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Link layer solutions for supporting real-time traffic over CDMA wireless mesh networks

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Summary

With recent advances in the development of wireless communication networks, wireless mesh networks (WMNs) have been receiving considerable research interests in recent years. The need to support integrated services and ensure quality of service (QoS) satisfaction for various applications is one of the fundamental challenges for successful WMN deployment. In order to provide differentiated services, medium access control (MAC) should have priority management at the link layer. In code division multiple access (CDMA)-based WMNs, the interference phenomenon and simultaneous transmissions must be considered. We propose two priority schemes for MAC in a distributed CDMA-based WMN, taking into account interference, multimedia services, QoS requirements, and simultaneous transmissions. The first priority scheme is within a node. Each node has an independent queue for each traffic class. According to QoS requirements, the queue that should be served first is determined. The second priority scheme is among neighbor nodes. It is proposed for multiple simultaneous transmissions in the CDMA network. This scheme gives a larger chance of correct transmission to high priority traffic than low priority traffic. In addition, we propose to use adaptive spreading gain and a frame structure to achieve high resource utilization. Simulation results demonstrate that the proposed schemes can achieve effective QoS guarantee. Copyright © 2009 John Wiley & Sons, Ltd.

KEY WORDS: wireless mesh networks; priority management; real-time traffic; MAC; QoS; CDMA

1. Introduction

A wireless mesh network (WMN) consists of a wireless mesh backbone and wireless mesh clients. The backbone is a collection of fixed nodes that work as routers (referred to as mesh routers), connected together *via* multi-hop wireless links without a central controller. In contrast, the mesh clients are mostly mobile nodes that can work as routers as well, but their hardware and software platforms are simpler than those of the mesh routers. Recently WMNs have been a subject of extensive research due to their characteristics, such as low cost, ease of maintenance, self organization, large coverage, ease of expansion, and robustness. These significant characteristics can help to support several applications in the public and private sectors. Irrespec-

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tive of military applications, many promising civilian applications have been presented [1,2], e.g., community networks, metropolitan area networks, broadband home networks, enterprise 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. 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 system bandwidth at all time, with unique codes for receiving and transmitting. CDMA-based networks have been shown to achieve a significant increase in network throughput and capacity, over the distributed coordination function (DCF) mode of IEEE 802.11 standard [3,4], and over networks based on the time division multiple access (TDMA) and frequency division multiple access (FDMA) techniques as in Reference [5].

One of the fundamental challenges in WMNs 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 accuracy. For real-time traffic, delay is another 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 interference, QoS requirements for each service class, and simultaneous transmissions. This paper investigates and develops such priority schemes as an extension to our previous work in Reference [6]. In this work, we study the performance parameters when there is heavy high priority traffic since the traffic load is expected to be high in WMNs. In addition, this work gives more details of the proposed scheme and more comprehensive performance results.

The remainder of this paper is organized as follows. The previous research work is reviewed in Section 2. Our system model is described in Section 3, including the network structure, medium access control (MAC) protocol, frame structure, adaptive transmission rates, and QoS requirements. Two priority schemes for supporting real-time traffic in a CDMA-based WMN are proposed in Section 4. Simulation results are presented in Section 5 to evaluate the proposed priority schemes. Finally, conclusions are drawn in Section 6.

2. Related Work

To the best of our knowledge, little work has been proposed for CDMA-based MAC to support real-time traffic in WMNs. In Reference [7], CDMA has been used to avoid collisions; however, the effects of interference have been ignored. Fantacci and Tarchi [8] propose two priority MAC schemes. First, each node has two queues that follow the first-in first-out (FIFO) approach. One queue is for priority traffic and the other is for non-priority traffic. The second scheme manages which node has the priority to contend for transmission. A node that has a priority packet to send enters the contention directly, whereas if a node has a non-priority packet to send, it has to know whether or not any other node has a priority packet to send. In this situation, the authors suggest that the node should broadcast a special packet to inform others about its priority status, which increases the interference in the network and consumes resources. In Reference [9], as discussed in the next section, a distributed MAC scheme is proposed for CDMA wireless networks. If a node sends its probe in a mini-slot time with a small ID number, the probability for getting the acceptance of transmission is high. 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 disappears. In addition, there are no results demonstrating the performance of the scheme to support various applications with different QoS requirements since the authors only provide a general idea but not a precise scheme for QoS support.

3. System Model

3.1. Network Structure

We consider a wireless mesh backbone with N fixed nodes. Each node can be a traffic source, destination, or router, and has one transceiver. With a focus on the link layer, we consider transmissions over single hops.

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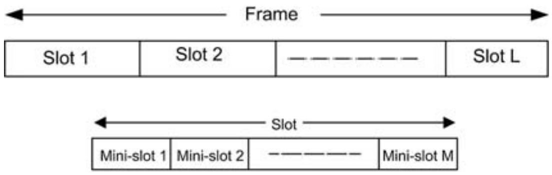

Fig. 1. The frame structure in the MAC protocol.

The network topology and traffic generation are random. Due to the node stationarity, each node has the location information of all the N nodes. The physical layer exploits CDMA technologies for supporting multiple access, where the spread spectrum bandwidth is the total system bandwidth excluding the narrow-band busy-tone channel (to be discussed). Each node has a unique sending code and a unique receiving code that are known by other nodes, thanks to the fixed network topology.

3.2. MAC Protocol

This section briefly describes the MAC protocol that we use as proposed in References [9–11], which is an interference aware distributed MAC protocol for CDMA-based wireless mesh backbone. The protocol is chosen due to its significant advantages such as fully distributed control, low information exchange overhead, high robustness, high scalability, accurate information estimation by receivers, and fine QoS support. As shown in Figure 1, time is partitioned into frames of constant duration. Each frame is divided into L slots, and each slot is divided into M mini-slots. The frame structure facilitates time-division transmissions, where links with low mutual interference can transmit in the same slot while those with large mutual interference should transmit in different slots. The system frequency spectrum is split into two bands: information and busy-tone bands.

As in Figure 2, 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 sending code at each time slot since each node cannot send and receive at the same time. In addition, at mini-slot 1 of each slot, node a measures the total interference it experiences at that 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 mini-slots 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 g_p. The transmit power level is $\xi_p P_{ab}$, where ξ_p is a very small value ($\xi_p \ll 1$) 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 a large spreading gain is to ease the demodulation of a probe at existing receivers (e.g., node c in Figure 2).

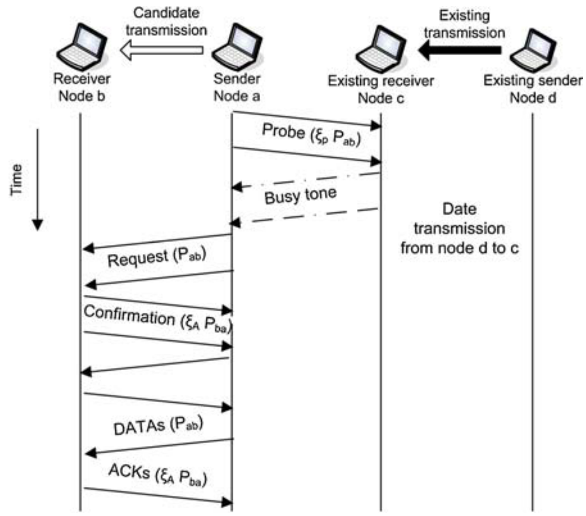

Fig. 2. The procedure of the MAC protocol.

- After the reception of a probe, the nearby existing receivers estimate the increased interference due to the potential new transmission, the transmission from node a to b, and decide either to allow or reject the transmission based on the following equation

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

where I_c is the experienced interference plus noise level measured at node c, 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 required E_b/N_o threshold of the data reception.

If inequality (1) does not hold, node c will send a busy-tone 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 the candidate transmitters (e.g., node a) that their transmission will corrupt other transmissions. Therefore, any node hearing a busy-tone has to select another slot for its transmission, and follows the preceding steps.

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Table I. QoS requirements for different services.

Service	Class	Delay bound	Packet dropping rate bound	E_b/N_o (dB)
Voice	1	150 ms	3%	5.31
Video	2	150 ms	1%	9.32
Data	3	—	0	2.94

- If no busy-tone is detected, at $S_{\min}$ of frame $\ell + 2$, node a (sender) sends a request message with power level P_{ab}. This message contains the information of the interference level that has been measured by node a. After the reception of the request message, node b (receiver) estimates the E_b/N_o of the candidate transmission. If the estimated ratio is above the required E_b/N_o, node b selects a slot (denoted by S_A) to send an ACK message. S_A has the minimal interference level for node a, and 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 $\xi_A P_{ba}$ (where $\xi_A \ll 1$) and a large spreading gain g_A.
- At $S_{\min}$ of the subsequent frames, node a transmits data with power P_{ab}. Meanwhile, node b transmits ACKs with power $\xi_A P_{ba}$ at slot S_A until the data transmission is completed.

3.3. Frame Structure

In the frame structure shown in Figure 1, a properly determined slot time for voice traffic is vital because voice traffic has a long packetization interval, which is the inactive interval between two adjacent packets. Each slot time should be equal to the transmission time of a voice packet in order to avoid wasting resources. As a result, the slot time is given by

$$t_T = \frac{g \cdot S_R \cdot P_I}{C} \tag{2}$$

where g is the spreading gain, C is the channel capacity (in bps), S_R is the source rate (in bps), and P_I is the packetization interval (in s).

3.4. Adaptive Transmission Rates

The usage of adaptive transmission rates is to maximize the link utilization especially under a light traffic load (e.g., there are only a few active links at each slot). At a fixed transmission rate, each link may transmit at a low rate even when the received E_b/N_o is higher than the target value. On the other hand, with adaptive transmission rates, each link transmits at the highest possible rate, under the constraint on the received E_b/N_o. The popular methods to adjust transmission rates in CDMA are to use multiple codes as in Reference [11] and variable spreading gain as in References [8,12].

Here, we use a variable spreading gain in an interval $[g_{\min}, g_{\max}]$ to achieve adaptive transmission rates. The up-to-date received E_b/N_o value is sent *via* the ACK message to the transmitter from the receiver. As a result, the sender determines the new spreading gain as

$$g = \max\left\{ g_{\min}, \min\left[g_{\max}, \left\lfloor \frac{(E_b/N_o)_{\text{new}}}{(1+\beta)\cdot \Gamma_D} \right\rfloor \right] \right\} \tag{3}$$

where $\lfloor x \rfloor$ is the floor function of value x.

3.5. QoS Requirements

The 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 Table I [13–15]. We consider best effort data service which is delay tolerant. Failed data transmissions are retransmitted since the delay requirement is not strict and the packet dropping rate is vital for data traffic.

4. Priority Schemes

4.1. Packet Priority Scheme Within a Node

Each node has three logical queues followed by a packet prioritizer as shown in Figure 3. Each queue has a single type of traffic, and follows the FIFO mechanism. The 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 is given by

$$t_{\text{due}_{ij}} = t_{g_{ij}} + t_{d_j} \tag{4}$$

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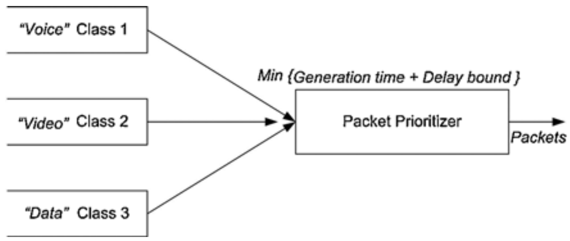

Fig. 3. Packet priority queueing system within a node.

where i is the packet's ID, j is the service type or class ID (voice or video), $t_{\mathrm{due}_{ij}}$ is the due time of packet i with class j, $t_{g_{ij}}$ is the generation time of packet i with class j, and t_{d_j} is the delay bound for class j. Since the delay is not urgent in class 3, we assume j is either 1 or 2.

4.2. Packet Priority Scheme Among Neighbor Nodes

After passing the packet prioritizer, a packet of either high priority or low priority is being selected. Consider that packets of class 1 and class 2 have high priority, while packets of class 3 have low priority. Because of the simultaneous transmissions in CDMA, four situations may occur:

(1) both the sender and its neighbors have a high priority packet to send;
(2) both 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.

In cases (1) and (2), packets from the sender and its neighbors have the same priority. In contrast, packets have different priorities in cases (3) and (4). Thereby, any node that has a low priority packet to send has to know whether or not its neighbor nodes have a high priority packet to send. The MAC procedure and the proposed priority management technique in the CDMA MAC protocol operate as follows.

4.2.1. MAC procedure for a node with a high priority packet to send

In contrast to Reference [9], where a node that has a high priority packet sends its probe at a mini-slot with a small ID, in our scheme, we divide the mini-slots, excluding mini-slots 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 is to send and detect a busy-tone (as to be discussed).

When node a has a high priority packet to send to node b as shown in Figure 2, node a at mini-slot 2 sends a busy-tone that should be heard by all of its two-hop neighbors in order to let all the nodes (which may potentially corrupt its receiver's reception and 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 cannot send and receive at the same time. As a consequence, in the second and third cases, the node cannot sense a busy-tone because it is either sending a busy-tone at the same time slot or receiving data from the information band.

With the busy-tone in mini-slot 2, mini-slots 3 to M are separated into two parts as shown in Figure 4. If M is an even number (M_{even}), from 3 to $(M+2)/2$ mini-slots are reserved for a probe sent by nodes that have high priority traffic, and from $(\frac{M+2}{2}+1)$ to M are reserved for a probe sent by nodes that have low priority traffic. If M is an odd number (M_{odd}), the high priority traffic has an extra mini-slot. That is, mini-slots from 3 to $(M+3)/2$ are reserved for a probe sent by nodes that have high priority packets, and from $(\frac{M+3}{2}+1)$ to M are reserved for a probe sent by nodes that have low priority packets.

After detecting a probe, an existing receiver determines whether or not the new transmission could corrupt its reception. If the packet reception of an existing

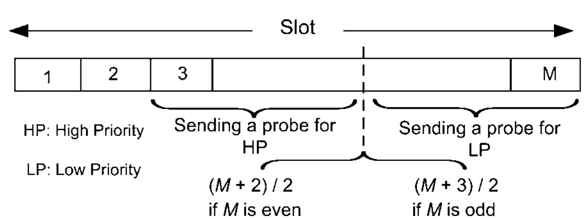

Fig. 4. Slot structure with service differentiation.

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receiver could be corrupted, the receiver sends a busy-tone at mini-slot $m + 1$ (when m < M), where m is the mini-slot of sending a probe. If no busy-tone is sensed, the candidate sender continues the MAC procedure by sending a request message and data in the subsequent frames as explained in Section 3.2.

4.2.2. MAC procedure for a node with low priority packets to send

When a node has a low priority packet to send, by sensing the channel at mini-slot 2, it knows whether or not among its neighbors a node has a high priority packet to send. If no busy-tone is 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 $m \in \{3, \ldots, M\}$. On the other hand, if a busy-tone is 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 $m \in \{\frac{M_{\text{even}}+2}{2} + 1, \ldots, M_{\text{even}}\}$ or $\{\frac{M_{\text{odd}}+3}{2} + 1, \ldots, M_{\text{odd}}\}$ in order to give advantage to the high priority traffic. The preceding steps are similar to what have been discussed in Subsection 3.2.

5. Performance Evaluation

To evaluate the performance of the proposed schemes, computer simulations are carried out using Matlab. At each node with a RAKE receiver, signal energy from multiple propagation paths can be collected [10]. As a result, we consider only distance dependent propagation attenuation without fading. Also, background noise is neglected as multiple access interference is dominant in CDMA networks [10]. The path loss exponent is $\alpha = 2.4$ [10,11]. We consider three traffic classes: data, voice, and video. The call arrivals at each node follow a Poisson process. All the traffic sessions are single-hop sessions.

The voice traffic follows the G.711 codec [16]. It 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 [17]. The CBR H. 263 codec is used for our video traffic [16]. It is designed for low bit rate communications such as radio communication links. 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 [18].

Table II. Simulation parameters.

L	3	M	16
$g_{\max}$	16 [15]	g_A & g_p	1600
$g_{\min}$	1	slot time	0.512 ms
B_c	50 Mcps	ξ_p & ξ_A	0.01
Voice rate	80 kbps	Video rate	160 kbps

We consider 55 source nodes and 55 destination nodes, among which 50 source nodes have one traffic session for each class, while 5 source nodes only have one data traffic session in order to study explicitly the impacts of data packets on real-time traffic. The senders and receivers are randomly located in a 10 km × 10 km area, and the transmission range is 500 m. The rest of our simulation parameters are specified in Table II.

Since we have a variable spreading gain, the required minimum and maximum transmission rate can be determined as

$$B_{b_{\min}} = \frac{B_c}{g_{\max}}, \quad B_{b_{\max}} = \frac{B_c}{g_{\min}} \tag{5}$$

where B_c is the chip rate.

As discussed in Subsection 3.2, 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 (or busy-tone) detection time, which depends on the communication hardware. A busy-tone signal detection is studied in Reference [19]. We assume that the detection time of a probe (or a busy-tone) is 32 μs.

A large number of mini-slots over a slot time is required. If the number of the mini-slots is very limited, the probability that more than one probe is transmitted at the same mini-slot is high, leading to more rejected transmissions. Here, we consider the minimum required length of a mini-slot in order to have the largest possible number of mini-slots for the given slot time.

For real-time traffic, we study the performance of our schemes in terms of average packet delay and packet dropping rate. We discuss our results in three parts. First, we compare the results between using the packet priority scheme within a node (called buffering priority) and without using it. Second, we discuss the performance improvement when we implement the proposed node priority scheme. Finally, we show the effect of having very heavy real time traffic.

5.1. Performance With Buffering Priority

In this subsection, we present the advantage of using our proposed buffering priority scheme. The total

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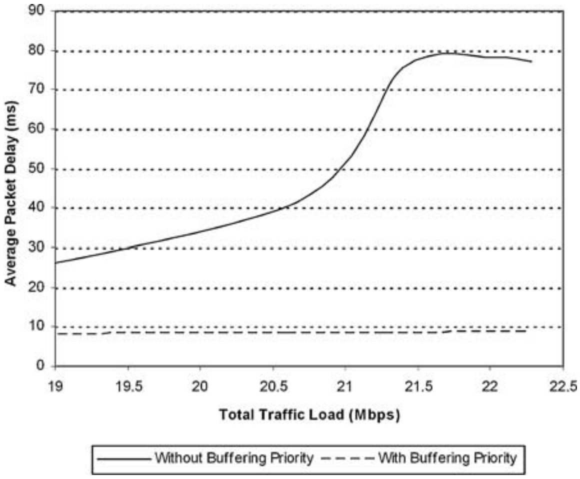

Fig. 5. Average voice packet delay *versus* total traffic load.

traffic load is defined as the total number of data, voice, and video transmitted bits (received and dropped bits) in the system over the simulation time, which is terminated when 25 000 voice packets are transmitted. Consider that each node transmits voice, video, and data as shown in Figure 3. The impact of increasing the traffic load on the average real-time packet delay is presented in Figures 5 and 6. The traffic load increases with increasing the arrival rate of data traffic, which varies from 80 to 600 kbps. Obviously in the figures, with buffering priority, we achieve a much shorter delay. With no buffering priority, the delay increases with the increase of traffic load, then remains around 77 ms; while with the buffering priority, the delay is around 9.5 ms regardless of traffic load. When the maximum spreading gain is used, the average delay reaches the maximum value, as observed in the simulation.

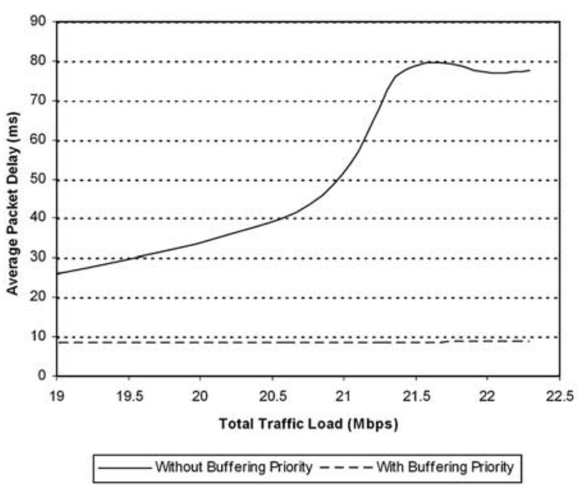

Fig. 6. Average video packet delay *versus* total traffic load.

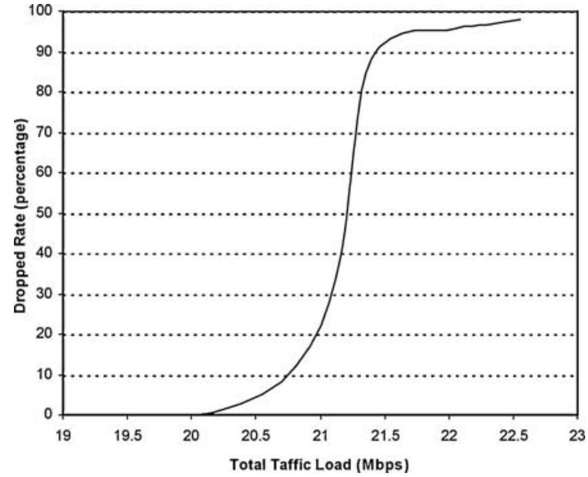

Fig. 7. Voice packet dropping rate due to over bounding delay without buffering priority.

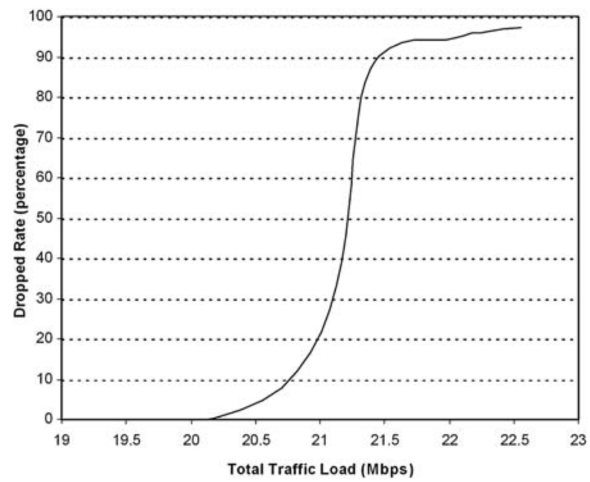

Fig. 8. Video packet dropping rate due to over bounding delay without buffering priority.

Real-time packets that experience a delay larger than the delay bound (150 ms) are dropped. It is clear from Figures 7 and 8 that, with the total traffic load higher than 21.5 Mbps, the packet dropping rates are above 90% with no buffering priority, while it is almost zero with the buffering priority scheme.

5.2. Performance of The Node Priority Scheme

In the following, we present and discuss the results when using our second proposed priority scheme, in addition to the proposed buffering priority scheme. Dropping rate of high priority packets due to high traffic load is the performance metric. Because a high

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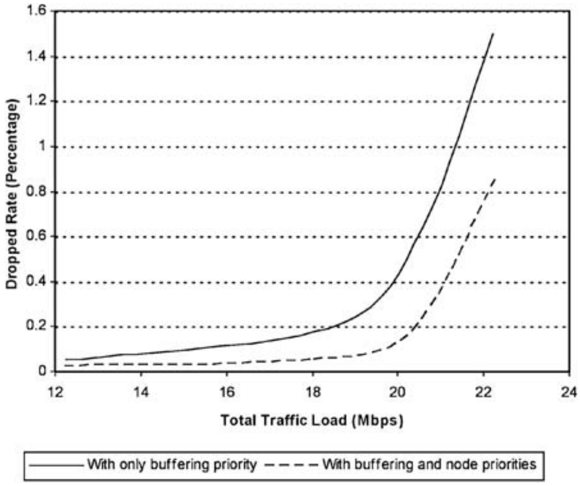

Fig. 9. Voice packet dropping rate due to high traffic load.

priority traffic load increases the interference among neighbors, the packet dropping rate increases with the increase of traffic load. As in Figures 9 and 10, with a light traffic load, the packet dropping rates with the node priority scheme are close to those without using the scheme. However, with a high traffic load, as nodes experience more interference, our proposed node priority scheme results in dropping less high priority packets than the scheme with only buffering priority. As a result, the advantage of the node priority scheme is significant when we have a high traffic load. As we know, the traffic is expected to be high in WMNs. On the other hand, the node priority scheme does not incur extra cost as compared with those without using the scheme.

5.3. The Effects of Increasing High Priority Traffic Load

In a wireless mesh backbone, the traffic load is usually high. Consequently, we should study the performance measures 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 while other traffic types have one source. In Subsections 5.1 and 5.2, 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 5, 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 11, the average packet delay starts from 9.5 ms and increases with increasing the number of voice sources, and it remains unchanged after adding more than four voice sources at each node. Without the buffering priority scheme, the delay is fixed around 77 ms because the transmission rate is fixed. With the buffering priority scheme, when the number of voice sources in each node is equal to or larger than four, 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 7, 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

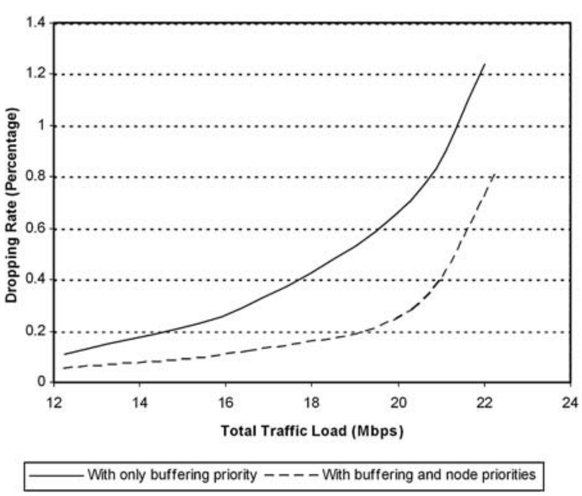

Fig. 10. Video packet dropping rate due to high traffic load.

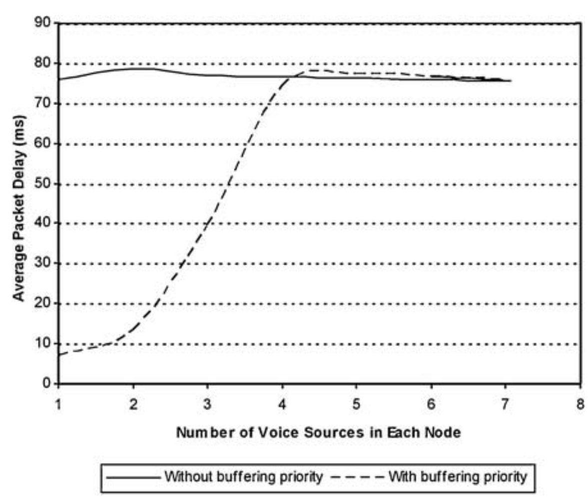

Fig. 11. Average voice packet delay.

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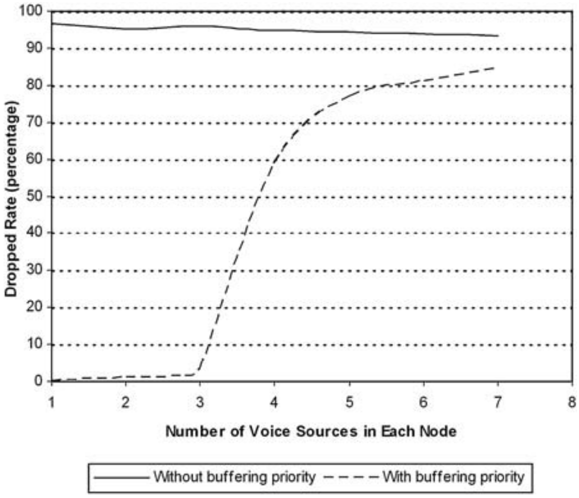

Fig. 12. Voice packet dropping rate due to over bounding delay.

proposed buffering priority scheme, while it is nearly 0% with the use of our scheme. As in Figure 12, 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 increase of the number of voice sources at each node. It is important to note that, after adding four voice sources, the average packet delay with and without using the buffering priority scheme is similar as in Figure 11. However, the scheme still gives better performance as shown in Figure 12, where the voice packet dropping rate using the buffering priority scheme is lower even if there are more than six voice sources at each node.

In Figure 13, the voice packet dropping rate due to a high traffic load is presented. As shown in Figure 9, 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. The same values are observed in Figure 13 with the corresponding number of voice source being one. Both curves in Figure 13 decrease with the increasing number of voice sources until four. On the other hand, the video packet dropping rate due to high traffic load increases with the number of voice sources as shown in Figure 14. This is because the voice packets dominate the transmission time 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

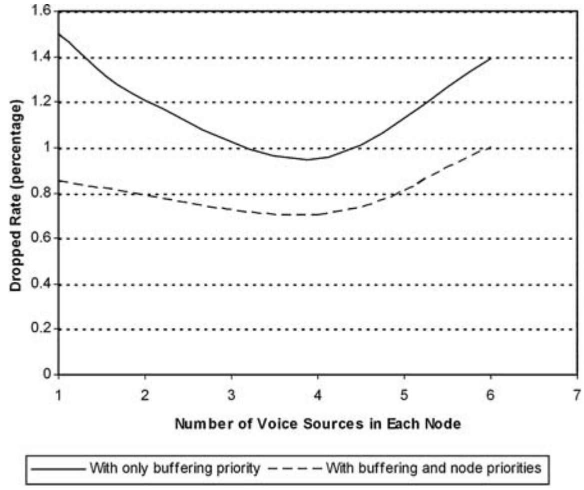

Fig. 13. Voice packet dropping rate due to high traffic load.

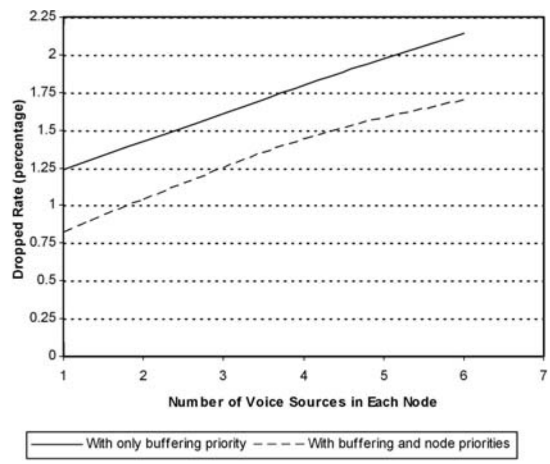

Fig. 14. Video packet dropping rate due to high traffic load.

packet generation time and the delay bound, the video packet has the priority to transmit. Adding more voice sources, the inter-arrival time of voice packets will be shorter. As a consequence, it is very likely 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 than four voice sources, the voice packet dropping rate increases.

Generally, the major trade-off of any priority scheme is a decrease of the chances of transmitting low priority traffic for fast transmission of high priority traffic, especially if there is a heavy high priority traffic load. Figure 15 shows the total received data bits from all the data users over the simulation time with the increasing voice traffic load, while the simulation time is terminated when 25 000 voice packets are transmitted. It is

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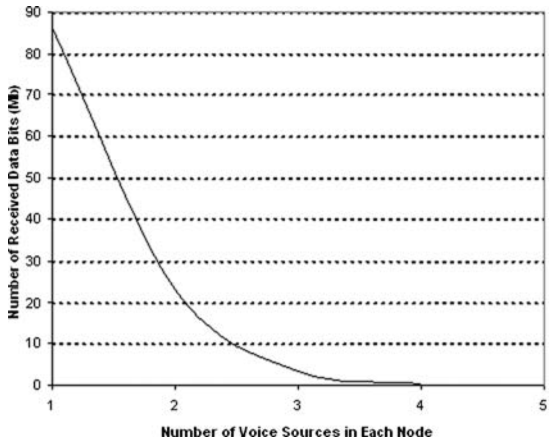

Fig. 15. Number of received data bits.

obvious from the figure that less data bits will be received when adding more voice sources.

6. Conclusions

This work was motivated by the fact that, to date, few priority management techniques have been proposed to support real-time traffic in CDMA-based WMNs. The previous techniques either do not take into account the interference phenomenon or have only a single service. We propose the buffering priority and node priority schemes for multimedia services in an interference limited CDMA network, taking into account QoS requirements for each class, the priority within a node, and the simultaneous transmissions in CDMA. We compare the performance of the MAC protocol in terms of average packet delay and packet dropping rate with and without using the proposed priority schemes. Simulation results demonstrate the significant performance improvement of our proposed schemes.

For future work, we will extend our current work to a multi-hop scenario. In this case, MAC should be jointly designed with routing and call admission control in order to achieve end-to-end QoS support.

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