, when one video call is served the maximum numbers of acceptable voice calls are denoted by No-m, since m voice calls are equivalent to 1 video call. The ratio of occupied slots for a video call to a voice call is given by m.

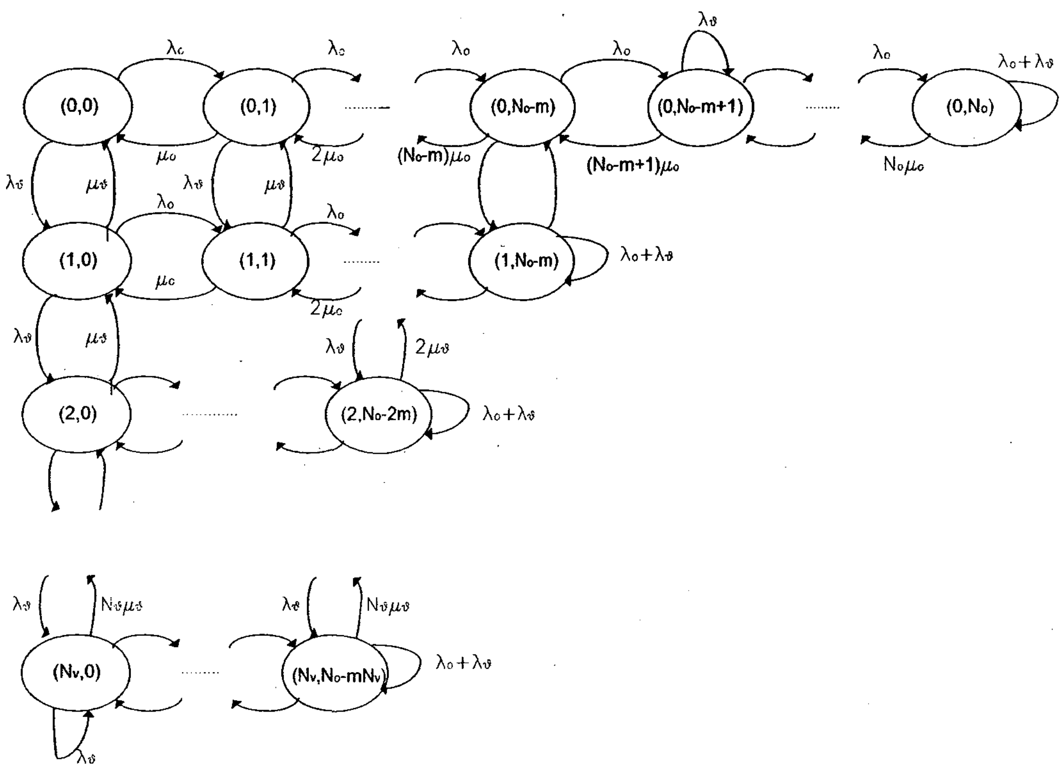


Figure 2.2-3 The state transition diagram

This ratio m is assumed to be more than 1 as the video calls are provided with more slots than the voice calls. $p_{i,j}$ is the joint probability (i.e. the steady state distribution for the number of voice calls and the video calls) where i is the number of video calls and j is the number of voice calls served. The balance equations for the two dimensional space of Figure 2.2-3 are given below

$$(\lambda_o + \lambda_g)p_{0,0} = \mu_o p_{0,1} + \mu_g p_{1,0} \qquad i = 0, j = 0;$$
$$(\lambda_o + \lambda_g + j\mu_o)p_{0,j} = \lambda_o p_{0,j-1} + (j+1)\mu_o p_{0,j+1} + \mu_g p_{1,j} \qquad i = 0, 1 \le j \le N_o - m;$$
$$(\lambda_o + j\mu_o)p_{0,j} = \lambda_o p_{0,j-1} + (j+1)\mu_o p_{0,j+1} \qquad i = 0, N_o - m + 1 \le j \le N_o - 1;$$
$$(\lambda_o + \lambda_g + j\mu_g)p_{i,0} = \mu_g p_{i-1,0} + (i+1)\mu_g p_{i+1,0} + \mu_o p_{i,1} \qquad 1 \le i \le N_g - 1, j = 0;$$
$$(\lambda_o + \lambda_g + j\mu_g + j\mu_o)p_{i,j} = \mu_g p_{i-1,j} + \lambda_o p_{i,j-1} + (j+1)\mu_o p_{i,j+1} + (i+1)\mu_g p_{i+1,j} \qquad 1 \le i \le N_g - 1, 1 \le j \le N_o - m(i+1);$$
$$(\lambda_o + j\mu_g + j\mu_o)p_{i,j} = \mu_g p_{i-1,j} + \lambda_o p_{i,j-1} + (j+1)\mu_o p_{i,j+1} \qquad 1 \le i \le N_g - 1, N_o - m(i+1) + 1 \le j \le N_o - im - 1;$$
$$(j\mu_g + j\mu_o)p_{i,j} = \mu_g p_{i-1,j} + \lambda_o p_{i,j-1} \qquad 1 \le i \le N_g - 1, j = N_o - im$$

The balance equations are necessary for further analysis of the average delay and the delay variance as in [49]. In Chapter 4, Code Division Multiple Access (CDMA) will be utilized as a MAC access technique alternative to these TDMA techniques for WMNs. Queuing models will be similarly utilized among other analysis tools.

2.3 Cross layer Design in Wireless Mesh Networks

Cross layer Design in Wireless mesh networks (WMNs) has been an important research issue for the past few decades. The MAC access and the routing capability of WMNs have put them in an advantageous position than the WLANs. In WMNs, the data packets can be routed through different nodes towards the destination. The WMNs operation combines both MAC layer and network layers. As a result, lot of researchers have focused their research on the joint network and MAC allocation schemes [34], [32], [44],[30]. The authors in [38] investigated potential routing techniques for ad hoc networks, while separately optimizing resources at the MAC level. At the MAC level [38], a conflict free MAC scheme named Spatial Time division Multiple Access (STDMA) was deployed for WMNs [5]. While in network layer, minimum hop routing algorithm (MHA) was utilized. MHA is one of the shortest path routing algorithms.

On the MAC level, STDMA [37] was presented which is a generalization of the TDMA protocol for multihop networks. A frame consists of slots which are assigned to some non-conflicting transmission links of the network. To check the operation of STDMA we consider the network shown in Figure 2.3-1.

An arc (link) denotes the presence of an available direct communication channel between two nodes e.g. with link 1, node 1 can transmit directly to node 2, while node 6 and node 2 cannot communicate directly. The transmission commences with the lowest link number (link 1) where node 1 transmits to node 2 in the first clique (time slot). The first slot is assigned to the links in the first clique. Now we investigate other neighbors of node 1, to see if they can transmit at the same time slot or not. So we start with the lowest node number of node 1's neighbors. In this case node 3 comes first. From node 3 we investigate one by one all the outgoing links (link 4 and 5) to see if one or more of them can be activated in the same time slot (slot 1).

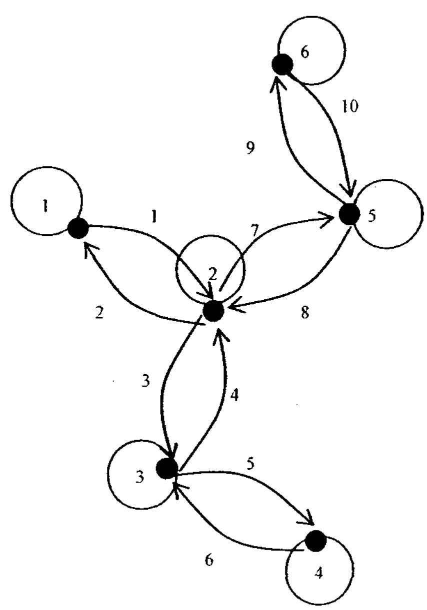


Figure 2.3-1 A Multihop Network

If we put node 3 in transmitting mode and activate link 4 i.e. the lowest link number, it will affect node 2's reception. This will lead to a conflict because a particular node can not receive simultaneously from two nodes. Now we look for the next lowest node number which is node 4. We can put node 4 in transmitting mode and activate link 6. Link 6 will not affect the reception of node 2. Also in the same slot, link 10 can be activated. Slot by slot, the algorithm investigates the links that can be activated in the same slot (due to space consideration). If one or more links still remain unserved in all slots, another slot is investigated and the frame ends when each link is guaranteed at last one slot per frame for its transmission activities. If we denote C_i as the ith clique, we can form a clique cover, C or corresponding a STDMA frame, which is a set of maximal cliques $C = \{C_1, C_2, \ldots\ldots\ldots\ldots C_k\}$ having the property that every link of the network is contained in at least one member of C. For example, the clique cover for the topology shown in Figure 2.3-1 is given by

$$C = \{C_1, C_2, C_3, C_4, C_5, C_6\}$$

Where

$C_1 = \{1,6,10\}$ $C_2 = \{2,5,9\}$ $C_3 = \{3,9\}$ $C_4 = \{4,10\}$ $C_5 = \{5,7\}$ $C_6 = \{6,8\}$

For the cross layer optimization in WMNs, this STDMA allocation can be utilized. In cross layer optimization, two or more layers are simultaneously optimized. This is not a violation of the layered architecture but an extra benefit. PHY, MAC and the Network layers are widely utilized for this cross layer design in WMNs. CDMA can be used as a MAC access technique alternative to STDMA. In

Chapter 5, cross layer designs are proposed utilizing STDMA and CDMA techniques.

Chapter 3

TABLE DRIVEN WIRELESS LOCAL AREA NETWORKS

In this chapter we seek new techniques for improving the overall QoS of integrated 4G systems. Towards this objective we start with the local low tier WLAN access.

In this phase, a new access mechanism for WLANs is developed, in which users use an optimum transmission probability obtained by estimating the number of stations from the traffic conditions in a sliding window fashion, thereby increasing the throughput compared to the standard DCF and RTS/CTS mechanism while maintaining the same fairness and the delay performance.

3.1 Operation of Table Driven Techniques

In the proposed protocol, if the nodes sense that the channel is idle for an interval called DIFS (DCF interframe space), they try to send a packet with a probability p, which is dependent on the traffic condition i.e. the number and activities of the nodes, as follows. The definition of the symbols highlighted in this section are given in 3.2.

A user continuously monitors the channel in each idle slot following the DIFS idle period. If the previous slot is idle, it calls a uniform random generator (0,1). If the value of this generator is less than or equal to p, it tries to start its packet transmission in the given next slot. If the value is larger than p, the user persist on listening and repeats trials as stated. However if the channel is sensed busy the user defers his transmission till the next DIFS idle period heard.

The users measure the number of collisions $C = I - 1$ and the length W_L by monitoring the channel over a large enough window (which is explained latter on). With the help of these values users can use the tables formulated in 3.2 to obtain the corresponding p and M.

Users having a non-empty queue start by monitoring the channel for the first n transmission periods (Figure 3.2-1). This active user will average the length of the idle period preceding the correct packet transmission over n transmission periods, i.e. $\overline{W}_L$ and $\overline{C}$, the average number of collisions over the same period. Aided with these values users obtain the operating values of p and M and uses p to control their transmission activities for the head of line packet in their queue. Active users continuously monitor the channel and use a sliding window technique to estimate W_L and I and hence obtain (M, p). For example the first sliding window averages

W_L and C of the first n transmission periods. The second window averages W_L and C of the $l = 2,3,.......n+1$ transmission periods. The sliding window averaging process reflects the changing traffic, so transmission activity of active users are dependent only on the current traffic and not on past history.

It is possible that the tables relating (M,p) to (W_L,I) yield more than one possibility for (M,p) for certain (W_L,I) measurement values from the sliding window. In this case, the user averages the obtained values of M and use Table 3.2-1 to find the optimum p at this averaged value of M. This Table 3.2-1 is obtained from Figure 3.2-2 in an evident manner. The operation of this table driven technique is similar to the DCF standard (IEEE 802.11) [13] except for using this optimized transmission probability p. Active users just estimate (M,p) from the traffic conditions (by sensing the channel) in a sliding window fashion one period after another.

We note that in the new technique old and new users both measure the traffic and adapt to the same traffic conditions fairly and obtain the same p. However having same p does not mean all users will repeatedly collide in the same slot because of feeding a random number generator with p.This is different than IEEE 802.11 where fresh users may use smaller backoff than old users.

The above procedure is followed for both table driven DCF and table driven RTS/CTS shown in Figure 3.2-1.

3.2 Analysis of Table Driven DCF and Table Driven RTS/CTS

Let p be the transmission probability of each node and M be the number of active stations. Assuming each user tries to transmit randomly in each slot following the DIFS period. According to table driven DCF (Figure 3.2-1) the probability of successful transmission, is thus given by 3.2.1.

$$P_s = Mp(1-p)^{M-1} \tag{3.2.1}$$

The probability of an idle slot in table driven DCF is

$$P_o = (1-p)^M \tag{3.2.2}$$

and the probability of unsuccessful transmission for table driven DCF is

$$P_{cc} = 1 - P_s - P_o \tag{3.2.3}$$

Let i be the number of idle periods (cycles) before a successful transmission as shown in Figure 3.2-1 and j be the number of idle slots in each idle period lengths $(W_1, W_2, \ldots\ldots)$. The throughput (η_1) is given by equation (3.2.7) for table driven DCF.

It is easily seen that the average length of each idle period except the last one before packet success in table driven DCF is given by

$$W_1 = W_2 = \ldots\ldots W_{L-1} = E(j) = \sum_{j=1}^{\infty} j(P_o)^j P_{cc}$$

$$= \frac{P_o P_{cc}}{(1-P_o)^2} \tag{3.2.4}$$

Equation (3.2.4) determines the number of idle slots before a collision.

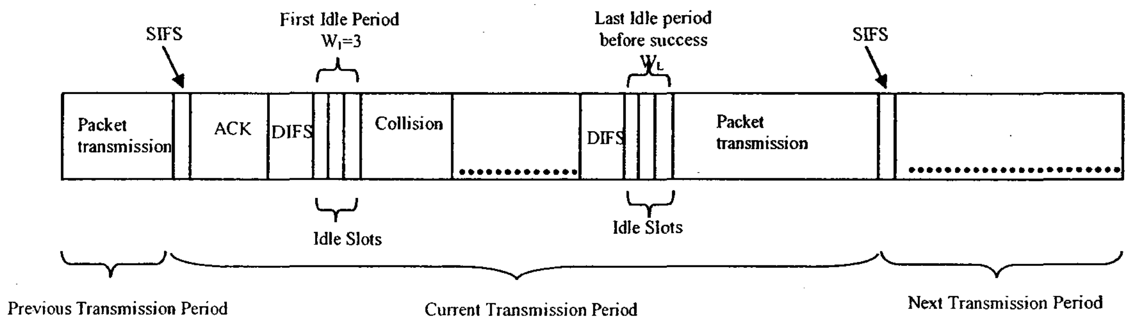
The last idle period before a success, has an average of

$$W_L = \frac{P_o P_s}{(1-P_o)^2} \tag{3.2.5}$$

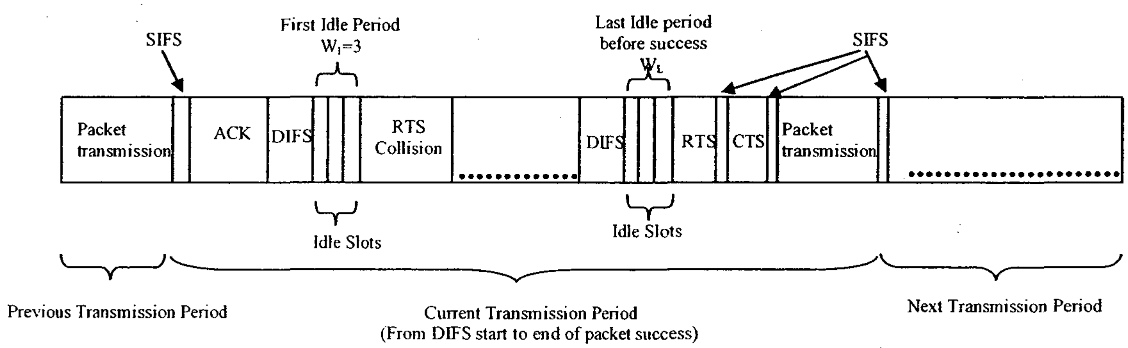
The average number of cycles is given by,

$$I = \sum_{i=1}^{\infty} iP_i$$

$$where \quad P_1(No.\,of\ cycles = 1) = P_s + P_0P_s + P_0^{\,2}P_s + \ldots\ldots$$
$$= \frac{P_s}{1-P_0}$$



(a)



(b)

Figure 3.2-1 Transmission Activity on the Wireless Channel for (a) table driven DCF and (b) table driven RTS/CTS

$$P_2(No.\,of\ cycles = 2) = P_{cc}\left[\frac{P_s}{1-P_0}\right] + P_0P_{cc}\left[\frac{P_s}{1-P_0}\right] + P_0^{\,2}P_{cc}\left[\frac{P_s}{1-P_0}\right]\ldots\ldots$$
$$= \frac{P_{cc}P_s}{(1-P_0)^2}$$

$$P_3(No.\,of\ cycles = 3) = \frac{P_{cc}^{\,2}P_s}{(1-P_0)^3}$$

$$P_i = \frac{P_{cc}^{\,i-1}P_s}{(1-P_0)^i}$$

Therefore

$$I = \frac{(1 - P_o)}{P_s} \tag{3.2.6}$$

$$\eta_1 = \frac{T_{Payload}}{(W_1 + W_2 + ..W_{L-1})T_{Slot} + (I-1)\{T_{DIFS}\} + T_{DIFS} + T_{ACK} + T_{SIFS} + T_{Payload} + W_L T_{Slot}} \tag{3.2.7}$$

$$\eta_2 = \frac{T_{Payload}}{(W_1 + W_2 + ..W_{L-1})T_{Slot} + (I-1)\{T_{RTS} + T_{DIFS} + T_{SIFS}\} + T_{DIFS} + T_{RTS} + T_{CTS} + T_{ACK} + 3T_{SIFS} + T_{Payload} + W_L T_{Slot}} \tag{3.2.8}$$

Let the number of collisions be $C = I - 1$. This C and W_L are calculated from different values of M and p which are stored in two different tables (see appendix). So for particular values of M and p there is a particular value of C and W_L. The throughput for table driven DCF (η_1) and table driven RTS/CTS (η_2) are given in equation (3.2.7) and (3.2.8) respectively based on the transmission activity on the wireless channel as shown in Figure 3.2-1.

The throughput η_1 (for table driven DCF) is calculated for different values of M and p as in Figure 3.2-2. Table 3.2-1 depicts the probabilities at which the maximum throughput occurs for different values of M .

Similarly for the table driven RTS/CTS, to calculate the C and W_L, equations (3.2.1)-(3.2.6) are used. However the throughput is calculated from equation (3.2.8) which includes the RTS/CTS frames (Figure 3.2-1).

For the case of table driven RTS/CTS all cycles leading to no success (RTS heard but no CTS) will each have a cost of $T_{RTS}+T_{DIFS}+T_{Slot}$ seconds.

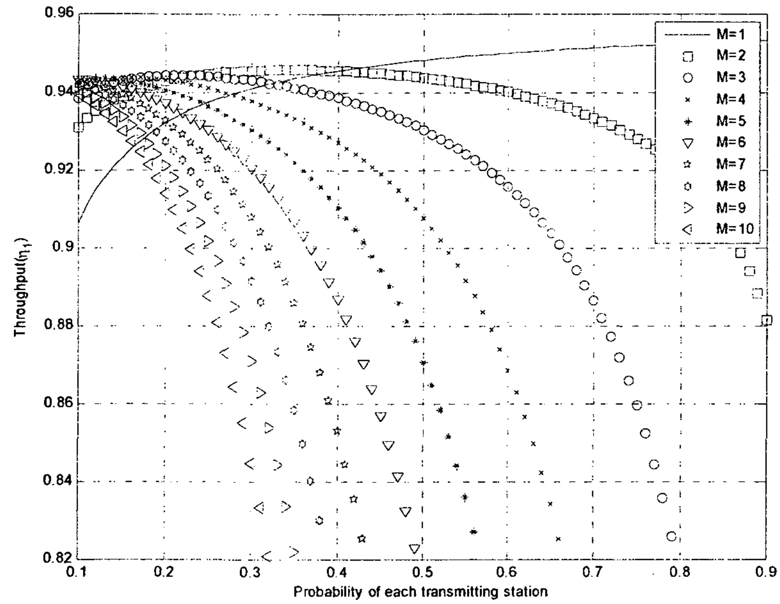


Figure 3.2-2 Throughput for different probabilities and different number of stations for DCF

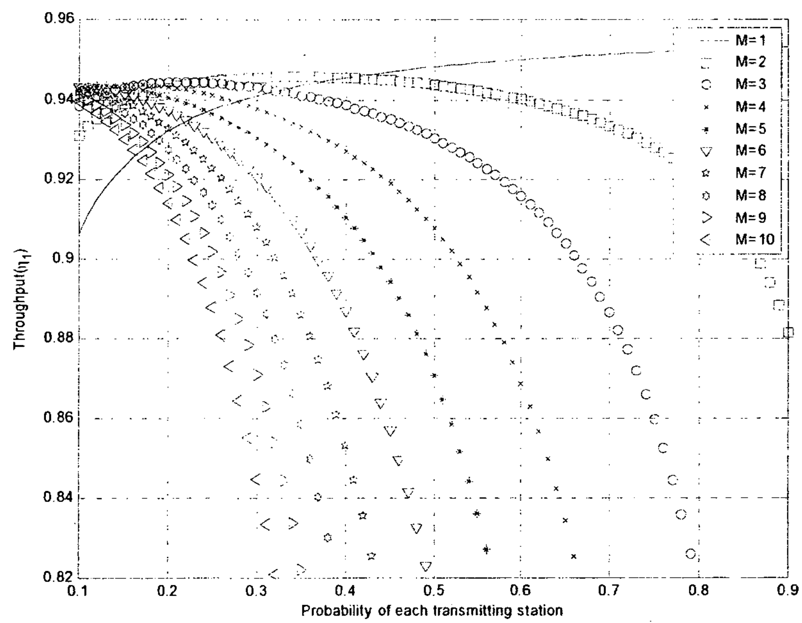


Figure 3.2-3 Throughput for different probabilities and different number of stations for RTS/CTS

Table 3.2-1 Optimum throughput for different probabilities and different number of stations for DCF

No of Stations	Probability	Optimum throughput
1	0.9000	0.9532
2	0.3400	0.9458
3	0.2200	0.9444
4	0.1600	0.9437
5	0.1300	0.9434
6	0.1000	0.9431
7	0.1000	0.9428
8	0.1000	0.9420
9	0.1000	0.9410
10	0.1000	0.9397

3.3 Simulation Results

For numerical calculations the following parameters are taken from "Bianchi" in [13]

$T_{Payload}$	*10msec*
PHYheader	*128bits*
ACK	*112bits+PHY header*
RTS	*160bits+PHY header*
CTS	*112bits+PHY header*
Channel bit rate	*1 Mbits/s*
Slot time (T_{Slot})	$50\ \mu s$
T_{SIFS}	$28\ \mu s$
T_{DIFS}	$128\ \mu s$
service rate	$\mu = \frac{1}{T_{payload}}$
offered traffic	λ *packets/sec*
	$M\lambda \le \mu$
No of stations	*M*

In the table driven DCF, as per the standards, following the observance of each DIFS, users try to transmit with probability p obtained from $\overline{W}_L$ and $\overline{C}$. If two or more stations try to transmit at the same time, collisions occur. If no stations transmit (Figure 3.2-1), the number of idle slots will increase. If one station is successful after certain number of idle and collision periods, the transmission period ends. As a result the total time for one successful packet transmission include T_{DIFS}, T_{SIFS} T_{Idle}, $T_{Payload}$. The throughput is calculated at the end of the simulation at certain values of M, λ and p i.e.

$$\eta = \frac{T_{Payload} \times No\ of\ Transmission\ Periods\ in\ the\ whole\ simulation}{Time^{(n)}}$$

where $Time^{(n)}$ is the total simulation time that depends on T_{DIFS}, T_{SIFS}, T_{Slot}, $T_{Payload}$. Initially $Time^{(n)} = T_{DIFS}$ and is subsequently increased based on the user's activity, e.g.

$$\begin{aligned} & Time^{(n)} = Time^{(n)} + T_{Slot}\ ;\ for\ each\ idle\ slot\ period \\ or\quad & Time^{(n)} = Time^{(n)} + T_{DIFS};\ for\ each\ collision \\ or\quad & Time^{(n)} = Time^{(n)} + T_{DIFS} + T_{SIFS} + T_{Payload};\ for\ each\ successful\ packet \end{aligned}$$

For the table driven RTS/CTS the total simulation time is calculated by the following equations

$$\begin{aligned} & Time^{(n)} = Time^{(n)} + T_{Slot}\ ;\ for\ each\ idle\ slot\ period \\ or\quad & Time^{(n)} = Time^{(n)} + T_{RTS} + T_{DIFS}\ ;\ for\ each\ collision \\ or\quad & Time^{(n)} = Time^{(n)} + T_{RTS} + T_{CTS} + T_{DIFS} + 3T_{SIFS} + T_{Payload}\ ;\ for\ each\ successful\ packet \end{aligned}$$

Figure 3.3-1 shows a comparison of the throughput between the table driven DCF and the standard DCF (IEEE 802.11) for 10 stations. The values of standard DCF are taken from [16] which uses the same parameters as in [13]. It is evident the table driven DCF performs better than the standard DCF (IEEE 802.11).

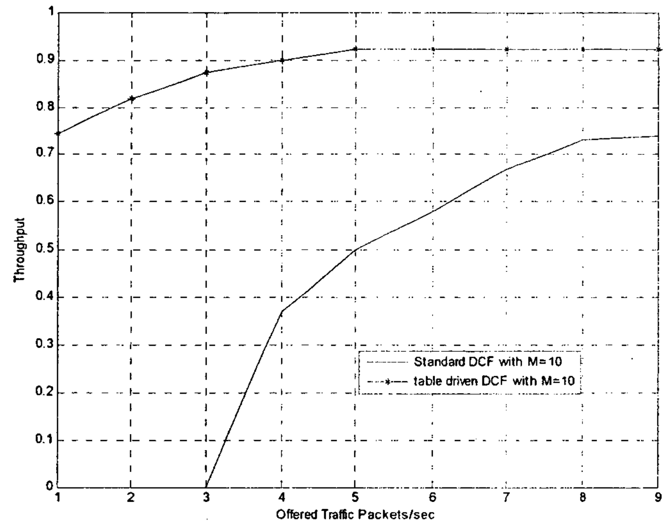


Figure 3.3-1 Throughput comparison between the table driven DCF and the standard DCF (IEEE 802.11)

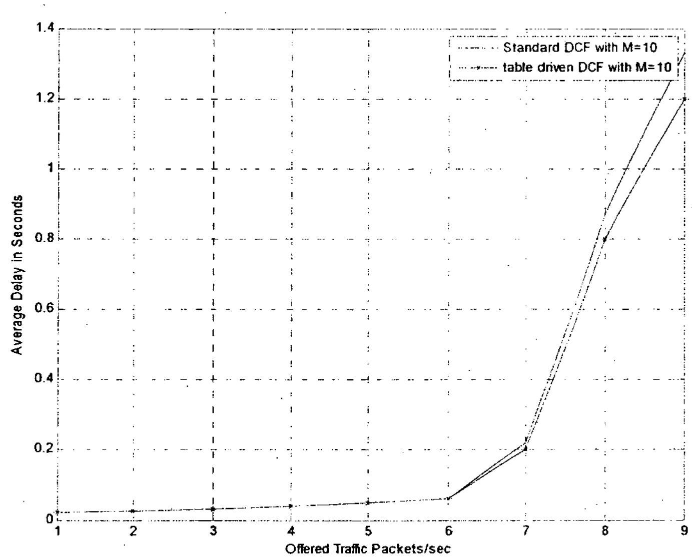


Figure 3.3-2 Average Delay Comparison between the table driven DCF and standard DCF (IEEE 802.11)

Figure 3.3-2 shows a comparison of average delay between the table driven and the standard DCF (IEEE 802.11). The values of standard DCF are again taken from[16] for comparison purposes. It is noticeable that the delay performances are almost the same.

Figure 3.3-3 shows the throughput curve for different offered loads for the table driven RTS/CTS technique. It shows that the throughput rises and becomes saturated at higher values of the load. The maximum throughput calculated by "Bianchi" in [13] for the standard RTS/CTS (IEEE 802.11) mechanism is 0.837281 when M=10.

From Figure 3.3-3 it is evident that the table driven RTS/CTS performs better than the standard RTS/CTS (IEEE 802.11) in terms of throughput.

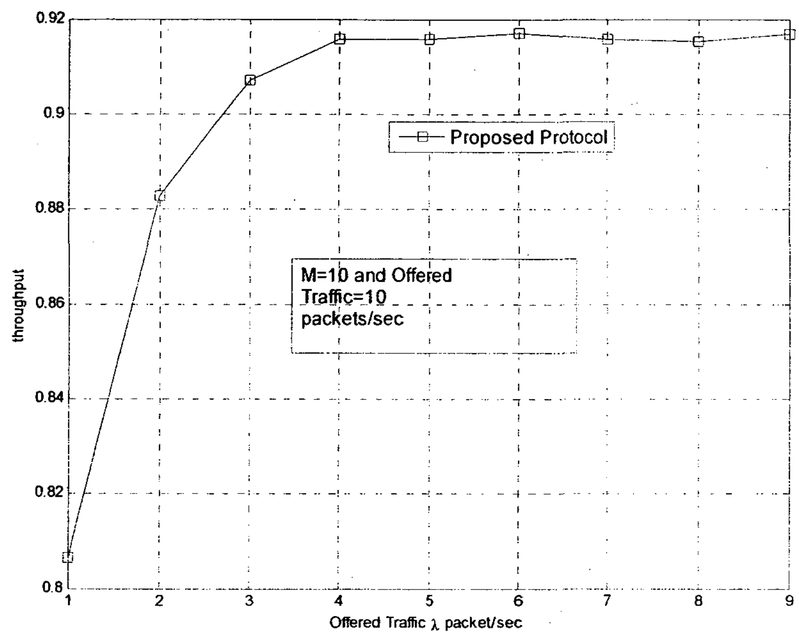


Figure 3.3-3 Throughput corresponding to different offered traffic

The table driven RTS/CTS technique has an extra advantage as it is a load adaptive system. It means that it has the capability to adapt to the input traffic as quickly as possible. Figure 3.3-4 shows a case where the input traffic suddenly increases from 5 packets/sec to 10 packets/sec. In this case the throughput $(\eta \times \mathit{Input\ traffic\ rate}(\lambda))$ is shown to follow the offered traffic λ.

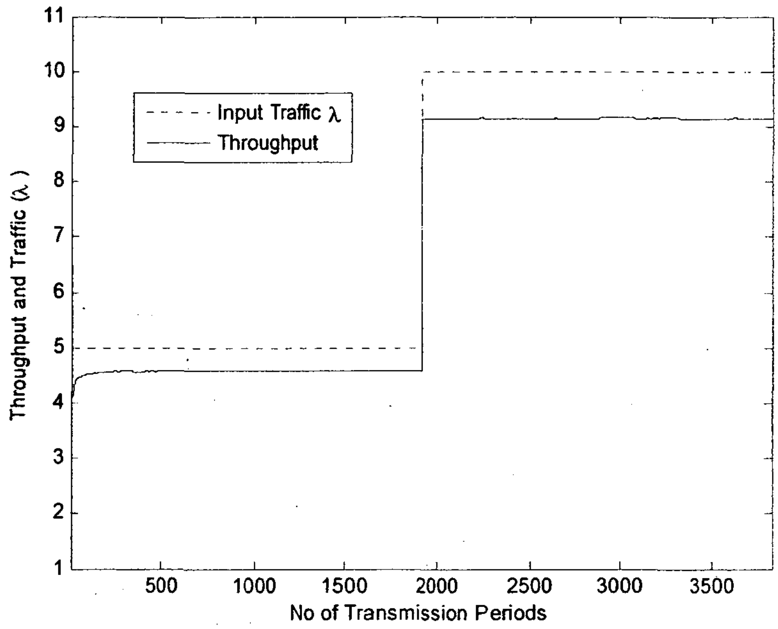


Figure 3.3-4 Throughput and Input Traffic corresponding to the number of Transmission periods (Table driven RTS)

Fairness (FI) is another important issue considered in this thesis. To express this, we take the fairness index defined in [51] and [14] to measure the fair packet capacity allocation. In [51] fairness index is represented as $\frac{\left(\frac{1}{n}\sum_{i=1}^{n} x_i\right)^r}{\left(\frac{1}{n}\sum_{i=1}^{n} x_i^r\right)}$. For example if m dollars are to be distributed among *n* people and we favor *k* people by giving them *m/k* dollars each and discriminate against *n-k* people, then the above function becomes $FI = \left(\frac{k}{n}\right)^{r-1}$. Favoring 10% would result in a fairness index of $FI = 0.1^{r-1}$ and discriminate index of $1 - 0.1^{r-1}$. Therefore *r* should be equal to 2. That is, $FI = \frac{\left(\sum_{i=1}^{n} x_i\right)^2}{n\left(\sum_{i=1}^{n} x_i^2\right)}$, where *FI* is the fairness index, n is the number of stations,

x_i is the packets transmitted by the i^{th} active station during the simulation time (current traffic in which the offered traffic λ is same for all stations) .

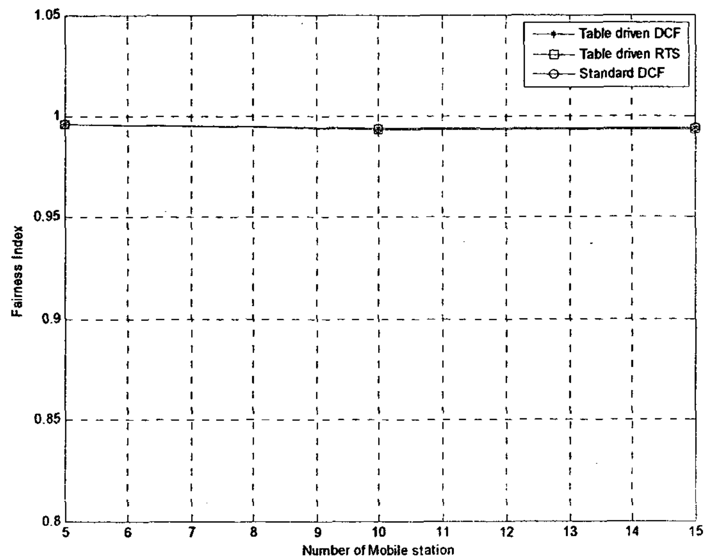


Figure 3.3-5 Fairness Index for different number of stations for the table driven DCF, table driven RTS/CTS and standard DCF (IEEE 802.11)

Figure 3.3-5 shows the comparison of the fairness index performance of table driven DCF, RTS/CTS and standard DCF. It can be observed that, for the three cases up to 15 active stations the performance is fair.

3.4 Conclusion

The table driven technique for DCF and RTS/CTS mechanism in WLAN outperforms the standards in terms of throughput, while maintaining the same delay and fairness performance. After the performance improvement of such techniques in the local low tier WLANs, we proceed towards the backbone mesh network which is given in the subsequent chapters. This WLAN access can also be extended and used for the communication among the backbone mesh nodes which can be compared with the recent developed mesh communication techniques. We left this approach for our future research work. Also, in the table driven technique we utilized a simple search mechanism to obtain the values of the transmission probability and the number stations from the two tables. An efficient search mechanism can be made for its better operation.

We can conclude that, in table driven technique less packets will be dropped and the users can utilize the channel more efficiently. This will allow the users to enjoy high speed wireless internet.

CHAPTER 4

CDMA/TDD APPROACH FOR WIRELESS MESH NETWORKS

For improving the overall QoS of integrated 4G systems and having started from the local low tier WLAN access, we move towards the backbone of mesh networks. In this chapter we investigate CDMA alternatives to the TDMA access typical of WMNs. We introduce a code division multiple access/Time division duplex technique CDMA/TDD for WMNs, we outline the proposed transmitter and receiver for the relay nodes and evaluate the efficiency, delay and delay jitter performances.

4.1 Introduction

Wide band mesh or star oriented networks have recently become a subject of greater interest. Providing wideband multimedia access for a variety of applications has led to the inception of mesh networks. Classic access techniques such as FDMA and TDMA have been the norm for such networks. CDMA maximum transmitter power is much less than TDMA and FDMA counter parts, which is an important asset for mobile operation. In this thesis we introduce a code division multiple access/Time division duplex technique CDMA/TDD for such networks. The CDMA approach is an almost play and plug technology for wireless access, making it amenable for implementation by the mesh network service station, SS. Further it inherently allows mesh network service stations to use a combination of turbo coding and dynamic parallel orthogonal transmission to improve network efficiency. We outline briefly the new transmitter and receiver structures then evaluate the efficiency, delay and delay jitter. By analysis we show the advantages over classic counter parts with respect to the total network efficiency achievable especially for larger number of hops.

4.2 System Model

Figure 4.2-1 shows the TDD operation of the proposed system. When the node (SS) powers ON, it is in passive mode and listens to HELLO message from nearby nodes and the lowest level in all HELLO message is j, it decides its own level as (j+1). For example U5 receives HELLO messages from U2 and U6 which indicate they are in level 1 and level 3, U5 decides its level to be 2. Such exchanged HELLO messages will also help the nodes to recognize the current topology, neighbors etc. as will be outlined in chapter 5. Typically a BS initiates the configuration of the TDD mode by transmitting HELLO messages. Such HELLO and other control messages follow the format shown in Figure 4.3-5.

In TDD mode nodes such as U1, U2, U3 in level 1, nodes U6, U7, in level 3, nodes U10 in level 5 all transmit in even numbered slots 0, 2 , 4, 6....while nodes BS and U4, U5 in level 2, node U8, U9 in level 4 all listen at same even slots. In odd slots the situation reverses, i.e. U1, U2, U3, U6, U7 and U10 all listen while BS, U4, U5, U8, U9 and U11 all transmit. Needless to say upon power on and listening to HELLO messages and determining the smallest heard level as before the node determines which slot is odd or even.

For proper TDD operation however, reception slot is slightly enlarged by the maximum one way propagation delay of a hop.

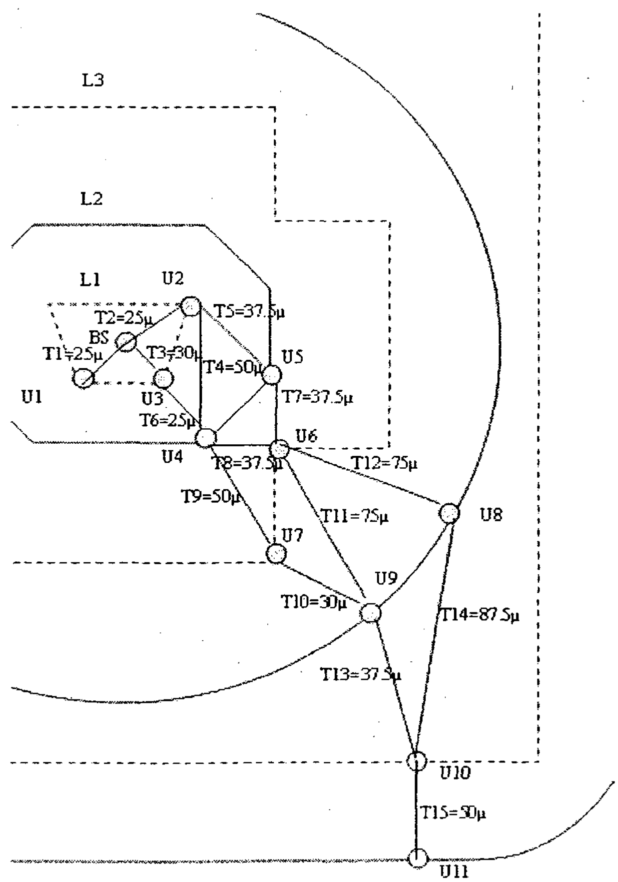


Figure 4.2-1 A typical TDD Mesh Network

4.3 Transmitter and Receiver Model

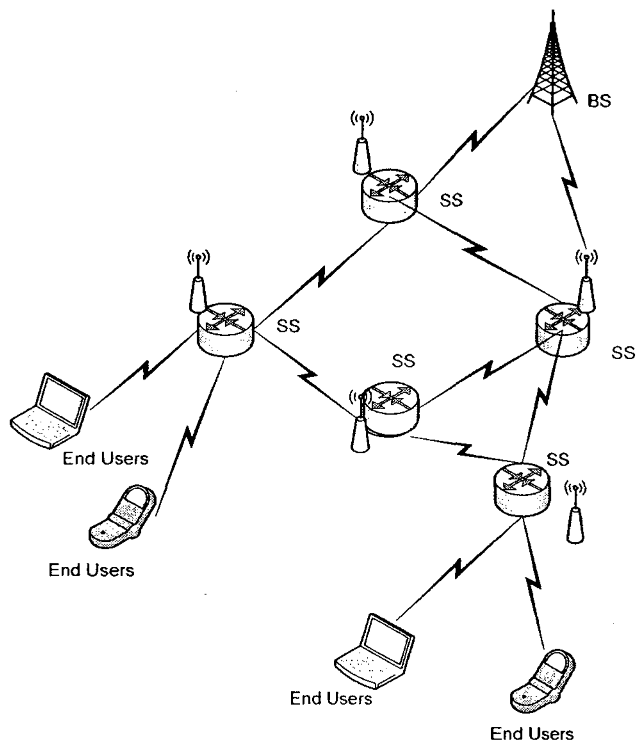


Figure 4.3-1 A Network Scenario with multiple SSs and multiple end users

Figure 4.3-1 shows a network scenario with multiple SSs and multiple end users. The mesh nodes (SS) consists of a transmitter and a receiver where the modulation and demodulation is performed. The synchronization between each mesh node and the mobile end users are maintained by a certain base station pilot signal. The cellular base stations can perform this task as in 4G. It is foreseen that mesh nodes (SS) will be interconnected with the cellular base stations (Internetworking UMTS-Mesh networks[1]). CDMA approach carries with it the versatility of parallel transmission of few packets from each user on same long scrambling code but each stream multiplies the stream bits by a different Walsh function, analogues to 3G standards[52],[5]. Parallel transmission from typical node, with each packet multiplying an orthogonal Walsh function is possible. The intermediate destination of parallel packets may be different thus allowing more transmission flexibility. We select Walsh function upon transmissions according to transmission rates needed, for example if a certain node X wishes to transmit a packet to node Y, then it will multiply the intended packet by the intended Walsh function and randomly selected scrambling code. The immediate destinations of nodes packets are determined by the route decisions which are exchanged via the control packets including HELLO and other control messages, as will be outlined in chapter 5. Source and destination of the subject sending node are inserted in the headers of transmitted packets using a predetermined Walsh function W1 and a randomly selected scrambling code C_s. Predetermined orthogonal Walsh functions are used to enable nodes to transmit many packets in parallel. Such packets will then become orthogonal to each other. Multiplying by the corresponding Walsh functions, receiving nodes can separate and demodulate each of the parallel packets. This will enable each receiving node to tune itself to the corresponding Walsh functions of this nearby transmitting node. Each receiving node will demodulate all packets heard but will route only the intended packets passing by this node as dictated by the routing policy (not all packets demodulated). This means that each node is actually a crossing road (switch) and it handles many data from many source destination pair at the same time. This implies that each node should have many parallel demodulation banks working at the same time and can process many packets of many neighboring nodes simultaneously. The intended receiving node demodulates all packets heard. However, nodes do not necessarily route all packets they hear from their neighbors. From the routing table (as in chapter 5), each node determines which heard packet it should route and it configures its receiver bank and demodulates accordingly.

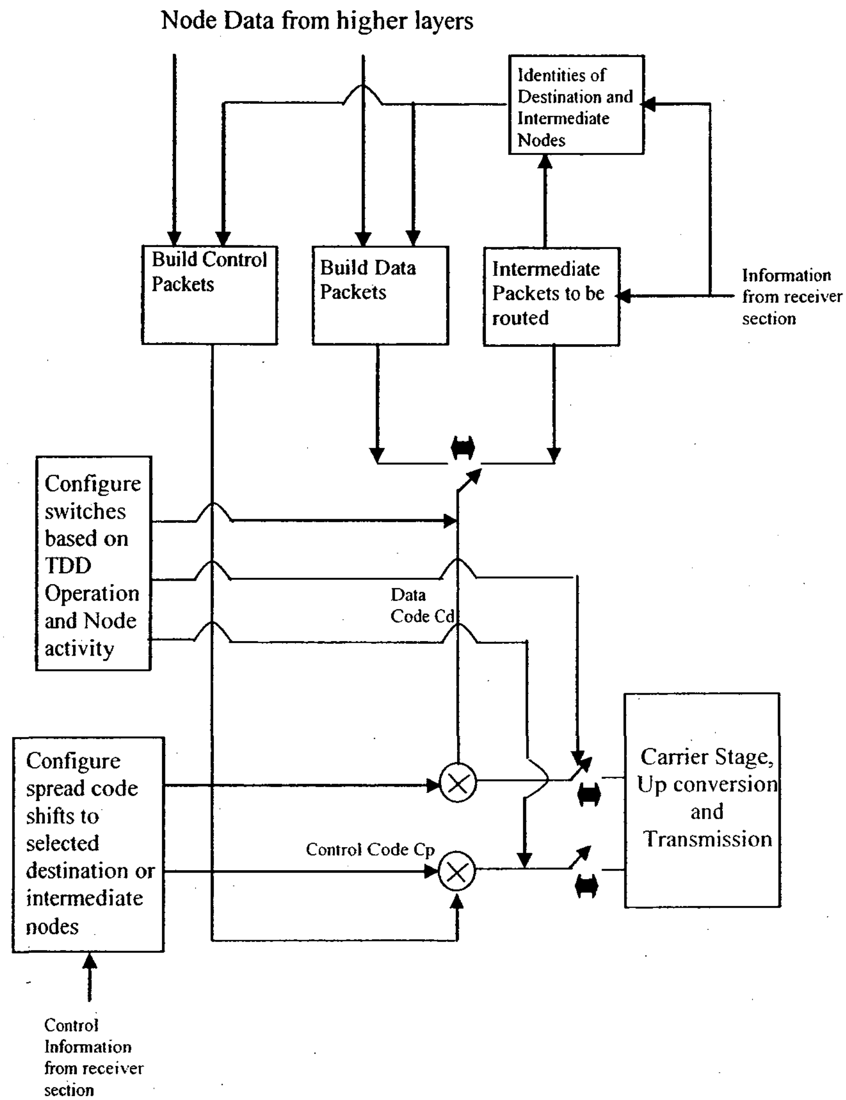


Figure 4.3-2 Block diagram of the Transmitter

For the end users our protocol can be very efficient as each mesh node can modulate and demodulate many packets in parallel thus increasing network throughput. For example, an end user is browsing a document from the internet and suddenly he receives voice or video. In this case our approach would be helpful for the users to accomplish both tasks simultaneously. Meaning multitasking is possible using our CDMA/TDD protocol. This technique applies to wireless cluster with moderate number of mesh routers. For example, upto hundred stations. This implies a large enough network spanning of maximum end to end distance of few hundred KM.

The subject network is packet switched and not circuit switched. It is connection oriented in the sense that SS's have to signal their destination (through control packets) before transmitting data packets. In this regard a priori establishment of the routes via exchanging control packets is essential. Many routing techniques have been designed for routing packets among peer nodes in ad hoc WLANs [41], [53], [54]. Most of these are applicable to mesh networks and are actually traditional shortest path routing operating in connection oriented mode. For rapid fault tolerance and autonomous load balancing other techniques can also be used [55],[56],[57]. In this thesis we only analyze the system performance considering the interaction between the MAC and PHY layers, while for the network layer one of the efficient load balancing routing mechanism stated above can be used.

Figure 4.3-2 shows the block diagram of the transmitter. Higher layer data and control packets are modulated by the spreading codes. In this work only one pilot code is used for all SS's for data and another pilot code for control as shown in Figure 4.3-5.

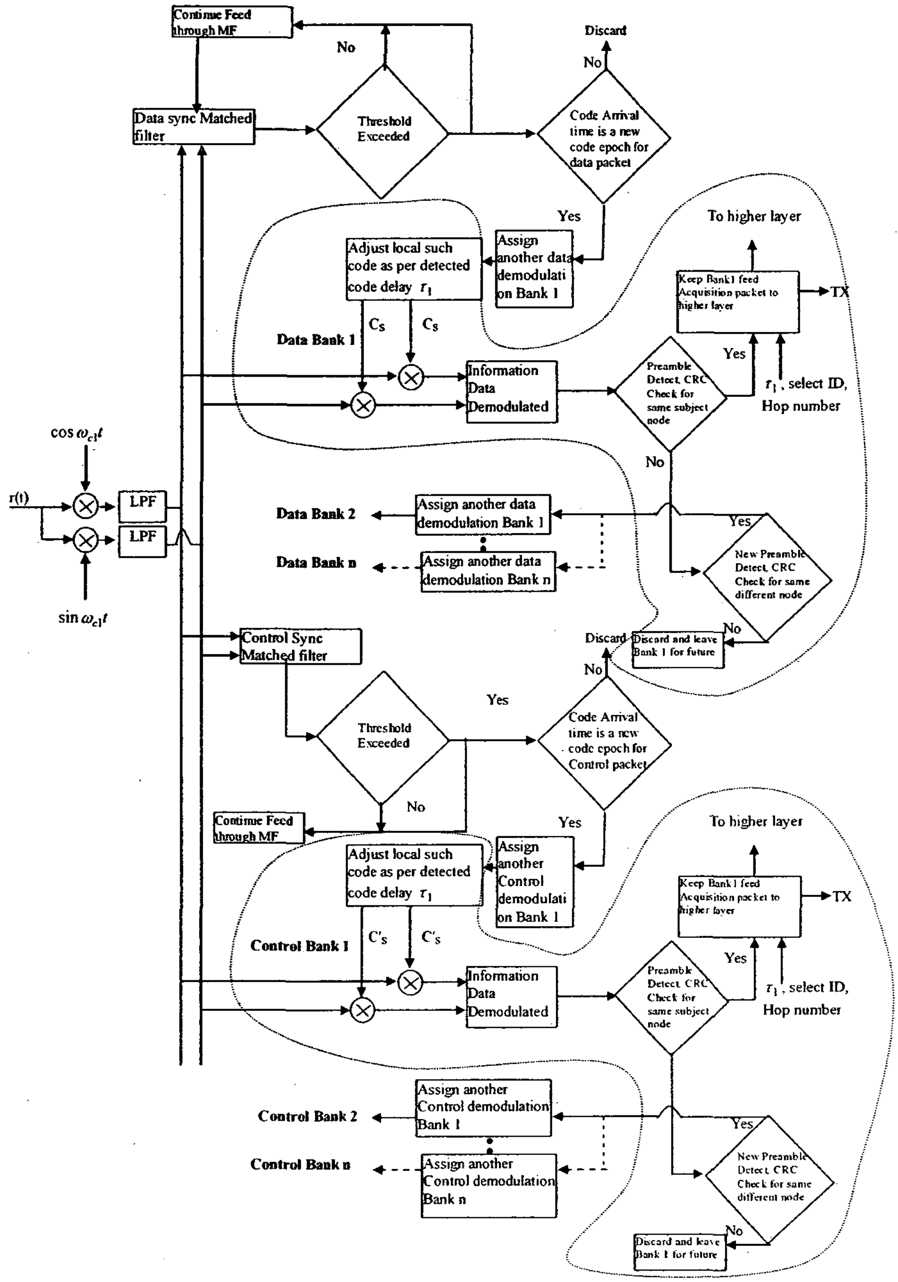


Figure 4.3-3 Block diagram of the receiver

Also one longer scrambling code C_S is used by all transmitting nodes but with different shifts to scramble the whole of the data packet except the pilot part as in Figure 4.3-4. Another longer scrambling code $C_S^{'}$ is used by all transmitting nodes but with different shifts to scramble the control packet except the pilot part as in Figure 4.3-5. Different parts of data packets from subject node or routed packets from intermediate nodes are spread by the pilot and scrambling codes then TDD gated according to the TDD operation and the node activities.

Figure 4.3-3 shows the block diagram of the physical layer constituents of the receiver. The received signal is fed through a filter matched to data pilot code spreading. If the signal from the matched filter exceeds the threshold, this signal is assigned a data bank and the data is descrambled and demodulated via the data scrambling code C_S. If the signal does not exceed the threshold we continue to feed it through the matched filter until a threshold is detected. After demodulation of the data information, the header is checked using the CRC for the same subject node. If a preamble is detected, this bank is kept and the packet is fed to higher layer and also to the transmitter to initiate the data pilot spreading code. If no preamble is detected, we check for possible preambles from other nodes. In this case if a new preamble is detected, another demodulation bank is assigned. If no preamble is detected, the signal is discarded and corresponding bank is left for future use. The Control section of Figure 4.3-3 precedes the same as the data section but the signal is demodulated by the pilot spreading code for a small part of the control packet (Figure 4.3-5) and by C_S' .

Figure 4.3-4 and Figure 4.3-5 show respectively the physical data and the physical control packet format. In addition to the short pilot spreading codes constituting the SYNC field of length 256 chips we also employ a longer scrambling code of length $(2^{15}-1)T_c$. The Starting phase is selected randomly by each user, same for all its subsequent packets. Each packet starts it's C_s from a certain initial condition. The initial state (starting phase of scrambling code) is conveyed by the field $C_d d_i$ of Figure 4.3-4 which is used by the receiver to be able to synchronize its local scrambling code to that of the received packet. These d_i bits are spread by the short pilot data code for data packets and spread by the short pilot control code for control packets.

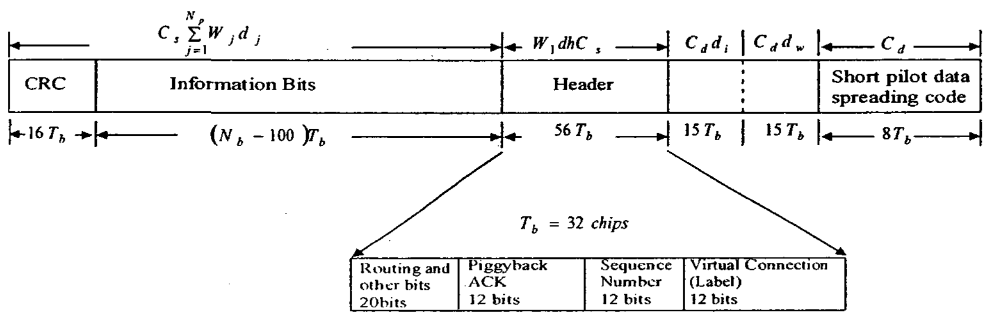


Figure 4.3-4 Physical Data Packet Format

The field d_w of Figure 4.3-5 conveys to the receiver, the current number of Walsh functions used in this packet, i.e. the number of packets it is transmitting parallel in time and not the identities of their Walsh functions which are assigned a priory in receiver oriented code assignment. Such identities are exchanged via HELLO and similar control messages. These d_w bits are spread by the short pilot data spreading code.

W_i=ith Walsh function used i=1,2, 3,4. A typical user employs a number of N_p Walsh functions at its receiver. d_w bits are the binary representation of N_p.

W_L=The Walsh function length=T_b sec, After indicating the SYNC is detected, users employ C_d to detect i.e. d_i bits the code starting bits initial LFSR shift of long C_s code and number of Walsh function i.e. N_p

d_w=bits indicate number of Walsh function i.e. number of packets transmitted in parallel.

d_i =bits indicate initial shift of C_s code.

C_d =short data pilot code. We note that the beginning of each packet is spread by the short pilot code while the header and data bits are spread by Walsh function and longer scrambling code.

Table 4.3-1 The parameters used for the Analysis

Bandwidth (W)	Number of bits/packet (Nb)	Bit rate Rb	Processing Gain (PG)	Turbo Code rate (r)	Rate Packets/sec	1 Packet in ms
32 M	1000 &2000	1M	32 (PG=W/Rb)	1/2	$\frac{32\times10^6\, bits/sec}{1000\, bits/packet}=32000$	$\frac{1}{32000}=0.03125ms$

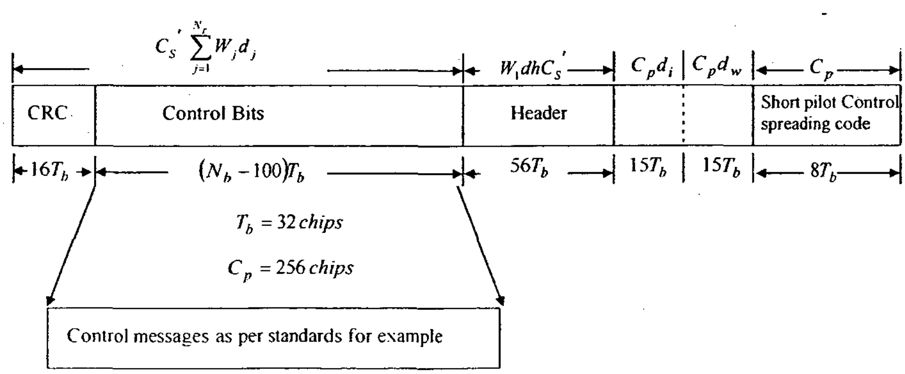


Figure 4.3-5 Physical Control Packet Format

4.4 Performance Analysis

As discussed in 4.3, the CDMA system is capable of transmitting packets in parallel. So for analysis purposes we consider a bulk arrival bulk service system shown in Figure 4.4-1 for packets in queue of a typical node.

From Figure 4.4-1 the state transition probabilities which are function of the probability packet arrival α_l and service probabilities β_l are given below:

$$a_{ij} = 0; \quad i \succ j; \quad i - j \succ B \tag{4.4.1}$$

$$a_{ij} = 0; \quad i \prec j; \quad j - i \succ B \tag{4.4.2}$$

$$a_{ij} = \sum_{l=0}^{\min(j+B-i,B)} \alpha_l \beta_{i+l-j}; \quad i \succ j; \quad i - j \leq B \tag{4.4.3}$$

$$a_{ij} = \alpha_0 + \sum_{l=1}^{B} \alpha_l \beta_l; \quad i = j = 0 \tag{4.4.4}$$

$$a_{ij} = \sum_{l=0}^{B} \alpha_l \beta_l; \quad i = j \neq 0 \tag{4.4.5}$$

$$a_{ij} = \sum_{l=0}^{B-(j-i)} \alpha_{j+l-i} \beta_l; \quad i \prec j; |j - i| \leq B \tag{4.4.6}$$

In Figure 4.4-1 each node i.e. a SS or logically a switching node receives packet from other nodes to forward (switch action) that adds to the node's intrinsic packet generation (self traffic). Assuming that during each packet time the user possible generation probabilities are:

$$\alpha_1 = \alpha_2 = \alpha_3 = \alpha_4 = p / B, \alpha_0 = 1 - p$$

We assume that the simultaneous generation of one or more packets by each node is equally likely. Here p is a given user combined packet generation parameter i.e. probability of generation at any packet time. For analysis convenience all input traffic to our wireless network herein has independent increments with no correlation between arrivals.

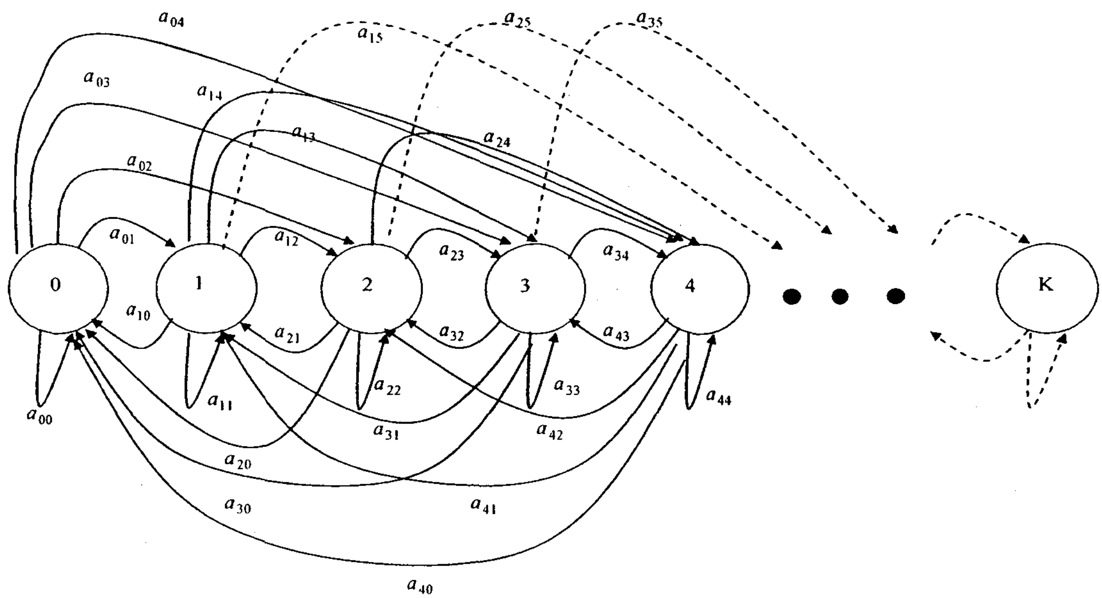


Figure 4.4-1 State transition diagram of bulk arrival bulk service switching node underlay the TDD/CDMA Label switched Network

The intended node probabilities of success of parallel transmission of one or two or B packets on the channel are $\beta_1, \beta_2, \beta_3, ... \beta_B$. These probabilities are obtained by averaging over the conditional transmission from other users; i.e.

$$\beta_k = \sum_{n=k}^{BU} \varphi_n P_k \left(\sum_{u_2=0}^{B} \sum_{\substack{u_3=0 \\ u_2+u_3=n-k}}^{B} P_{u_2} P_{u_3} \right) \tag{4.4.7}$$

P_k is the probability that the intended user transmits k packets in parallel (to be stated shortly after) P_{u_2} is the probability that another user transmits u_2 packets in parallel. P_{u_3} is similarly defined and in equation (4.4.7) we average over all u_2 , u_3

values such that $k+u_2+u_3=n$. Here $\varphi_n = (1-P_{b_n})^{Nb}$; where φ_n is the probability of packet success, n is the total number of parallel packets transmitted on the channel, N_b is the number of bits/packet and k is the number of packets transmitting in parallel (1 or 2 or 3 or 4) and P_{b_n} is the probability of bit error given n parallel packets on the channel. For convenience, equation (4.4.7) is for three users. For example if the total number of parallel successful packets is 6 (probability of success= φ_6), and if the intended user transmits 2 packets in parallel ($P_k = P_2$), other 4 packets must come from the other two users ($2+u_2+u_3=6$).

Here we take worst case analysis scenario. The k packets transmitting in parallel by intended node will not act as interfering signals at the receiving node due to the orthogonality provided by Walsh functions. However all u_2, u_3 i.e. transmitting parallel packets are actually interfering signals (Conveying from different users where Walsh orthogonality can not be maintained).

For different number of packets n, equation (4.4.8) will be used to find the values of SNR and subsequently the values of Pb from the turbo coding results of [58]. We assume a simple turbo code of rate ½ and 256×256 interleaving.

$$(SNR_n)^{-1} = \left[\frac{1}{E_b/N} + \frac{n-1}{PG} \right] \tag{4.4.8}$$

Here $n=1,2,3,....BU$ and E_b/N is considered for the thermal noise which is assumed as 0.001. Equation (4.4.8) models the cumulative effects of n spread packets received in parallel at the intended user as an equivalent AWGN with interface density equal to $S(n-1)/WW$; where WW is the spread spectrum bandwidth;

$$PG = \frac{T_b(Bit duration)}{T_c(Chip duration)} = WWT_b$$

The probability of a typical node transmitting one or more packets on the channel $(P_0, P_1 .... P_B)$ is made dependent on its queue content $(\xi(0), \xi(1), \xi(2).....)$ which finally come to be the function of generation probability p and the probability of packet success on the channel i.e. $\beta_1, \beta_2, \beta_3, ... \beta_B$. If the steady state distribution of the number of packets queued in a typical node is $\xi(0), \xi(1), \xi(2).....$it follows by the buffer adaption policy that

P_0=probability of node transmitting no packet on the channel=$\xi(0)$ (4.4.9)

P_1=probability of node transmitting one packet on the channel=$\xi(1)$ (4.4.10)

Meaning switching node will transmit one packet if they have one packet in the queue..........

P_B=probability of node transmitting B packets on the channel=$1 - \sum_{i=0}^{B-1} \xi(i)$ (4.4.11)

Meaning nodes will transmit B packet in parallel if there are B or more packets in the queue, the essence of adaption to queue contents.

P_0, P_1, P_2, P_3, P_4 refers the transmission action of a switched SS. As the network configuration changes, so will the queue distribution of ξ, hence P_0, P_1, P_2, P_3, P_4 are dynamically adjusted by the node every packet so as to react to network configuration and node activity. $\xi(0), \xi(1)........\xi(BU)$ are the steady state distribution of the values of packets in the queue.

We assume a maximum queue size which is at least equal to BU i.e. $K \geq BU$. B is the maximum number of packets simultaneously sent in parallel from a typical node (B=4). U is the maximum number of users per hop (i.e. it is three in first hop of Figure 4.2-1) which are taken as 5, 6 and 7 in this work.

Starting with a certain estimated distribution for the number of packets in the queue $\xi(0),\xi(1)........\xi(BU)$ values of P_0,P_1,P_2,P_3,P_4 can be computed from equations (4.4.9)-(4.4.11). Based on corresponding values of P_0,P_1,P_2,P_3,P_4 all α_l and β_l are computed from equations (4.4.1)-(4.4.6). Next we solve the state equation $\xi = [A][\xi]$. Where a_{ij} are the elements of A and we obtain a solution ξ. This is again used iteratively to get new values of P_0,P_1,P_2,P_3,P_4 and new values of a_{ij} then a new solution ξ and the process repeats till the steady state i.e. values of ξ do not change iteration to iteration. To find the steady state solution of the number of packets at the intended user generation i.e. $\xi(0),\xi(1),.............\xi(UB)$, we initially take estimated values such as 0.5,0.2,0.1,0.1,0.1. For queue stability we require the following condition

$$\left(\frac{B+1}{2}\right)p \prec \bar{\beta}$$

Where $(1-p)0+1\left(\frac{p}{B}\right)+2\left(\frac{p}{B}\right)+....=\left(\frac{B+1}{2}\right)p$ is the average sum of number of packets generated or routed. $\bar{\beta}$ is the average packet service probability of a given node is given by:

$$\bar{\beta} = 1\beta_1 + 2\beta_2 + ...+ B\beta_B$$

Since our Network is connection oriented, call establishment time is a crucial measure. The call establishment time is determined by the sum of the buffer occupancy of all SS on the most crowded route taken by call establishment control packets. To determine this, one has to work at the statistics of this most crowded request link.

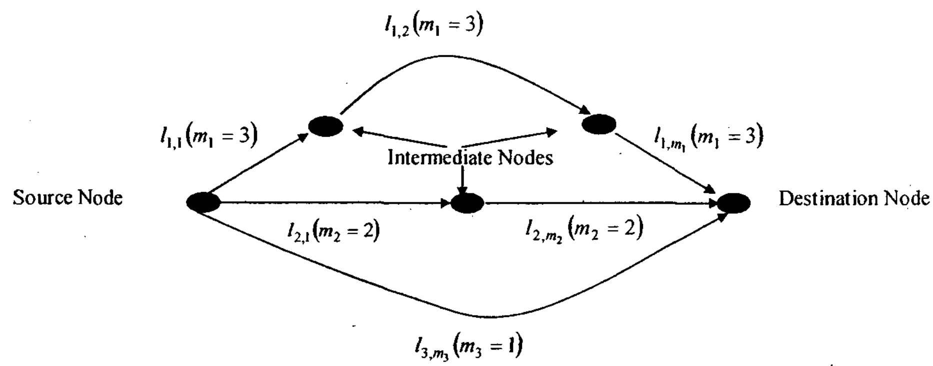


Figure 4.4-2 Network scenario of the longest (upper) and shortest route (lowest)

Table 4.4-1 Distribution of Packets in different paths that corresponds to the choice of path 1 is the most crowded path

r_i =0							r_i at Path 1	r_i at Path 2	r_i at Path 3
							0	0	0
r_i =1	r_i at Path 1	r_i at Path 2	r_i at Path 3	r_i at Path 1	r_i at Path 2	r_i at Path 3	r_i at Path 1	r_i at Path 2	r_i at Path 3
	1	0	0	1	1	0	1	1	1
r_i =2	2	0	0	2	2	0	2	2	2
	2	1	0	2	2	1	-	-	-
	2	0	1	-	-	-	-	-	-
	2	1	1	-	-	-	-	-	-

Figure 4.4-2 shows such a network having a source and a destination with few intermediate nodes. For the call establishment (as a worst case analysis) we work with the most crowded request link in Figure 4.4-2 which is the path having

three links ($l_{Path\,Number,link\,Number}$). Since the selection of best route is done based on call establishment phase on the other hand, once the call is established, for data packets the least crowded path is then typically selected (like source bridge technique).

From Figure 4.4-2, to determine the maximum or minimum delay required for the most and least crowded request links, it is necessary to find the probability of total sum of packets in queue on the i^{th} path having m_i links and total of links having r_i number of packets, which can be written as:

$$b_{i,r_i,m_i} = \sum_{l_{i,1}=0}^{k} \sum_{l_{i,2}=0}^{k} \cdots\cdots \sum_{\substack{l_{i,m_i}=0 \\ l_{i,1}+l_{i,2}\cdots\cdots+l_{i,m_i}=r_i}}^{k} \prod_{j=1}^{m_i} \xi_{i,j}(l_{i,j})$$

where $\xi_{i,j}(l_{i,j})$=Probability of j^{th} queue of i^{th} path having $l_{i,j}$ packets.

In Table 4.4-1 a typical distribution of packets in different paths is shown which is used to calculate the maximum delay for the control packets. In this case, for convenience and without losing generality three paths $(n_p = 3)$ have been chosen, where n_p is the value of the number of paths available. From Table 4.4-1 a generalized form for different number of packets in the most crowded path (worst case control packets) can be written:

We start with the easy case, $\alpha_0\big|_{n_p,m_1,m_2,\cdots m_{n_p}} = \prod_{a=1}^{n_p} b_{a,0,m_a}$ which multiply the probability of paths, each with zero packets to give the probability that the most crowded path has zero packets

Similarly,

$$\alpha_1\Big|_{n_p,m_1,m_2,\cdots m_{n_p}} = \sum_{a=1}^{n_p} b_{a,1,m_a} \prod_{\substack{i=1 \\ i\neq a}}^{n_p} b_{i,0,m_i} + \sum_{a=1}^{n_p}\sum_{\substack{c=1 \\ c\succ a}}^{n_p} b_{a,1,m_a} b_{c,1,m_c} \prod_{\substack{i=1 \\ i\neq a\neq c}}^{n_p} b_{i,0,m_i} + \prod_{a=1}^{n_p} b_{a,1,m_a}$$

$$\alpha_n\Big|_{n_p,m_1,m_2,\cdots m_{n_p}} = \sum_{a=1}^{n_p} b_{a,n,m_a} \prod_{\substack{i=1 \\ i\neq a}}^{n_p}\sum_{l=0}^{n-1} b_{i,l,m_i} + \sum_{a=1}^{n_p}\sum_{\substack{c=1 \\ c\succ a}}^{n_p} b_{a,n,m_a} b_{c,n,m_c} \prod_{\substack{i=1 \\ i\neq a\neq c}}^{n_p}\sum_{l=0}^{n-1} b_{i,l,m_i} + \prod_{a=1}^{n_p} b_{a,n,m_a} ;n\succ 0$$

α_n the probability that the most crowded path taken by the control packets has a sum of n packets in all its queues. First term of α_n corresponds to the intended crowded path number a having a total sum of n packets from all queues, while sum of all packets in other paths is less than n. For the second term of α_n, two paths can have the same sum of packets on all queues i.e. n while the third one has less than n. $n=1,2,\ldots l_{max}BU$; where l_{max} is the maximum number of links in one path.

In reality we don't know the values of $n_p, m_1, m_2, \cdots m_{n_p}$,so we fit probability distribution to their random variables. It follows that the probability of finite number of packets (n) averaged over a random number of paths (n_p) each having random number of links (m_i) is given by the following equation:

$$\bar{\theta}_n = \sum_{n_p}\sum_{m_1}\sum_{m_2}\ldots\ldots\sum_{m_i}\alpha_n\Big|_{n_p,m_1,m_2..m_i} \times p(m_1,m_2..m_i)\times p(n_p)$$

Where $p(m_1, m_2\ldots m_i)= \dfrac{1}{\sum m_1 \sum m_2 \ldots \sum m_i}$;is the probability of jointly uniformly distributed $m_1, m_2\ldots m_i$ links on the 1st, 2nd,... ith paths, and $p(n_p)=1/$*(Total possible number of paths* $n_p(max)$*);* is the probability of the uniformly distributed 1st , 2nd ,...ith paths respectively.The average delay for the call establishment i.e. maximum sum of packets in all queues for the most crowded path then becomes:

$$\bar{\lambda} = \sum_{n=1}^{l_{max}BU} n\bar{\theta}_n$$

The delay jitter for the control packets can be expressed as follows:

$$\sigma_\theta = \sqrt{\sum n^2\bar{\theta}_n - \left(\bar{\lambda}\right)^2}$$

Now for the data packets, which commences after call establishment, the shortest path have been discovered. So the statistics of the least crowded path should be followed by all data packets. Analogous to the aforementioned steps a generalized form for different number of packets in the least crowded path (data packets) can be written as:

$$\gamma_n = \sum_{a=1}^{n_p} b_{a,n,m_a} \prod_{\substack{i=1\\ i\neq a}}^{n_p}\left(1-\sum_{l=0}^{n-1} b_{i,l,m_i}\right) + \sum_{a=1}^{n_p}\sum_{\substack{c=1\\ c>a}}^{n_p} b_{a,n,m_a} b_{c,n,m_c} \prod_{\substack{i=1\\ i\neq a\neq c}}^{n_p}\left(1-\sum_{l=0}^{n-1} b_{i,l,m_i}\right) + \prod_{a=1}^{n_p} b_{a,n,m_a}$$

Where the first term of γ_n corresponds to the intended least crowded path having n packets on all queues while the sum of all packets in other paths is greater than n. For the second term of γ_n, two paths have equal sum of packets on all queues i.e. n while the third one has greater than n packets. The third term is the multiplication of the probability of paths, each with n packets to give the probability that the least crowded path has n packets.

The probability of a finite number of packets (n) averaged over a random number of paths (n_p) each having random number of links (m_i) is given by the following equation: $\bar{\varepsilon}_n = \sum_{n_p}\sum_{m_1}\sum_{m_2}......\sum_{m_i} \gamma_n\Big|_{n_p,m_1,m_2.......m_i} \times p(m_1, m_2..m_i) \times p(n_p)$

The average delay for the data packets i.e. minimum number of packets in all queues of least crowded path and the delay jitter are expressed in equation (4.4.12) and (4.4.13).

$$\overline{D} = \sum n\bar{\varepsilon}_n \qquad (4.4.12)$$

$$\sigma_\varepsilon = \sqrt{\sum n^2\bar{\varepsilon}_n - (\overline{D})^2} \qquad (4.4.13)$$

4.5 Results and Discussion

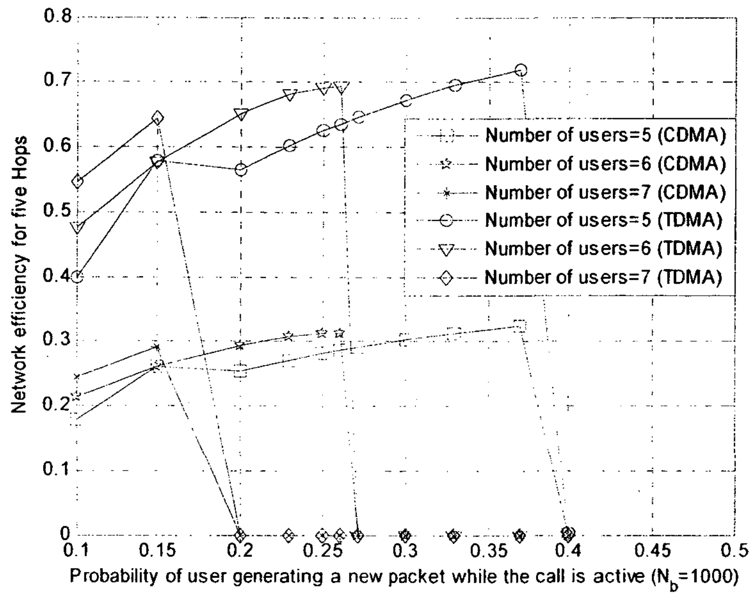


Figure 4.5-1 Network efficiency for TDMA and CDMA systems

In a CDMA system the network efficiency is given by equation (4.5.1) which is to be compared with equation (4.5.2) the network efficiency for the TDMA system [5].

$$\eta\eta_{CDMA} = \frac{U}{2WW}\bar{\beta}.R_b.r'.number\ of\ hops\left(\frac{N_b - Overhead}{N_b}\right) \qquad (4.5.1)$$

Here $U\bar{\beta}R_b$ is the successful number of bits/sec for all users; r' is the FEC rate which reduces the bandwidth efficiency by ½, *number of hops* is an independent parameter which accounts for the automatic band width reuse, $\left(\frac{N_b - Overhead}{N_b}\right)$ is considered for the overhead effects and *WW* is the available spread spectrum bandwidth. Due to half duplex TDD operation we multiply the efficiency by ½.The efficiency of a TDMA system carrying the same traffic as the CDMA system while ignoring the channel errors, overhead bits and assuming minimum FEC.

$$\eta_{TDMA} = \frac{U.number\,of\,hops.\bar{\beta}.R_b}{2WW} \tag{4.5.2}$$

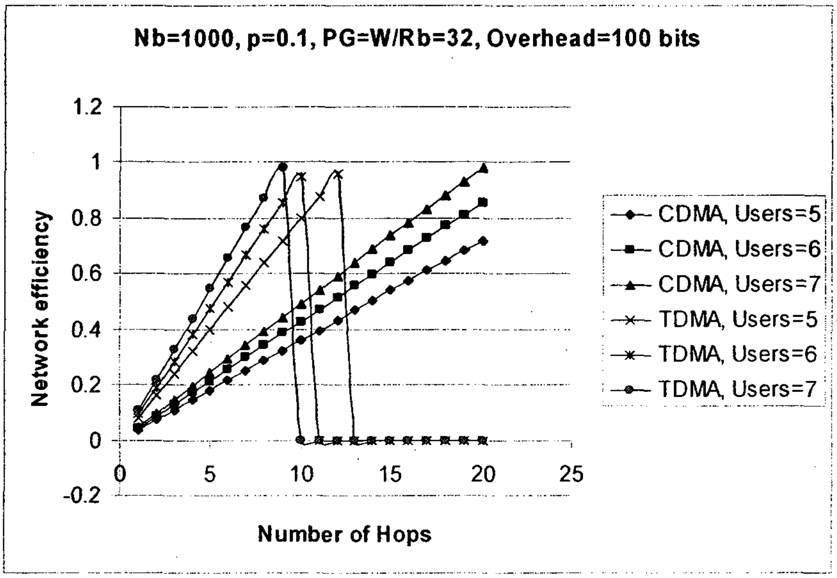


Figure 4.5-2 Network efficiency for TDMA and CDMA systems

Equation (4.5.1) implies the dependence of η_{CDMA} on the channel conditions and correct packet detection probability through $\bar{\beta}$. This takes into account certain worst case assumptions where the number of nodes and load increases. In such case probability of correct packet detection will deteriorate with parallel transmission permitted. However even with this deterioration CDMA still outperforms TDMA counterparts shown in Figure 4.5-3.

Figure 4.5-1 show the network efficiencies of CDMA and TDMA systems for different values of p (i.e. users generating a new packet while the call is active) for 5 hops. It is evident that for this number of hops the network efficiency for TDMA out performs CDMA.

Figure 4.5-2 shows the same comparison of CDMA and TDMA systems for different number of hops at a constant p. Here it is clear that, while giving much advantage (i.e. not considering overheads, channel errors, FEC (Bandwidth efficiency reduction effect)) to TDMA system, still CDMA system out performs TDMA counterparts.

The average packet delay of CDMA system (equation 4.4.12) is compared with the TDMA system given by the following well known equation [5]:

$$Delay_{TDMA} = \frac{U'.\eta_{TDMA}}{2(1-\eta_{TDMA})} + \frac{U'}{2} + 1 \quad Packets$$

Where the effective carried users $U' = \frac{2.U.\bar{\beta}.number\ of\ hops.R_b}{W}$

The delay jitter of the CDMA system (equation 13) is compared with the TDMA system as given by the following equation:

$$\sigma_i = \sqrt{\sum_{i=1}^{BU} i^2 P_i - (\bar{d})^2}$$

Where $\bar{d} = \sum_{i=1}^{BU} iP_i$ and $P_i = (1-\rho_{TDMA})\rho^i_{TDMA}$ where $\rho_{TDMA} = \eta_{TDMA}$ is the load on the system (arrival rate/service rate). In this case we take M/M/1 queuing [5] as a worst case scenario due to its higher delay value compared to M/D/1 queuing.

From the delay values for both cases in Figure 4.5-3, it is evident that when the number of hops increases, the delay increases faster in the case of TDMA. Also as we increase the number of hops the efficiency of CDMA system improves and may exceed 1 due to automatic band reuse while for TDMA, as the number of hops increases η_{TDMA} will increase and it will reach to only 1. This leads to an unstable situation yielding very high delays. TDMA problems culminate when the total number of users in all hops of the CDMA system will have to share a large single TDMA frame. Besides large delays peak transmission power will then increase tremendously in TDMA systems compared to CDMA counterpart which is a well known fact in wireless transmission. It is however possible in principle to divide the total number of users from all hops over many TDMA frames each frame for a specific geographic region. In such case gateways will still be needed to relay the packet between different TDMA frames. This entails accumulating complete packets and subsequent queuing. Meaning the end to end TDMA delay will arise from the individual TDMA frames (Sub networks) as well as delays at the gateway making TDMA delay even worse than the single frame for all case. Also needless to say TDMA access needs much higher transmission power than CDMA counterparts). FDMA based WIMAX system have synchronization, power etc. constraints similar to TDMA system and so will perform closely the same [23].

In [35], the performance analysis of a multi hop mesh network is proposed where the average delay and delay jitter is calculated for a 3-hop wireless path. The average delay for such 3 hop mesh is 8.4 ms which is much higher than the average delay (CDMA) shown in Figure 4.5-3. Also our delay jitter (Figure 4.5-4) performance for CDMA system outperforms the delay jitter calculated in [35] (where the delay jitter is 135.5 ms). In this thesis the delay and delay jitter performance is calculated in packets. However, translation from packets to mili seconds is given in table 1.

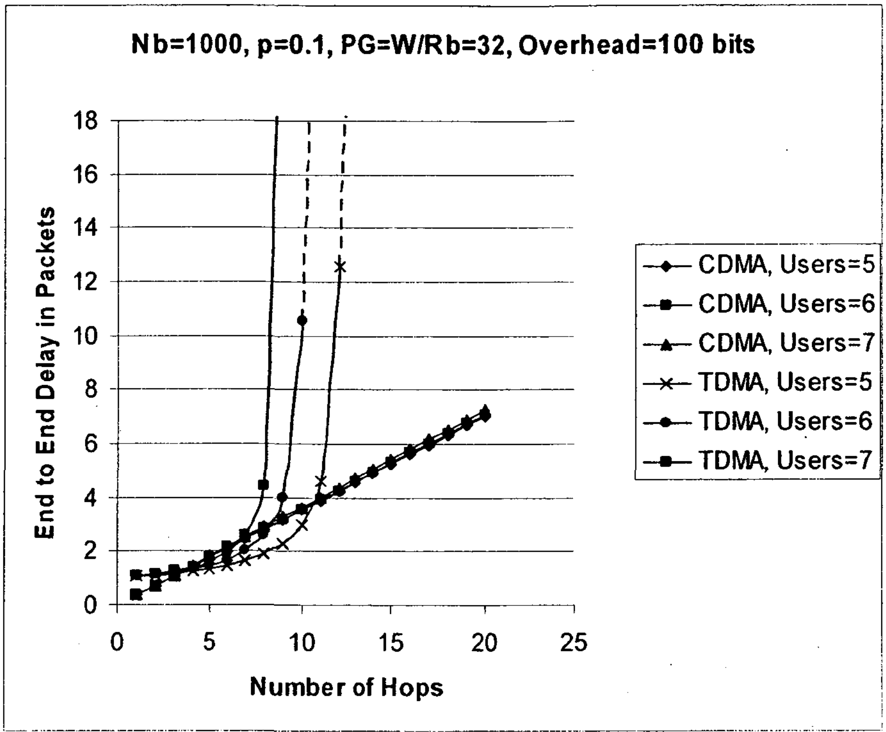


Figure 4.5-3 End to End Delay for TDMA and CDMA systems

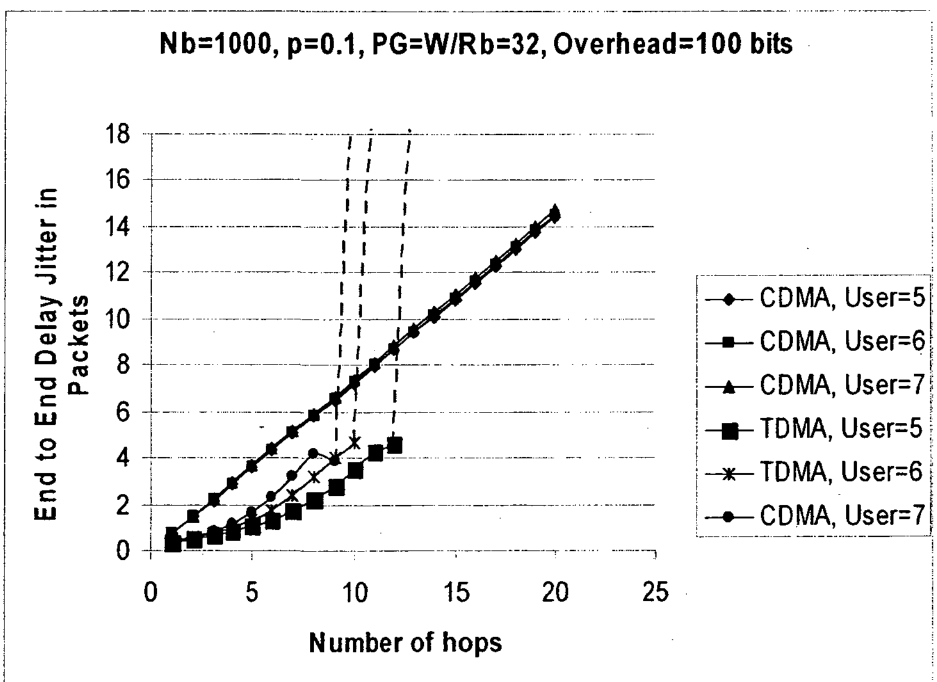


Figure 4.5-4 End to End Delay Jitter for TDMA and CDMA systems

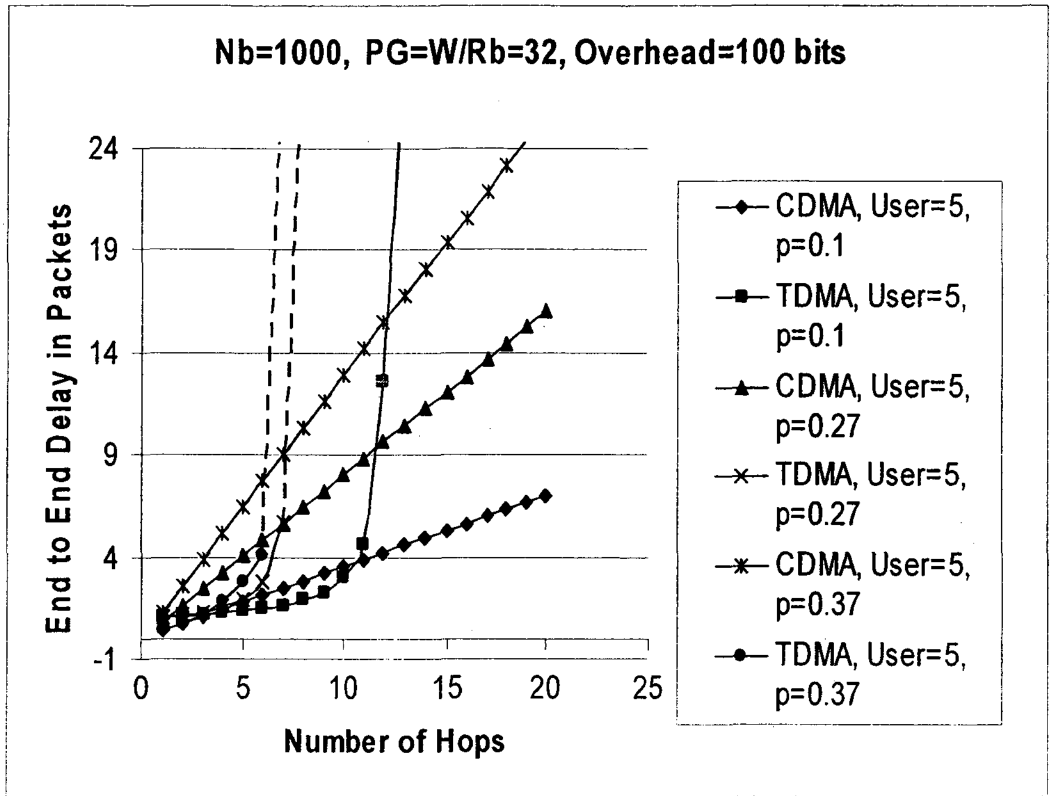


Figure 4.5-5 End to End Delay for TDMA and CDMA systems

Information Bit duration in CDMA is considered as (1/Rb) i.e. 1microsecond. While for the equivalent TDMA, it compresses to 1/W; where W is the total available bandwidth 32M in this analysis. In the course of comparison between CDMA and TDMA WMNs we have put TDMA at relative advantages, for example we ignored the channel errors, overhead bits and had no FEC rate loss. Yet CDMA delay performance is better.

In [49], the delay performance for wireless mesh network (Collision free MAC) was calculated as 2.5ms at 40 packets/sec which translate into a packet generation probability of p=0.1. At p=0.1 our CDMA delay performance (Figure 4.5-3) outperforms such delay in [31] at comparable number of hops, p value etc.

Figure 4.5-6 compares the end to end delay jitter for both CDMA and TDMA systems. It is evident that the delay jitter increases and the system become unstable for higher values of p, for the TDMA system. While at higher value of p, the CDMA system can still be operated.

Figure 4.5-7 shows a comparison of the network efficiency between the two systems for different number of Bits/packet. It can be seen that there is not much variation in the network efficiency in the case of TDMA system since overhead is neglected. While for the CDMA system the network efficiency varies widely due to overhead changes.

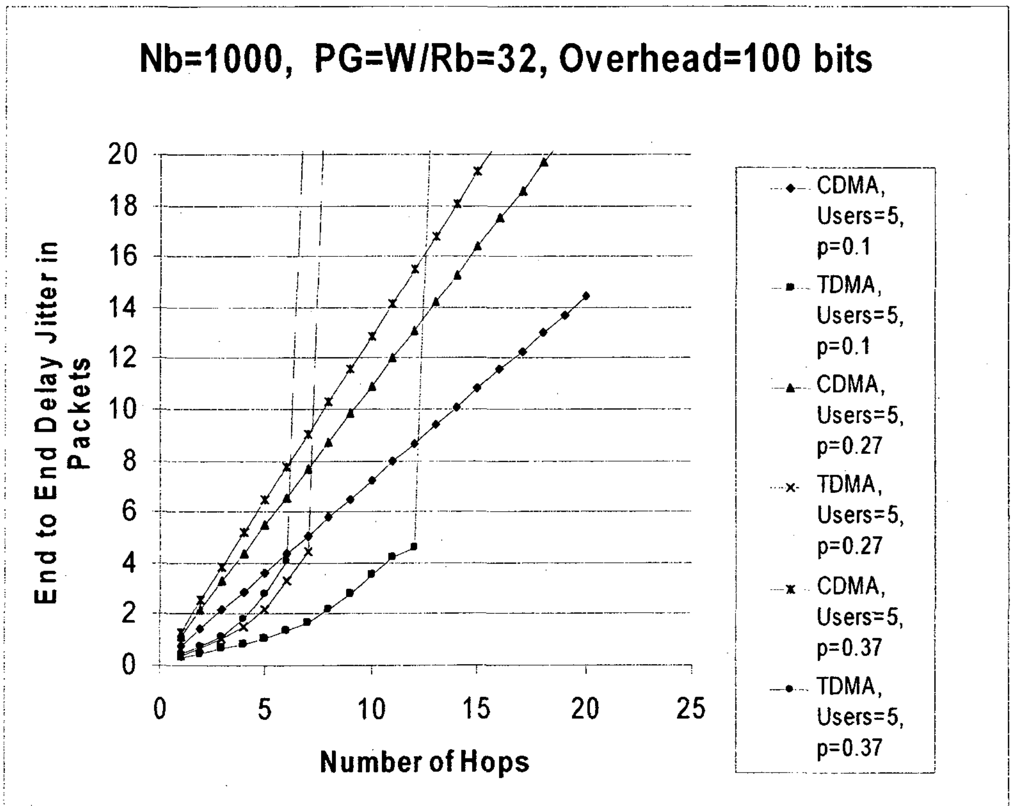


Figure 4.5-6 End to End Delay Jitter for TDMA and CDMA systems

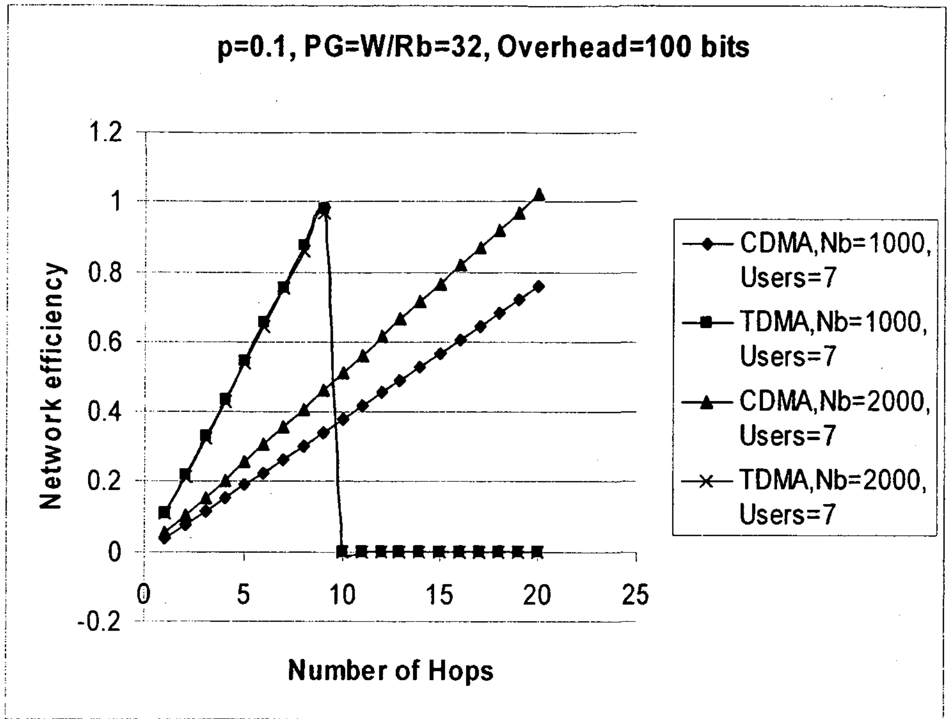


Figure 4.5-7 Network efficiency for TDMA and CDMA systems

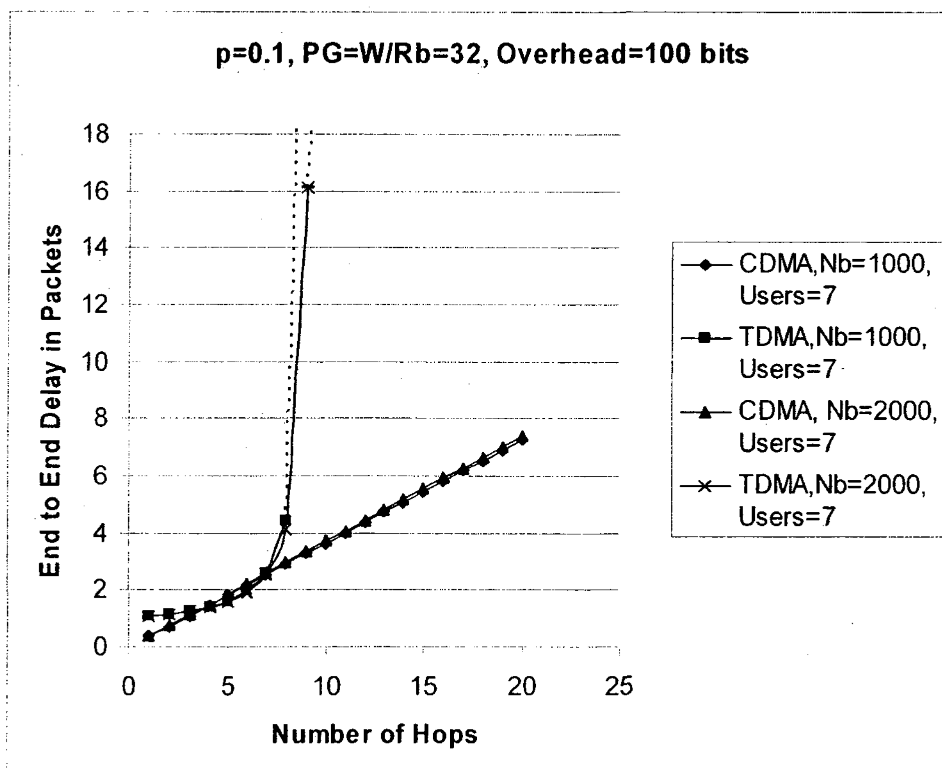


Figure 4.5-8 End to End delay for TDMA and CDMA systems

Figure 4.5-8 shows the end to end delay performances of the two systems for different number of bits/packet. For the CDMA case the delay does not change much. The effect of number of bits/packet is less on the probability of successful packet transmission on the channel. From this figure it is evident that Delay performance is much better in the case of CDMA system.

Figure 4.5-9 shows the Delay jitter for the systems at low value of p. It is evident that at low values of p, the number of bits/packet has little effect.

Figure 4.5-10 shows a comparison of the two systems at higher value of p. The variation in this case is visible. This means that the value of p (i.e. users generating a new packet while the call is active) affects the two systems. The system becomes unstable more noticeably in the case of the TDMA system.

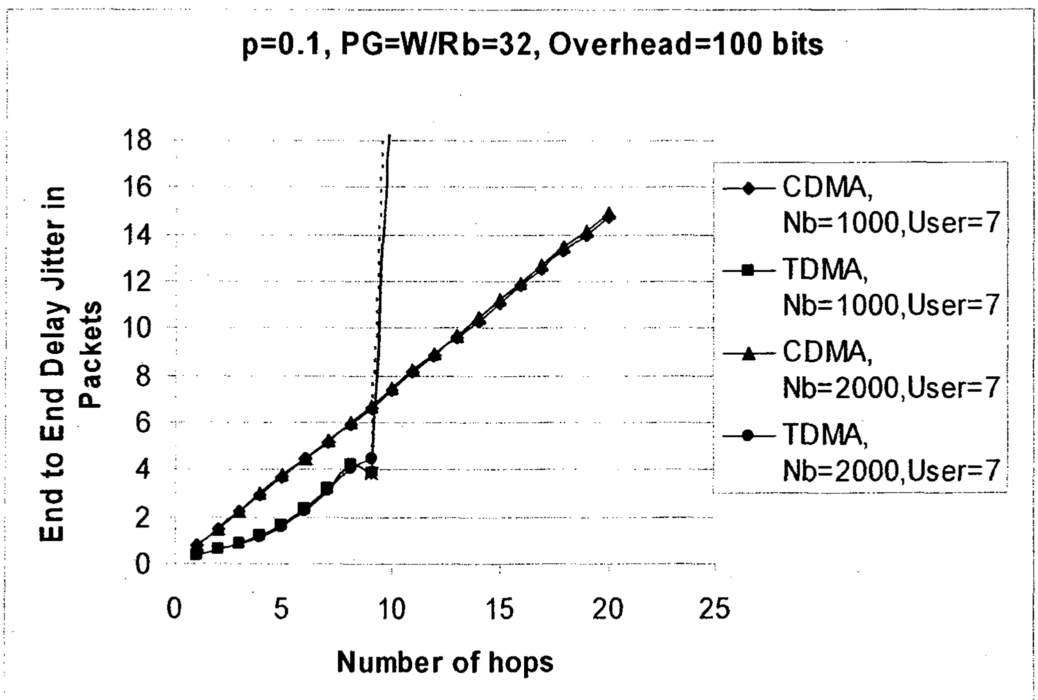


Figure 4.5-9 End to End delay Jitter for TDMA and CDMA systems

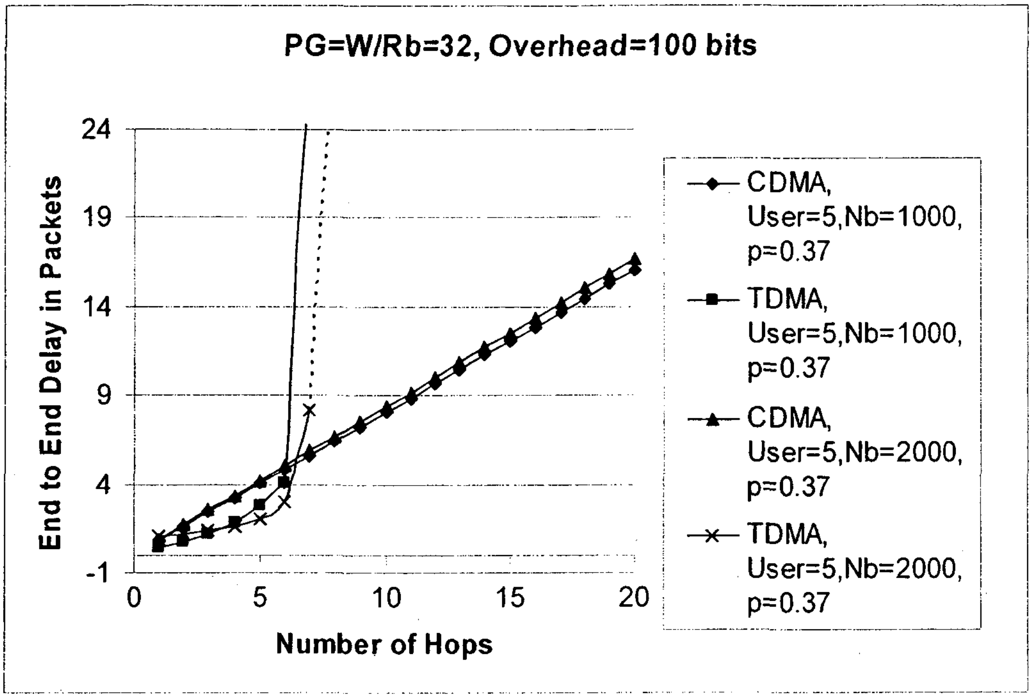


Figure 4.5-10 End to End delay for TDMA and CDMA systems

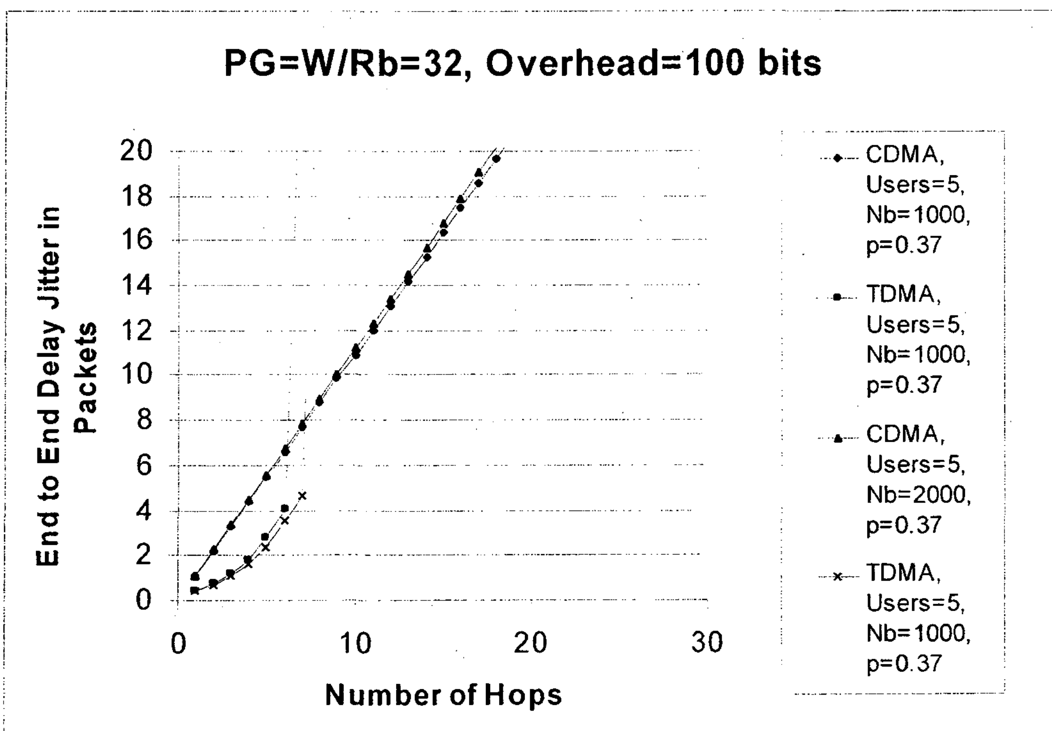


Figure 4.5-11 End to End delay Jitter for TDMA and CDMA systems

Figure 4.5-11 shows the comparison of the delay jitter for the two cases. Again it is evident that the delay jitter performance gets worse as p increases for both cases. The TDMA system gets unstable more rapidly at higher value of p.

4.6 Conclusion

In this chapter we have introduced a code division multiple access/Time division duplex technique CDMA/TDD for WMNs. We chose such system because CDMA maximum transmitter power is much less than TDMA and FDMA counter parts, which is an important asset for wireless networks. Further because of easy configuration and amenability to mobility, the CDMA approach is an almost play and plug technology for wireless access, making it cost effective for implementation by WMNs subscriber station (SS). Along with the introduction to CDMA/TDD system we also analyzed its performance. We make comparisons with the TDMA based systems in terms of network efficiency, delay and delay jitter. Results show that the CDMA system outperforms its TDMA counterparts.

With the help of turbo coding, the new network uses parallel transmission from the SS to improve the network QoS. Meaning, this technique will help the service providers to provide the users with a better wireless internet with low cost due to increased capacity.

In the next chapter we investigate the interactions between the MAC, PHY and Network layer in WMNs which is defined in the literature as cross layer design.

CHAPTER 5

CROSS LAYER OPTIMIZATION IN WIRELESS MESH NETWORKS

A cost effective way of improving the QoS in WMNs and hence in 4G is to build an efficient interaction mechanism amongst the layers of the network hierarchy. For this reason we move towards the cross layer design of WMNs where different layers such as MAC, PHY and network layers help each other to improve the QoS. In this chapter we design a cross layered architecture where the MAC, PHY network layers are involved. As MAC access techniques we use Spatial Time Division Multiple Access (STDMA) and CDMA. For both access techniques we optimize the routes in the network layer utilizing the information received from the MAC and PHY layers.

5.1 Introduction

In this thesis we combine a new adaptive Spatial TDMA and optimum routing for cross layer design in wireless mesh networks. For wireless media access we investigate uniform STDMA as well as a new adaptive version thereof. An overall optimization routine finds simultaneously the best route and the best capacity allocation (termed as uniform and adaptive STDMA allocation) to various nodes. Figure 5.1-1 shows the design of cross layer architecture. In our analysis, the optimization routine finds the best capacity allocation in the MAC. Simultaneously, it utilizes the information of the correct packets from the physical layer to optimize the routes of all calls generated by each node.

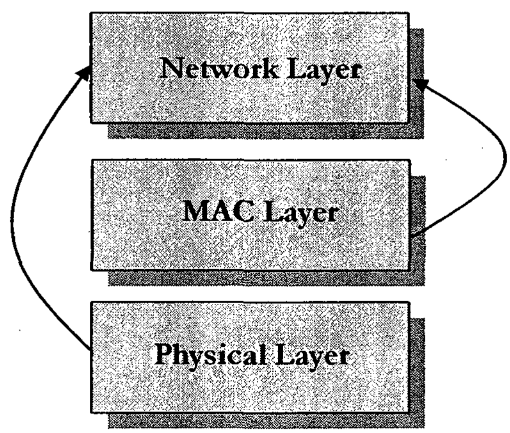


Figure 5.1-1 Cross layer architecture for wireless mesh networks

This optimization routine minimizes the average end to end packet delay over all calls subject to various constraints. We also investigate the level of connectivities on the various performance criteria taking two cases of possible topologies. Given a certain wireless channel and hence average packet transmission success probability (typically relayed from PHY layer to routing network layer in cross layer design), we compute the end to end average delay and delay jitter over all calls yielded by the cross layered optimization algorithm.

5.2 System Description

Figure 5.2-1 and Figure 5.2-2 show the two instances of topologies to be investigated.

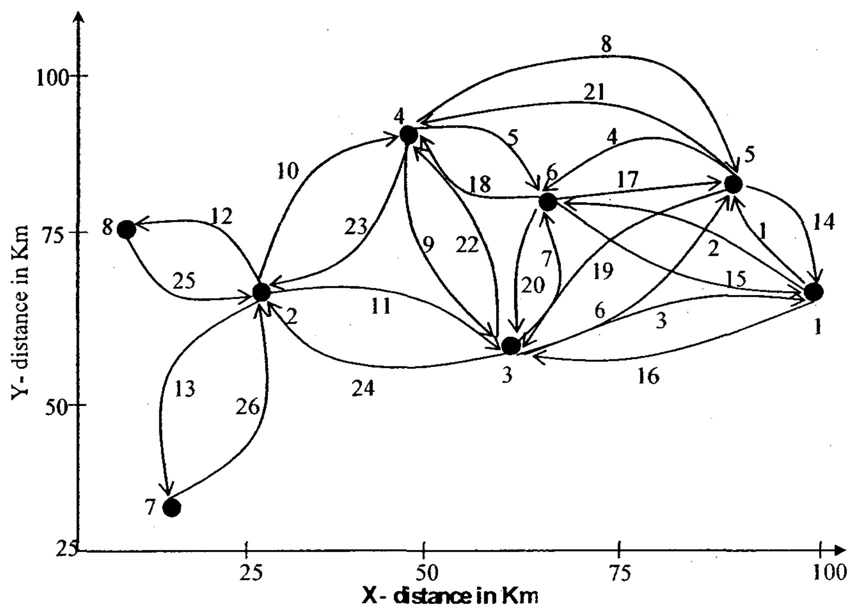


Figure 5.2-1 Topology 1 High Power nodes: High connectivity

All the cross layered routing, STDMA and optimization techniques are general in nature and apply as well to other topologies. In the first topology the subscriber stations (nodes) have high connectivities (possibly due to their large powers). As the connectivity gets higher, distant nodes can communicate with each other through lower number of hops. However, for the second topology the scenario is reversed due to their low power. In the second topology distant nodes will communicate via a larger number of hops.

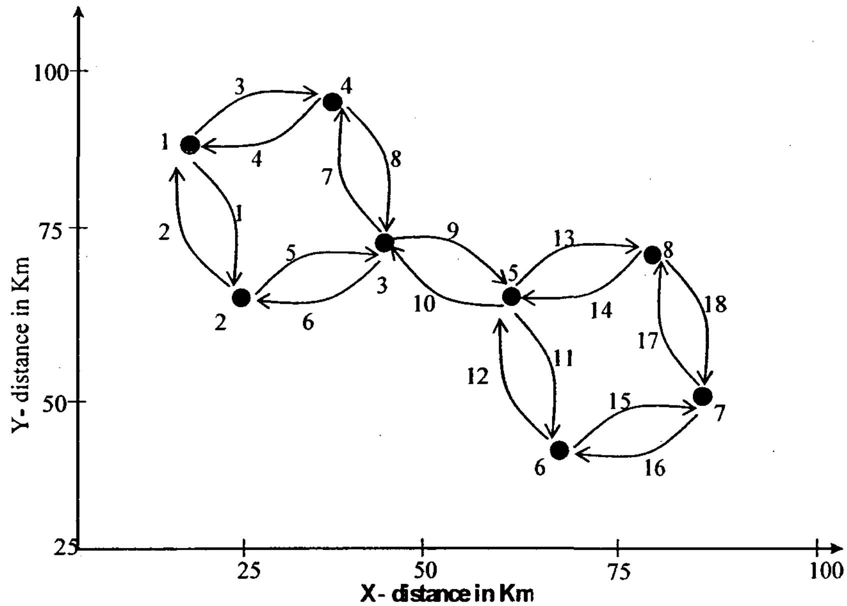


Figure 5.2-2 Topology 2 Low Power nodes: Low connectivity

The subject topologies are packet switched and not circuit switched. Also they are assumed to be connection oriented in the sense that end to end nodes have to call and establish a virtual connection with their destinations before transmitting data packets. In this regard efficient routing of the data packets is essential. Many routing techniques have been designed for routing packets among peer nodes in ad hoc WLANs [53]. However in most of the cases shortest path is chosen to reduce the complexity. In this thesis each node optimally finds the best routes over the whole network while considering routing decisions by all other nodes.

We assume that all nodes have the full topology information (link and node number) of the mesh network as well as the correct packet detection probabilities for all links and a reference timing starting point for the STDMA frames. This is achieved via HELLO and other control packets as outlined before. Reference timing can be obtained from a certain base station pilot signals. The nodes and the links information are exchanged amongst the nodes such that they know distributively each of their particular transmission time slot (after running the cross layered optimization in this thesis). Network topology information can be obtained at all nodes by using AODV [19] routing protocol for example. After such protocol updates the network topology information, our cross layered optimization in this thesis will then be applied at all nodes so as to find the best slots and routes allocation. Such mechanisms are not the subject of this thesis. To initially obtain network topology, classic clustering, and routing of local networks have been the subject of intense research [27], [28] which we do not address in this thesis. Added, implementation process at the nodes due to optimization etc. are also not addressed. Moreover for cross layer design the PHY layer in each node will estimate and deliver the correct packet success probability Pc value to the network layer. Mechanisms for such estimation (e.g. known preamble verification or cyclic redundancy check CRC etc.) are beyond the intended scope of this thesis.

5.3 MAC Layer (STDMA Operation)

The nodes have information about the whole network topology as discussed in the previous section (all node identification and all link identification) from the frequent handshaking among the nodes. For both subject topologies the following rules are adopted for the conflict free MAC operation of all links[37]:

We take topology 1 for describing our MAC operation. An arc (link) denotes the presence of an available direct communication channel between two nodes e.g. with link 1, node 1 can transmit to node 5 of course with a certain packet success probability. So for other links. Consequently node 6 and node 2 cannot communicate directly. Typically [37] transmission commences with the lowest link number (link 1) where node 1 transmits to node 5 in the first clique (time slot). It is an easy exercise to show for any topology that starting from another node, the number of cliques in the whole network and links transmitting in parallel within each clique will be the same regardless of the starting node. The first slot is assigned to the links in the first clique. Now we investigate other neighbors of node 1, to see if they can transmit at the same time slot or not. So we start with the lowest node number of node 1's neighbors. In this case node 3 comes first. From node 3 we investigate one by one all the outgoing links (link 3, 6, 7, 22 and 24) to see if one of them can be activated in the same time slot (slot 1). If we put node 3 in transmitting mode and activate link 3 i.e. the lowest link number, node 5's reception will be affected by the unintentional transmission of node 3 (link 6). This will lead to a conflict because a particular node can not receive simultaneously from two nodes. We try links 6, 7, 22 and 24 and they lead to similar conflicts. Consequently node 3 can not transmit in slot 1.

Now we investigate other neighbors of node 1 (node 6 excluding node 5 because it is already in the receiving mode) in which case similar conflicts repeat.

Now we move to other nodes which are not the neighbors of node 1. In this case we start with the lowest node number among them. This is node 2 which is investigated next. If we activate any of the outgoing links of node 2 such as link 10, 12, 13 excluding link 11, there will be no conflict of node 5's reception. If we activate link 11, node 3 will be in receiving mode and will simultaneously receive from node 1 and node 2. This will lead to similar conflict stated above. Among the three links (link 10, 12, 13) we start with the lowest link number 10 which can be activated in clique 1. Proceeding further with nodes at 2 or more hops from node 1 (e.g. node 4, 7 and 8) does not reveal any further possibility of non conflicting transmission in clique 1 (first slot).

Now we come to clique 2 (second slot) where we start with the next lowest link number which is link 2 (node 1 transmitting to node 6). By inspecting node 1's neighbors and according to previously stated rules none of the neighboring nodes of node 1 can be in transmitting mode in clique 2. So in the next step we investigate nodes at 2 hops from node 1 which is node 2 (not an immediate neighbor of node 1) which again can be in the transmitting mode (link 10, 12, 13- any of them can be activated excluding link 11 which will create similar conflict as stated in clique 1). Now we pick link 12 (since it is the next lowest link number after link 10) as link 10 is already selected in clique 1(first slot). Proceeding further does not reveal any more possible parallel transmissions in clique 2.

According to the previous rules for topology 1, if we denote C_i as the ith clique, we can form a clique cover, C, which is a set of maximal cliques $C = \{C_1, C_2, ...C_k\}$ having the property that every link of the network is contained in at least one member of C. Further a typical link usage parameter which is the total number of occurrences of a certain link in all cliques should be equalized as far as possible over all links. Traffic of these cliques are transmitted in corresponding time slots of STDMA frames. For example, the clique cover for the topology 1 shown in Figure 5.2-1is given by

$$C = \begin{Bmatrix} C_1, C_2, C_3, C_4, C_5, C_6, C_7, \\ C_8, C_9, C_{10}, C_{11} ....C_{24} \end{Bmatrix}$$

Where

$C_1 = \{1,10\}\, C_2 = \{2,12\}\, C_3 = \{3,25\}$.....

Where in C_1 (slot 1) link 1 and link 10 can transmit in parallel. Similarly for other cliques.

....

$C_{23} = \{23,14\}\, C_{24} = \{24,17\}$

While link usage parameters are:

$U_1 = 2; U_2 = 2; ....U_{12} = 4.......$
$U_{22} = 1; ......U_{25} = 4; U_{26} = 4$

Meaning link 1 appears in two cliques, link 2 appears in two cliques, link 12 appears in four cliques and so on. Here we see a near equalization of U_i values of all links as per the clique assignment algorithm. It is done to achieve such equalized distribution of links i.e. at each investigations of a node of possible links transmission at a certain clique e.g. links 10,12, 13 at clique 2, the clique selects link 12 with minimum value of $U_i\ (U_i = U_{12})$. Whenever a link is enabled during a certain clique, its usage parameter is incremented i.e. $U_i = U_i + 1$.

For topology 2 the cliques are given below according to the previously stated rules:

$C_1 = \{1,8,12,17\}\, C_2 = \{2,9,16\}$.....

....

$C_6 = \{6,4,11,18\}\, C_7 = \{7,13,15\}$

Also for topology 2 the usage parameters are determined in a similar manner as topology 1.

Since the link usage parameters are equalized as far as possible during the links assignments to different cliques, they are never exactly equal; consequently the number of packets transmitted over each link in a certain STDMA frame is never the same for all links. Needless to say, the link usage parameter also gives the number of packets which a certain link transmits in a typical STMDA frame. The above MAC assignment we call uniform STDMA. It is also conceivable that the different links would be transmitting a number of packets per frame proportional to their instantaneous traffic. The latter case we call Adaptive STDMA, and will discuss it later.

We should note that synchronization in packet radio network implies that each slot is expanded by a small guard time which is used to offset the varying geographical distances between nodes. The capacity of the channel lost to this guard band is not taken into account in our calculations of delay performance. So the system delay is lower than what we would obtain if this lost capacity were accounted in this analysis.

5.4 Traffic flow Characterization

In this section we discuss the uniform STDMA allocation. Let II_l be the number of calls simultaneously travelling through l^{th} link. For each topology, a link may carry a volume of calls less than or equal to the total number of calls generated by all the nodes. For example, the first topology has 8 nodes which can create maximum of N=(8*7)56 calls at a certain time. So a link can carry maximum of N=(8*7)56 calls if all calls happen to be routed through that link. However optimum routing policy tries to equalize the traffic over all links according to the control variable X which is a binary decision value of either 1 or 0. If a call takes a certain route the value of X is 1 and zero otherwise, e.g. $X(3,5)=1$ means call number 3 has taken route number 5, $X(4,6)=0$ means call number 4 is not taking route number 6. Consequently the traffic over l^{th} link can be expressed as

$$II_l = \sum_{c=1}^{CC'} \sum_{R=1}^{R_c'} \theta_c'(c) X(c,R) \tag{5.4.1}$$

Where, CC' is the total number of calls generated by all nodes, R_c' = Total number of routes for a certain call and θ_c' is the call traffic intensity of call number c. for c=1, the percentage of time spent on such call is given by

$$\theta_1' + \theta_2' + \theta_3' \dots\dots\dots + \theta_7' = \theta_1$$

For c=1; θ_1' gives the call probability from node 1 to node 2, θ_2' gives the call probability from node 1 to node 3 and so on. While θ_1 is the total probability of node 1 making any such calls.

Similarly we can write the following for the calls from other nodes, thus obtained for topology 1

$$\begin{aligned}
&\theta_8' + \theta_9' + \theta_{10}' \dots\dots\dots + \theta_{14}' = \theta_2 \\
&\theta_{15}' + \theta_{16}' + \theta_{17}' \dots\dots\dots + \theta_{21}' = \theta_3 \\
&\vdots \\
&\vdots \\
&\theta_{50}' + \theta_{51}' + \theta_{52}' \dots\dots\dots + \theta_{56}' = \theta_8
\end{aligned}$$

Each call has an initial source and destination which makes a source destination pair. The packets from source to destination nodes may travel through intermediate nodes. Therefore, the traffic load in l^{th} link is given by $\lambda' \times \Pi_l$ (packets/sec). Here we assume that all calls carry a uniform traffic λ' packets/sec. However all links do not have the same traffic load$(\Pi_1 \neq \Pi_2 .... \neq \Pi_{26})$. If we now take the effect of packet success probability P_{c_l} (estimated and relayed from the PHY layer to the network layer) on l^{th} link, the traffic load becomes

$$\lambda_l = \lambda' \frac{\Pi_l}{P_{c_l}} \text{ (Packets/sec)} \qquad (5.4.2)$$

Equation (5.4.2) represents a geometric distribution to number of packet time till success in which case number of packets transmitted for each packet becomes $\frac{1}{P_c}$ [5].

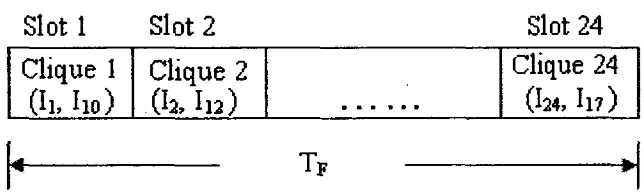


Figure 5.4-1 Links occupying the slots according to our MAC algorithm for

Uniform STDMA

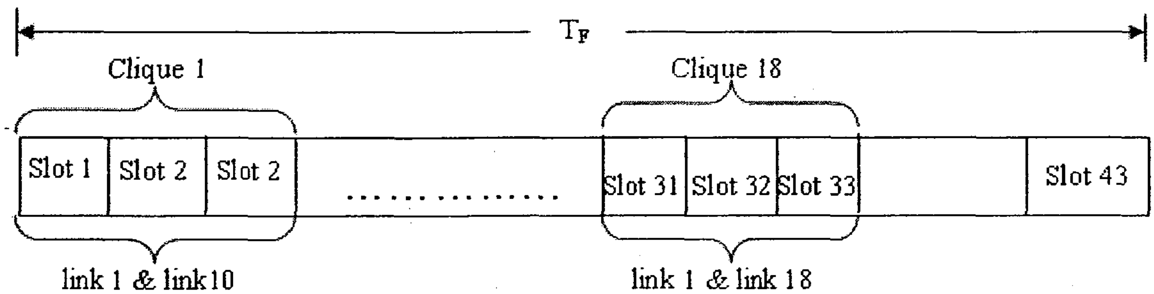


Figure 5.4-2 Links occupying the slots according to our MAC algorithm for Adaptive STDMA

In the worst case all packets of all N=56 calls will pass by subject link.

$$\frac{\lambda'(II_l)_{\max}}{P_c} \leq \frac{WW}{N_b \times Total\ No\ of\ Calls}\ packets/\sec$$

Where WW is the bandwidth assumed 100 Mbits/sec and N_b is 10000 bits/packet. Which says that the per link traffic service rate is less than or equal to link capacity.

For analysis purposes we take a random value of P_c for l^{th} link which is generated as a uniformly distributed random variate with a given mean and variances.

5.5 Adaptive and Uniform STDMA Allocation

In adaptive STDMA allocation we consider that the different links would serve (transmit) a number of packets per frame proportional to their instantaneous traffic. To accomplish this, we first compute the per clique service portion of the total channel capacity. Shortly after we use this to compute the per link capacity allocation. The per clique service is expressed as the ratio of the maximum amount of traffic of all links in a certain clique to the summation of maximum traffic in all cliques, i.e.

$$d_1 = \frac{Max(II_1, II_{10})}{Max(II_1, II_{10}) + Max(II_2, II_{12}) + \cdots\cdots Max(II_{24}, II_{17})}$$

$$d_2 = \frac{Max(II_2, II_{12})}{Max(II_1, II_{10}) + Max(II_2, II_{12}) + \cdots\cdots Max(II_{24}, II_{17})}$$

$$\vdots$$

$$\vdots$$

$$d_{24} = \frac{Max(II_{24}, II_{17})}{Max(II_1, II_{10}) + Max(II_2, II_{12}) + \cdots\cdots Max(II_{24}, II_{17})}$$

We note that $d_1 + d_2 + d_3 + \cdots\cdots d_{24} = 1$ i.e. the normalized channel capacity. Traffic values of $d_1, d_2, d_3, \cdots\cdots d_{24}$ will be estimated at the data link layer and then conveyed to the network layer by cross layer design after which $d_1, d_2, d_3, \cdots\cdots d_{24}$ are computed. Needless to say, in the uniform STDMA allocation we force $d_1 = d_2 = d_3 = \cdots\cdots d_{24} = \frac{1}{Total\ Number\ of\ Cliques}$ independent of the traffic needs.

Figure 5.4-1 gives definition of frame, slots, and cliques for the uniform STDMA system. The time frame T_F (expressed in number of slots not in seconds) is divided into 24 slots as there are 24 cliques in the first topology. A clique is a moment when several non conflicting links can be activated. For example link 1 and link 10 are activated in clique 1 (operating slot 1) and link2 and link 12 are activated in clique 2 (operating slot 2). In each slot one packet of Nb bits can be transmitted by each link. For example in slot 1 link 1 and link 10 operate, so they can each send 1 packet in that particular slot.

Now for adaptive STDMA allocation and for convenience and clarity we find the minimum of $d_1, d_2, d_3, \cdots\cdots d_{24}$ and assign one frame slot to this minimum clique service such that clique with smallest d value gets one slot and the STDMA frame T_F becomes

$$T_F = \left\lceil \frac{1}{d_{\min} = \min(d_1, d_2, d_3, \cdots\cdots d_{24})} \right\rceil \text{ slots}$$

Figure 5.4-2 shows an example of the adaptive STDMA frame where $d_1 = 0.07$ $d_2 = 0.023$ $d_3 = 0.023$ $d_4 = 0.07$ $d_5 = 0.023$ $d_6 = 0.023$ $d_7 = 0.023$ $d_8 = 0.023$ $d_9 = 0.09$ $d_{10} = 0.04$ $d_{11} = 0.023$ $d_{12} = 0.07$ $d_{13} = 0.04$ $d_{14} = 0.023$ $d_{15} = 0.04$ $d_{16} = 0.023$ $d_{17} = 0.023$ $d_{18} = 0.08$ $d_{19} = 0.023$ $d_{20} = 0.023$ $d_{21} = 0.023$ $d_{22} = 0.09$ $d_{23} = 0.07$ $d_{24} = 0.04$ and $T_F = 43$ slots.

For topology 1, links will transmit during the following number of slots:

$$\zeta_1 = \lceil T_F(d_1 + d_{18}) \rceil slots$$
$$\zeta_2 = \lceil T_F(d_2 + d_{10}) \rceil slots$$
$$\vdots$$
$$\vdots$$
$$\zeta_{26} = \lceil T_F(d_4 + d_9 + d_{19} + d_{21}) \rceil slots$$

We note that $\zeta_1 + \zeta_2 + \cdots \zeta_{26}$ is typically greater than or equal to T_F slots. The actual physical frame size seen by all links is still T_F. Thanks to the blessings of parallel transmission of many links at same slot as per the STDMA operation. Also we note that in the uniform STDMA case, $d_1 = d_2 = d_3 = \cdots\cdots = d_{24}$, but still one link can transmit in more than one clique as the adaptive case.

$\zeta_1 = \lceil T_F(d_1 + d_{18}) \rceil$ means that link 1 transmits in both cliques 1 and 18 as per Figure 5.4-2. Similarly other links will take their share by the STDMA frame. In the uniform STDMA (Figure 5.4-1) link 1 occupies 2 slots out of 24 slots as it is activated in clique 1 and clique 18. For the adaptive STDMA and taking minimum of $d_1, d_2, d_3, \cdots\cdots d_{24}$ and assigning one frame slot to this minimum clique service gives a frame nearly twice the frame size of uniform STDMA. Moreover it allows link 1 to transmit in 6 slots shown in Figure 5.4-2 which is obtained from the optimization process depending on the traffic needs. This means the optimization routine has a flexibility to assign more service to links depending on the traffic needs in adaptive STDMA which is absent in uniform STDMA.

We note that frames could be different in the two cases (Uniform and adaptive STDMA). Total number of cliques remains the same for the both cases. The slot size for both cases remain the same since it carries 1 packet of Nb bits for each link in both cases. The file size also remains same for the both cases. Corresponding values of capacity allocation to different cliques are different in the two cases.

When the applicable queued packet jumps to the top of the queue it may be lucky to be synchronized with the assigned consecutive $\zeta_1 = 4$ slots of each frame of T_F=24 slots. Why consecutive will be shortly answered? In this case it gets served immediately in which case the STDMA frame synchronization time would be zero. However, the time slot when this applicable packet jumps to the top of the queue may coincide with the remaining 24-4=20 consecutive slots where it can not be served immediately. In this case the packet will encounter a delay of 1 slot or 2 slots ….. or up to 20 slots to be served. Averaging, the frame synchronization time becomes

$$\frac{(1+2+3+4+5+6......+20)}{20} slots$$

We note that the synchronization time above assumes consecutive assigned service slots to each node. In reality service slots assigned to the nodes are scattered throughout the frame, it is easy to show by taking few examples that the frame synchronization time in the latter case will be actually less than the case where assigned slots are consecutively placed. This means that having a scattered slot service will reduce the waiting time which results in lower delay. Therefore the consecutive service slot assignment shown in the following equation can be considered as on upper bound. To add to this upper bound, we note that not all packets encounter synchronization time, e.g. a packet that jumps to the front of the queue may find a slot ready for service, and in reality the frame synchronization will be less than this upper bound, for link 1 of topology 1 we obtain the adaptive STDMA frame synchronization time expressed in units of slots where each can take one data packet i.e.,

$$F_{d1} = \left\{ \frac{1+2+...(T_F - \lceil T_F(d_1 + d_{18}) \rceil)}{(T_F - \lceil T_F(d_1 + d_{18}) \rceil)} \right\}$$

This reduces to the following

$$F_{d1} = \frac{(T_F - \lceil T_F(d_1 + d_{18}) \rceil) \times \{(T_F - \lceil T_F(d_1 + d_{18}) \rceil) + 1\}}{2(T_F - \lceil T_F(d_1 + d_{18}) \rceil)}$$

$$F_{d1} = \frac{\{(T_F - \lceil T_F(d_1 + d_{18}) \rceil) + 1\}}{2}$$

Now the total number of slots (packets) required for the synchronization and the transmission times of a typical packet on l^{th} link is given by

$$F_{Dl} = F_{dl} + 1 \text{ Packets} \qquad (5.5.1)$$

So the average end to end packet delay for the adaptive STDMA allocation is given by

$$Delay = \frac{sum\ of\ all\ packet\ delays\ of\ all\ files\ of\ all\ calls\ in\ the\ network}{F \times Number\ of\ Calls}$$

$$Delay = \frac{\sum_{c=1}^{CC}\sum_{R=1}^{R_c} X(c,R) \sum_{l \in L.c.R} \left(\frac{\rho_l}{1-\rho_l} \times F_{dl} + \frac{\lambda_l}{\lambda_l + \nabla} \times F \times F_{Dl} \right)}{F \times Number\ of\ Calls} \text{ Packets} \qquad (5.5.2)$$

Where $\rho_l = \frac{\lambda_l}{\mu_l}$, The definition of λ_l is given earlier in 5.4. The service rate of l^{th} links share in the adaptive STDMA allocation (packets/sec) is given by

$$\mu_l = \frac{\zeta_l}{T_F} \times \frac{WW}{N_b}$$

Where WW is the total bandwidth in (100 Mbits/sec) Mbits/sec and N_b is in bits/packet (10000 bits/packet).

Now for the uniform STDMA allocation the service rate of link 1 simply becomes

$$\mu_1 = \frac{(d_1 + d_{18}) \times T_F}{T_F} \times \frac{WW}{N_b}$$

which can be written generally for l^{th} link i.e.,

$$\mu_l = \frac{Number\ of\ Slots\ used\ by\ l^{th}\ link}{Total\ Number\ of\ Slots\ in\ a\ certain\ frame} \times \frac{WW}{N_b}$$

And

$$F_{dl} = \frac{T_F - Number\ of\ slots\ used\ by\ link\ l + 1}{2}$$

The number of slots used by l^{th} link for uniform STMDA allocation can be found from the link usage parameters described earlier in 5.3. The total number of slots in a certain frame is the total number of cliques in a clique cover which is also explained in 5.3.

Back to equation (5.5.2) which applies to both adaptive and uniform STDMA allocation, where $\frac{\rho_l}{1-\rho_l} \times F_{dl}$ represents the queuing delay in units of slots (for simplicity we take M/M/1 queue [5]) and $\frac{\lambda_l}{\lambda_l + \nabla} \times F \times F_{Dl}$ ($\nabla = very\ small\ value, e.g., 0.00001$) represents the synchronization and transmission delay of a typical packet, and we assume all F packets of the file has the same F_{Dl}.

Equation (5.5.2) applies to both adaptive and uniform STDMA allocations with the only differences is that; in the uniform STDMA case all $d_1, d_2, d_3, \ldots d_{24}$ are equal. Also it is necessary to multiply F_{Dl} by the factor $\frac{\lambda_l}{\lambda_l + \nabla}$ in equation (5.5.2) since some links may not carry any traffic as per the optimum routing decision. This factor is approximately equal to 1 which does not change the cost function except for taking care of the above zero traffic in service links. Without such term, some links encounter a packet delay in Equation 5.5.2 even if they have no traffic. F= File size in packets;

$L_{c,R}$ = Set of consecutive links that are used by a certain call to establish a connection according to the route select decision R as decided by the optimum routing policy. In our work we consider the following optimization problem:

We minimize the average end to end packet delay given in equation (5.5.2). Subject to the route constraint

$$\sum_{c=1}^{CC'}\sum_{R=1}^{R_c'} X(c,R) = N \qquad (5.5.3)$$

X is the control variable described earlier in 5.4. We recall the definition of $X(c,R)$, for a given call c, $X(c,R)$ is 1 for only a specific value of R since the call selects only a specific single route out of R_c' . Values of $X(c,R)$ for other R values (Possible routes) are all zero. The control variable X was obtained by running the optimization commercial software package named LINGO. In this program we insert all the possible routes and the constraints for all calls. The program automatically minimizes the cost function by selecting a specific route for each call and hence obtains the best values of $X(c,R)$ for each call. Well known mathematical optimization techniques are utilized as opposed to trial and error or inspection of all possible routes and trying to fit the best route for each call. The LINGO program uses branch and bound technique which is designed to make building and solving linear, integer, nonlinear and global optimization models.

The average end to end packet delay is computed in seconds.

In this thesis, we calculate the delay jitter of l^{th} link from its corresponding load which is $\left(\frac{\lambda_l}{\mu_l}\right)$. For the l^{th} link the delay jitter is given by [59]:

$$\sigma_l = \sqrt{\sum_{i=1}^{K} i^2 P_i - (\bar{d})^2}$$

Where $\bar{d} = \sum_{i=1}^{K} iP_i$, $P_i = \left(1-\frac{\lambda_l}{\mu_l}\right)\left(\frac{\lambda_l}{\mu_l}\right)^i$ and K is the number of packets. For analysis convenience we take M/M/1 queuing [5].

After computing the delay jitter of all links we average over all. So the delay jitter for topology 1 which has 26 links becomes

$$\sigma = \frac{\sigma_1 + \sigma_2 + \ldots\ldots\ldots\sigma_{26}}{26}$$

5.6 Network Layer

5.6.1 Minimum hop algorithm (MHA)

In this section we use the classic minimum cost routing where the cost metric for all links are equal to one. In this case the cost per link is considered as one (means one hop). Here we adopt the following optimization problem

Minimize $\sum_{c=1}^{CC'} \sum_{R=1}^{R_c'} X(c,R) \sum_{l \in Lc} 1$

Subject to the constraint

$$\sum_{c=1}^{CC'} \sum_{R=1}^{R_c'} X(c,R) = N$$

After solving this, and obtaining the X(c,R) values, i.e., finding for each calls which route contains the minimum number of hops. We insert these routes in equation (5.6.1) to find the corresponding performance i.e. end to end average packet delay. Here we use uniform STDMA allocation where $d_1 = d_2 = d_3 = \cdots\cdots d_{24} = \frac{1}{Total\ Number\ of\ Cliques}$.

$$Delay = \frac{\sum_{c=1}^{CC'} \sum_{l \in L'_c} \left(\frac{\rho_l}{1-\rho_l} \times F_{dl} + \frac{\lambda_l}{\lambda_l + \nabla} \times F \times F_{Dl} \right)}{F \times Number\ of\ Calls} \qquad (5.6.1)$$

Where L'_c is the optimum set of links of classic shortest path route of call c as implied by the optimum values of X(c,R).

5.6.2 Random Routing

In the random routing approach we randomly choose the routes i.e. values of X(c,R) (for example for call c=3 the process may yield X(3,1)=0, X(3,2)=0.....X(3,7)=1.....X(3,10)=0; where X(c,R) is one for only one value of R) of all calls and insert those routes in the corresponding performance in equation (5.6.1) and calculate the average end to end packet delay for both networks.

5.7 Results and Discussion

Figure 5.7-1 shows the delay comparisons for the two topologies using two types of allocation (uniform and adaptive STDMA) where the packet success probability (Pc) in the PHY layer is taken as unity. Also we consider uniform call traffic intensity where $\theta_1 = \theta_2 = \cdots\cdots = \theta_8 = 0.7$ and $\theta_1' = \theta_2' = \theta_3' .... = \theta_7' = 0.1$. We compare these results with the results obtained from random routing. In case of random routing, we calculate the delay for all calls from all nodes choosing random routes for each of the calls stated in 5.2. In this thesis we also evaluate the delay using the minimum hop algorithm. From the Figure 5.7-1 it is evident that the delay is more in case of topology 1 where the network is dense. The reason is, in this case the number of cliques is large and in each clique less number of links can operate in parallel using the STDMA allocation. In topology 2, where the network is sparse, the nodes are scattered and large number of links can be accommodated in each clique while using lower number of cliques in a certain frame. In both networks, the delay is less for optimum routing compared to the random routing. We also observe that in topology 2 the variation of the delay between the schemes (optimum, random and MHA) is less than topology 1. This is due to having few routes from each source to destination for topology 2. Therefore, changing the policy schemes (optimum, random and MHA) in topology 2 doesn't lead to large variations in the delay performances as in case of topology 1.

Further, the minimum hop algorithm which is considered as a non- cross layered approach performs better than the random routing in both cases (Topology1 and Topology 2). However it does not perform as well as our cross layered approach (in both Uniform and Adaptive STDMA).

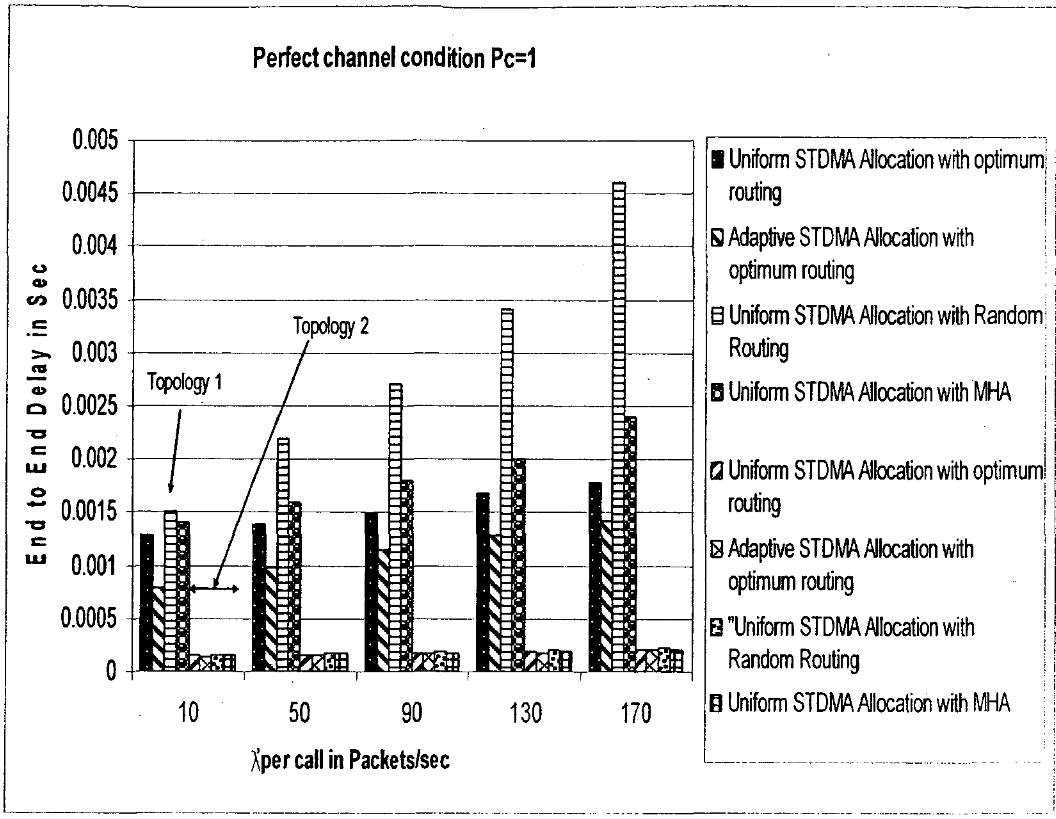


Figure 5.7-1 Delay comparisons of the two topologies

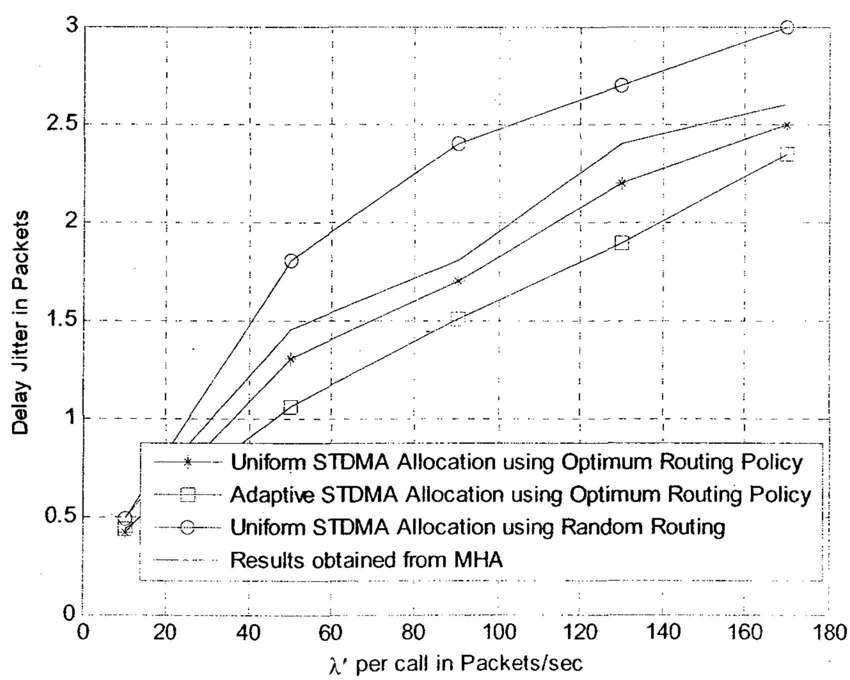


Figure 5.7-2 Delay Jitter Performance of Topology 1(Perfect channel Condition)

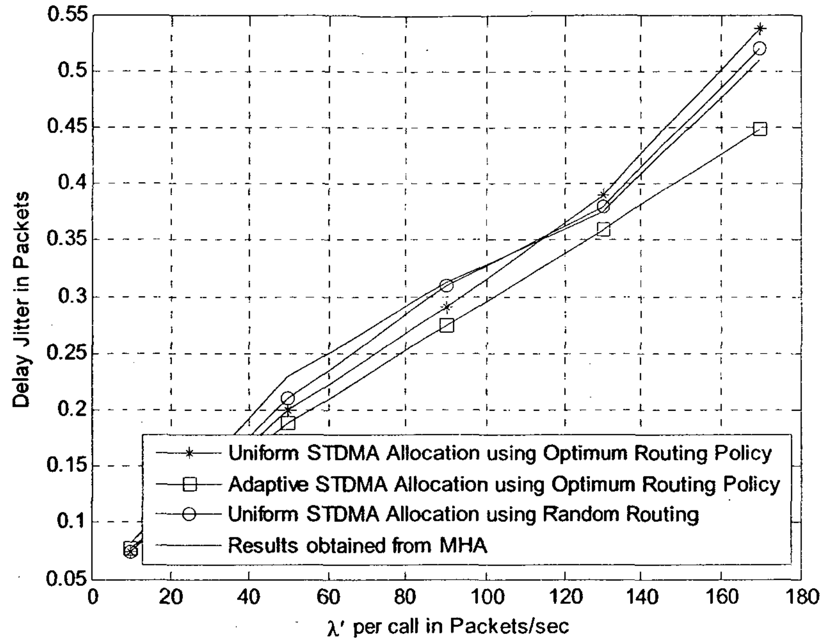


Figure 5.7-3 Delay Jitter Performance of Topology 2 (Perfect Channel Condition)

We plot the delay jitter at different traffic rate for the two topologies which are depicted in Figure 5.7-2 and Figure 5.7-3. From Figure 5.7-2 and Figure 5.7-3 it is evident that the delay jitter is more in case of topology 1. For the same reasons described earlier, i.e. in topology 1 few links can be operated in parallel. Moreover, wider differences are observed between different schemes of Figure 5.7-2 than Figure 5.7-3 in the case of dense networks (Figure 5.7-2). This is due to the fact that our optimum routes select the nodes which are not congested. Accordingly adaptive STDMA gives better performance than the uniform STDMA especially for the dense networks (Topology 1). Further the variation in the jitter performance for topology 2 is less. This is because the number of available routes with different number of links is less in topology 2 compared to topology 1.

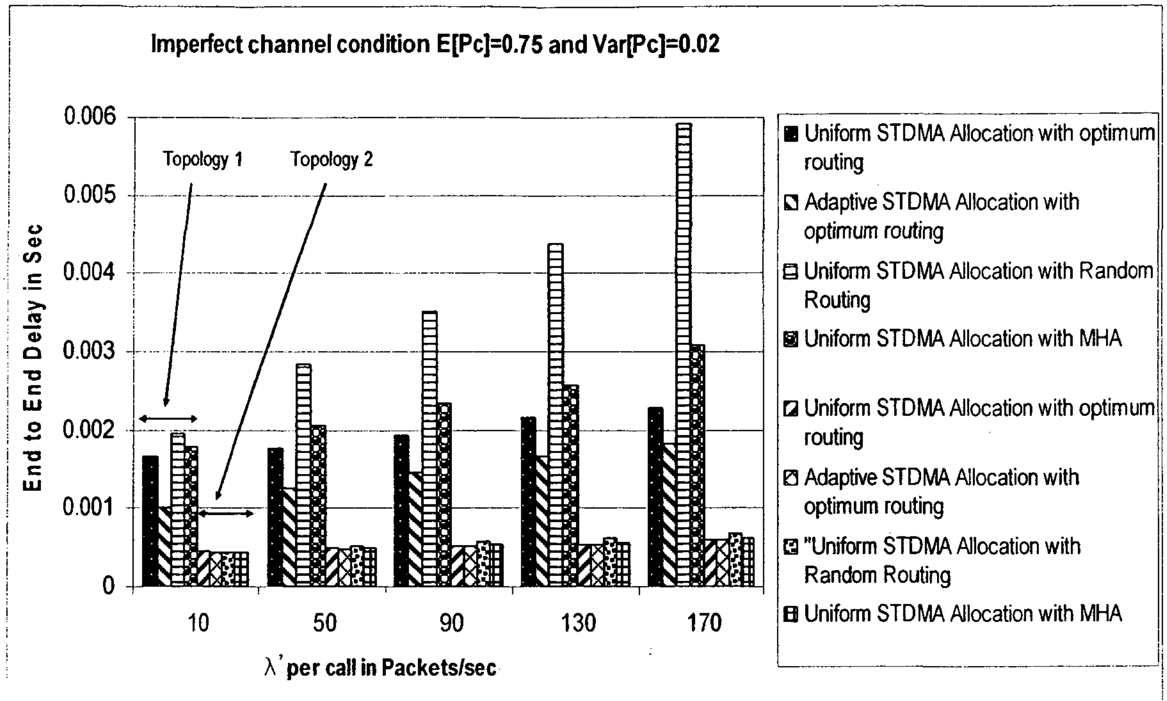


Figure 5.7-4 Delay comparisons for the two topologies

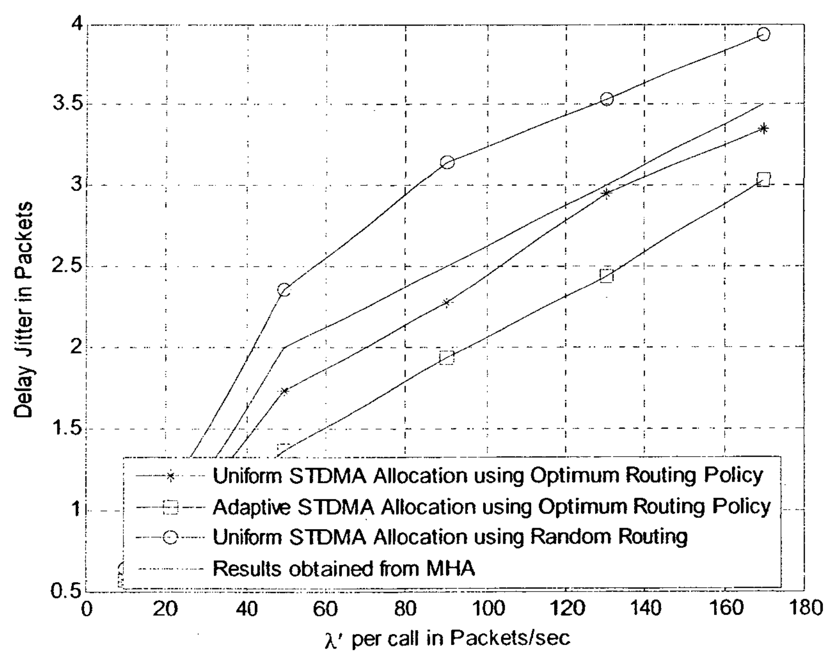


Figure 5.7-5 Delay Jitter Performance of Topology 1 (Imperfect Channel:Condition E[Pc]=0.75 and Var[Pc]=0.02)

Figure 5.7-4 shows the delay for both topologies at a lower packet transmission success probability (P_c). The value of P_c for each link is a uniform random value based on a certain mean and variance. The delay rises for all schemes when the mean value of P_c decreases. The reduction of transmission success probability will increase the traffic in each link thereby increasing the end to end delay performance for all schemes (optimum, random and MHA). However corresponding to Figure 5.7-1 and Figure 5.7-4 reveals that, less tightly coupled topologies (sparse) suffer more in channel with lower Pc values (delays are almost four times their values with higher Pc).

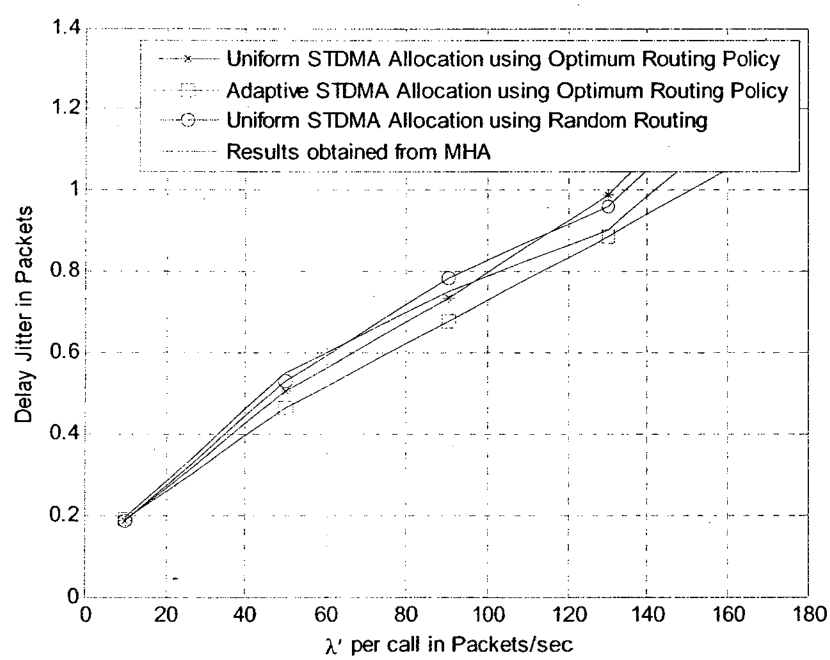


Figure 5.7-6 Delay Jitter Performance of Topology 2 (Imperfect Channel Condition: E[Pc]=0.75 and Var[Pc]=0.02)

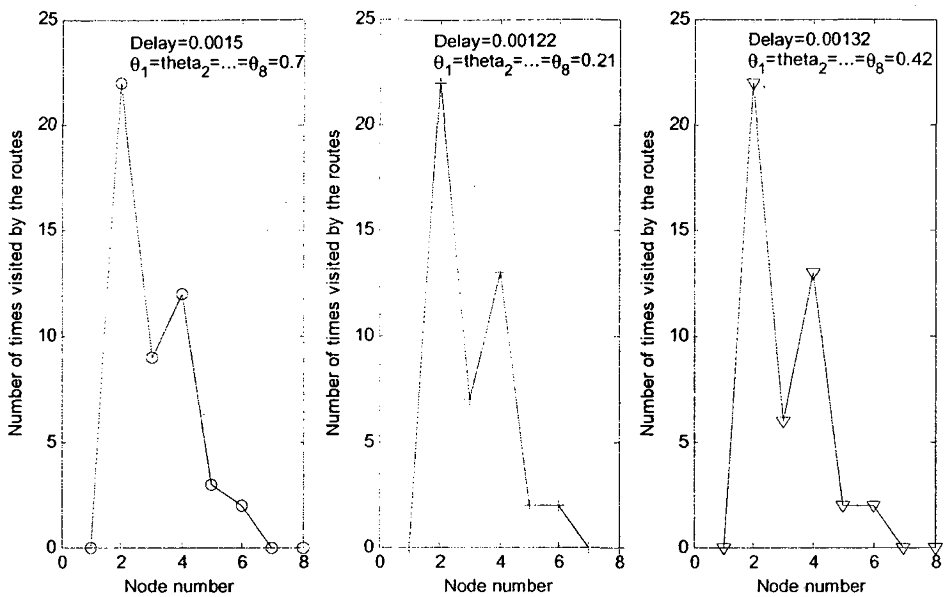


Figure 5.7-7 Cross layered node usage for different values of flow θ (Uniform) for Uniform STDMA Allocation (Topology 1)

Comparisons of Figure 5.7-5 to Figure 5.7-2 and Figure 5.7-6 to Figure 5.7-3 reveal similar conclusions for the delay jitter in topologies 1 and 2. As Pc degrades so will the delay and delay jitter but topology 2 seems to suffer more.

Figure 5.7-7 shows the utilization of different nodes. From the figure of topology 1 and Figure 5.7-7, it is evident that the bottle neck nodes (most highly visited) are 2, 3 and 4. Figure 5.7-7 also reveals that in uniform STDMA allocation, if the call traffic intensity θ_c decreases the end to end average delay decreases.

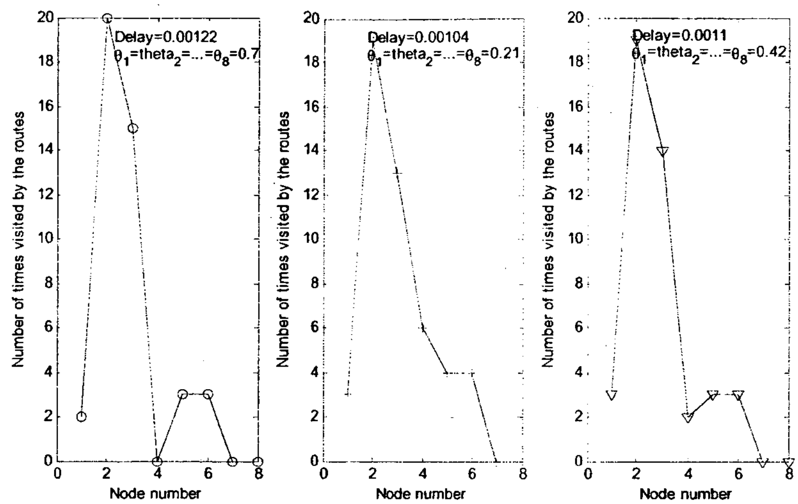


Figure 5.7-8 Cross layered node usage for different values of flow θ (Uniform) for Adaptive STDMA Allocation (Topology 1)

From Figure 5.7-8 using adaptive STDMA allocation the concentration in node 4 is reduced (routing calls through other nodes) which results in lower delay for the same traffic rate (90 packets/sec) in perfect channel conditions. This proves that if the calls can avoid node 4 the total end to end delay can be reduced. In Figure 5.7-9, we assign non-uniform call traffic intensity at different nodes and compute the corresponding end to end average delay for all calls. Here we assign higher call traffic intensity to the subject node implying that the rest will have lower call traffic intensity e.g.,

$\theta_1 = 0.7, \theta_2 = \theta_3 = ...\theta_8 = 0.21$ (Subject node= node 1); such that $\sum_l \theta_l = 2.17$

$\theta_2 = 0.7, \theta_1 = \theta_3 = ...\theta_8 = 0.21$ (Subject node= node 2); and so on for cases of subject nodes 3, 4, 5, 6, 7 and 8.

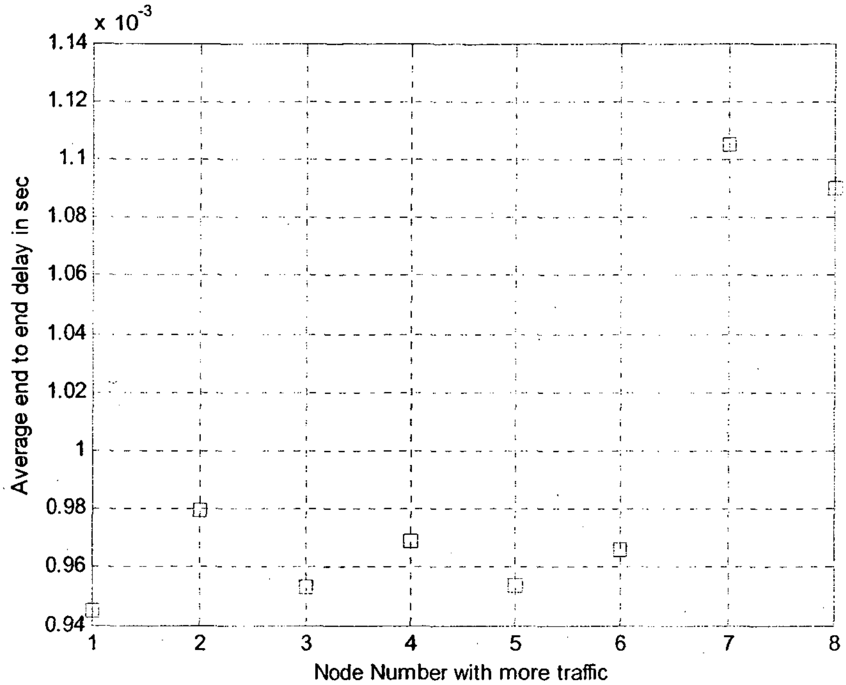


Figure 5.7-9 Average end to end delay performance over all calls using non-uniform call traffic intensity for topology 1 (Adaptive STDMA Allocation (cross layered))

Figure 5.7-9 state that, if the subject node is 7 or 8 (with higher traffic flow i.e. by forcing θ_7 and θ_8 to be high), it will yield a higher end to end average delay for all calls. This means that even with cross layer optimization, certain bottle neck node with intrinsic traffic may still lead to worsened performance. In topology 1, calls can not bypass node 2 which is one of the bottle necks described earlier. However if nodes 1, 2, 3, 4, 5, 6 (i.e. have higher intrinsic traffic values of their own) are selected as subject nodes, the calls can avoid other bottle necks, which considerably reduces the end to end average delay. Similar observations were found for topology 2.

5.8 Cross Layer Design Using CDMA/TDD Approach

Here, we combine CDMA/TDD and optimum routing for cross layer design in wireless mesh networks. An overall optimization routine finds simultaneously the route to various nodes. This optimization routine minimizes the average end to end packet delay over all calls subject to various constraints. We also investigate the effects of level of like user interference on the various performance criteria taking two cases of possible topologies. We compute by analysis the end to end average delay, delay jitter and efficiency over all calls yielded by the cross layered optimization algorithm. Moreover we compare these results with the results obtained from the STDMA allocation techniques.

5.9 System Description

Figure 5.9-1 and Figure 5.9-2 show the two instances of topologies to be investigated. All the cross layered routing and optimization techniques are general in nature and apply as well to other topologies. In both topologies we fix a number of groups depending on the geographic location. Depending on the interference we assign these groups with a number of nodes (SS). For fair comparison we divide the same number of nodes to different groups in the two topologies. According to the concentration of the nodes we consider topology 1 (Figure 5.9-1) as dense and topology 2 (Figure 5.9-2) as sparse.

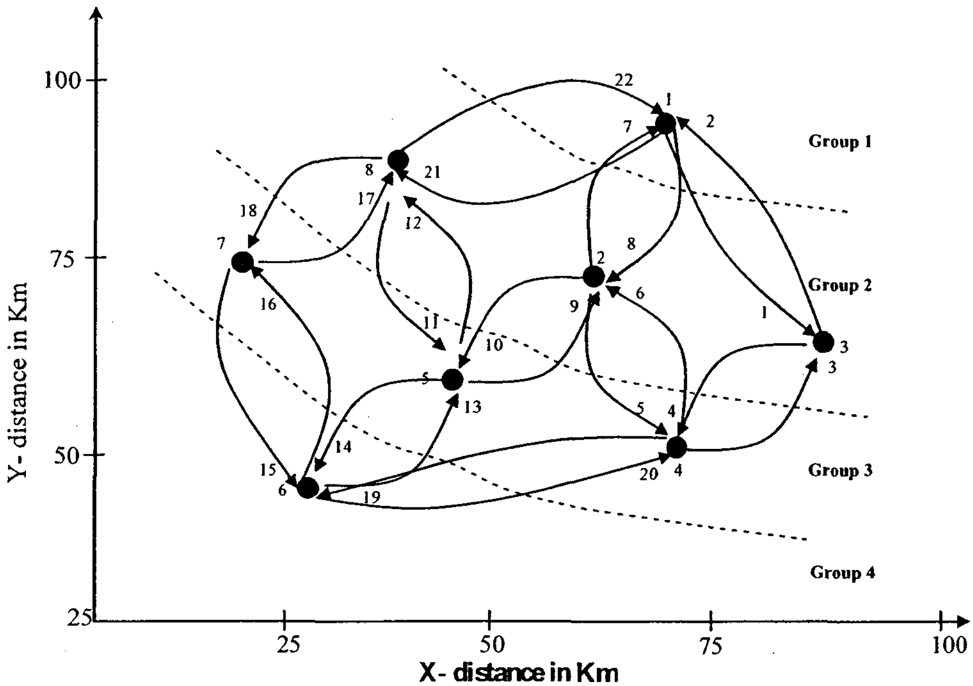


Figure 5.9-1 Dense Topology-High connectivity

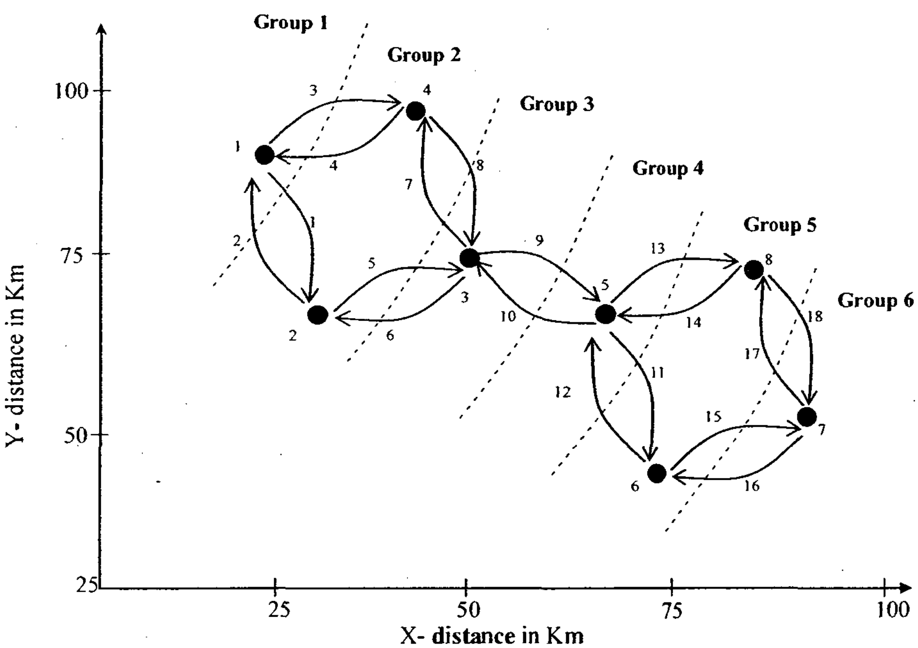


Figure 5.9-2 Sparse Topology-Low connectivity

In this thesis we consider a code division multiple access/Time division duplex technique where the multiple access is provided by a short pilot spreading code. Time is divided into frames where each frame consists of uplink and downlink slots. These slots contain the packets of equal length. These packets are transmitted at a constant bit rate which means that the processing gain remains constant. In this thesis we consider an optimized MAC protocol [49], which is not the subject of this thesis.

The subject topologies are packet switched and not circuit switched. Also they are assumed to be connection oriented in the sense that end to end nodes have to call and establish a virtual connection with their destinations before transmitting data packets. In this regard efficient routing of the data packets is essential. Many routing techniques have been designed for routing packets among peer nodes in ad hoc WLANs [53], [54]. However in most of the cases shortest path is chosen to reduce the complexity. In this thesis each node optimally finds the best routes over the whole network while considering routing decisions by all other nodes.

We assume that all nodes have the full topology information (link and node number) of the mesh network as well as the reference timing starting point for the frames. Such information will be exchanged among the nodes during the frequent handshaking periods between the nodes in a manner analogous to that used in interconnected WLANs [60], [61].While reference timing can be obtained from the base station pilot signals. Such mechanisms are not the subject of this thesis. Further in this thesis we do not explain such Hello, Route request route reply and other handshaking messages [53] incorporated within the clustering and routing techniques for WLANs .

5.10Physical Layer Operation

In cross layer design the PHY layer in each node will estimate and deliver the correct packet success probability P_c value to the network layer. To determine the packet success probability of each link Equation (5.10.1) will be used to find the values of SNR and subsequently the values of Pb from the turbo coding results of [58]. We assume a simple turbo code of rate ½ and 256X256 interleaving [58].

$$(SNR_l)^{-1} = \left[\frac{1}{E_b/N} + \frac{n-1}{PG} \right] \tag{5.10.1}$$

Here *n=number of interfering nodes to the subject link* and E_b/N is considered for the thermal noise which is assumed as *0.001*.

$$PG = \frac{T_b(Bit duration)}{T_c(Chip duration)} = WWT_b$$

The value of P_c for each link can be calculated from equation (5.10.2)

$$P_{c_l} = (1 - P_{b_l})^{N_b} \tag{5.10.2}$$

Where *WW* is the bandwidth assumed 100 Mbits/sec and N_b is 1000 bits/packet.

5.11Traffic

According to section 5.4 the traffic over l^{th} link can be expressed as

$$\Pi_l = \sum_{c=1}^{CC'} \sum_{R=1}^{R_c'} \theta_c{}'(c) X(c, R) \tag{5.11.1}$$

Each call has an initial source and destination which makes a source destination pair. The packets from source to destination nodes may travel through intermediate nodes. Therefore, the traffic load in l^{th} link is given by $\lambda' \times II_l$ (packets/sec). Here we assume that all calls carry a uniform traffic λ' packets/sec.

Now for the CDMA/TDD approach the service rate of l^{th} link simply becomes

$$\mu_l = P_{c_l} \frac{WW}{N_b \times PG}$$

Where P_{c_l} is the probability of correct packets of each link transferred from the Physical layer to the network layer.

The average end to end packet delay of CDMA/TDD system becomes

$$Delay = \frac{\sum_{c=1}^{CC'} \sum_{R=1}^{R_c'} X(c,R) \sum_{l \in L_{c,R}} \left(\frac{1/\mu_l}{\left(1 - \lambda_l/\mu_l\right)} \right)}{P_{c_l} \times Number of\ Calls} \text{ secs} \qquad (5.11.2)$$

Where $\frac{\lambda_l}{\mu_l}$ is the load on l^{th} link. $D_l = \frac{1/\mu_l}{\left(1 - \lambda_l/\mu_l\right)}$ represents the queuing delay of the CDMA/TDD approach (for simplicity we take M/M/1 queue[5]). Equation (5.11.2) is divided by P_{c_l} due to the retransmissions of the data packets e.g. see the ARQ analysis in [5].

$L_{c,R}$ = Set of consecutive links that are used by a certain call to establish a connection according to the route decision decided by the optimum routing policy.

In our work we consider the following optimization problem:

We minimize the average end to end packet delay given in equation (5.11.2). Subject to the route constraint

$$\sum_{c=1}^{CC'}\sum_{R=1}^{R_c'} X(c,R) = N \tag{5.11.3}$$

X is the control variable described earlier in this section. Here the average end to end packet delay is computed in seconds.

In this thesis, we calculate the delay jitter from the following equation

$$\sigma = \frac{\sum_{l=1}^{L}\left(\overline{D} - D_l\right)}{L-1}$$

Where $\overline{D} = \frac{\sum_{l=1}^{L} D_l}{L}$, and L is the number of links in the whole topology. For analysis convenience we take M/M/1 queuing [5].

The efficiency of the CDMA/TDD system is given by

$$\eta = \frac{\left(\sum_{c=1}^{CC'} P_c\right) \times R_b}{2WW} \tag{5.11.4}$$

Where R_b is the bit rate which is equal to $\lambda' \times N_b$. Equation (5.11.4) is multiplied by ½ because of using half rate turbo code.

5.12 Minimum hop algorithm (MHA)

In this section we use the classic minimum cost routing where the cost metric for all links are equal to one. In this case we adopt the following optimization problem

Minimize $\sum_{c=1}^{CC'}\sum_{R=1}^{R_c'} X(c,R)\sum_{l\in Lc} 1$

Subject to the constraint

$$\sum_{c=1}^{CC'}\sum_{R=1}^{R_c'} X(c,R) = N$$

After solving this, and obtaining X(c,R) values, i.e., finding for each calls which route contains the minimum number of hops. We insert these routs in equation (5.11.2) to find the end to end average packet delay.

5.13 Results and Discussion

Figure 5.13-1 shows the delay comparison between the two topologies (both Optimum and Minimum Hop Algorithm) having different processing gains.

From Figure 5.13-1 it is evident that at higher traffic, optimum routing performs better than the Minimum hop routing. This is due to the optimum cross layered algorithm where the routes of each source to destination is achieved by the information received from the physical layer to the network layer. In case of MHA the end to end delay is calculated using the shortest paths among the nodes. The shortest path may not be the optimum path from a certain source to destination due to congestions on each node.

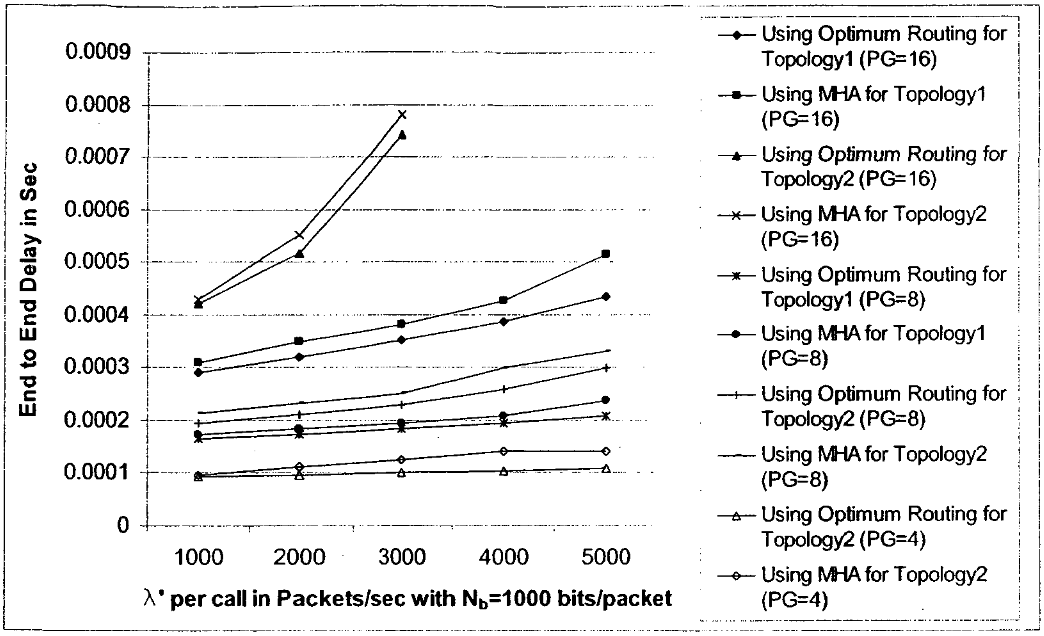


Figure 5.13-1 End to end delay comparison between the optimum routing and the minimum hop routing

Moreover, our cross layered algorithm performs better for the case of topology 1 (dense) than topology 2(sparse) at different processing gains. This due to the fact, in the CDMA/TDD system the routings dominates where topology 1 has more routes from each source to destination than topology 2.

Further, dense topologies are not suitable for low processing gain due to high bit error rates. This also reflects on Figure 5.13-1.

Figure 5.13-2 shows the delay jitter comparison between the two topologies (both Optimum and Minimum Hop Algorithm) having different processing gains. In this case same characteristics trend is noticeable as the previous figure.

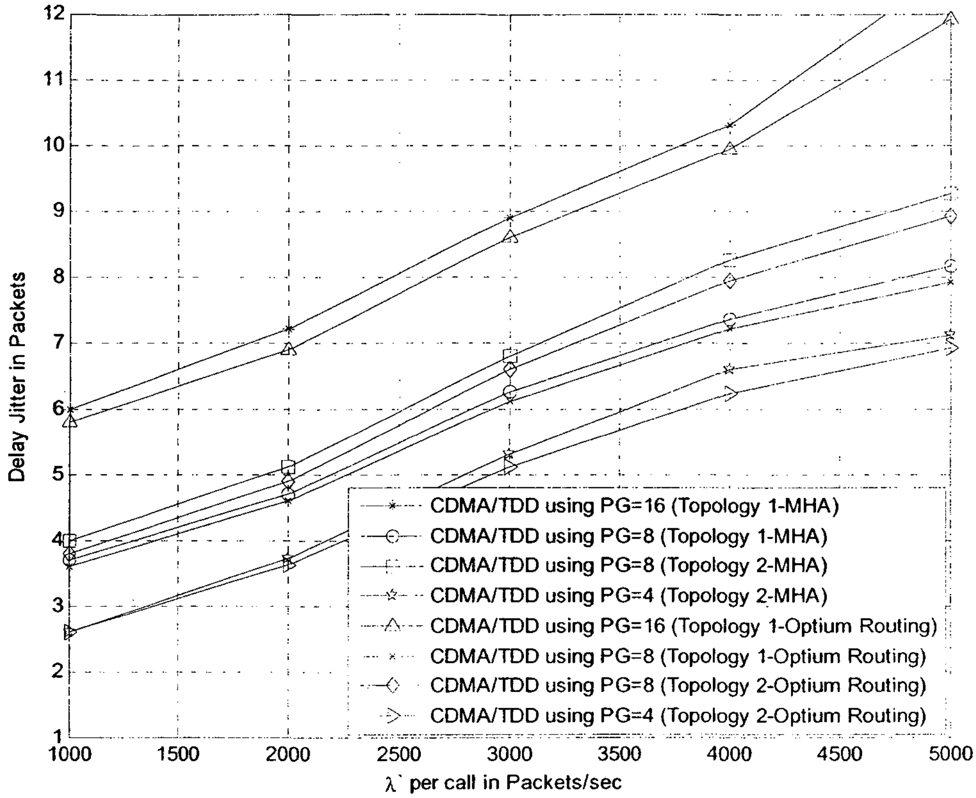


Figure 5.13-2 End to end delay jitter comparison between the optimum routing and the minimum hop routing

Figure 5.13-3 shows the delay comparison of optimum cross layered CDMA and optimum cross layered STDMA approaches. From the figure it is evident that both of the approaches have significant impact on the system topologies. For the CDMA approach our optimum cross layered technique performs better in case of dense topologies. However this is reversed in case of STDMA approach. In STDMA approach the MAC layer in which different links are scheduled in different slots dominate the system performance. However in CDMA case the nodes are divided into groups operating in a Time division duplex manner in the MAC layer has relatively lower domination than the physical layer on the system performance.

Moreover, for both sparse and dense topologies at lower Nb CDMA/TDD approach with a lower processing gain gives better delay performance than the STDMA approach.

For large Nb the delay performance in all cases are better then lower Nb. In this case STDMA performs better than CDMA/TDD for sparse topology (Figure 5.13-4).

Figure 5.13-5 and Figure 5.13-6 shows the delay jitter performances at higher and lower Nb for CDMA and STDMA approaches. At lower Nb for both sparse and dense topologies CDMA approach with a lower processing gain performs better than the STDMA approach. At higher Nb for case of topology 2 CDMA and STDMA performs the same. For case of topology 1 CDMA with a lower processing gain performs better than the STDMA approach.

From the Table 5.13-1 it is evident that for higher Nb and higher PG the efficiency is better. For lower Nb this trend is less noticeable for dense network. For sparse network increasing the PG does not improve efficiency much in two cases of small and large Nb. Using MHA same trend as Optimum routing is observed. However the efficiency is less in case of using MHA. We note that efficiency is affected by mainly two factors, correct packets which favors higher PG values and data rate R which favors lower PG values.

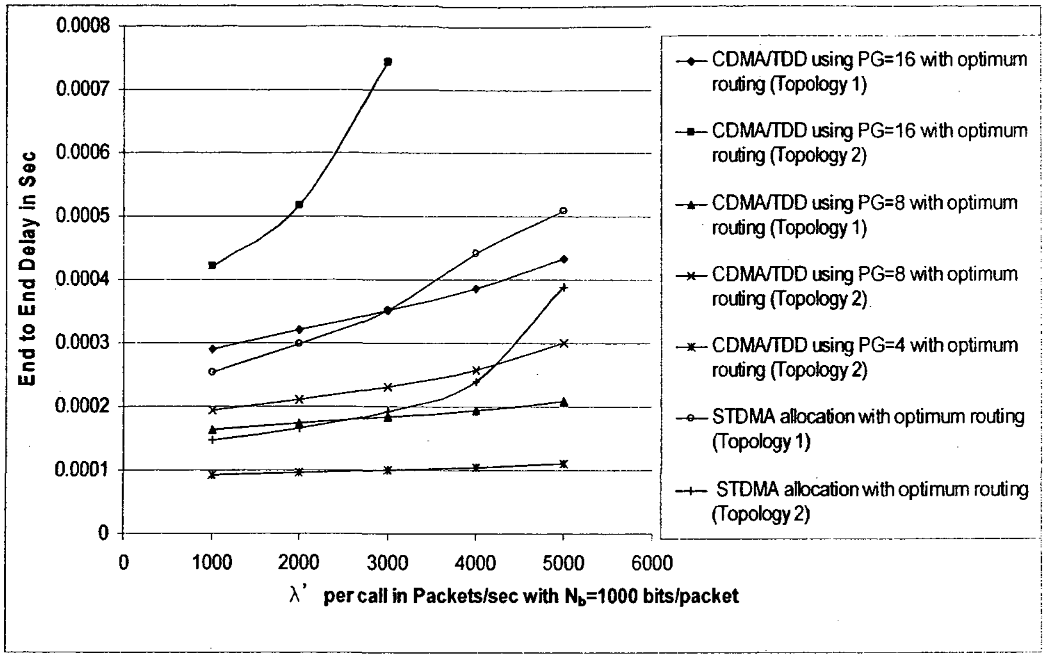


Figure 5.13-3 Delay comparison with CDMA and STDMA approach at lower Nb

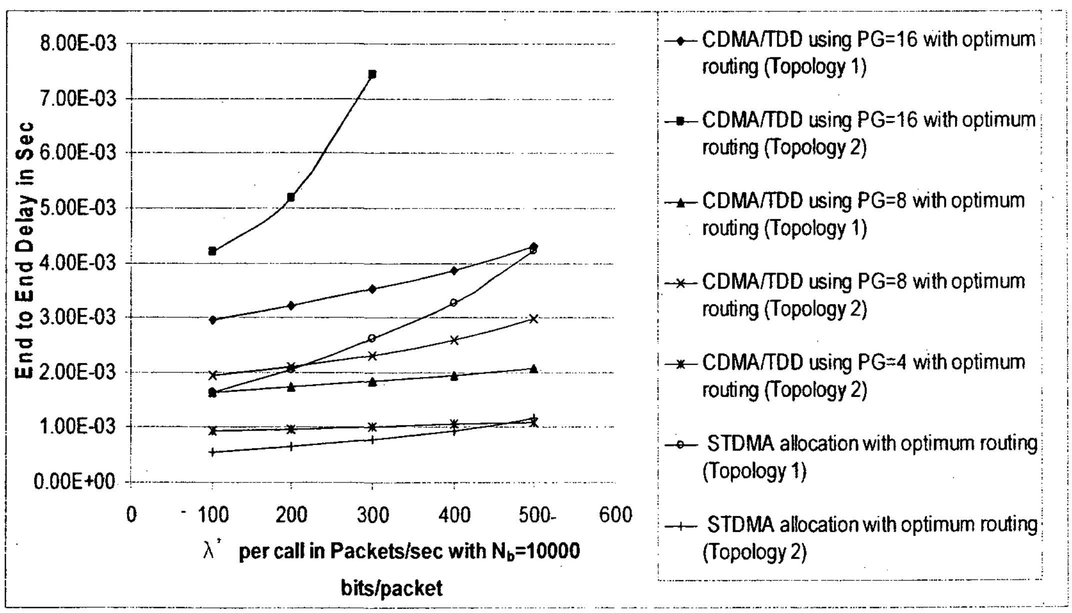


Figure 5.13-4 Delay comparison with CDMA and STDMA approach at higher Nb

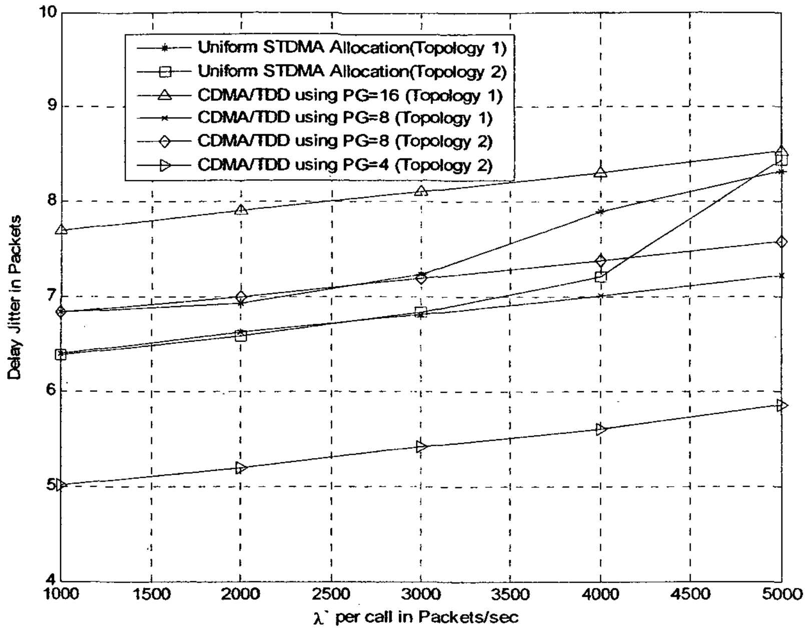


Figure 5.13-5 Delay Jitter comparison with CDMA and STDMA approach at lower Nb

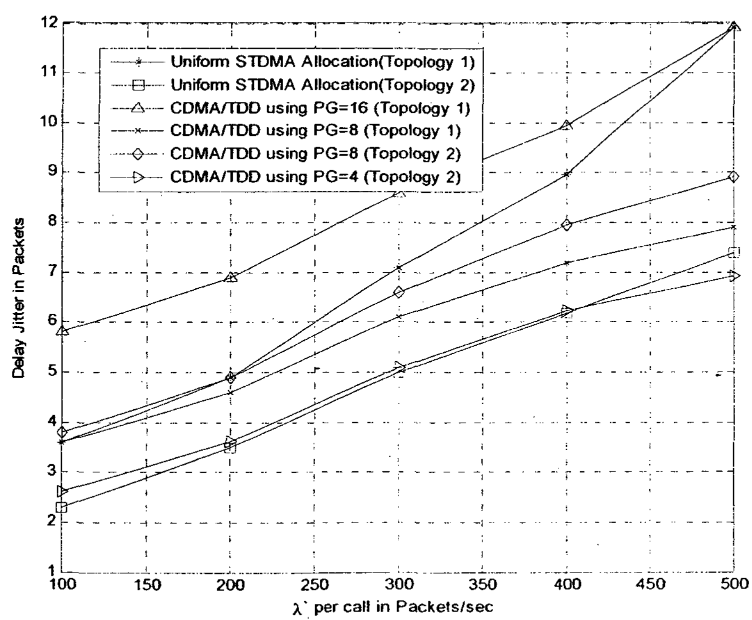


Figure 5.13-6 Delay jitter comparison with CDMA and STDMA approach at higher Nb

Table 5.13-1 Efficiency comparisons for both topologies at lower and higher Nb

λ` per call		Dense Topology				Sparse Topology					
		Optimum PG=16	MHA PG=16	Optimum PG=8	MHA PG=8	Optimum PG=16	MHA PG=16	Optimum PG=8	MHA PG=8	Optimum PG=4	MHA PG=4
Nb =10000	100	0.2699	0.256	0.158	0.149	0.275	0.269	0.26	0.254	0.255	0.25
	200	0.5398	0.5213	0.3475	0.312	0.55	0.498	0.52	0.5013	0.5	0.49
	300	0.8098	0.7907	0.515	0.495	0.825	0.793	0.81	0.7912	0.79	0.78
	400	1.079	0.998	0.6878	0.66			0.99	0.972	1.97	0.963
	500	1.349	1.312	0.8414	0.822			1.25	1.21	1.2	1.19
Nb =1000	100	0.2699	0.256	0.2597	0.232	0.275	0.269	0.265	0.257	0.259	0.258
	200	0.53998	0.5213	0.519	0.489	0.55	0.498	0.54	0.51	0.53	0.52
	300	0.8099	0.7907	0.77	0.74	0.825	0.793	0.815	0.8	0.8	0.78
	400	1.0799	0.998	1.035	0.981			1.0	0.986	0.99	0.97
	500	1.3499	1.312	1.29	1.23			1.3	1.279	1.23	1.21

5.14 Conclusion

In this chapter we introduced cross layer architectures using two different access schemes such as STDMA and CDMA and analyzed their performances. Analysis shows that, CDMA outperforms STDMA in terms of delay. The cross layer architecture is proved to give better performance than the layered architecture in terms of delay and delay jitter. This will eventually increase the QoS in mesh networks. As a result, our CDMA approach with cross layer architecture will help the users to enjoy better wireless internet service through the mesh nodes.

In chapter 4 we proved CDMA's superiority over the classic access technique. These demonstrate the versatility of CDMA system in 4G wireless networks. For this reason, in future we will pursue our research towards the development of CDMA system in future wireless networks.

CHAPTER 6

CONCLUSIONS AND FUTURE WORK

6.1 Conclusions

For overall effectiveness of 4G networks, both local and backbone constituting networks should be optimized. For improving the performance of the local networks i.e. WLAN we have developed table driven techniques for DCF and RTS/CTS mechanism. While maintaining the same delay the table driven DCF outperforms the standard DCF (IEEE 802.11) in terms of throughput. The table driven RTS/CTS also demonstrates that its throughput is more than the standard RTS/CTS mechanism. Moreover the table driven DCF and table driven RTS/CTS gives very good fairness performance. In the table driven technique (for both DCF and the RTS) a simple search mechanism is used to find the values of M and p from $\overline{W}_L$ and $\overline{C}$. However an efficient lookup mechanism is required for this purpose.

In this phase we can conclude that, using table driven technique users will enjoy high speed wireless internet utilizing the channel more efficiently because of high throughput and less delay.

For improving the performance of the backbone networks and in the second phase of this thesis a new CDMA/TDD system has been proposed for WMNs. New CDMA transmitter and receiver structures were introduced. The new CDMA system allowed multi service rate to each node and simultaneous multiple reception as well. This was followed by a comparison between CDMA/TDD and TDMA/TDD based networks. The system building blocks for both control and data planes were introduced. Next QoS criteria were evaluated for both control and data packets. Analysis shows that the versatility of CDMA approach is better suited to larger number of hops (i.e. large number of SS) compared to the TDMA counterpart in terms of network efficiency, delay, and delay jitter. Moreover CDMA is a plug and play i.e. instantly reconfigured operation with low power consumption which is an important asset towards mesh networks.

In this phase we conclude that, with the help of CDMA/TDD approach our new network will improve the QoS which allows the service providers to provide the users with a better wireless internet with increased capacity. The capacity is increased due to the parallel transmission of packets among the mesh nodes.

In the third phase of this thesis we combine a new adaptive Spatial TDMA and optimum routing for cross layer design in wireless mesh networks which takes into account both MAC and physical layer entities. We take two topologies for our main investigation. In the cross layer algorithm the routes from each source to destination is controlled by the information received from the MAC and PHY layers. This architecture outperforms the non cross layered one (MHA) in terms of delay and delay jitter. We apply our uniform and adaptive algorithms on those two topologies and measure different performance metrics such as delay and delay jitter. These two performance matrices are crucial to voice-video transmissions over the wired and wireless links. Our first observation is that, STDMA cannot perform well when the mesh network is dense. However it performs well when the network is sparse. This observation leads to the fact, if the nodes use less power, the connectivity will be less and the mesh will look sparse thereby allowing more links to operate in parallel in each clique which will give a better performance. Further it was found that for sparse mesh networks, the delay and delay jitter performance degrades more than the dense topology case as the channel condition worsens.

Moreover in this phase we developed an optimum cross layered routing design for WMNs using CDMA/TDD approach. The cross layered algorithm performs better for the case of topology 1 (dense) than topology 2(sparse) at different processing gains. However, this is reversed in case of STDMA approach. With a lower processing gain CDMA system can perform better than the STDMA system. In the STDMA approach the MAC layer in which different links are scheduled in different slots dominates the system performance. In CDMA case the nodes are divided into groups operating in a Time division duplex manner in the MAC layer has relatively lower domination than the Physical layer on the system performance.

In this phase, we can conclude that our cross layer architecture using CDMA/TDD has better performance than the layered architecture in terms of delay and delay jitter. This will eventually increase the QoS in 4G wireless mesh networks. Also the capacity will increase due to the parallel transmissions of the mesh nodes. As a result, the service providers can accommodate large number of user.

6.2 Future Work

6.2.1 WLAN Access

We developed the table driven technique for single hop WLANs. A search mechanism was used to find the transmission probability and the number of station from the two static tables. In future an efficient search mechanism can be developed for better performance. With an efficient search mechanism this protocol can be used for the communication among the mesh nodes. The performance of table driven technique in mesh networks will be compared with CDMA and TDMA techniques.

6.2.2 Cross layer design in Wireless Mesh Networks

In future we will compute the network efficiency for the STDMA technique for WMNs and compare the results with the CDMA system and with the mesh network formed by the table driven technique.

In future, we will also compare these cross layer techniques with the adaptive OFDM system for WMNs. Moreover, the effect of coding on cross layer design for mesh networks using both regular FEC and network coding will be investigated.

Finally the effects of the control plan i.e. invoking the HELLO and other messages for clustering and topology formulation are to be investigated for different mesh topologies.

Appendix

Table Driven WLANs

In the Table driven technique, we developed two tables to calculate the number of collisions (A1) and the length of the last idle period (A2) before the successful transmission on the channel. The values are computed for different number of stations and different transmission probability.

A 1 Number of collisions before a successful transmission on the channel

Number of Stations/Transmission Probability	0.01	0.02	0.03	0.04	0.05	0.06	0.07
1	1.105	1.105	1.105	1.105	1.105	1.105	1.105
2	1.111	1.116	1.122	1.128	1.134	1.14	1.147
3	1.116	1.128	1.14	1.152	1.164	1.177	1.19
4	1.122	1.139	1.158	1.176	1.196	1.216	1.236
5	1.128	1.151	1.176	1.201	1.228	1.256	1.285
6	1.133	1.163	1.194	1.227	1.261	1.297	1.335
7	1.139	1.175	1.213	1.253	1.296	1.341	1.388
8	1.145	1.187	1.232	1.28	1.332	1.386	1.445
9	1.151	1.2	1.252	1.308	1.369	1.434	1.503
10	1.157	1.212	1.272	1.337	1.407	1.483	1.565

A 2 Length of the idle period before the successful transmission on the channel

Number of Stations/ Transmission Probability	0.01	0.02	0.03	0.04	0.05	0.06	0.07
1	89.58	44.34	29.26	21.72	17.19	14.18	12.02
2	44.34	21.72	14.19	10.42	8.161	6.656	5.583
3	29.26	14.19	9.17	6.664	5.163	4.165	3.454
4	21.73	10.43	6.668	4.793	3.673	2.929	2.402
5	17.21	8.172	5.17	3.677	2.786	2.197	1.78
6	14.19	6.672	4.176	2.937	2.2	1.715	1.373
7	12.04	5.602	3.469	2.412	1.786	1.376	1.088
8	10.43	4.801	2.94	2.022	1.48	1.126	0.88
9	9.178	4.18	2.532	1.721	1.245	0.936	0.722
10	8.176	3.684	2.207	1.483	1.06	0.787	0.599

Reference:

[1] V. W. S. W. Arslan Munir, "Interworking Architectures for IP Multimedia Subsystems," *Mobile Networks and Applications,* vol. 12, pp. 296-308, 2008.

[2] S. Guojun and S. Shuqun, "Video Streaming Transmission Over Multi-Channel Multi-Path Wireless Mesh Networks," in *Proc. WiCOM '08*, 2008, pp. 1-4.

[3] Y. Andreopoulos, N. Mastronarde, and M. van der Schaar, "Cross-Layer Optimized Video Streaming Over Wireless Multihop Mesh Networks,", *IEEE J. Sele. Areas Commun.,* vol. 24, pp. 2104-2115, 2006.

[4] C. H. Liu, A. Gkelias, and K. K. Leung, "A Cross-Layer Framework of QoS Routing and Distributed Scheduling for Mesh Networks," in *Proc. VTC, 2008*, pp. 2193-2197.

[5] I. W. Alberto Leon-Garcia, *Communication Networks, Fundamental concepts and Key Architectures*: Mcgraw-Hill College, 2000.

[6] K. A. Noordin and G. Markarian, "Cross-Layer Optimization Architecture for WiMAX Systems," in *Proc. PIMRC, 2007,* pp. 1-4.

[7] V. Srivastava and M. Motani, "Cross-layer design: a survey and the road ahead," *IEEE Commun. Mag.,* vol. 43, pp. 112-119, 2005.

[8] J. R. Madsen, D. J. Tebben, A. Dwivedi, P. Harshavardhana, and W. Turner, "Cross layer optimization in assured connectivity tactical mesh networks," in *Proc. MILCOM,* 2008, pp. 1-5.

[9] V. Corvino, V. Tralli, and R. Verdone, "Cross-Layer Radio Resource Allocation for Multicarrier Air Interfaces in Multicell Multiuser Environments," *IEEE Trans. Vehicular Technology,* vol. 58, pp. 1864-1875, 2009.

[10] Z. Peifang and S. Jordan, "Cross layer dynamic resource allocation with targeted throughput for WCDMA data," *IEEE Trans. Wireless Communications,*, vol. 7, pp. 4896-4906, 2008.

[11] G. Kannan, S. N. Merchant, and U. B. Desai, "Cross Layer Routing for Multihop Cellular Networks," in *Proc. AINAW,* 2007, pp. 165-170.

[12] X. C. Xuejun TIAN, Tetsuo IDEGUCHI and Yuguang FANG, "Improving Throughput and Fairness in WLANs through Dynamically Optimizing Backoff," *IEICE Trans. Commun.,* vol. E88-B(11), pp. 4328-4338, 2005.

[13] G. Bianchi, "Performance analysis of the IEEE 802.11 distributed coordination function," *IEEE J. Sele. Areas Commun.,* vol. 18, pp. 535–547, 2000.

[14] L. Zhang, "Performance Improvement for 802.11 Based Wireless local Area networks," *IEICE Trans. Commun.,* vol. E90-B, 2007.

[15] V. Bharghvan, "Performance evaluation of algorithms for wireless medium access," in *Proc. IPDS, 1998* pp. 86–95.

[16] D. Das, "Throughput and Delay Evaluation of a Proposed-DCF MAC Protocol for WLAN," in *Proc. Indicon*, 2004.

[17] M. C. F. Cali and E. Gregori, "Dynamic tuning of the IEEE 802.11 protocol to achieve a theoretical throughput limit," *IEEE/ACM Trans. Netw.,* vol. 8, pp. 785–799, 2000.

[18] Y. F. Y. Kwon and H. Latchman, "A novel MAC protocol with fast collision resolution for wireless LANs," in *Proc. INFOCOM,* 2003.

[19] H. W. J. P. E. J. Weinmiller, and A. Wolisz, "Analyzing and tuning the distributed coordination function in the IEEE 802.11 DFWMAC draft standard," in *Proc. MASCOT,* 1996.

[20] F. Eshghi and A. K. Elhakeem, "Performance analysis of ad hoc wireless LANs for real-time traffic," *IEEE J. Sele. Areas Commun.,* vol. 21, pp. 204-215, 2003.

[21] S. Khurana , A. Kahol, S.K.S. Gupta and P.K. Srimani, , "Performance evaluation of distributed co-ordination function for IEEE 802.11 wireless LAN protocol in presence of mobile and hidden terminals," in *Proc. MASCOT*, 1999, pp. 40-47.

[22] C. L. *Fullmer* and *J. J.* Garcia-Luna-Aceves, "Complete single-channel solutions to hidden terminal problems in wireless LANs," in *Proc. ICC, 1997,* pp. 575 - 579.

[23] J. Yun and M. Kavehrad, "PHY/MAC Cross-layer Issues in Mobile WiMax," *Bechtel Telecomm Technical Journal,* Vol. 4, No. 1,2006.

[24] "IEEE Std 802b - 2004 IEEE Standard for Local and Metropolitan Area Networks: Overview and Architecture," *IEEE Std 802b-2004 (Amendment*

to IEEE Std 802-2001, as amended by IEEE Std 802a-2003), pp. 0_1-4, 2004.

[25] W. Hung-Yu, S. Ganguly, R. Izmailov, and Z. J. Haas, "Interference-aware IEEE 802.16 WiMax mesh networks," in *Proc. VTC*, 2005, pp. 3102-3106.

[26] H. Bo, T. Fung Po, L. Lidong, and J. Weijia, "Performance Evaluation of Scheduling in IEEE 802.16 Based Wireless Mesh Networks," in *Proc. MASS, 2006*, 2006, pp. 789-794.

[27] P. Djukic and S. Valaee, "Distributed Link Scheduling for TDMA Mesh Networks," in *Proc. ICC*, 2007, pp. 3823-3828.

[28] M. G. Nekoui, A. Esfahani and S.N. Soltan, "Iterative Cross Layer Schemes for Throughput Maximization in Multi-Channel Wireless Mesh Networks," in *Proc. ICCCN*, 2007.

[29] V. Loscri, "Evaluating the Impact of Multiple Paths in a Wireless Mesh Network with Distributed Scheduling Schemes," in *Proc. PIMRC, 2007*, pp. 1-5.

[30] P. S. Mogre, N. d'Heureuse, M. Hollick, and R. Steinmetz, "A Case for Joint Routing, Scheduling, and Network Coding in TDMA-based Wireless Mesh Networks: A Cross-layer Approach," in *Proc. MASS*, 2007, pp. 1-3.

[31] Z. Yuanyuan, X. Bo, Z. Ziming, and W. Hao, "Bandwidth Guaranteed Shortest Path Routing in Wireless Mesh Networks," in *Proc. ChinaCom*, 2006, pp. 1-3.

[32] C. Hongju, "Access Scheduling on the Control Channels in TDMA Wireless Mesh Networks," in *Proc. ICISS*, 2008, pp. 3-12.

[33] Z. Rui, B. Walke, and M. Einhaus, "Constructing efficient multi-hop mesh networks," in *Proc. LCN*, 2005, pp. 166-173.

[34] S. M. Abd El-atty, "Efficient Packet Scheduling with Pre-defined QoS using Cross-Layer Technique in Wireless Networks," in *Proc. ISCC*, 2006, pp. 820-826.

[35] Y. Chen , J. Chen , Y. Yang, "Multi-hop Delay Performance in Wireless Mesh Networks," *International journal of Mobile Networks and Applications,* vol. 13, pp. 160-168, 2008.

[36] Y. Wang, Y. Wang, and L. Wang, "A Cross Layer Design of Fair Resource Allocation Algorithm for Downlink OFDM System," in *Proc. WiCom,* 2007, pp. 373-376.

[37] L. K. R. Nelson, "Spatial TDMA: A Collision –free Multihop Channel Access Protocol," *IEEE Trans. Commun.,* vol. COM-33, 1985.

[38] J. O. Somarriba, "Analysis of Capacity for Spatial TDMA in Wireless Ad Hoc Networks with Variable Power and Rate Control," in *Proc. VTC*, 2006, pp. 906-910.

[39] J. O. Somarriba, "Evaluation of heuristic algorithms for scheduling, routing and power allocation in traffic sensitive spatial TDMA Wireless Ad Hoc Networks," in *Proc. WiOPT*, 2008, pp. 462-466.

[40] P. Bjorklund, P. Varbrand, and Y. Di, "Resource optimization of spatial TDMA in ad hoc radio networks: a column generation approach," in *Proc. INFOCOM*, 2003, pp. 818-824.

[41] C. K. Toh, *Ad Hoc Mobile Wireless Networks: Protocols and Systems*: Prentice Hall, 2001.

[42] Y. Kobayashi, K. Mori, and H. Kobayashi, "Radio resource assignment scheme for asymmetric traffic in CDMA/shared-TDD cellular packet communications," in *Proc. VTC*, 2004, pp. 939-943 Vol. 2.

[43] C. Kun-nyeong, L. Ki-dong, and V. C. M. Leung, "An efficient local predictive method for distributed timeslot allocation in CDMA/TDD," *IEEE Trans. Wireless Commun.*, vol. 8, pp. 2341-2351, 2009.

[44] K. Zhen, W. Jiangzhou, and K. Yu-Kwong, "A New Cross Layer Approach to QoS-Aware Proportional Fairness Packet Scheduling in the Downlink of OFDM Wireless Systems," in *Proc. ICC*, 2007, pp. 5695-5700.

[45] H. T. Cheng, H. Jiang and W. Zhuang "Distributed medium access control for wireless mesh networks," *Wireless Communications & Mobile Computing*, vol. 6, pp. 845 - 864 2006.

[46] K. T. Phan, J. Hai, C. Tellambura, S. A. Vorobyov, and R. Fan, "Joint medium access control, routing and energy distribution in multi-hop wireless networks," *IEEE Trans. Wireless Commun.*, vol. 7, pp. 5244-5249, 2008.

[47] A. Abdrabou and Z. Weihua, "A Position-Based QoS Routing Scheme for UWB Ad-Hoc Networks," in *Proc. ICC*, 2006, pp. 3578-3584.

[48] I. F. Akyildiz and W. Xudong, "A survey on wireless mesh networks," *IEEE Commun. Mag.*, vol. 43, pp. S23-S30, 2005.

[49] W. Ping and Z. Weihua, "A collision-free MAC scheme for multimedia wireless mesh backbone," *IEEE Trans. Wireless Commun.*, vol. 8, pp. 3577-3589, 2009.

[50] H. J. a. W. Z. Ho Ting Cheng, "Distributed medium access control for wireless mesh networks," *Wireless Communications & Mobile Computing,* vol. 6, pp. 845–864, 2006.

[51] D. C. a. W. H. R. Jain, "A quantitative measure of fairness and discrimination for resource allocation in shared computer systems," 1984.

[52] H. H. a. A. Toshkala, *WCDMA for UMTS, Radio access for third generation mobile communications*: John Wiley & Sons, 2004.

[53] C. Perkins, "Ad-hoc on-demand distance vector routing," in *Proc. MILCOM,* 1997.

[54] C. E. P. a. E. M. Royer, "Ad-hoc on-demand distance vector routing," in *IEEE Workshop on Mobile Comp. Sys. and Apps*, 1999.

[55] J. Sangsu, M. Kserawi, L. Dujeong, and J. K. K. Rhee, "Distributed potential field based routing and autonomous load balancing for wireless mesh networks," *IEEE Communications Letters,* vol. 13, pp. 429-431, 2009.

[56] R. Bo, Q. Yi, L. Kejie, and R. Q. Hu, "Enhanced QoS Multicast Routing in Wireless Mesh Networks," *IEEE Wireless Communications,* vol. 7, pp. 2119-2130, 2008.

[57] L. Yajun, Y. Yuhang, and C. Chengyu, "A novel routing algorithm in distributed IEEE 802.16 mesh networks," *IEEE Communications Letters,* vol. 13, pp. 761-763, 2009.

[58] C. Berrou, A. Glavieux, and P. Thitimajshima, "Near Shannon limit error-correcting coding and decoding: Turbo-codes. 1," in *Proc. ICC*, 1993, pp. 1064-1070 vol.2.

[59] I. Al-wazedi and A. K. Elhakeem, "CDMA/MPLS platforms for WIMAX mesh networks," in *Proc. LCN*, 2008, pp. 851-858.

[60] M. Lott, A. Sitalov, E. Linsky, and L. Hui, "Performance analysis of multicast transmission in WLAN," in *Proc. VTC*, 2003, pp. 1223-1227 vol.2.

[61] K. A. Mohamed and L. Pap, "Inter-cell interference in spread-spectrum wireless LANs employing handshake protocols," in *Proc. ICC*, 1998, pp. 151-155 vol.1.