Link Layer Priority Management Techniques for Supporting Real Time Traffic in CDMA Based Wireless Mesh Networks

by

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I hereby declare that I am the sole author of this thesis. This is a true copy of the thesis, including any required final revisions, as accepted by my examiners.

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Abstract

With the recent advances in the development of wireless communication networks, Wireless Mesh Networks (WMNs) have been receiving considerable research interests in recent years. Many challenges need to be addressed for successful WMN deployment. One of the fundamental challenges is the need to support integrated services and provision different Quality of Service (QoS) for various applications. In order to allow differentiated services, Medium Access Control (MAC) has to provide priority management techniques at the link layer. In Code Division Multiple Access (CDMA) based WMNs, the interference phenomenon and the simultaneous transmissions must be considered. We propose two priority schemes for a distributed CDMA-based MAC WMNs. We take into account interference, multiple services, QoS requirements for each type of traffic, and the simultaneous transmission in CDMA. The first priority scheme is within a node. Each node has independent queues for each traffic class. According to QoS requirements, the queue that should be served first is determined. The second priority scheme is among neighbour nodes. It is proposed for possible multiple simultaneous transmissions with CDMA. This scheme gives a higher chance of correct transmission to high priority traffic than low priority traffic. In addition, we propose to use an adaptive spreading gain and a frame structure to achieve high resource utilization. Simulation results demonstrate that the proposed schemes can achieve effective QoS guarantee.

Acknowledgements

To my parents...

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WLAN	Wireless Local Area Network
MS	Mobile Station
BS	Base Station
MSC	Mobile Switching Center
WMN	Wireless Mesh Network
WMR	Wireless Mesh Router
WMC	Wireless Mesh Client
CDMA	Code Division Multiple Access
DCF	Distributed Coordination Function
TDMA	Time Division Multiple Access
FDMA	Frequency Division Multiple Access
QoS	Quality of Service
E_b/N_o	energy to interference plus noise density ratio
MAC	Medium Access Control
CSMA	Carrier Sense Multiple Access
RTS	Request To Send

CTS	Clear To Send
MACAW	Multiple Accesses with Collision Avoidance Wireless
ACK	Acknowledgement
MARCH	Multiple Access with Reduced Handshake
DCA	Dynamic Channel Assignment
MMAC	Multi-channel MAC
ATIM	Ad hoc Traffic Indication Message
PCL	Preferred Channel List
ATIM-RES	ATIM-Reservation
FPRP	Five Phase Reservation Protocol
RR	Reservation Request
CR	Collision Report
RC	Reservation Confirmation
RA	Reservation Acknowledgment
P/E	Packing and Elimination
DTDMA	Distributed Spatial TDMA
MCSMA	Multi-channel CSMA
MSI	Maximum Sustainable Interference
HIPERLAN	High Performance Radio Local Area Network
UMTS	Universal Mobile Telecommunication System
FIFO	First Input First Output
WMB	Wireless Mesh Backbone
CBR	Constant bit rate

IP	Internet Protocol
UDP	User Datagram Protocol
RTP	Real-time Transport Protocol
ITU	International Telecommunication Union
VBR	Variable Bit Rate
BER	Bit Error Rate

Chapter 1

Introduction

1.1 Wireless Networks

Wireless networks are a collection of nodes that connect with each other without using wires. These networks are having significant impacts on everyday life and they have many applications in military and civilian environments. Wireless networks can be classified into two classes: infrastructure-based networks and non-infrastructure-based networks. Each classe can be divided into two subclasses: fixed nodes and mobile nodes.

Wireless infrastructure-based networks have fixed control centers such as base-stations in cellular networks and access points in Wireless Local Area Networks (WLANs). Cellular networks consist of two kinds of links: wireless and wired links; moreover, two kinds of stations: mobile and fixed stations. As in Figure 1.1, cellular networks consist of three parts: Mobile Stations (MSs), Base Stations (BSs), and Mobile Switching Centers (MSCs). The communications between mobile stations and base stations are the only wireless communi-

cation links. In addition, the only moving stations are the mobile stations. The basic idea of cellular networks is that each base station has its own coverage area (cell). Therefore, when a mobile station initiates a phone call, it sends a phone call request to the base station that covers its area. Another infrastructure-based wireless network is WLANs. They consist of fixed wireless access points and wireless clients. As in Figure 1.2, access points work as base stations that transmit and receive the information to and from clients, which might be mobile devices such as laptops or fixed devices such as desktop computers.

Non-infrastructure-based wireless networks do not have control centers. Each node connects with its neighbors directly. For example, ad hoc networks are collections of mobile nodes connected together without access points. These kinds of networks are called peer-to-peer or distributed networks. Another example of non-infrastructure-based wireless networks is Wireless Mesh Networks (WMNs), which have fixed and/or mobile nodes connected together using multi-hop wireless links without control centers. The next section has more details and explanations about WMNs.
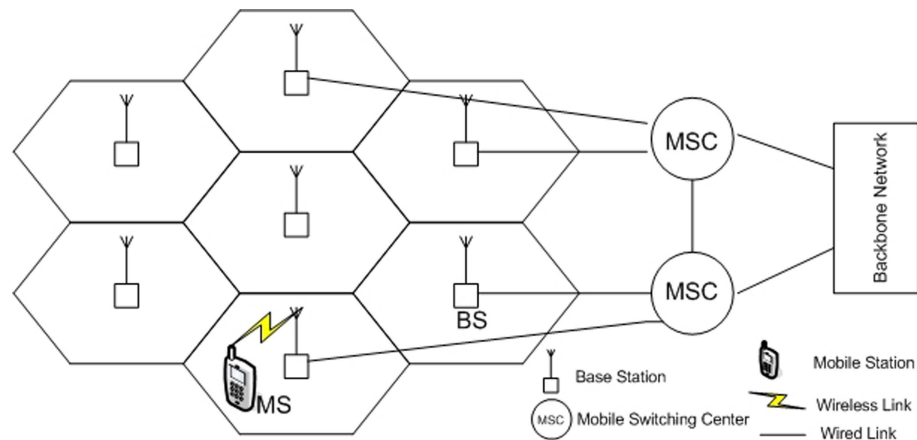
Figure 1.1: Cellular network architecture

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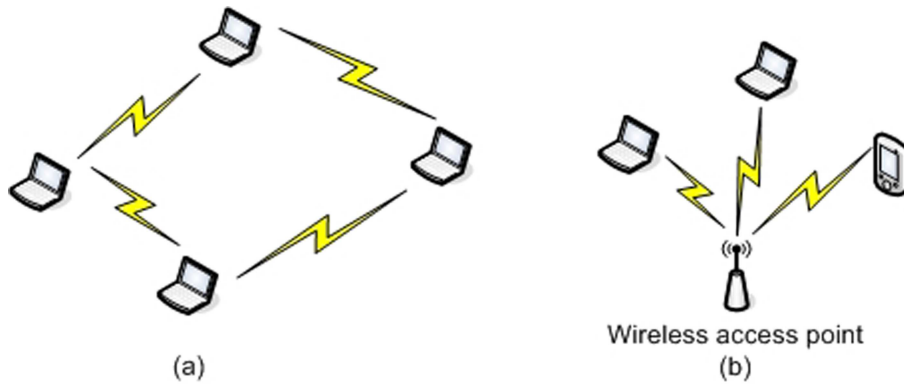
Figure 1.2: (a) A distributed network and (b) A centralized network

1.2 Wireless Mesh Networks

WMNs are comprised of a wireless mesh backbone and wireless mesh clients. A wireless mesh backbone is a collection of fixed nodes that work as routers, called Wireless Mesh Routers (WMRs), connected together using multi-hop wireless links without control centers. In contrast, Wireless Mesh Clients (WMCs) mostly are mobile nodes that work as routers as well, but their hardware platform and software are simpler than those for WMRs. Consequently, there are two types of nodes in WMNs: wireless mesh routers and wireless mesh clients. Table 1.1 compares those types of nodes.

Table 1.1: Characteristics of wireless mesh routers and clients

Wireless Mesh Routers	Wireless Mesh Clients
Form the wireless mesh backbone	Form peer-to-peer network
Have gateway/bridge functions in order to support the integration of WMNs with other networks	Do not have gateway/bridge functions due to the simplicity of their hardware and software platform
Fixed nodes	Mobile nodes

The architecture of WMNs is shown in Figure 1.3 that combines a wireless mesh backbone and wireless mesh client architectures. WMRs form a wireless mesh backbone. They have no mobility, so their power constraints are reduced and their locations are known. Some WMRs have gateway or bridge functions in order to authorize the integration of WMNs with various other networks and conventional clients. In comparison, WMCs can be mobile nodes, so the power consumption and location information are very important issues. The lack of a hardware platform and software in these kinds of nodes does not allow them to have gateway or bridge functions.

The significant characteristics of WMNs, such as low cost, ease of maintenance, self organization, large coverage, ease of expansion, and robustness, help to support several applications in the public and private sectors. Irrespective of military applications, many promising civilian applications have been presented [1, 2]; for example, community networks, metropolitan area networks, broadband home networks, enterprize networks, and transportation systems. Several applications are already in place [2]. In the San Francisco Bay area, the San Matteo Police Department uses mesh networking technology in a public safety application by outfitting its vehicles with laptops and PDAs, applying the IEEE 802.11b/g standard. Another example of commercial applications using wireless mesh networking technology is the metro-scale broadband city network that provides public internet access in the city of Cerritos, California.

Because of the significant characteristics of WMNs, many challenging issues need to be addressed, from the physical layer to the application layer. For instance, the physical layer has the challenge of achieving high transmission rates with affordable software and hardware radio techniques. Scalability issues in the MAC and network layers must be

studied. The transport layer must deal with different transport control protocols due to the integration between WMNs and other wireless networks. A significant algorithm or software must be developed in the application layer, taking into account the integration of WMNs with various wireless networks, the multi-hop transmission, and the distributed control characteristics. There are also cross-layer design, security, capacity, and network management issues.
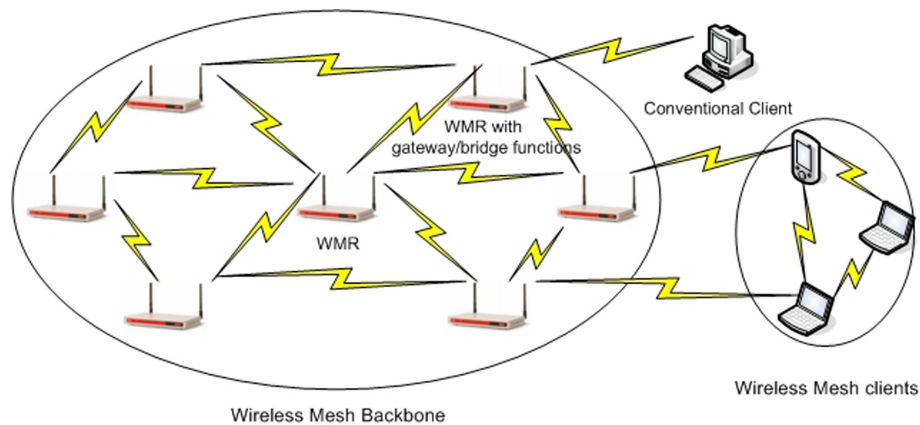
Figure 1.3: Wireless mesh network architecture

1.3   Motivation

Recently WMNs have been a subject of extensive research. The significant characteristics of WMNs have attracted attention from the academic and industrial sectors. Currently, many efforts are underway to standardize protocols for the operation and management of WMNs.

Code Division Multiple Access (CDMA) is a spread spectrum technique, in which each user can use the whole bandwidth at all times, and each user has unique codes for receiving

and transmitting. CDMA-based networks have many advantages, for example, multiple access can be applied, the administration of time and frequency domains is simplified, and the security is increased [3]. In particular, CDMA-based networks have been shown to achieve a significant increase in network throughput and capacity, compared to the Distributed Coordination Function (DCF) mode of IEEE 802.11 standard [4] as in [5, 6], and compared to the networks based on the Time Division Multiple Access (TDMA) and Frequency Division Multiple Access (FDMA) techniques as in [7].

One of the fundamental challenges in WMNs research is how to support real time traffic with quality of service (QoS) provisioning. The priority techniques are essential to manage different services with different QoS requirements. Because all users transmit on the same bandwidth, serious interference can be generated. Therefore, maintaining the required signal bit energy to interference plus noise density ratio (E_b/N_o) is very important for transmission quality in terms of transmission accuracy. For real time traffic, delay is a very important QoS parameter. As a result, supporting real time traffic in the CDMA-based link layer for WMNs requires priority management techniques that take into account the effect of the interference, QoS requirements for each class, and the simultaneous transmissions. This research is to investigate and develop such priority schemes.

1.4 Thesis Organization

The remainder of this thesis is organized as follows. Chapter 2 reviews the background and provides literature survey of MAC protocols for WMN link layer. Chapter 3 defines our system model, including the network structure, CDMA system, MAC protocol, traffic

model, and QoS requirements. Chapter 3 also proposes two priority schemes for supporting real time traffic in the CDMA based WMN. Simulation results are presented in Chapter 4 to evaluate the proposed priority schemes. In Chapter 5, conclusions and further research work are discussed.

Chapter 2

Background and Literature Survey

2.1 MAC Protocols for Wireless Mesh Networks

Medium Access Control (MAC) protocols are very important to the performance of a network. MAC protocols represent how each node can share efficiently the limited wireless bandwidth medium. For example, many nodes may send packets simultaneously over the same medium. In this situation, we need a MAC protocol to solve this contention problem.

MAC protocols are divided into two classes: centralized and distributed MAC protocols. The centralized MAC protocols require a control center for the protocols; for instance, an access point in Wireless Local Area Networks (WLANs) and a base station in cellular networks. The distributed MAC protocols do not require any control center such as ad hoc networks. Our focus in this chapter is on the distributed MAC protocols.

Medium access control for Wireless Mesh Networks (WMNs) should incorporate the network's characteristics, which require some changes to the classical MAC such as:

- MAC protocols for WMNs have more than one hop communication while classical MAC protocols are limited to one hop communication.

- MAC protocols for WMNs do not have a centralized controller. Consequently, multi point-to-multi point communications should be established among nodes.

- In WMC networks (ad hoc networks), nodes have the ability to move, which affects the performance of MAC protocols.

- Network self-organization is needed to achieve better cooperation between neighboring nodes and nodes across multi-hop distances.

2.2   Performance Metrics

The following widely used metrics should be considered to compare and evaluate the MAC protocols:

- Throughput – defined as the percentage of the channel capacity used for data transmission. Our goal is to increase the throughput.

- Delay – the average time spent by the packet in a network. Minimizing the delay is one of the MAC protocols's objectives.

- Fairness – measuring how fair the channel allocation is among the different nodes.

- Power consumption – Since most wireless devices have limited battery power, providing some power saving features for MAC protocols is very important.

- Multimedia support – the ability of MAC protocols to accommodate multimedia traffic such as voice, video, and data. For a brief survey on the capability to support multimedia transport for multi-hop networks over IEEE 802.11 MAC protocol, see [8].

2.3 WMN MAC issues

The most popular issues in wireless mesh network design that should be considered are the hidden and exposed node problems in the carrier sense multiple access (CSMA) technique, the capture problem, radio link vulnerability, and self-contention.

2.3.1 Hidden and Exposed node Problems

In Figure 2.1, if nodes a and c want to communicate with node b at the same time, the steps according to CSMA (to be discussed in section 2.4.1) are as follows: nodes a and c sense the medium as idle and initiate a transmission to node b. A collision occurs at node b, but both a and c are unaware of the collision since they are out of each other's range. In this case, we can say that node a is hidden from node c with reference to a transmission to node b. As a result, hidden nodes reduce the capacity and the performance of the network by causing collisions at receivers without the transmitter knowing about these collisions [9, 10].

In the CSMA technique, suppose node b is transmitting to node a as shown in Figure 2.2, and node c has a packet to be transmitted to node d. Node c senses the medium and finds it busy because of node b's transmission. Therefore, node c refrains from transmitting

to node d although this transmission would not cause a collision at node a. In this case, we can state that node c is exposed to a transmission from node b. As a consequence, exposed nodes are more conservative in their transmission attempts, thus losing throughput as discussed in [11, 12].
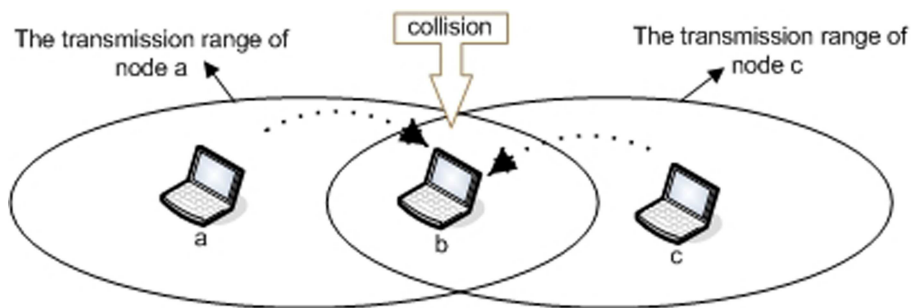
Figure 2.1: The hidden node problem
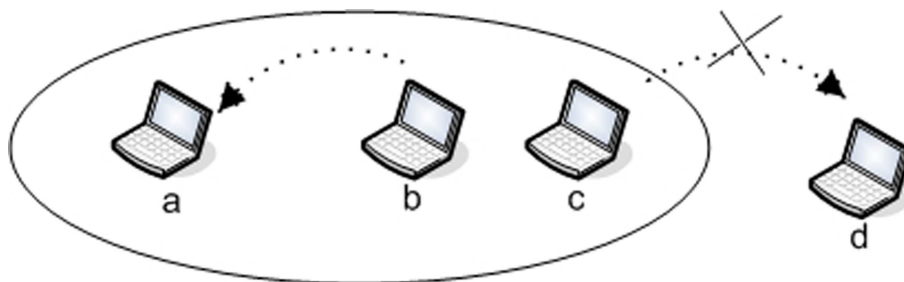
Figure 2.2: The exposed node problem

2.3.2 Capture problem

The capture problem is shown in Figure 2.3. Suppose node a and node b transmit simultaneously to node c. All of the nodes a, b, and c are within the same transmission range. The signal strength received from node b is much higher than that from node a because node

b is closer to node c than node a if we assume all nodes have the same power. Thus, node b's transmission can be decoded without errors while node a transmits. The advantage of the capture is that it improves the utilization of the channel and, therefore, the protocol performance. However, it causes unfairness among nodes [13].
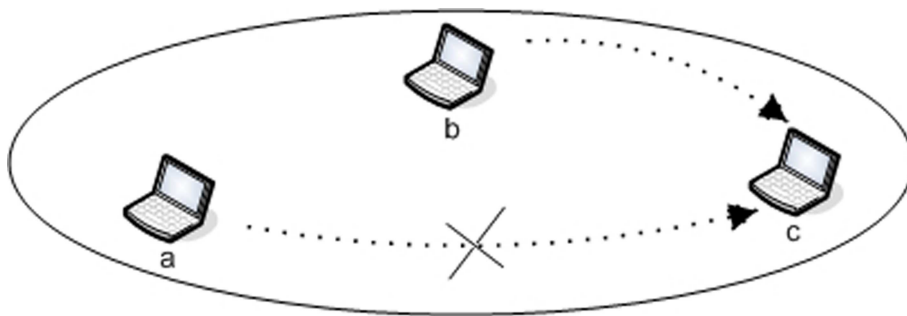
Figure 2.3: Capture problem

2.3.3 Radio link vulnerability

The effects of noise, interference, shadowing, fading, and other effects over wireless channels cause high bit-error-rate, which limits the channel capacity. In fact, the radio link vulnerability affects the utilization of the channel and the fairness among nodes as discussed in [14].

2.3.4 Self-contention

If a MAC protocol is unaware of the transport layer connection that a packet belongs to, packets belonging to the same connection contend for local spectra during transmission at neighboring nodes [15].

2.4 Approaches for Designing Wireless Mesh Networks MAC Protocols

2.4.1 Modifying CSMA MAC protocols

CSMA is a very popular MAC mechanism used to reduce the number of collisions. The mechanism is as follows when node a wants to send a packet to node b: First, node a listens to the channel to ensure that no other node is transmitting. If the channel is clear, node a transmits the packet; otherwise, node a chooses a random "back off value" that determines the amount of time the node must wait until it is allowed to transmit its packet. When the back off value reaches zero, the node retries to transmit the packet. Therefore, when the probability is small that two nodes choose the same back off factor, the probability of packet collisions is low. However, the wireless medium characteristics generate complex phenomena such as hidden and exposed node problems.

The Request To Send and Clear To Send (RTS/CTS) handshaking mechanism has been proposed to reduce the hidden node problem. Figure 2.4 shows the RTS/CTS mechanism. Node a requests the access of channel through the RTS frame. Node b replies with a CTS frame, indicating that it is ready to receive node a transmission. Node c receives a CTS frame from node b and thus refrains from transmitting for the duration indicated in the CTS frame. Although node a and node c are hidden from each other, the RTS/CTS mechanism ensures that a collision at node b does not occur. To reduce the exposed node problem, the transmitter's neighbors will listen to the RTS frame. As in Figure 2.4, node e will hear the RTS frame from node a. This protocol states that any node receiving the

RTS frame, but not the CTS frame, is permitted to transmit to other neighboring nodes. However, node c can not transmit to node d because it is exposed to a transmission from node b.

Multiple Accesses with Collision Avoidance Wireless (MACAW) [15] was proposed to offer a delivery guarantee by adding an acknowledgement message (ACK). Many schemes [16, 17] have been proposed to enhance the MACAW scheme.

For multi-hop ad hoc networks, using handshaking scheme with much control signalling as in Figure 2.5 will reduce the performance of the network, especially when the number of hops increases. Some schemes have been proposed to gain better performance such as the Multiple Access with Reduced Handshake (MARCH) scheme [18] that tries to minimize control signalling as shown in Figure 2.6. Suppose node a has a packet to transmit to node e. By using a path $a - b - c - d - e$, node a sends RTS_a to the next hop of the path, which is node b. When node b replies with CTS_b, node c hears this message, so it knows that node b is going to receive data from node a; thus, node c will reply with CTS_c to node b and node d will hear that and so on.
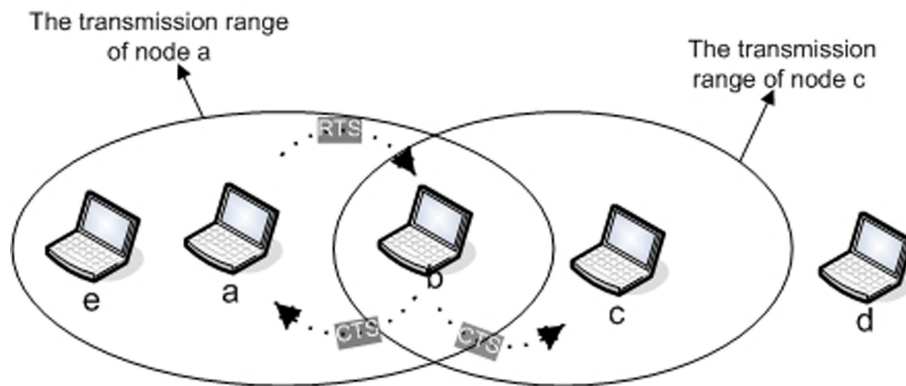
Figure 2.4: The RTS/CTS mechanism
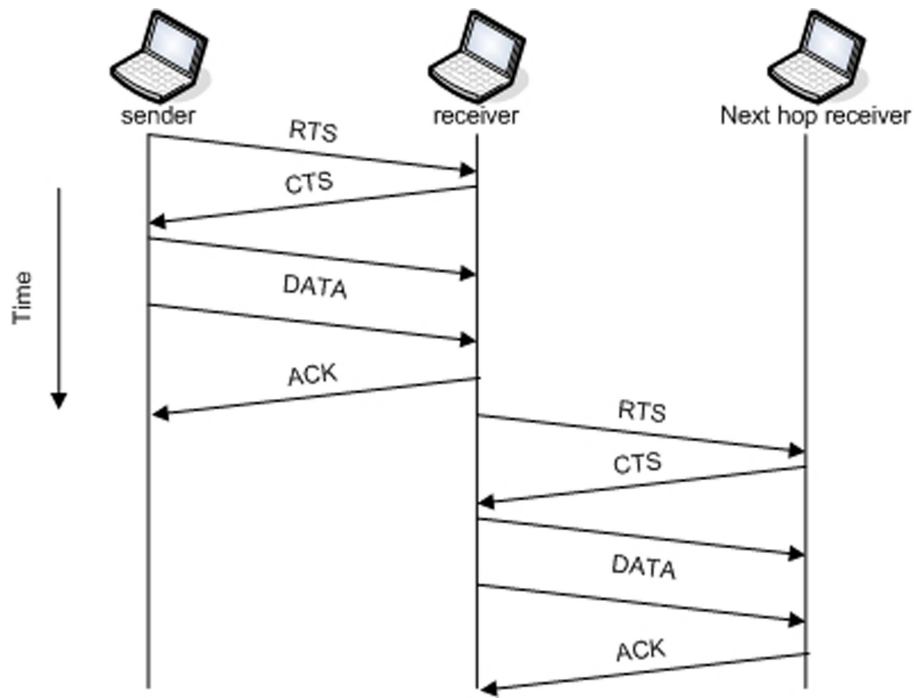
Figure 2.5: Multi-hop handshaking
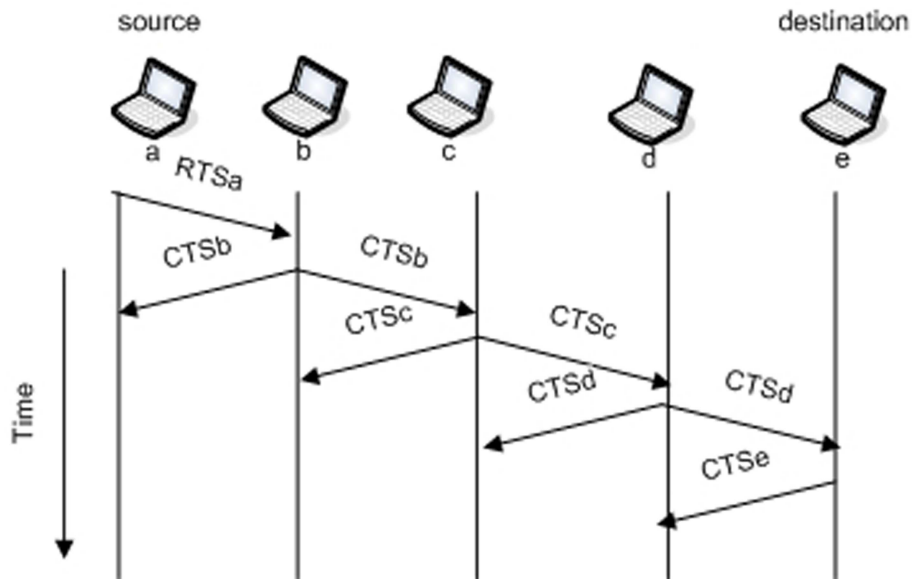
Figure 2.6: Reduced handshaking in MARCH

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Extra frame transmission is a proposed mechanism for multi-hop ad hoc networks [19] in order to utilize the unusable channel (unusable due to missing the CTS) by identifying CTS-Timeout. If the CTS-Timeout is gone, the sender picks a frame from the sending queue and immediately transmits it to the alternate receiver. This scheme increases the throughput up to 10% compared with IEEE 802.11.

Quick-exchange and fast-forward are two layer mechanisms to reduce the effect of self-contention [15]. Quick-exchange is an efficient mechanism for exchanging two data packets between adjacent nodes. As in Figure 2.7, the dialogue RTS-CTS-DATA1-ACK1 has extended by an additional data packet transmission DATA2 from the RTS-receiver. For instance, nodes a and b have data packets for exchanging. Node a sends an RTS frame indicating the duration required for DATA1 transmission. Node b replies with a CTS frame indicating the extra duration needed for DATA2 transmission. The neighbours of node a are notified of the extended channel reservation by the increased duration indicated in DATA1 while the neighbours of node b are notified on receipt of ACK1-DAT2 frame. Consequently, quick-exchange avoids transmitting RTS and CTS frames, and eliminates the back off time that is required by IEEE 802.11 before the transmission of DATA2. DATA2 transmission is free from channel contention, so the throughput will improve.

Fast-forward is the mechanism that tries to forward a packet immediately upon receipt [15]. As shown in Figure 2.8, when a packet is received, the receiver identifies the next hop for the packet and uses its ACK frame as an RTS frame for the next hop. Fast-forward avoids the RTS frame for the forwarded transmission and eliminates the back off time; thus, it increases the channel utilization. This mechanism is free from back off time, so it improves the throughput.
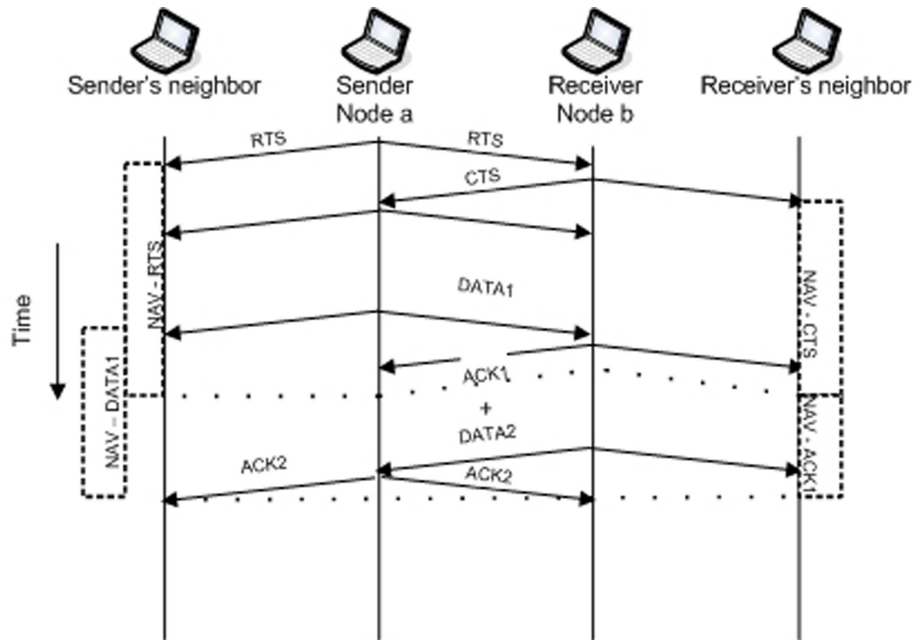
Figure 2.7: Quick-exchange mechanism
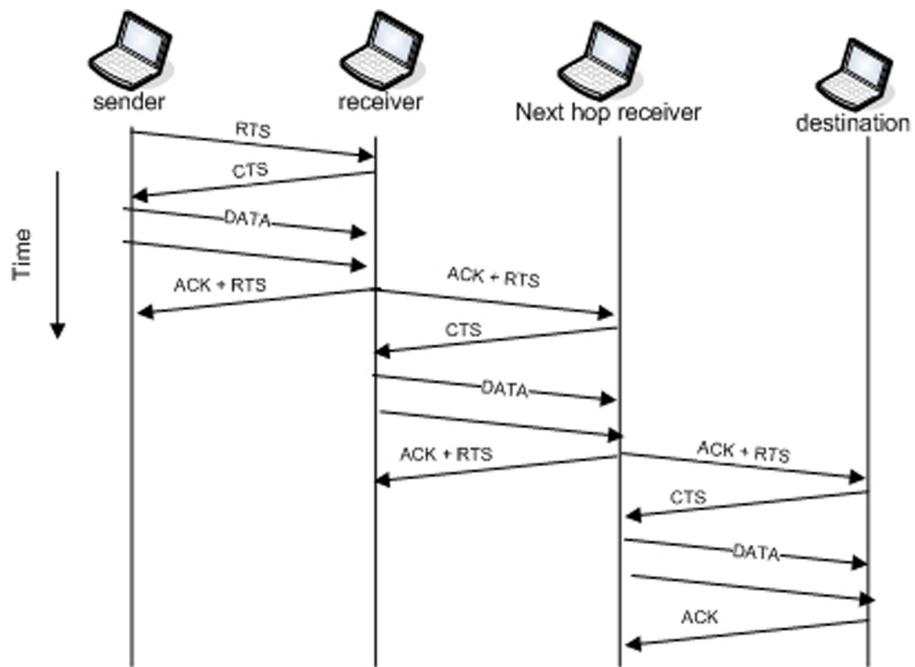
Figure 2.8: Fast-forward mechanism

The advantage of using the CSMA scheme in multi-hop ad hoc networks is its simplicity. However, the performance is not sufficient when we increase the number of hops. In [11], it is shown that the hidden and exposed node problems become worse in multi-hop ad hoc networks using IEEE 802.11. In summary, CSMA MAC protocols in multi-hop ad hoc networks still need further enhancements to achieve a good performance.

2.4.2   Multi-channel MAC

Multi-channel MAC is more complex and expensive to implement than a single-channel MAC. However, using multi-channels, we can achieve higher network throughput than by using one channel because multi-channel MAC protocols increase the number of simultaneous active users.

The multi-channel hidden node problem is an essential issue in multi-channel environments. To illustrate this problem, we will assume a simple protocol, which has one channel that is dedicated for exchanging control messages and all the other channels are for data. For example, in Figure 2.9, channel one is the control channel and the others are for data. Suppose node a wants to send a packet to node b. Node a sends an RTS frame on channel one to node b. Node b selects channel two for data communication and sends back a CTS frame. Channel two should be reserved by RTS and CTS frames in the transmission ranges of nodes a and b, so no collision will happen. However, node c will not hear the CTS frame from node b because it is busy receiving on channel three. Consequently, node c does not know that node b is receiving over channel two, so node c might initiate a communication with node d and select channel two. As a result, a collision at node b will happen.

The Dynamic Channel Assignment (DCA) protocol is proposed to solve the multi-

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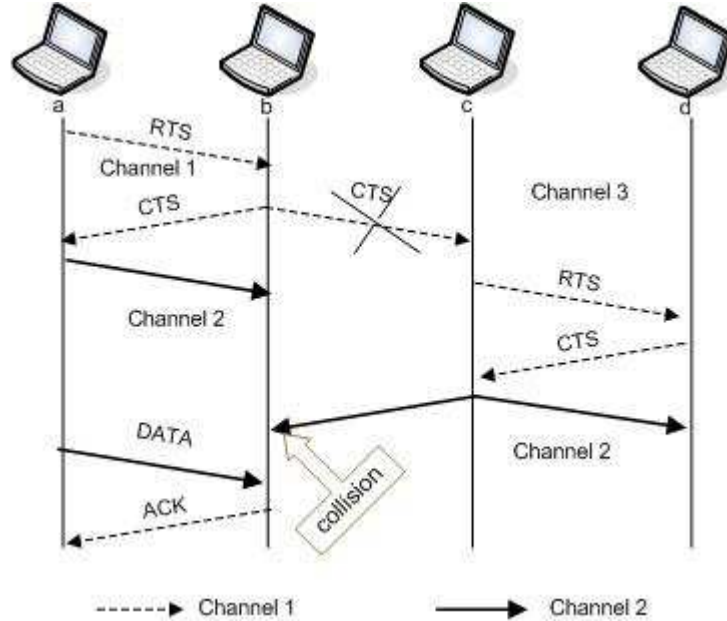
Figure 2.9: The hidden node problem in a multi-channel environment

channel hidden node problem [20]. This protocol is similar to what is explained in Figure 2.9 except that each node has two transceivers, so each node can listen simultaneously to the control and data channels. Since one of the two transceivers is always listening on the control channel, the multi-channel hidden node problem does not occur.

A multi-channel multi-transceiver MAC requires a high cost. If the concern is about cost and compatibility, a multi-channel single-transceiver MAC is preferred. In a multi-channel single-transceiver MAC, only one channel is active at a time in each network node. However, different nodes may operate on different channels simultaneously; thus, system capacity will be improved. Some protocols have been proposed to coordinate transmission between nodes in a multi-channel single-transceiver MAC protocol such as the multi-

channel MAC (MMAC) in [21].

The MMAC protocol enables nodes to use multiple channels by switching them dynamically, so the throughput will be increased. This protocol requires one transceiver per node; moreover, it solves the multi-channel hidden node problem.

In the MMAC protocol, time is divided into multiple fixed beacon intervals. Every node starts each beacon interval at the same time, so the nodes are synchronized. The beginning of every interval has a small Ad hoc Traffic Indication Message (ATIM) window, where every node should be in the awake state. ATIM packets are exchanged among nodes, so they can select the appropriate channel. Furthermore, every node maintains a Preferred Channel List (PCL), which stores the available channels within its transmission range and determines priorities for those channels.

The channel selection manner in MMAC is as follows: suppose node a wants to send to node b. Node a sends an ATIM frame to node b within an ATIM window while the PCL of node a is included in the frame. Upon receiving the ATIM, node b decides which channel to use during the beacon interval based on its PCL and the PCL of node a. After the decision, node b sends an ATIM-ACK frame to node a in order to specify the chosen channel. Then, node a decides if it can choose the channel that has been specified in the ATIM-ACK frame or not. If yes, node a will send an ATIM-Reservation (ATIM-RES) frame to node b with node a's selected channel specified in the frame; otherwise, it will not send an ATIM-RES frame to node b.

The problem in the MMAC protocol is that even if the nodes have already finished exchanging the ATIM frames, they can not exchange data frames during the ATIM window. Therefore, changing the size of the ATIM window dynamically based on the traffic condition

is a challenging issue.

2.4.3   MAC based on Multiple Access Techniques

MAC protocols based on multiple access techniques in WMNs have been proposed to resolve the issue of low end-to-end throughput. The most popular multiple access techniques are Time Division Multiple Access (TDMA), Frequency Division Multiple Access (FDMA), and Code Division Multiple Access (CDMA) techniques. In the TDMA technique, each node transmits in a particular time slot using the entire system bandwidth. The second technique, FDMA, allocates different data channels, used at all times, for each node. In CDMA, each node can use the whole bandwidth at all times. However, each node has unique codes; therefore, nodes are able to recognize each other's packets.

A Five Phase Reservation Protocol (FPRP) is one of the oldest protocols that have been proposed based on TDMA for ad hoc networks [22]. Nodes use a contention mechanism to acquire time slots. A time slot is divided into two slots: a reservation slot and an information slot. If a node wants to send a packet, it has to reserve an information slot by contending for it during the reservation slot. The reservation slot consists of five phases: Reservation Request phase (RR), Collision Report phase (CR), Reservation Confirmation phase (RC), Reservation Acknowledgment phase (RA), and Packing and Elimination phase (P/E). The five phase dialog ensures that the protocol is free from contention because once a reservation is made by a node, it achieves sole access to the slot within its neighborhood. In FPRP, the reservation process is simple. In contrast, one of the recent schemes that have been proposed is an Adaptive and Distributed Spatial TDMA (DTDMA) [23] to find the maximum non-interference link set. Based on the enhanced RTS-CTS scheme, fixed

ACK, and neighborhood information, DTDMA can avoid the hidden and exposed node problem. The enhanced RTS-CTS scheme allows a node to send its RTS frame after it hears another RTS frame if its transmission link does not interfere with the other RTS's link. ACK has been fixed in order to avoid collisions between data and ACK frame under the enhancement of RTS-CTS scheme.

As mentioned, FDMA divides the system bandwidth into different data channels. Nasipuri et al. use the FDMA technique in a multi-channel CSMA (MCSMA) protocol to reduce collisions [24]. Every node has a list of idle channels. When a node intends to transmit a packet, it attempts to access the last used channel if it is available; otherwise, it picks a channel from its list. FDMA has been applied in the RTS/CTS handshaking mechanism in [25, 26]. Each node can choose among the available channels, considering that one channel is for exchanging control messages and others are for exchanging data.

In MAC based on CDMA, two basic issues have to be considered: code assignment and interference. Code assignment is classified into three types [27]: common code, receiver-based code, and transmitter-based code. Common code means that nodes transmit with a common code. Receiver-based code means that nodes transmit with a unique receiving code of the receiver. Finally, nodes use a unique transmitting code of the transmitter for their transmission in the transmitter-based code. As mentioned, in CDMA, all nodes use the same bandwidth. As a result, each new transmission adds further interference to other concurrent transmissions that are within its range. The interference affects the network size, network density, traffic load, and consequently network throughput [28]. Researchers have proposed many approaches in order to control interference and address the near-far problem by exchanging a Maximum Sustainable Interference (MSI) [29, 30], which is the

maximum additional interference that can be endured. Monks et al. used a busy tone to exchange MSI, taking into account that MSI is inversely proportional to the busy tone power level [31]. The RTS/CTS handshaking mechanism has been used to exchange the MSI information in [30]. In the approaches mentioned in this paragraph, each sender makes the decision of transmitting or not by estimating the MSI, whereas Jiang suggested that existing receivers are the decision maker (to be discussed in details in Section 3.1.3) [32].

Some proposed approaches combine multiple access techniques. For example, in the High Performance Radio Local Area Network (HIPERLAN) [33], developed by the European Telecommunications Standards Institute, uses both the FDMA and TDMA techniques. The system bandwidth has been split into at most five channels, each with a rate of 23.5 Mbps. Before reserving a channel, nodes have to contend in three phases that their lengths and structures depend on fixed time frames and slots. Another example of MAC based on the multiple access techniques is CDMA and TDMA used in wireless mesh networks. In [32], each node has a unique sending and receiving code. Time is split into fixed frames. Each frame is divided into fixed L slots for sending data packets. Each slot is partitioned into fixed M mini-slots for sending probes (to be discussed in section 3.1.3).

2.5 MAC Protocols in CDMA Based WMNs Supporting Real Time Traffic

Before transmission, CDMA uses unique sending sequences to spread the bandwidth of baseband signals [34, 35]. At the receiver side, the same spreading sequences are used to despread the desired signals. The spreading factor is the ratio of the chip rate, the inverse

of each spreading chip's period, to the baseband data rate, given by

$$Spreading\,Factor = \frac{Chip\,Rate}{Baseband\,Data\,Rate}. \qquad (2.1)$$

Using the CDMA technique in wireless networks has superior advantages that have attracted much attention, for instance, using the spectrum more efficiently, simplifying the administration in time and frequency domains, and increasing security [3]. Since using CDMA has been proposed in a third-generation standard for mobile communication Universal Mobile Telecommunication Systems (UMTS) [35, 36], it is no surprise that it has been proposed for WMNs, both WMCs and WMRs.

Many research works have proposed MAC based on CDMA and compared it with IEEE 802.11 for WMNs in terms of network throughput [5, 37, 6, 38], and results show an improvement by applying the CDMA techniques. Successful wireless communication services need an excellent capability to support integrated traffic and provide several applications with different Quality of Service (QoS) requirements.

Supporting real time traffic, video and voice applications, is a challenge. Delay is the most important performance parameter for real time traffic. To our knowledge, little work has been proposed using CDMA based MAC to support real time traffic for WMNs. In [39], CDMA has been used to avoid collisions; however, the effects of interference have been ignored. Fantacci et al. in [40] propose two priority schemes for MAC based on CDMA in ad hoc networks. First, each node has two queues that follow the First Input First Output (FIFO) approach. One queue is for priority traffic and the other is for non-priority traffic. The second scheme manages who has the priority to contend for transmission. A node

that has a priority packet to send enters in the contention directly, whereas if a node has a non-priority packet to send, it has to know if any other node has a priority packet to send or not. In this situation, the authors suggest that the node broadcasts a special packet to inform about its priority status. However, they did not consider the interference that will be added in their network, thus wasting some resources. In [41], as is discussed in the next section, if a node sends its probe in a mini-slot time with a large ID, its probability to get an acceptance for transmission is low; on the other hand, it is high if a node sends its probe in a mini-slot time with a small ID. Therefore, a node that has a high priority packet sends its probe at a mini-slot time with a small ID; in contrast, a node that has a low priority packet sends its probe at a mini-slot time with a large ID. However, if there are only low priority packets to send, the advantage of sending a probe at a mini-slot time with a small ID will be lost.

2.6 Summary

In this chapter, we present an overview of the state of the art in MAC protocols for wireless mesh networks. First, we discuss the network characteristics that should be incorporated into MAC protocols, the popular metrics that used for evaluating MAC protocols, and the most important issues that should be addressed in order to achieve high end-to-end efficiency. Next, we review the previous work related to designing wireless mesh network MAC protocols. For example, modifying MAC protocols based on single-channel CSMA, MAC protocols in a multi-channel network, and MAC protocols based on multiple access techniques. After that, we discuss the previous work on MAC protocols for CDMA-based

wireless mesh networks supporting real time traffic.

Chapter 3

System Model and Priority Schemes

The priority mechanisms are very essential to support different applications with different Quality of Service (QoS) requirements. For real time traffic, delay is a very important QoS parameter. A long delay can make a received real time packet useless. Therefore, classifying traffic according to priority helps to meet their QoS requirements. Consider two traffic classes with high priority and low priority traffic. Since reducing delay in data traffic is not as urgent as in real time traffic, real time is the high priority traffic, while data is the low priority traffic. As a result, real time traffic can be transmitted in preference to data traffic.

We propose two priority schemes: priority scheme within a node (called buffering priority) and priority scheme among neighbor nodes (called node priority). The first proposed mechanism has independent queues for each type of traffic, and determines which queue should be served first according to service QoS requirements. In [42], a packet prioritizer followed by queues is used to reduce packet delay and packet loss ratio. The priority

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parameter is the ratio of the number of remaining packets in a queue to the remaining time of the candidate packet for transmission in the same queue. The remaining time is defined as the difference between the due time of the packet and the current time in the system. This definition does not consider the generation time of a packet. Therefore, the delay of a packet in its queue is not taken into account, leading to an inaccurate priority mechanism. As will be discussed in Section 3.2, our first priority scheme is based on the timeout of a packet, which is the summation of packet generation time and packet delay bound. The second mechanism is proposed for CDMA systems. In a CDMA system, many nodes can transmit simultaneously. However, in a dense neighborhood, packets can be dropped because of severe interference. Consequently, a priority mechanism among neighbor nodes should be considered in order to give a higher chance of correct transmission to high priority traffic than low priority traffic.

3.1 System Model

3.1.1 Network Structure

We consider a Wireless Mesh Backbone (WMB) with N fixed nodes. Each node acts as a wireless router. As shown in Figure 3.1, WMBs are a collection of wireless routers that connect with each other. Moreover, wireless routers can work as intermediate nodes to support multi-hop connections. Some wireless routers have gateway or bridge function in order to integrate with various other networks and conventional clients. Usually, WMBs have aggregate traffic.
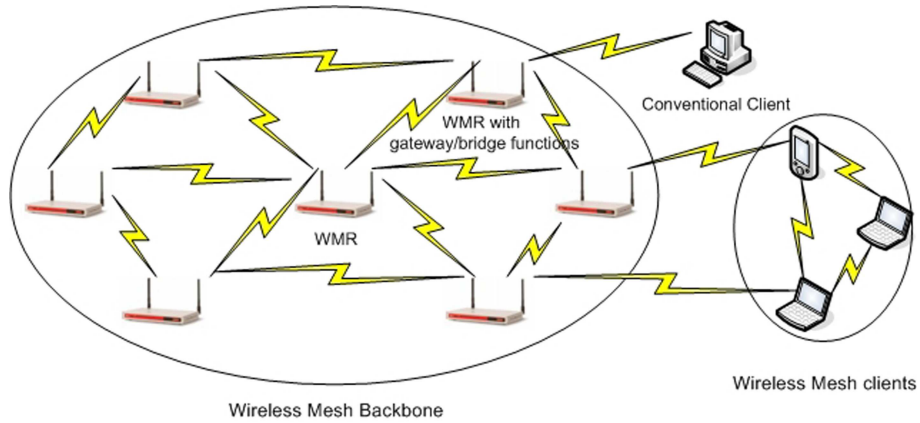
Figure 3.1: Wireless mesh network architecture

Because we deal with the link layer, we consider single-hop connections as in Figure 3.2. The generation of the network topology is random. Due to the node stationarity, each node has the location information of other nodes. The number of senders and receivers are equal. The senders and receivers are uniformly distributed, where the source and destination nodes are specified randomly.

3.1.2   CDMA System

CDMA is a spread spectrum system that transforms narrowband signals to a wideband signal using unique spreading sequences (codes) [43, 44] as in Figure 3.3. However, the power spectral density is decreased by a factor called the spreading gain. Figure 3.4 shows the spreading and dispreading process of the coded information bits in the time domain. Each signal is spread using a unique spreading sequence, then shares the medium. At the receiver, the signal from the intended transmitter (say user 1) is despread using its code.
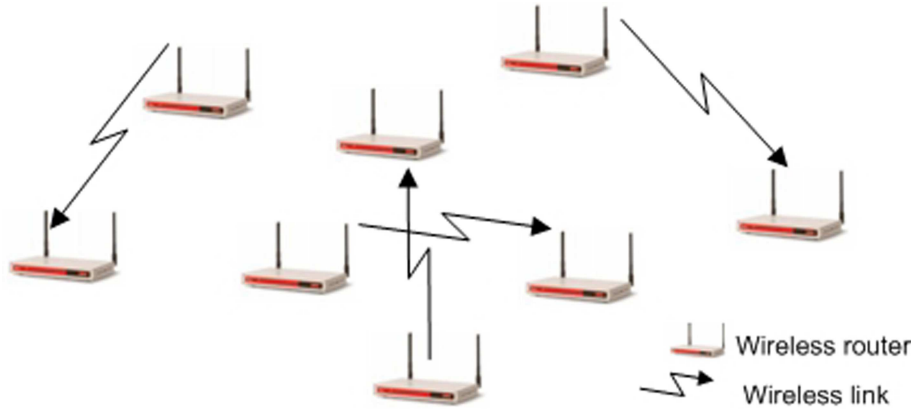
Figure 3.2: Wireless single-hop network

A spreading gain can be defined as

$$g = \frac{T_b}{T_c} = \frac{B_c}{B_b} \tag{3.1}$$

where T_b is the bit time (s), T_c is the chip time (s), B_b is the baseband data bit rate (bps), and B_c is the chip rate (cps).

Another definition of the spreading gain is the ratio of the signal bit energy to interference plus noise density ratio (E_b/N_o) after despreading over that before despreading. It is important to note that the spreading gain affects the transmission rates and the vulnerability to the interference. A high spreading gain gives lower transmission rates but higher invulnerability to the interference. In contrast, a low spreading gain gives higher transmission rates but lower invulnerability to the interference.

The CDMA technique is used for supporting multiple accesses. We consider the case where the spread spectrum bandwidth is the total system bandwidth. Each node has a unique sending code and a unique receiving code. In addition, the sending code and
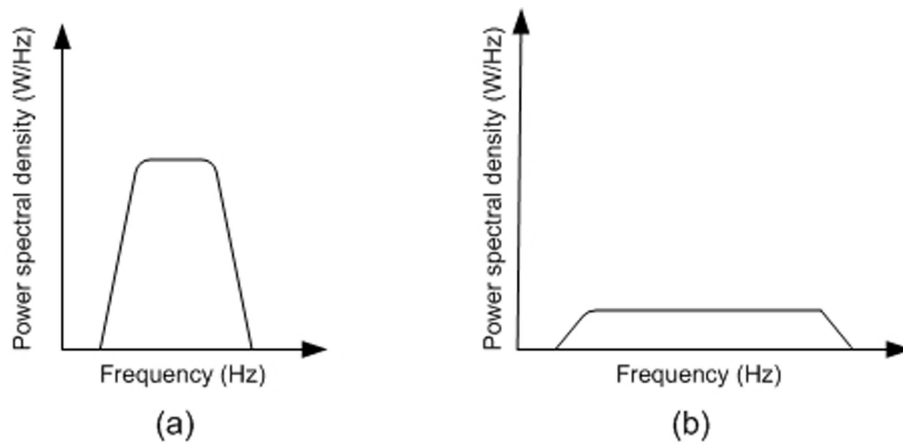
Figure 3.3: Signal power spectral density: (a) before spreading (b) after spreading
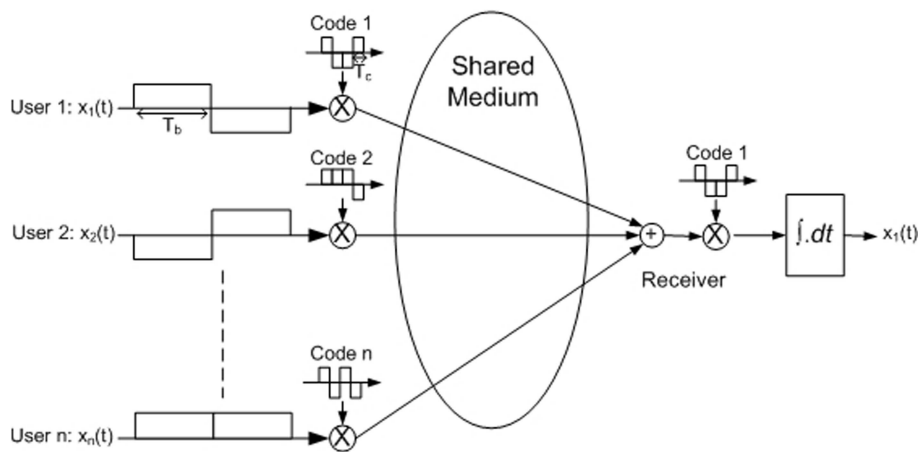
Figure 3.4: CDMA transmission system

receiving code of each node are known by other nodes because of the fixed network topology. We assume each node has a RAKE receiver. Since a RAKE receiver can collect signal energy from different path in CDMA-based networks [45], we assume no fading and the attenuation of transmit power is because of the path loss. We ignore background noise because the effect of multiple access interference is dominant in CDMA transmission [45].

3.1.3 MAC Protocol

This section briefly describes the MAC protocol that we use as proposed in [32, 41, 45]. This MAC is an interference aware distributed MAC protocol for CDMA-based WMB. The reasons of choosing this protocol are its significant characteristics such as: fully distributed, low information exchange overhead, high robustness, high scalability, accurate information estimation by receivers, fine QoS support, and simultaneous transmissions. A slotted time frame structure has been considered in order to make links with low mutual interference transmit at the same slot. On the other hand, those with large mutual interference should transmit at different slots. As shown in Figure 3.5, each frame is divided into L slots. Moreover, each slot is divided into M mini-slots. Frequency band is split into two bands: information and busy tone bands.

The Procedure of the MAC Protocol

As in Figure 3.6, suppose node a has a packet to send to node b. The procedure of the MAC protocol is as follows:

- At the first available frame, denoted by ℓ (where $\ell \geq 1$), node a scans node b's
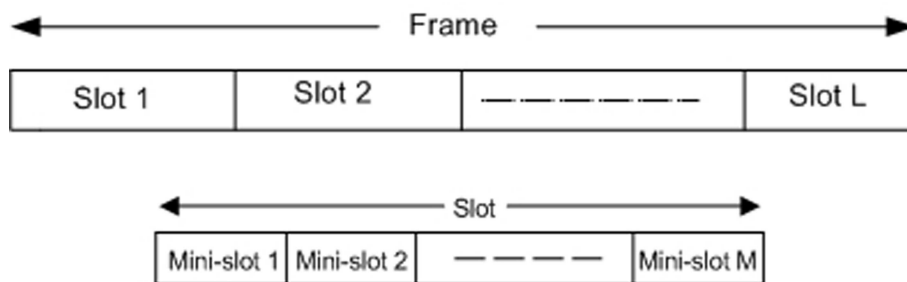
Figure 3.5: The frame structure in the MAC protocol

sending code at each time slot because each node can not send and receive at the same time, thus node a must not transmit to b at a time that node b transmits. In addition, at mini-slot 1 of each slot, node a measures all experienced interference at each slot, then selects the slot that has the minimal interference (say S_{min}).

- At the next frame $(\ell + 1)$, slot S_{min}: node a randomly selects a mini-slot from 2 to M. At the selected mini-slot (say m), node a transmits a probe via a common probe code with a very large spreading gain. The transmit power level is $\xi_p \cdot P_{ab}$, where ξ_p is a very small value ($\xi_p << 1$) and P_{ab} is the transmit power level for data transmission from node a to b. The reason of using a small power level is to avoid corrupting concurrent transmissions and the use of large spreading gain is to ease the demodulation of a probe at existing receivers (e.g. node c in Figure 3.6).

- After the reception of a probe, existing receivers decide whether the new transmission, the transmission from node a to b, will corrupt its own transmission or not (to be discussed). If yes, node c sends a busy tone in the busy tone band at mini-slot n, where n equals $m + 1$ in the same slot if $m < M$, or equals 1 at the next slot if $m = M$. The reason of sending a busy tone is to inform candidate transmitters (e.g.

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node a) that their transmission will corrupt other transmissions. Therefore, any node listens to a busy tone has to select another slot for its transmission, and follows the steps above.

- If no detected busy tone, at S_{min} of frame $\ell + 2$, node a (sender) sends a request message with power level P_{ab}. This message has the interference level that has been measured by node a (sender). After the reception of the request message, node b (receiver) estimates the signal bit energy to interference plus noise density ratio (E_b/N_o) of the transmission. If the estimated ratio is above the required E_b/N_o (denoted by Γ), node b selects a slot (denoted by S_A) to send an ACK message. S_A has the minimal interference level, and it is used neither for transmission by node a nor for reception by node b. After that, node b sends a confirmation message to node a with a small power level $(\xi_A \cdot P_{ba})$ and a large spreading gain.

- At S_{min} of the next frames, node a transmits data with power P_{ab} at slot S_{min}. In contrast, node b transmits ACKs with power $\xi_A \cdot P_{ba}$ at slot S_A until the data transmission is completed.

Power Assignment

A link that experiences high interference needs high transmit power level. Due to the effect of the transmission length on the path attenuation, the transmit power level should increase or decrease with respectively increasing or decreasing the path attenuation. For

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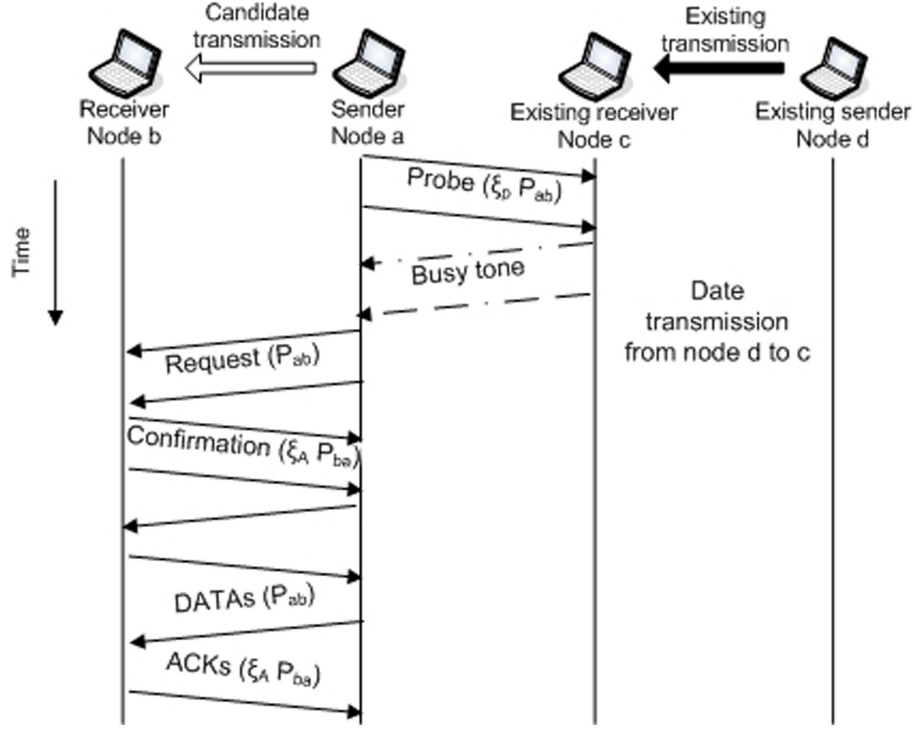
Figure 3.6: The procedure of the used MAC protocol

instance, when node a wants to transmit to node b, the proposed power level is

$$P_a = P \cdot d_{ab}^{\alpha} \sum_{c \neq a, c \neq b}^{N} d_{cb}^{-\alpha} \tag{3.2}$$

where P_a is the power level at node a, P is a constant, d_{ab}^{α} is the path attenuation of the link from node a to b, α is a path attenuate exponent, d_{ab} is the distance from node a to b, and $\sum_{c \neq a, c \neq b}^{N} d_{ab}^{-\alpha}$ is the interference level generated by other nodes, where N is the number of nodes.

E_b/N_o **Estimation and Busy Tone Level**

This part can be divided into two parts:

1. *The procedure of interference estimation at an existing receiver*

As in Figure 3.6, node c (existing receiver) determines its experienced interference (say I_c) from other transmissions at mini-slot 1. Node c receives a probe at mini-slot m ($m \in \{2, ..., M\}$) with power level $P_p^R(m)$. Upon the reception of the probe, node c measures the received probe power, then it estimates the power level to $P_p^R(m)/\xi_p$, where ξ_p is the ratio of probe to data transmission power. After that, node c decides either to allow or reject the new transmission based on the following equation

$$\frac{g_c \cdot P_c^R}{I_c + \frac{1}{\xi_p} \cdot \sum_{i=2}^{m-1} P_p^R(i) \cdot (1 - f(i)) + \frac{P_p^R(m)}{\xi_p}} \geq (1 + \beta)\Gamma_D \qquad (3.3)$$

where g_c is the spreading gain of the reception at node c, P_c^R is the power level of the desired signal received at node c, β is a margin value, $f(i)$ equals 1 if a busy tone is detected at mini-slot $i+1$ and equals 0 otherwise, and Γ_D is the E_b/N_o threshold of the data.

If the left side of Equation (3.3) is less than the right side, node c will send a busy tone at mini-slot n.

When there are more than one probe (say j probes) being sent at the same mini-slot

(say m), the received probe power level is

$$P_p^R(m) = \Sigma_{i=1}^j \xi_p \cdot P_i^T \cdot h_{ic}. \tag{3.4}$$

As a result, the interference generated by probe's senders is given by

$$\frac{P_p^R}{\xi_p} = \Sigma_{i=1}^j P_i^T \cdot h_{ic} \tag{3.5}$$

where P_i^T is the data transmission power level at node i and h_{ic} is the path gain from node i to c.

2. *The busy tone power level if the potential links should be rejected*

A high busy tone power could block some probes's senders that do not corrupt other nodes from the transmission. Then, the senders retry to send another probe. As a result, the delay will be increased because of sending many probes. On the other hand, a low busy tone power could not block senders that do make corruption. Therefore, the busy tone power level has been determined as

$$P_c^{bt} = \frac{P^{bt}}{\sum_{i=1,i=c}^N d_{ci}^{-\alpha}}. \tag{3.6}$$

Equation (3.6) indicates that the busy tone power level decreases if the existing receiver is in a crowded neighborhood, where senders that generate large interference are mostly close to the existing receiver. In contrast, the busy tone power level increases if the existing receiver is in a light neighborhood, because it is most probable

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that senders are far from the existing receiver.

3.1.4   Frame Structure

As shown in Figure 3.5, time is divided into fixed size frames. Each frame consists of a number (L) of slots, while each slot comprised of a number (M) of mini-slots. Adjusting the slot time for voice traffic is vital because voice traffic has a long packetization interval, which is the inactive interval between generation packets. Consequently, each slot time should be equal to the transmission length of a voice packet in order to avoid wasting some resources. For example, if the slot time is longer than the sending voice packet's transmission time, we waste the residual time in the slot because the next voice packet will be sent after the inactive period.

Based on Equation (3.1), we can calculate the transmission time of a voice packet as

$$t_T = \frac{g \cdot P_p}{C} \tag{3.7}$$

where

$$P_p = S_R \cdot P_I \tag{3.8}$$

t_T is the transmission time of a packet (s), P_p is a packet payload (bits), C is the channel capacity (bps), S_R is the source rate (bps), and P_I is the packetization interval (s).

3.1.5 Adaptive Transmission Rates

Adaptive transmission rates are needed to maximize the utility of the link especially in a light traffic load. In our situation, we need to have adaptive transmission rates especially when we have few active links at each slot. In fixed transmission rates, each link transmits at a low rate although the received E_b/N_o can be higher than the target value. On the other hand, in adaptive transmission rates, each link transmits at the highest possible rate, such that the target E_b/N_o is met. The popular methods to adopt transmission rates in CDMA are to use multiple codes as in [32] and variable spreading gain as in [40, 46].

In our work, we use a variable spreading gain in an interval $[g_{min}, g_{max}]$ to achieve adaptive transmission rates. An update of the received E_b/N_o value is sent within the ACK message to the transmitter by the receiver. As a result, the sender determines the new spreading gain as

$$g = max\{g_{min}, min[g_{max}, \lfloor \frac{(E_b/N_o)_{new}}{(1 + \beta) \cdot \Gamma_D} \rfloor]\} \tag{3.9}$$

where g_{max} is the maximum spreading gain and $\lfloor x \rfloor$ is the floor function of value x.

3.1.6 Traffic Model

We consider three types of traffic: data, voice, and video. The call arrival at each node is a Poisson process. Voice and video follow the G.711 and H.263 codecs, respectively. Codecs are to digitize, compress, and encode the analog signals into digital. After that, the digital signals are packed by packetizers [47, 48]. Our traffic models are as follows:

- Data traffic is to simulate a low delay service (non-real time traffic). We consider the best-effort data class. The size of each packet is fixed. The packet generation is a Poisson process with different average arrival rates in order to see how the performance changes with the traffic load.

- In general, voice traffic is represented by an on-off model. The on-off model is a two state process, such that on state is in talk spurt, and off state is in silence [49, 50]. Our voice traffic is considered as a Constant Bit Rate (CBR) flow because of two reasons. First, silence suppression schemes are not used in many voice codecs. Second, although the silence suppression scheme is used, some packets are transmitted intermittently during off period to obtain better voice quality [51]. G.711 is an International Telecommunication Union (ITU) standard codec [52]. It is supported by most VoIP providers because it gives superior voice quility [53]. G.711 packetizes the signal into CBR at 64 $kbps$. A new voice packet is generated every 20 ms. The payload size of each packet can be calculated as 20 $ms*$ 64 $kbps = 1280$ $bits$ [54, 47], so we can determine the packet rate as (64 $kbps/$ 1280 $bits$) = 50 $packets/s$. By accounting the IP, UDP, and RTP headers, the bit rate and payload size will be 80$kbps$ and 1600 bits (with 25% overhead), respectively [55].

- Since video signals have a large bandwidth, video compression or video coding technology is required to reduce the bandwidth before transmission. H. 263 is an ITU standard. It was designed for low bit rate communications such as radio communication links. H. 263 can support compression for video conferencing and video-telephony applications. The basic H.263 encoder generates a variable bit rate (VBR)

traffic. However, the encoder can map the VBR to CBR by carrying out rate control [56]. If the bit rate before the encoder is too high, the encoder increases the compression. In contrast, the compression is reduced if the bit rate is low. As a result, H. 263 can send a compressed video packets at CBR less than 64 kbps and multiples of 64 kbps [57]. We consider CBR at 160 kbps (128 $kbps + 25\%$ overhead) in our model.

3.1.7   QoS Requirements

Our QoS requirements of each service type are a guaranteed E_b/N_o at the receiver, delay, and packet dropping rate. These requirements for each class are shown in Tables 3.1 and 3.2 [58, 54].

Table 3.1: Delay bound requirements

Service	Class	Delay bound
Voice	1	150 ms
Video	2	150 ms

Table 3.2: Packet dropping rate and E_b/N_o requirements

Service	Class	Packet dropping rate bound	E_b/N_o (dB)
Voice	1	3%	5.31
Video	2	1%	9.32
Data	3	0	2.94

The one way end-to-end delay for a voice or a video packet should not exceed 150 ms;

otherwise, the packet will be dropped. We consider best effort data service which is delay tolerant. Packet dropping rate is a very important parameter for data traffic. For data traffic, since the delay requirement is not strict and the packet dropping rate is vital, failed data transmissions will be retransmitted. The E_b/N_o requirements for each service are determined based on the Bit Error Rate (BER) requirements as in [42].

3.2 Packet priority scheme within a node

Each node has three logical queues that follow packet prioritizer as shown in Figure 3.7. Each queue has a single type of traffic, and follows the FIFO mechanism. Packet prioritizer is needed in order to provide the priority of packets in different queues depending on a packet's due time. If there is real time traffic, the packet prioritizer will choose a packet that has the minimum due time among real time classes; otherwise, non-real time traffic will be chosen. The due time for a packet can be calculated as:

$$t_{due_{ij}} = t_{g_{ij}} + t_{d_j} \tag{3.10}$$

where i is the packet's ID, j is the service type or class ID (voice or video), $t_{due_{ij}}$ is the due time of packet i with service type j, $t_{g_{ij}}$ is the generation time of packet i with service type j, and t_{d_j} is the delay bound for class j. Since the delay is not very urgent in class 3, we assume j is either 1 or 2.
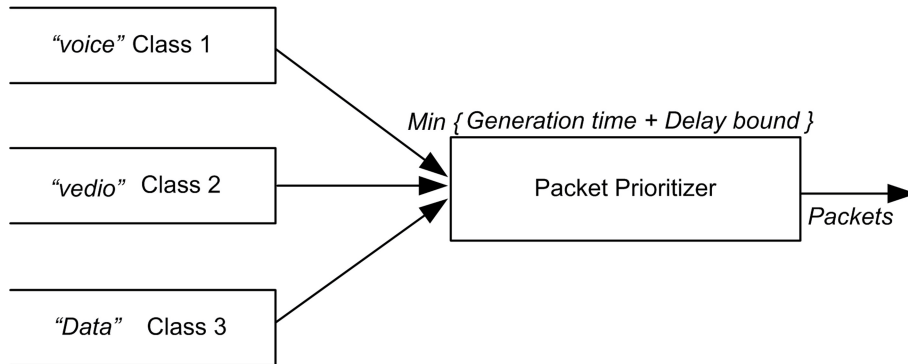
Figure 3.7: Packet priority within a node

3.3   Packet priority scheme between neighbor nodes

After passing the packet prioritizer, a packet of either high priority or low priority packet is being selected. We consider packets from class 1 and class 2 have high priority, while class 3 packets have low priority. For example, as in Figure 3.8, when a certain node a has to send a packet to another node b, this packet is either high or low priority packet. Because of the simultaneous transmissions in CDMA, four cases might occur:

1. The sender (e.g. node a) and its neighbors have a high priority packet to send;

2. The sender and its neighbors have a low priority packet to send;

3. The sender has a low priority packet and at least one of its neighbors has a high priority packet;

4. The sender has a high priority packet and at least one of its neighbors has a low priority packet.

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In cases one and two, packets from the sender and its neighbors have the same priority. In contrast, packets have different priorities in cases three and four. Thereby, any node that has a low priority packet to send has to know if its neighbor nodes have a high priority packet to send or not. The next subsections discuss the MAC procedure and the proposed priority management technique in the CDMA MAC protocol.
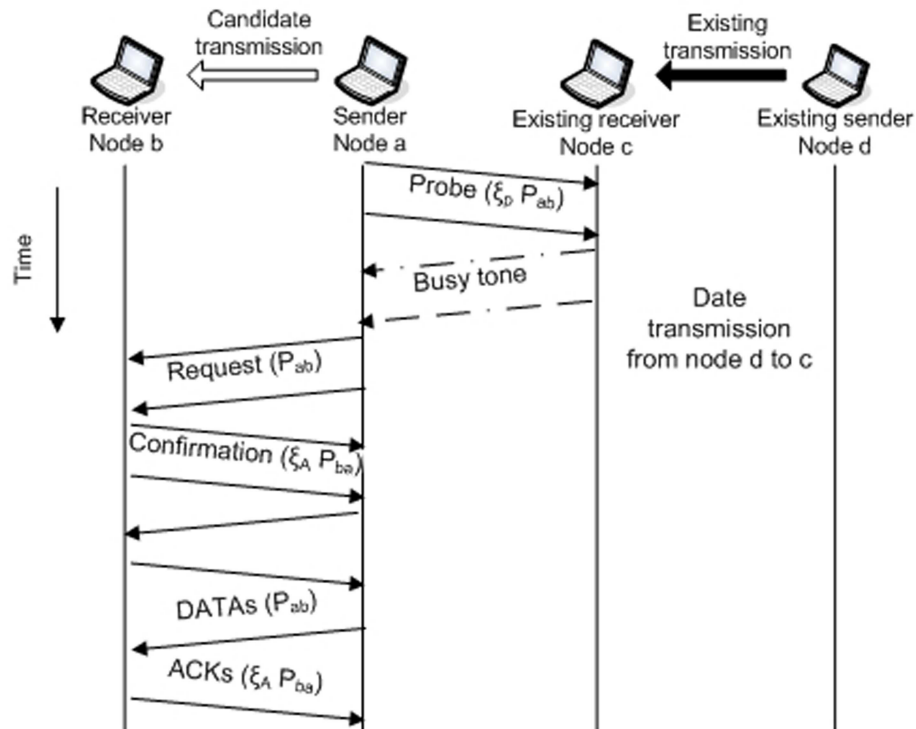

Figure 3.8: The procedure of the used MAC protocol

3.3.1 MAC procedure for a node with a high priority packet to send

As discussed in [41], if a node sends its probe in a mini-slot time with a large m, its probability to get an acceptance for transmission is low; on the other hand, the probability is high if a node sends its probe in a mini-slot time with a small m. Therefore, a node that has a high priority packet sends its probe at a mini-slot time with a small m; in contrast, a node that has a low priority packet sends its probe at a mini-slot time with a large m. In our scheme, we divide the mini-slots, excluding mini-slot 1 and 2, into two halves: the first half is for high priority traffic, and the second one is for low priority traffic. Mini-slot 1 is used to measure the interference, while mini-slot 2 to send and detect a busy tone (as will be discussed).

When node a has a high priority packet to send to node b as shown in Figure 3.8, node a at mini-slot 2 sends a busy tone that should cover its two-hop neighbors in order to let the existing receiver's neighbors (e.g. node d) that have a low priority packet know about the high priority packet.

For each transmitter, we define the neighborhood coverage as a circle centered at the transmitter with radius being the distance to its neighbor with the longest distance.

A neighbor of the busy tone sender may be one of the following:

1. a source node having a low priority packet;

2. a source node having a high priority packet;

3. a receiving node.

Only nodes in the first case sense the busy tone at mini-slot 2. Since we have only one transceiver in each node, each node can not send and receive at the same time. As a consequence, in the second and third cases, the node can not sense a busy tone because it is either sending a busy tone at the same time slot or receiving data from the information band.

After sensing the busy tone, mini-slots 3 to M will be separated into two parts as shown in Figure 3.9. If M is an even number, from 3 to $(M + 2)/2$ mini-slots are reserved for a probe by nodes that have high priority traffic, and from $(\frac{M+2}{2} + 1)$ to M are reserved for a probe by nodes that have low priority traffic. If M is an odd number, it is impossible to divide the slot into equally two parts of mini-slots. In this case, the high priority traffic has the preference to have the extra mini-slot. Mini-slots from 3 to $(M + 3)/2$ are reserved for a probe by nodes that have high priority traffic, and from $(\frac{M+3}{2} + 1)$ to M are reserved for a probe by nodes that have low priority traffic.

After sending a probe, an existing receiver determines whether or not the new transmission will corrupt its reception according to Equation (3.3); however, the required E_b/N_o is different for each type of service. If an existing receiver has a corrupted packet, it sends a busy-tone at mini-slot $m + 1$, where m is the mini-slot of sending a probe. If no busy tone is sensed, the candidate sender will continue the MAC procedure by sending request message and data in the next frames as explained in Section 3.1.3.

3.3.2 MAC procedure for a node with low priority packet to send

When a node has a low priority packet to send, it has to know by sensing a busy tone at mini-slot 2 if among its neighbors a node has a high priority packet to send or not. If no
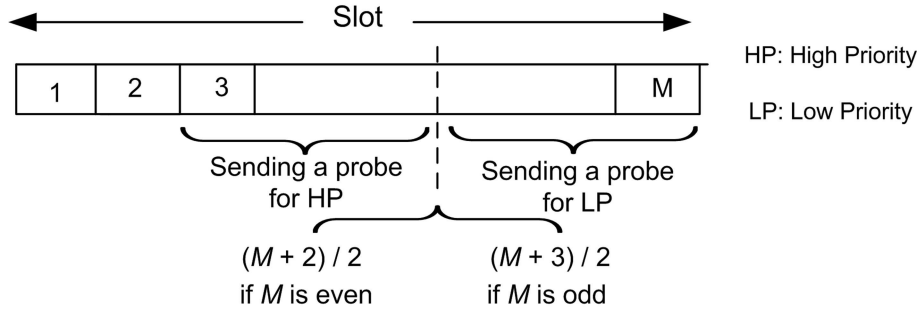
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Figure 3.9: Slot structure with service differentiation

busy tone has been detected at mini-slot 2, which means no high priority packets will be transmitted by its neighbor nodes, the node selects randomly a mini-slot $m \in \{3, ..., M\}$. On the other hand, if the busy tone has been detected at mini-slot 2, which means there are high priority packets to be sent by neighbor nodes, the node selects randomly a mini-slot $m \in \{\frac{M+2}{2} + 1, ..., M\}$ in order to give the advantage to the high priority traffic. The proceeding steps are similar to what have been discussed in Section 3.1.3.

3.4 Summary

In this chapter, we presented our system model and our priority schemes. The system model includes the network structure, the CDMA system, the MAC protocol, the frame structure, the transmission rates, the traffic model, and the QoS requirements. The proposed priority schemes are the buffering priority and the node priority. The first scheme is to prioritize packets within a node based on the due time of class 1 and class 2 (voice and video) packets. The second scheme is to manage the priority among neighbor nodes. The busy tone is used to inform nodes that have a low priority packet to send about the existence of a node with a

high priority packet to send. These schemes take into account the interference phenomenon, QoS requirements for each type of service, and the simultaneous transmissions in CDMA.

Chapter 4

Performance Evaluation Based on Simulation

To evaluate the performance of the proposed schemes, computer simulations are carried out using Matlab [59]. The simulations are performed for wireless mesh backbones, which have fixed nodes. CDMA is used to support multiple accesses. The chip rate is 50 Mcps. We assume no fading, no propagation and jitter delay, and the attenuation of transmit power is because of the path loss with exponent $\alpha = 2.4$ [32, 45]. We consider three traffic classes: data, voice, and video. The arrival at each node is a Poisson process. The generation of the network topology and traffic are random. The simulation runs are carried out with various seeds.

We consider 110 nodes (55 links), among which 100 nodes can transmit the three classes, while 10 nodes have only data traffic in order to notice explicitly the impacts of data on real time traffic. The senders and receivers are uniformly distributed in $10km \times 10km$,

where the transmission range is 500 m. The rest of our simulation parameters are specified in Table 4.1.

Time is divided into fixed frames. Increasing the frame length will increase the delay. Since delay is the most important for real time traffic, choosing a suitable frame length is very important to achieve the delay requirements.

Table 4.1: Simulation Parameters

slot number L	3
mini-slot number M	16
slot time	0.512 ms
mini-slot time	32 μs
path loss exponent	2.4
chip rate B_c	50 Mcps
channel capacity C	50 Mbps
maximum spreading gain g_{max}	16 [42]
minimum spreading gain g_{min}	1
spreading gain for a probe g_p	1600
spreading gain for an ACK g_A	1600
the ratio of probe to service transmission power ξ_p	0.01
the ratio of ACK to service transmission power ξ_A	0.01
Required (E_b/N_o) [42]	
$(E_b/N_o)_{voice}$	5.31 dB
$(E_b/N_o)_{video}$	9.32 dB
$(E_b/N_o)_{data}$	2.94 dB
Bit rate	
voice	80 kbps
video	160 kbps

Since we have a variable spreading gain, the required minimum and maximum trans-

mission rate can be determined as

$$B_{b_{min}} = \frac{B_c}{g_{max}}. \tag{4.1}$$

$$B_{b_{max}} = \frac{B_c}{g_{min}}. \tag{4.2}$$

We assume the channel capacity equals the chip rate. Therefore, $B_{b_{min}} = $ (50 Mcps / 16) $= 3.125$ Mbps, and $B_{b_{max}} = 50$ Mbps.

As discussed in Section 3.1.3, at a mini-slot, either a busy tone or a probe is sent. The minimum required length of a mini-slot equals to the probe and busy tone detection time, which depends on the communication hardware. A busy tone signal detection is studied in [60]. We assume the detection time of a probe and a busy tone is 32 μs.

A slot with a large mini-slot number is required. Senders select randomly a mini-slot to send its probe. If we have a few mini-slots, the probability that more than one probe transmits at the same mini-slot is high, thus more rejected transmissions. In contrast, a large number of mini-slots decrease the probability of sending probes at the same mini-slots. For instance, assume nodes a and b have a probe to send to node c. Two cases can occur: both a and b send at the same mini-slot (say $m = 4$) or different mini-slots (say node a at $m = 3$ and b at $m = 4$). Assume node c sends a busy tone at mini-slot 5. In the first case, nodes a and b transmissions will be rejected. In the second case, only node b transmission will be rejected. In our model, we consider the minimum required length of a mini-slot in order to have the largest possible number of mini-slots because, given a slot time, the mini-slot number is inversely proportional to the mini-slot time.

The E_b/N_o requirements for each service are determined based on its Bit Error Rate

(BER) requirement as in [42]. The required BER for voice is 10^{-3}, for video is 10^{-5}, and for data is 0. As discussed in the traffic model (Section 3.1.6), voice traffic is generated at 80 kbps and video at 160 kbps, respectively, including the header overhead.

For real time traffic, we study the performance of our schemes in terms of average packet delay, packet dropping rate, and the number of received bits. We discuss our results in three parts. First, we compare between the results with using the packet priority scheme within a node (called buffering priority) and without using it. Second, we discuss the improvement of the performance when we add our second priority scheme. Finally, we show the effect of having very heavy real time traffic. Our simulation results are discussed in the next sections.

4.1 Performance with buffering priority

In this section, we present the amelioration of using our proposed buffering priority scheme. The traffic load is defined as the total number of transmitted bits (received and dropped bits) in the system over the simulation time, which is terminated when 25 thousands voice packets are transmitted. Consider each node transmits voice, video, and data as shown in Figure 4.1. The impact of increasing the traffic load on the average real time packet delay is presented in Figures 4.2 and 4.3. The traffic load increases with increasing the arrival rate of data traffic. Obviously in the Figures 4.2 and 4.3, with buffering priority, we achieve a much shorter delay. With no buffering priority, the delay increases with increasing the traffic load, then saturates around 77 ms; while with the buffering priority, the delay saturates around 9.5 ms. The saturation state means that the spreading gain is

maximum. We run a simulation when the spreading gain is fixed at 16, and we get the same results as the saturation state for an adaptive spreading gain.
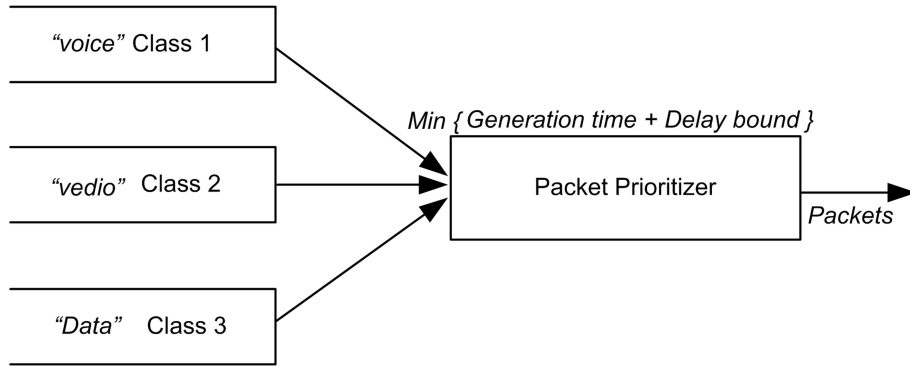

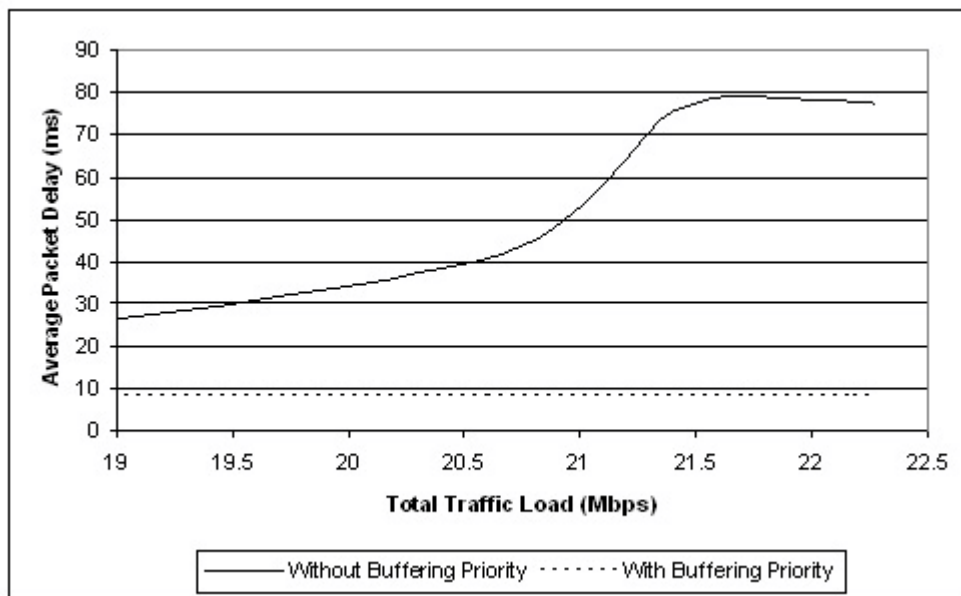Figure 4.1: Packet priority within a node

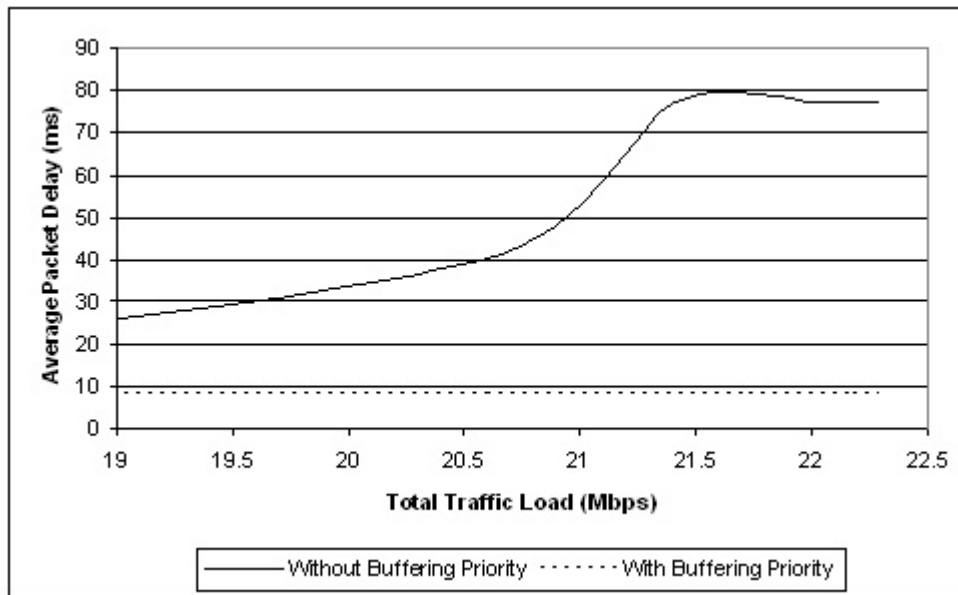Figure 4.2: Average voice packet delay versus total traffic load

Figure 4.3: Average video packet delay versus total traffic load

Packets that go over the delay bound (150 ms) are dropped because they are useless. Figures 4.4 and 4.5 show the real time packet dropping rate after and before adding the proposed buffering priority. It is clear from the figures that, above 21.5 Mbps traffic load, the packet dropping rates are above 90% with no buffering priority, noting that it is almost zero with our buffering priority scheme.

Since the average packet delay and the packet dropping rate with buffering priority are much better than those without using the proposed buffering priority scheme, it is obvious that more real time bits will be received when using the proposed buffering priority scheme as in Figures 4.6 and 4.7.
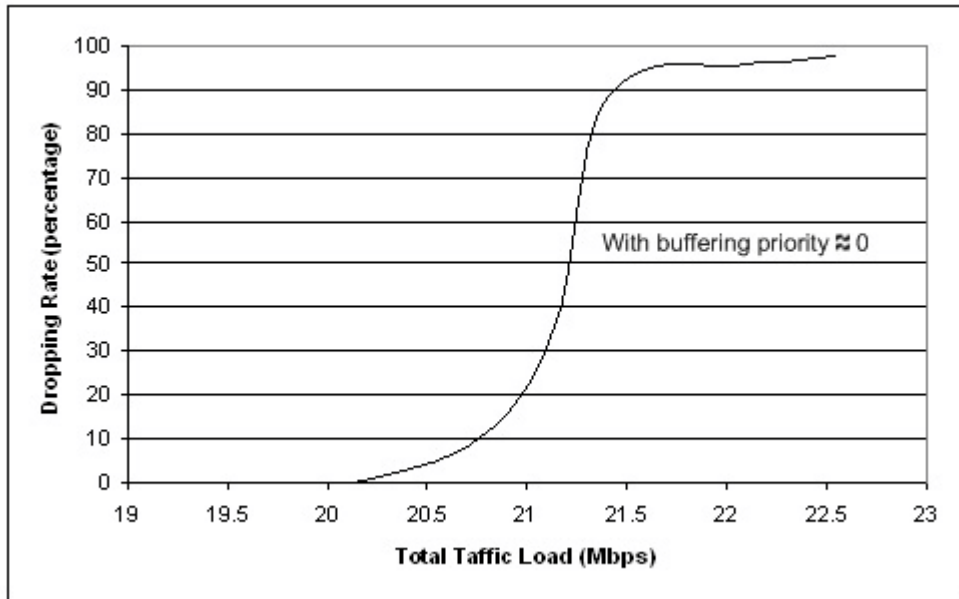
Figure 4.4: Voice packet dropping rate due to over bounding delay without buffering priority
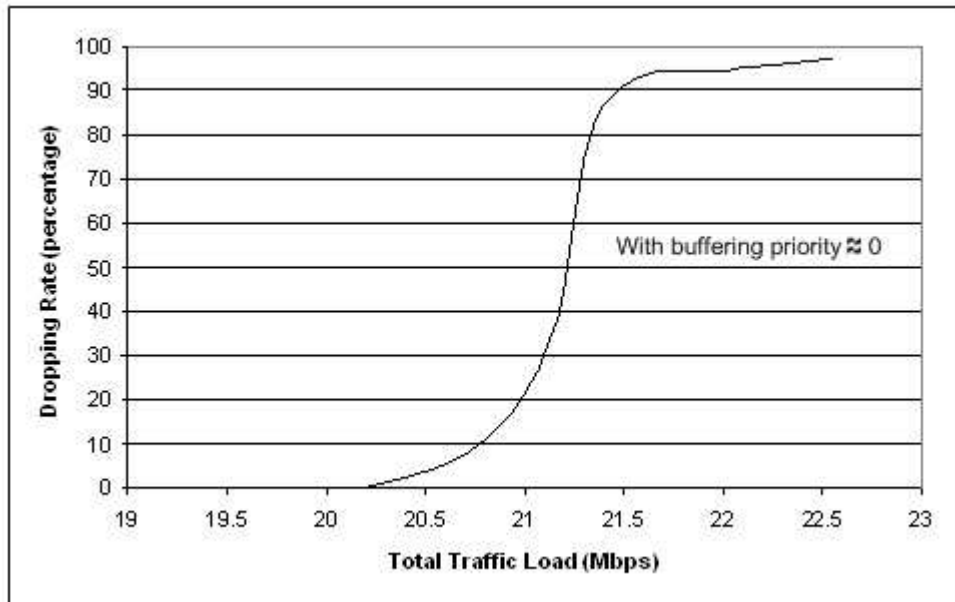

Figure 4.5: Video packet dropping rate due to over bounding delay without buffering priority
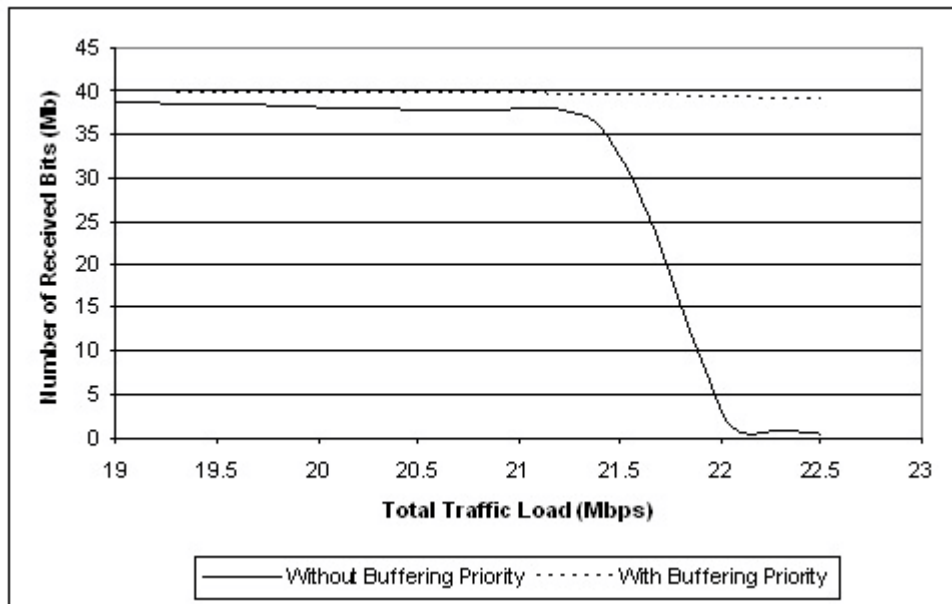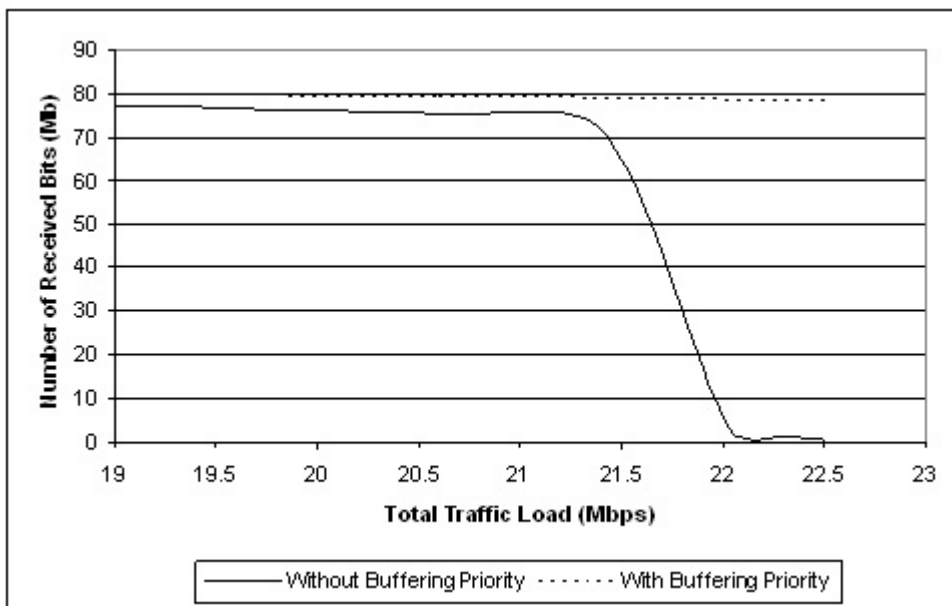
Figure 4.6: Number of received voice bits

Figure 4.7: Number of received video bits

4.2 Performance of the node priority scheme

In this section, we present and discuss the results with using our second proposed priority scheme (called node priority). The proposed buffering priority scheme is used. Dropping rate of high priority packets due to severe interference and the number of received high priority bits are the performance parameters. The objective of using this scheme is to reduce the packet dropping rate for high priority traffic in order to receive more high priority traffic.

Because a high priority traffic load increases the interference among neighbors, the packet dropping rate increases with increasing the traffic load. As in Figures 4.8 and 4.9, in a light traffic load, the packet dropping rates with using the node priority scheme are close to the packet dropping rates without using this scheme. However, in a dense traffic load, nodes experience more interference. In this case, our proposed node priority scheme results in dropping less high priority packets than the scheme with only buffering priority.

In Figures 4.10 and 4.11, the numbers of received high priority bits versus the traffic load are presented. In a light traffic load, the numbers of received bits are high because the dropping rates are low. In addition, with using our proposed node priority scheme, the numbers of received high priority bits are slightly higher than those with using only the buffering priority. On the other hand, the numbers of received high priority bits decrease in a dense traffic load because the dropping rate increases. In this situation, the improvement of using our proposed node priority scheme is noticeable.
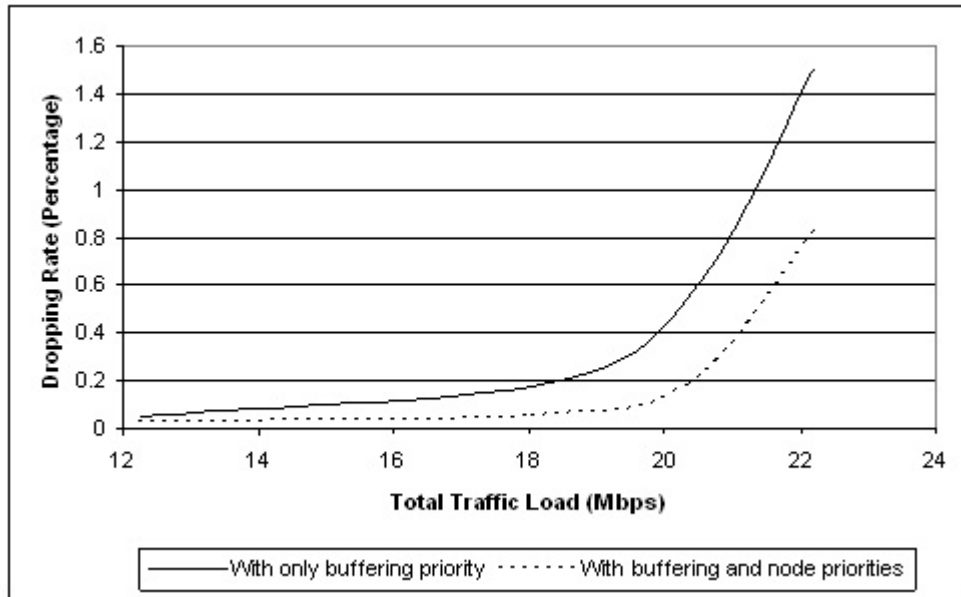
Figure 4.8: Voice packet dropping rate due to severe interference
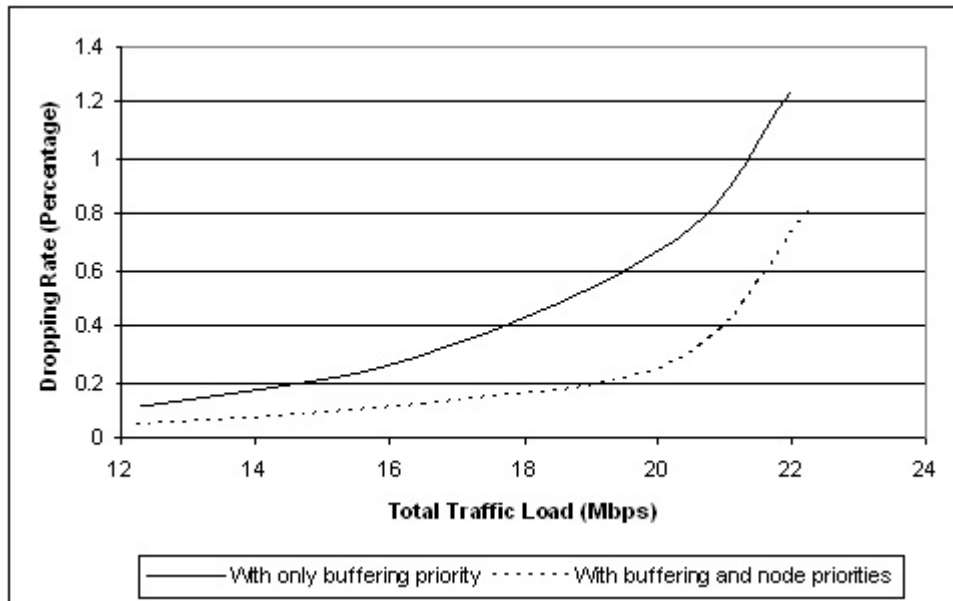

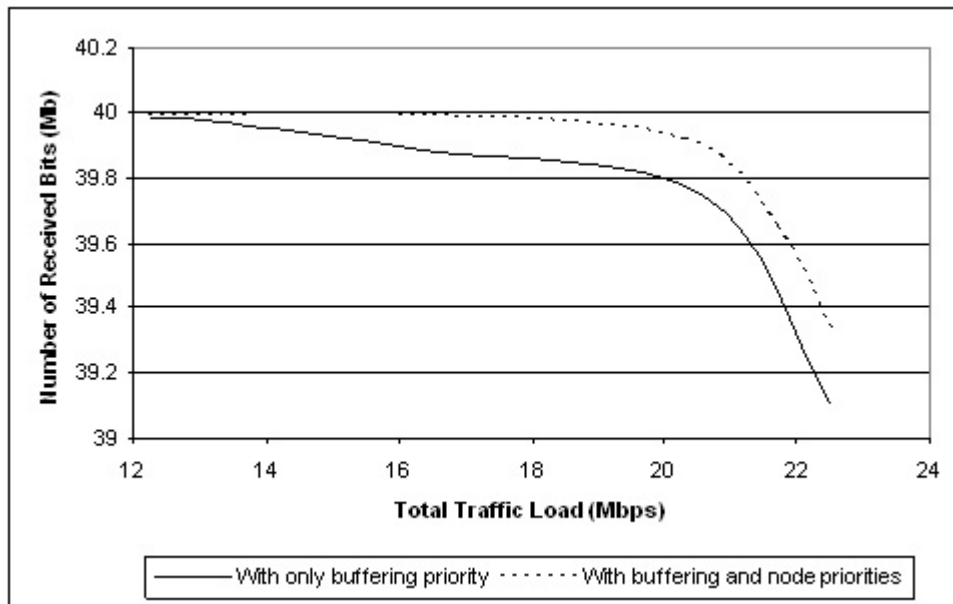Figure 4.9: Video packet dropping rate due to severe interference
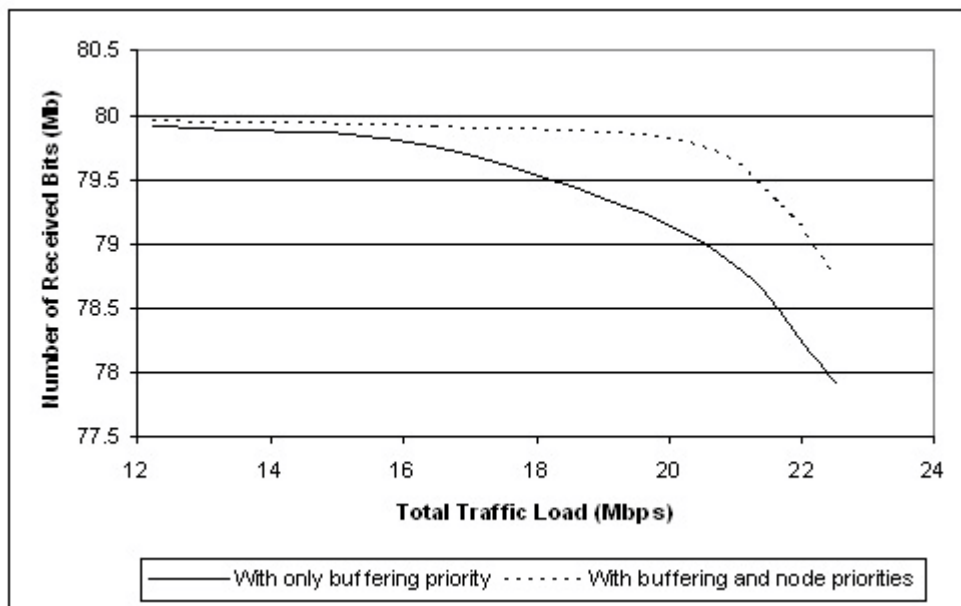
Figure 4.10: Number of received voice bits

Figure 4.11: Number of received video bits

4.3 The effects of increasing high priority traffic load

In wireless mesh backbones, the traffic load is usually high. Consequently, we should study the performance parameters when there is heavy high priority traffic. We have two types of high priority traffic: voice and video. We take voice traffic as an example to study the impacts of having dense high priority traffic. We increase the high priority traffic load by increasing voice traffic load. Voice sources are added at each node in order to increase the voice traffic load. In Sections 4.1 and 4.2, the results are obtained with only one voice source at each node.

The arrival data rate at each node is fixed at 500 kbps, where the traffic load is 22 Mbps. From Figure 4.2, at 22 Mbps and one voice source, the average packet delay is around 9.5 ms if we use the buffering priority scheme, and around 77 ms if we do not use the buffering priority scheme. In Figure 4.12, the average packet delay starts from 9.5 ms with the use of our proposed buffering priority scheme and 77 ms without its use. Without the buffering priority scheme, the delay is fixed because the transmission rate is fixed. In contrast, with the buffering priority scheme, the delay increase with increasing the number of voice sources, and it saturates after adding more than 4 voice sources at each node. The delay value becomes similar to the delay without using the buffering priority scheme because voice packets dominate in each node, thus the waiting time is longer before transmission.

As in Figure 4.4, when the traffic load is 22 Mbps with one voice source, the voice packet dropping rate due to over bounding delay is above 90% without using our proposed buffering priority scheme, while it is nearly 0% with the use of our scheme. As in Figure
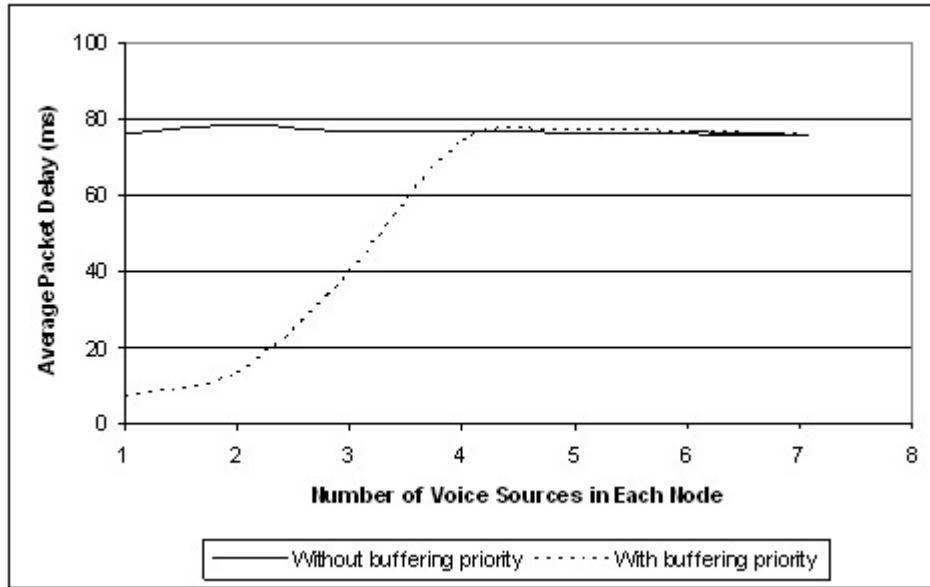
Figure 4.12: Average voice packet delay

4.13, most transmitted voice packets are dropped if we do not use the buffering priority scheme. If we use it, the voice packet dropping rate starts at slightly above 0% with one voice source. After that, it increases with the number of voice sources at each node. It is important to note that after adding 4 voice sources, the average packet delay with and without using the buffering priority scheme is similar as in Figure 4.12. However, the scheme still gives better performance as shown in Figure 4.13, where the voice packet dropping rate using the buffering priority scheme is lower even if we add more than 6 voice sources.

In Figure 4.14, the voice packet dropping rate because of severe interference is presented. As shown in Figure 4.8, when the traffic load is 22 Mbps, the voice packet dropping rate is about 0.85% using our proposed node priority scheme, and it is about 1.5% without using the node priority scheme. These values are obtained when we have only one voice
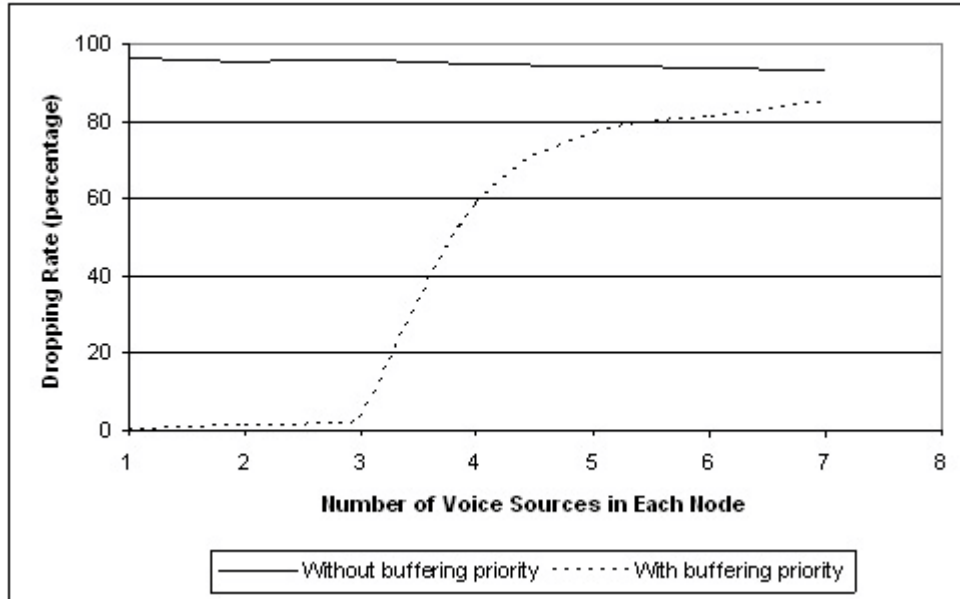
Figure 4.13: Voice packet dropping rate due to over bounding delay

source at each node as shown in Figure 4.14. Both curves in Figure 4.14 decrease with the increasing number of voice sources until 4. On the other hand, the video packet dropping rate because of severe interference increases with the number of voice sources as shown in Figure 4.15. This is because the voice packets dominate within a node. In other words, the transmissions of voice packets are increased, while the transmissions of other classes are decreased. For example, with one voice source, assume node a has a video packet generated at time t_1 and a voice packet at time t_2, where $t_1 < t_2$. Based on the buffering priority scheme, which depends on the generation packet time and the delay bound, the video packet has the priority to transmit. Adding more voice sources, the generation time of voice packets will be shorter. As a consequence, it is most probable that at node a, a voice packet will be generated before the video packet; thus, this voice packet will have the priority for transmission. However, with a further voice traffic load increase, $i.e$ more

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than 4 voice sources, the voice packet dropping rate increases.

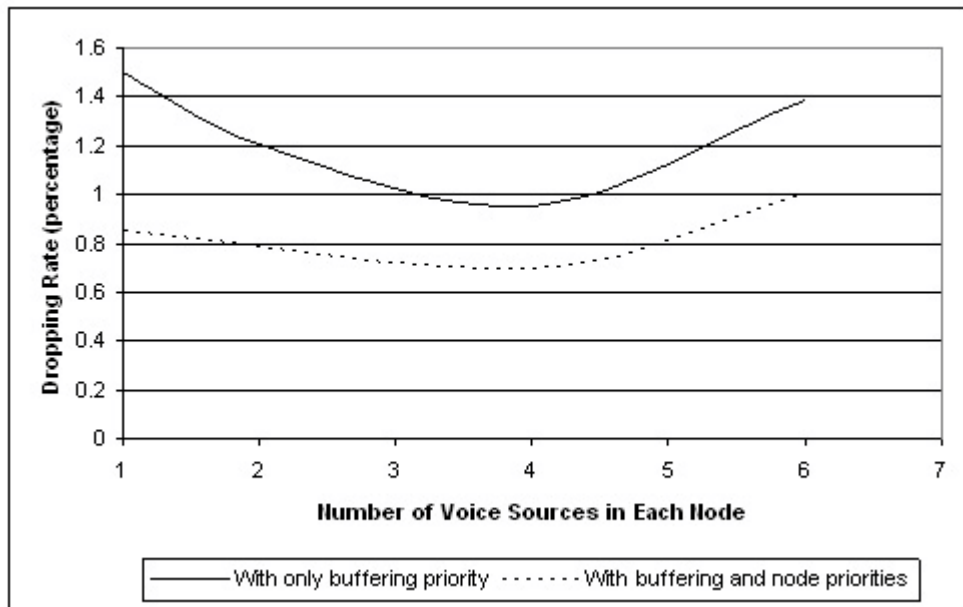Figure 4.14: Voice packet dropping rate because of severe interference
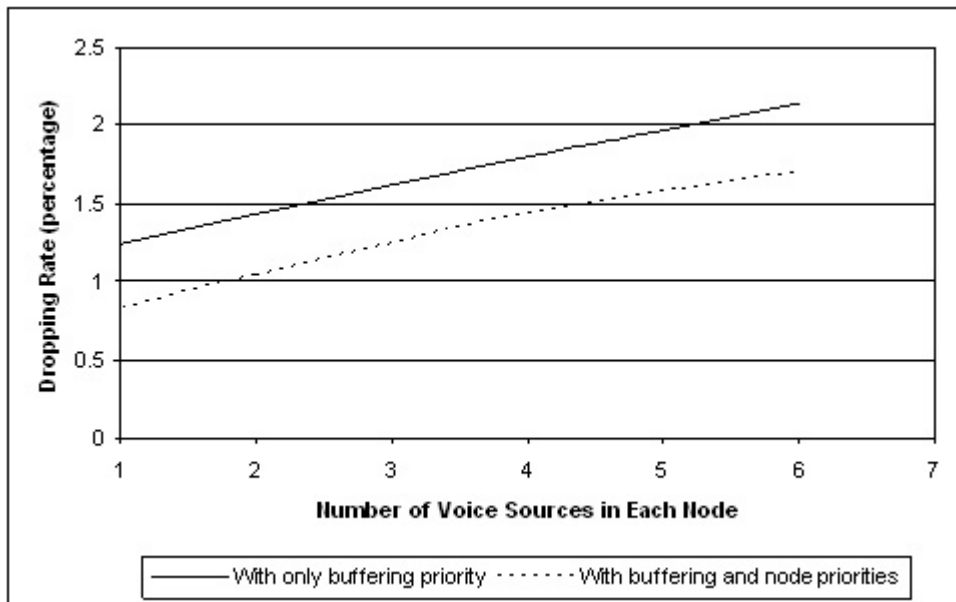
Figure 4.15: Video packet dropping rate because of severe interference

4.4 Summary

In this chapter, we present our simulation results in three sections. First, we compare between results with and without using our buffering priority scheme. For high priority traffic, the results show that the proposed buffering priority scheme can achieve a shorter average packet delay, no packet dropping due to over bounding delay, and more received bits. Then, we discuss the numerical results for high priority traffic with and without using our proposed node priority scheme. The results with a dense traffic load demonstrate the effectiveness of using the proposed scheme such that packet dropping rate is reduced and more bits are received on time. In the last section, we study the effect of having a high voice traffic load at each node with and without using the proposed node priority scheme. The results demonstrate that increasing a high priority traffic load increases its average packet delay, its packet dropping rate due to over bounding delay, and decreases temporarily its packet dropping rate due to severe interference, and then increases it with the traffic load. However, the other classes packet dropping rate due to severe interference increases since the dominant traffic has more chances for transmissions.

Chapter 5

Conclusions and Future Work

5.1   Summary of Contributions

The main contributions of this thesis are listed as follows:

- An effective packet priority scheme within a node is proposed;

- A packet priority scheme among neighbor nodes is proposed for CDMA systems where multiple users can transmit simultaneously;

- It is demonstrated that the priority schemes reduce the average packet delay, the packet dropping rate, and increase the number of the received bits for high priority traffic;

- A busy tone is used to indicate the existence of a node that has a high priority packet to send; thus, the awareness of the interference is not needed since a busy tone does not carry any information and has a sperate band;

- A frame structure is designed in order to reduce signaling overhead as much as possible;

- Adaptive transmission rates are exploited to fully utilize the transmission link;

- QoS for real time traffic is enhanced using the proposed priority schemes.

5.2 Thesis Summary and Concluding Remarks

In this thesis, we considered the problem of supporting real time traffic in CDMA based wireless mesh networks. The work was inspired by the fact that to date, little priority management techniques have been proposed for CDMA systems to support real time traffic for WMNs or wireless ad hoc networks. The previous techniques either do not take into account the interference phenomenon or have only single service.

We address these problems through two priority schemes. We considered the interference phenomenon, multiple services, QoS requirements for each type of traffic, the priority within a node, and the simultaneous transmissions in CDMA. We compare the MAC protocol using the proposed buffering priority scheme with the MAC protocol without using the scheme in terms of average packet delay, packet dropping rate, and the number of received bits. We find that our proposed scheme has much better performance. Moreover, we also compare the MAC protocol using only the buffering priority scheme with the protocol using the buffering and node priority schemes, and we find that adding the node priority gives much better performance. The numerical results show the performance improvement of our proposed schemes.

In addition, some concluding remarks are as follows:

- Classifying traffic according to priority is necessary to support several services with different QoS requirements;

- The node priority technique is needed in a distributed CDMA-based MAC protocol in order to give preference to the high priority traffic, since all users can share the same frequency at the same time;

- Interference is the most significant factor in the CDMA systems; therefore, it has to be considered in the design of priority schemes;

- In a network with fixed topology such as wireless mesh backbone, each node has other neighbors's information in terms of location, sending code, and receiving code;

- A node that has a high priority packet to send has to inform nodes that have a low priority packet to send about its priority status by sending a busy tone. In other words, a node that has a low priority packet to send has to know by sensing a busy tone from its neighboring nodes that have a high priority packet to send;

- Resources such as time slot, number of mini-slot, transmission power, and rate have been allocated to utilize the link as much as possible.

5.3   Future Work

In the course of our work, several avenues can be followed:

Adaptive slot time

Since we have adaptive transmission rates using variable spreading gain, the transmission time of packets is a variable based on the spreading gain. In fixed time slots, the transmission time of a packet at a slot might be shorter than the slot time, thus wasting some resources. As a result, finding a solution to achieve an adaptive slot time depending on the transmission rate will fully utilize the time.

Multi-hop connections

For a more realistic scheme in WMNs, multi-hop connections should be considered. In this case, the priority of packets should consider their hop lengths because packets that have to traverse many hops suffer larger delays. The hop length information is supported by the routing protocol.

Frequency band

Since our system has two frequency bands: information and busy tone, the transmitter and receiver circuits should be able to switch between the bands. Consequently, the design of these circuits might be more complex. In summary, for simplicity and low implementation cost, proposing one band MAC and priority schemes to achieve the same performance as that having a busy tone band is preferred.

Multimedia traffic

With multimedia traffic, the QoS requirements for each type of services are different. Therefore, the priority management technique should take into account these various re-

quirements.

Analytical model

One way to validate the results is to compare the performance evaluation carried out by simulation with the performance evaluation carried out by analytical models. An extension of this work is to provide analytical models.

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