A Cellular Wireless Local Area Network with QoS Guarantees for Heterogeneous Traffic

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Abstract

A wireless local area network (WLAN) or a cell *with quality-of-service (QoS) guarantees for various types of traffic is considered. A centralized (i.e., star) network topology is adopted as the topology of a cell which consists of a base station and a number of mobile clients. Dynamic Time Division Duplexed (TDD) transmission is used, and hence, the same frequency channel is time-shared for downlink and uplink transmissions under the dynamic control of the base station. We divide traffic into two classes: class I (real-time) and II (non-real-time). Whenever there is no eligible class-I traffic, class-II traffic is transmitted, while uplink transmissions are controlled with a reservation scheme. Class-I traffic is handled with the framing strategy [1] combined with the admission test for adding new class-I connections. Finally, we present the performance (average delay and throughput) evaluation of the reservation scheme for class-II traffic using both analytical calculations and simulations.*

1 Introduction

Wireless LANs (WLANs) are emerging as an attractive alternative or complementary to wired LANs [2,3], because they allow us to set up and reconfigure LANs easily without incurring the cost of wiring. They are generally characterized as high-speed wireless systems which cover relatively small areas compared to other wireless systems such as cellular, PCS, and mobile data radio systems. It is expected that they will meet the growing demand that mobile clients should have access to the existing high-speed wired networks. As the interest in broadband multimedia communications involving digital audio and video grows, a number of researchers have been looking into ways of providing QoS guarantees in wired point-to-point WANs [1,4,5] and LANs [6].

In this paper, we consider how to provide QoS guarantees for heterogeneous traffic on a WLAN. The following three types of QoS are considered: (1) maximum packet delivery delay; (2) transmission throughput; and (3) packet loss tolerance. We categorize various traffic into two classes: (1) class-I real-time traffic like real-time data, video, and voice which requires bounded delay and guaranteed throughput, but is usually loss-tolerable; and (2) class-II non-real-time traffic like the conventional data traffic which requires zero loss, but requires no bounded delay nor guaranteed throughput. Class II is also divided further into two subclasses: (i) delay-sensitive class II-A like FTP and remote log-in; and (ii) delay-tolerable class II-B like paging and e-mail. Class II-A has priority over class II-B.

We adopt a reservation scheme which is similar to the reservation ALOHA [7] or PRMA [8] for uplink class-II traffic transmissions. (The reservation scheme proposed in this paper differs from the previous work, but appears similar in the sense of adopting collision-based reservation schemes.) This reservation scheme is a promising multiple access protocol for class-II traffic, as it provides higher throughput and smaller average delay than other collision-based random access protocols like ALOHA as in [9]. Basically, class-II traffic is transmitted when there is no class-I traffic to be transmitted since it has a lower priority than class I.

The framing strategy [1], which was originally proposed as a framework for congestion management in integrated service packet networks, is used with some modifications. The framing strategy is composed of a smoothness traffic model and stop-and-go queueing, and provides both packet-delay bound and guaranteed transmission throughput. Each connection of class I should follow a smoothness traffic model, and each new connection needs to pass the *a priori* admission test with a traffic model, implying that the framing strategy reserve slots for class-I connections according to their traffic model. To implement the framing strategy, it is necessary to schedule the uplink and downlink transmissions.

The paper is organized as follows. Section 2 shows the specifications and assumptions of the WLAN under consideration. Section 3 describes the proposed protocol, including the reservation scheme for class-II traffic. Section 4 considers the framing strategy with QoS guarantees for class I and defines the admission test for establishing a new connection. Section 5 presents the analysis and simulation results of the performance of the reservation scheme for class II. Finally, the paper concludes with Section 6.

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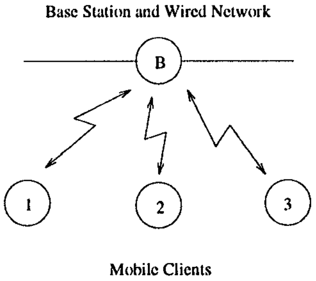


Figure 1: A centralized wireless network with a base station.

2 Specifications and Assumptions of System

As shown in Fig. 1, the WLAN under consideration consists of a base station (denoted by B) and several mobile clients (denoted by numbers) forming a star network, called a *cell.* The base station is connected to a wired high-speed network. In this topology, the uplink (mobile-to-base) is not a broadcast channel while the downlink (base-to-mobile) is. Hence, mobile clients are not able to listen directly to other mobiles using the same frequency channel. This assumed situation can occur in real world due to the existence of *hidden* terminals [3].

The entire wireless network may consist of several cells, and mobile clients may move from one cell to another. However, we will in this paper focus on the communication within a single cell, hence the uplink and downlink transmissions only. Since wireless links usually have much less bandwidth than the wired counterpart, the former might become a bottleneck. Since the base stations are connected to a wired network, the other communicating party of each mobile in a cell can be a node in the wired network, or a mobile in another cell, or another mobile in the same cell. In any case, the wireless link in the cell is considered as the end-most link (for downlink) or the front-most link (for uplink) of the entire multi-hop communication. Note that the downlink traffic comes from the wired network or mobiles in the same cell, and the uplink traffic is generated by mobiles.

Dynamic Time Division Duplexed (TDD) transmission is used in the network, and hence, the base station multiplexes the uplink and downlink packet transmissions dynamically according to the traffic load over a frequency channel. We could instead use frequency division duplexed (FDD) transmission, in which two different frequency channels are allocated for uplink and downlink transmissions, or static TDD in which a portion (usually a half) of each time frame is allocated for uplink and the other part for downlink. FDD is the common duplexing mode in cellular systems, and static TDD was adopted in the DECT system [2]. But, dynamic TDD allows for more efficient link utilization in the case of unbalanced uplink and downlink traffic, e.g., non-interactive data transmissions, as shown in [9]. We assume all packets, like ATM cells, to be of the same fixed size. Throughout the remainder of this paper, we will ignore the packet-propagation delay, because it is usually small relative to the other delay components like queuing and transmission delays in the cell.[1]

Since the wireless channel is inherently unreliable (due to noises, interferences, and multipath fadings), we need a special means to ensure the error-free delivery of packets through each wireless link. Usually, a combined channel coding and diversity scheme [2] is used to meet this need. To handle various types of traffic in our system, we can apply error-handling schemes adaptively. We adopt an ARQ (Automatic Retransmission reQuest) scheme for class II to ensure virtually error-free transmission of data. But, it is ruled out for class I because of its difficulty in making delivery-delay guarantees. We use an FEC (Forward Error Correction) scheme for class I, instead.[2] To this end, the receiver is equipped with a dual-mode channel decoder: a received packet is decoded by an error-correction decoder (if class I) or an error-detection decoder (if class II). The dual-mode receiver is expected to work well since using a channel code, the decoder can detect more errors than those correctable. We will not consider error-combating techniques any more since they are not within the primary scope of this paper, but we assume that a packet is received correctly unless that packet collides with concurrent packets.

3 Protocol Description

When a mobile wants to send a packet, regardless whether it is destined for another client in the same cell or for a remote cell, it must send the packet to its base station first, which will then forward the packet to the final destination, sometimes via other base stations. We adopt two different strategies for class I and II. First, class-I traffic is transmitted via connections, i.e., for each of class-I (downlink or uplink) connections, a finite number of slots are reserved to meet the required QoS. Each connection (between the base station and a mobile) is identified by: (1) for which client and (2) for downlink or uplink. For the QoS provision, class-I traffic has priority over class-II traffic, where the transmission is controlled by the framing strategy (to be discussed in the next section). Class-II traffic doesn't need the concept of connection, but if there is a pending message (which consists of a number of packets), it will be transmitted when there are available slots, i.e., when no class-I traffic is being transmitted over the link. For uplink class-II traffic, a request of slot reservation for transmission is made for each message.

A *slot* and a *control mini-slot* alternate continuously as shown in Fig. 2. In a slot of duration T_s, a packet is transmitted. By dynamic TDD transmission, each slot can be used for either downlink or uplink transmission under the control of the base station. A control mini-slot of duration T_{ms} is used to transmit a control packet. Control packets are used by the base station

[1] A cell in this paper refers to a *micro-cell,* which has coverage of the order of a few hundred meters, or a *pico-cell,* which covers small indoor areas [2].

[2] Although combined FEC and diversity seems to be the only way for error-protection of class I, it is extremely difficult to guarantee the virtually error-free transmission of packets of these classes over the wireless link due to the error-correcting capability limit of the underlying FEC scheme. So the proposed scheme here might not be applicable for reliability-critical real-time data traffic of class I.

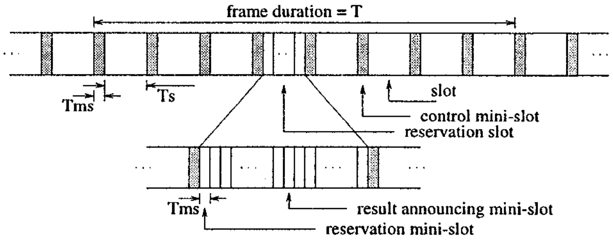
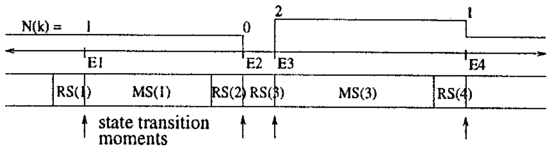


Figure 2: Dividing the time-axis into mini-slots, slots, and reservation slots. A frame includes a number of slots.



RS(k) = the k-th reservation slot
MS(k) = message transmission after the k-the reservation slot
N(k) = # of downlink messages or uplink requests in the base station service queue at the end of k-th reservation slot

Figure 3: Timing diagram of class-II communications during the absence of class-I traffic.

to announce to the mobiles information of the next slot (1) for downlink or uplink, (2) for class I or II, and (3) for which client. These regularly alternating slots and mini-slots are expected to help each mobile synchronize to the global transmission system. We will henceforth use T_{ms} as a basic time unit. Assume that a slot duration is an even number multiple of a mini-slot duration, i.e., $K = T_s/T_{ms}$ is an even number.

There exist slots, called *reservation slots*, which are used for requesting an uplink class-I connection establishment or an uplink class-II message transmission. A reservation slot consists of K mini-slots of duration T_{ms}. It is divided into two parts: (1) the first half is a set of $K/2$ *reservation mini-slots* used by mobiles to request uplink transmissions; and (2) the second half is a set of $K/2$ *result-announcing mini-slots* for each of the corresponding previous reservation mini-slots. The reservation mini-slots are accessed by a slotted ALOHA-like random access protocol: when a reservation slot is issued by the base station, each mobile with a pending request chooses one of $K/2$ mini-slots randomly, and then sends the request in that chosen mini-slot with the traffic information. The result of each of $K/2$ mini-slots can be success (of which mobile[3]), or collision, or empty/unused. Using each of the next $K/2$ downlink mini-slots, the result of the corresponding reservation mini-slot is announced.

If a reservation slot contains only "collided" reservation mini-slots, the base station will issue reservation slots consecutively until a successful reservation mini-slot appears in a reservation slot. Using this policy, the base station will obtain at least one successful reservation request for mobiles who want to make a slot reservation. A mobile whose reservation request collided with others will retransmit the request again in the subsequent reservation slots with the probability q_r until it is successful. (The retransmission probability q_r can be determined adaptively according to the results of all of $K/2$ reservation mini-slots.) If a reservation request is successful, the base station will be informed that the mobile who made the request wants (1) to send a pending class-II message, or (2) to establish a class-I connection.

For dynamic TDD transmission, the base station needs to multiplex between downlink and uplink transmissions. The base station does not know if a mobile has a pending message without receiving a reservation request. Basically, a reservation slot is issued after completing the transmission of a (downlink or uplink) class-II message as shown in Fig. 3, where only class-II traffic (without subclasses) exists. Two first-in-first-out (FIFO) base station class-II service queues are implemented for two subclasses of class II, in which both the uplink requests (from mobiles) and the downlink messages are queued together. The contents of the queues are updated at the end of every reservation slot (marked with arrows in Fig. 3): at the end of the k-th reservation slot, E_k, all the downlink messages which arrived at the base station between E_{k-1} and E_k, and all the uplink requests which were successfully received during the k-th reservation slot are queued in a random order. By this random queueing policy, the uplink transmission achieves fairness since the uplink reservation requests might suffer excessive delays compared to the downlink transmission due to the collision-based reservation request access. The second queue for class II-B can be served whenever the first queue for class II-A is empty. When both queues are empty, the base station issues a reservation slot for each available slot.

With the scheme explained above, the maximum achievable throughput of class II might be very low depending on the average message length, i.e., the smaller the average message length, the smaller maximum achievable throughput. To solve this problem, we assign the Minimum Next Reservation Slot Length (*MNRSL*) defined as the minimum number of slots between two consecutive reservation slots. Assume that a (downlink or uplink) message transmission was completed after a reservation slot. If less than *MNRSL* packets were transmitted since the reservation slot, the base station will serve more messages until the entire transmitted packets exceed *MNRSL*. Thus, the maximum achievable throughput is guaranteed to be $\geq (MNRSL \cdot T_s)/((MNRSL + 1)(T_s + T_{ms}))$.

4 Framing Strategy for QoS Provision

In this section, we describe the framing strategy to guarantee QoS for class-I traffic. The time axis is divided into frames, each of which is composed of a finite number of slots (and so mini-slots) as shown in Fig. 2. If there are N slots and mini-slots in a frame of time duration T, then $T = N \cdot (T_s + T_{ms})$.

4.1 Traffic Model

For each connection i of class-I traffic, we adopt the (M_i, T_i)-smooth model, i.e., during each frame of length T_i, no more than M_i packets arrive (or be generated) for connection i. If connection i is for uplink transmissions, the mobile regulates its uplink traffic to follow the (M_i, T_i)-smooth model using the packet admission

[3] Due to the *capture effects* [3], a reservation request can be transmitted successfully even in the presence of concurrent reservation requests from other mobiles.

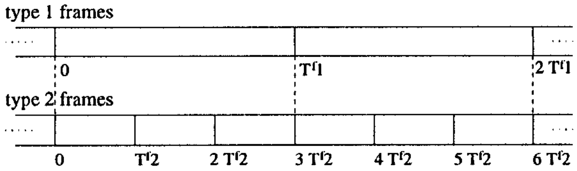


Figure 4: Two frames with duration $T^f_1 = 3T^f_2$.

policy, by which any packet which violates the smoothness is assumed not to be generated until the beginning of the next frame. In the wired part of the network, we assume the existence of a traffic regulator like the (M_i, T_i)-smooth admission policy or leaky-bucket [4] in the source end nodes and a flow/congestion control or packet scheduling scheme such as the framing strategy and Weighted Fair Queueing (or PGPS) [4] in intermediate nodes. Thus, the traffic arriving at the base station from the wired network will have the smoothness property, which can then be converted to the (M_j, T_j)-smooth model. Moreover, the downlink traffic from a mobile within the cell (in case of the intra-cell communications) will also have the the smoothness property (as explained later). So, it is possible to adopt the (M_j, T_j)-smooth model for a downlink connection j as well.

Suppose there are G frame sizes, $T^f_1, T^f_2, \cdots, T^f_G$, and each frame size is a multiple of smaller frame sizes, i.e.,

$$T^f_g = I_g \cdot T^f_{g+1}, \quad g = 1, 2, \cdots, G-1, \qquad (1)$$

for some integer I_g. For all g,

$$T^f_g = K_g \cdot (T_s + T_{ms}), \qquad (2)$$

for some integer K_g, i.e., there are a finite number of slots in each frame. Each frame of duration T^f_g is called a *type-g frame*. For each connection i, $T_i = T^f_g$ for some g, and the connection is called the *type-g connection*. Fig. 4 shows the case of $G = 2$ and $T_1 = 3T_2$. Note that all frames are incorporated into a single frequency channel. As shown in the next subsection, the packets in a type-g connection will be guaranteed to have a delivery delay bound $2T^f_g$, implying the existence of G delivery delay bounds.

4.2 Stop-and-Go Queueing

<u>Downlink Transmissions:</u> The transmission from the base station to mobile clients can be viewed as taking place over a single link in a wired network since it is broadcast-type communication. Stop-and-go queueing is used for downlink transmissions with the following rules.

R1. A downlink packet of a type-g connection that has arrived at the base station during a frame does not become *eligible* until the beginning of the next frame. During a frame, the set of eligible packets of the corresponding type are transmitted.

R2. Any eligible packet of a type-g connection, $g = 2, 3, \cdots, G$, has priority over eligible packets of type $g' < g$.

R3. The wireless link should not be left idle whenever there are eligible packets in the queues.

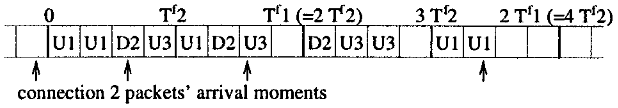


Figure 5: An example of stop-and-go queueing.

To implement the above rule, the base station is equipped with G FIFO queues.

<u>Uplink Transmissions:</u> During each type-g frame, the base station issues (via mini-slots) up to $\sum_{\{T_i = T^f_g\}} M_i$ slots for type-g uplink connections with the same priority given in **R2**. Within the same type, uplink connections have priority over downlink connections, and so in each frame, uplink slots are issued first, and then eligible downlink packets are transmitted. For uplink connection i, the base station will issue up to M_i uplink slots or until the corresponding mobile sends a packet marked as the last packet. Then, the next connection is served. When a slot is issued for connection i in a frame, the connection i mobile transmits a packet which was generated during the previous frame via the issued slot, and marks the last packet arrived in the previous frame. Now, the traffic arrived at the base station from each mobile client also has the (M_i, T_i)-smoothness.

Using the above transmission rules, all of the connection-i packets, which conform to the smoothness, will be transmitted until the end of the next type-g frame (when $T_i = T^f_g$), and thus are guaranteed to be transmitted within a delay of $2T_i$. For each connection i, up to M_i packets can be transmitted over a frame of duration T_i, and so, it is guaranteed to have the throughput of $M_i \cdot T_s / T_i$.

Example 1 *We ignore the mini-slots assuming that $T_{ms} \ll T_s$. Suppose there are three class-I connections, where connection 1 and 3 are for the uplink and connection 2 is for the downlink with the smoothness parameters $\{(M_i, T_i)\} = \{(2, 4T_s), (1, 4T_s), (2, 8T_s)\}$. So, there are two frame sizes, i.e., $G = 2$, where $T^f_1 = 8T_s$ and $T^f_2 = 4T_s$. In Fig. 5, an example of stop-and-go queueing for this set of connections is shown. The slots for connection 1, 2, and 3 are marked with $U1$, $D2$, and $U3$, respectively. The packet arrivals of connection 2 are also marked by arrows. One can see that connection 1 has priority over connection 2 since connection 1 is uplink. Connection 3 has the lowest priority. Through the blank slots in the figure, class-II traffic, if any, will be transmitted by the rules presented in Section 3.* □

4.3 Admission Tests

If a new connection is to be added, it has to pass the following simple admission test depending on the frame-size constraints. Downlink connections requests come from the wired network (or from a mobile originating a connection within the same cell) with the traffic characteristics (M_i, T_i), while uplink connection requests come from mobile clients via a reservation slot. The admission test is given as

$$\sum_{g=1}^{G} M^f_g \cdot (T_s + T_{ms}) / T^f_g \leq 1, \qquad (3)$$

where $M_g^f = \sum_{\{T_i=T_g^f\}} M_i$ is the number of the reserved slots within a type-g frame, in which the existing connections and the newly-requested connection are included. If the results of the above test is positive, it is possible to provide the required QoS to the new connection without compromising the existing connections' guarantees, and so, the base station starts to serve the new connection beginning at the next frame. The basic idea of the admission test is the total reserved throughput for class-I connections plus the redundancy of mini-slots should be less than, or equal to, one. The readers are referred to [10] for a formal proof. Note that in Example 1, the summation in Eq. (3) is exactly one, implying that all of the slots be reserved for the three class-I connections. It might sometimes be desirable to set aside a certain portion of throughput for class-II traffic (say S). In such a case, Eq. (3) should be modified by replacing one with $1 - S(T_s + T_{ms})/T_s$.

5 Performance Analysis of Class-II Communications

This section analyzes the performance of the reservation scheme for class-II communications, where smaller average delay and larger throughput are desirable. Here, all traffic belongs to class II, where class II-A and II-B are not differentiated for simplicity, and hence, only one base station service queue is implemented. As described in Section 3, when there exists class-I traffic, the reservation scheme considered here is activated whenever there are no eligible class-I packets to be transmitted.

For uplink accesses using the reservation scheme, we use the model of K_u clients with the following assumptions.

A1. Message length has a geometric distribution with parameter p_l measured in the number of packets.

A2. Downlink messages arrive from the wired network according to a Poisson process with the overall arrival rate λ_d (messages/mini-slot).

A3. Messages are generated at each of the K_u clients according to independent Poisson processes with the generation rate λ_u/K_u (messages/mini-slot).

A4. Each collided request must be retransmitted in a later reservation mini-slot until the request is successfully received.

A5. Closed-loop behavior of clients, i.e., backlogged clients will discard newly-generated messages until the successful transmission of the request.

A client is said to be *backlogged* when it was notified by the base station to have a collided request and hence must retransmit it. Note that from **A1**, we consider the inter-cell communications only since downlink messages are assumed to arrive exclusively from the wired network. We also make the following simplifications of the scheme to facilitate the derivation.

S1. Even if a reservation slot contains only collided reservation mini-slots, the base station will not issue another reservation slot.

S2. *MNRSL* will be set to 1.

S3. The retransmission probability q_r will be assigned a fixed constant.

By **S1** and **S2**, a message transmission and a reservation slot will alternate continuously when the base station service queue is not empty.

Markov Chain Modelling: The pair $(M(k), N(k))$ is modelled by a 2-dimensional Markov chain, where $M(k)$ is the number of the backlogged clients requesting uplink message transmission and $N(k)$ is the number of the downlink messages or uplink requests in the base station service queue at the end of the k-th reservation slot. Fig. 3 shows a timing diagram of class-II communications with the state transition moments and the change of state $N(k)$. Each of the $M(k)$ backlogged clients will transmit a request in the $(k+1)$-th reservation slot, independently of each other, with probability q_r. Each of the $K_u - M(k)$ other clients will transmit a request in the $(k+1)$-th reservation slot if one (or more) such messages are generated since the last reservation slot. $T_{nr}(k)$ $(= L_{nr}(k)(T_{ms} + T_s))$ is the time period from the k-th reservation slot to the $(k+1)$-th reservation slot, and so $L_{nr}(k)$ is the number of corresponding slots including the reservation slot during $T_{nr}(k)$ with the following conditional distribution given $N(k) = n$:

$$q_l(l|n) = \begin{cases} 1, & \text{if } l = 1, n = 0, \\ 0, & \text{if } l \neq 1, n = 0, \\ 0, & \text{if } l < 2, n \neq 0, \\ p_l(1-p_l)^{l-2}, & \text{if } l \geq 2, n \neq 0. \end{cases} \tag{4}$$

The distribution of the number of the downlink message arrivals, $N_a(k)$, from the end of the $(k-1)$-th reservation slot to the end of the k-th reservation slot given $L_{nr}(k-1) = l$ is

$$q_a^l(i) = e^{-\lambda_d l(T_{ms}+T_s)} \frac{(\lambda_d l(T_{ms}+T_s))^i}{i!}. \tag{5}$$

The probability that a non-backlogged client requests in the k-th reservation slot given $L_{nr}(k-1) = l$ is

$$q_g^l = 1 - e^{-\lambda_u l(T_{ms}+T_s)/K_u}. \tag{6}$$

Let $Q_g^l(i, m)$ be the probability that i out of $K_u - m$ non-backlogged clients transmit requests in the k-th reservation slot, and let $Q_r(i, m)$ be the probability that i of m backlogged clients transmit requests given $M(k-1) = m$ and $L_{nr}(k-1) = l$, then

$$\begin{aligned} Q_g^l(i, m) &= \binom{K_u - m}{i}(1 - q_g^l)^{K_u - m - i}(q_g^l)^i, \\ Q_r(i, m) &= \binom{m}{i}(1 - q_r)^{m-i}(q_r)^i. \end{aligned} \tag{7}$$

Now, $N_r(k) + N_g(k)$ clients will transmit requests in the k-th reservation slot. Accordingly, we obtain the following state transition relationship:

$$\begin{aligned} N(k) &= N(k-1) + N_s(k) + N_a(k) - T(k-1), \\ M(k) &= M(k-1) + N_g(k) - N_s(k), \end{aligned} \tag{8}$$

where $T(k) = 0$ if $N(k) = 0$ and 1 if $N(k) \geq 1$, and $N_s(k)$ is the number of the successful requests during the k-th reservation slot.

The probability $P_u(\hat{j}, \hat{k}, \hat{l})$ that $\hat{j}$ out of $\hat{k}$ clients succeed in the k-th reservation slot (with $\hat{l}$ reservation request mini-slots) is given by

$$P_u(\hat{j}, \hat{k}, \hat{l}) = \begin{cases} 0, & \text{if } \hat{j} > \hat{l} \text{ or } (\hat{j} = \hat{l} \text{ and } \hat{k} > \hat{l}), \\ \binom{\hat{k}}{\hat{j}} \frac{\hat{l}!}{(\hat{l}-\hat{j})!} A(\hat{k} - \hat{j}, \hat{l} - \hat{j}) / \hat{l}^{\hat{k}}, & \text{otherwise,} \end{cases} \tag{9}$$

where $A(k', l')$ is the number of cases such that k' clients requested during one of l' mini-slots, and all of them failed:

$$A(k', l') = \begin{cases} 1, & \text{if } k' = 0, \\ 0, & \text{if } k' = 1, \\ \sum_{g=1}^{\lfloor k'/2 \rfloor} \binom{l'}{g} \sum_{\mathbf{C}(\mathbf{n})} \binom{k'}{n_1 n_2 \cdots n_g} & \\ \cdot \binom{g}{m_1 m_2 \cdots m_{g'}}, & \text{if } k' \geq 2, \end{cases} \tag{10}$$

where $\binom{k'}{n_1 n_2 \cdots n_g}$ $(= k'!/n_1!n_2! \cdots n_g!)$ is the g-th order multinomial coefficient, and the condition $\mathbf{C}$ of the g-th order vector $\mathbf{n} = \{n_1, n_2, \cdots, n_g\}$ is: (i) $\sum_{i=1}^{g} n_i = k'$; (ii) for all i, $n_i \geq n_{i+1}$; and (iii) for all i, $n_i \geq 2$. The g'-th order vector $\mathbf{m} = \{m_1, m_2, \cdots, m_{g'}\}$, which directly depends on the vector $\mathbf{n}$, satisfies: (i) $\sum_{i=1}^{g'} m_i = g$; (ii) $g' = \max_{i=1}^{g} n_i$; and (iii) m_i is the number of n_j's such that $n_j = i$.

Finally, we can easily derive the state transition probabilities of $M(k)$ and $N(k)$, respectively, given $(M(k), N(k), L_{nr}(k)) = (m, n, l)$:

$$\begin{aligned} P_{m,m+i}(m, n, l) &= \sum_{g=0}^{K_u - m} \sum_{r=0}^{m} Q_g^l(g, m) Q_r(r, m) \\ &\quad \cdot P_u(g - i, g + r, L_{ms}), \end{aligned} \tag{11}$$

for $K_u - m \geq i \geq -L_{ms} + 1$ if $m > L_{ms}$ and $K_u - m \geq i \geq -m$ if $m \leq L_{ms}$, where L_{ms} is the number of the reservation mini-slots in a reservation slot, i.e., $L_{ms} = K/2 = T_s/(2T_{ms})$, and

$$\begin{aligned} & P_{n,n+j}(m, n, l) \\ &= \begin{cases} \sum_{a=0}^{j} q_a^l(a) Q_s(j - a | m, l), & \text{if } n = 0, \\ \sum_{a=0}^{j+1} q_a^l(a) Q_s(j + 1 - a | m, l), & \text{if } n > 0, \end{cases} \end{aligned} \tag{12}$$

for $j \geq 0$ if $n = 0$ and $j \geq -1$ if $n > 0$, where $Q_s(i|m, l)$ is the probability of $N_s(k) = i$ given $M(k-1) = m$ and $L_{nr}(k-1) = l$:

$$Q_s(i|m, l) = \sum_{g=0}^{K_u - m} \sum_{r=0}^{m} P_u(i, r+g, L_{ms}) Q_g^l(g, m) Q_r(r, m). \tag{13}$$

The conditional state transition probability of the 2-dimensional Markov chain $(M(k), N(k))$ given $L_{nr}(k) = l$ is:

$$P_{(m,n),(m+i,n+j)}^l = P_{m,m+i}(m, n, l) P_{n,n+j}(m, n, l). \tag{14}$$

Averaging the effect of the condition $L_{nr}(k)$, we obtain the state transition probability:

$$P_{(m,n),(m+i,n+j)} = \sum_{l=1}^{\infty} P_{(m,n),(m+i,n+j)}^l P_{L_{nr}(k)|N(k)}(l|n). \tag{15}$$

Finally, we can obtain the steady-state probability:

$$\pi_{m,n} = \lim_{k \to \infty} P(M(k) = m, N(k) = n). \tag{16}$$

Uplink Request Success Rate: The request success rate from the $(k-1)$-th reservation slot to the k-th reservation slot given $N_s(k) = i$ and $L_{nr}(k-1) = l$ is:

$$R_u^s(i, l) = \frac{i}{l(T_{ms} + T_s)}. \tag{17}$$

By averaging $N_s(k)$ and $L_{nr}(k)$, we get the uplink request success rate given $M(k-1) = m$ and $N(k-1) = n$.

$$R'_s(m, n) = \sum_{i=1}^{L_{ms}} \sum_{l=1}^{\infty} R_u^s(i, l) Q_s(i|m, l) q_l(l|n). \tag{18}$$

We define two new continuous-time processes $\hat{M}(t) = M(k)$ and $\hat{N}(t) = N(k)$, if $t \in [E_k, E_{k+1})$, where E_k is the end time of the k-th reservation slot. Note that $\hat{M}(t)$ denotes the number of backlogged clients at time t. We can obtain the steady-state probability of this process as follows:

$$\hat{\pi}_{m,n} = \frac{\pi_{m,n} E[L_{nr}^n]}{\pi_0^{bsq} E[L_{nr}^0] + (1 - \pi_0^{bsq}) E[L_{nr}^1]}, \tag{19}$$

where $E[\cdot]$ is the expectation of a random variable, $\pi_n^{bsq} = \lim_{k \to \infty} P(N(k) = n) = \sum_m \pi_{m,n}$, and L_{nr}^n is the number of the slots between two consecutive reservation slots, $L_{nr}(k)$, given $N(k) = n$. It is easily shown to be $E[L_{nr}^0] = 1$ and $E[L_{nr}^1] = 1 + 1/p_l$. For a given time t, if $\hat{M}(t) = n$ and $\hat{N}(t) = m$, then the conditional request success rate is $R'_s(m, n)$. Thus, by averaging this over time, we get the average request success rate:

$$R_u^s = \sum_m \sum_n R'_s(m, n) \hat{\pi}_{m,n}. \tag{20}$$

Average Request Success Delay: We derive the delay from the generation of a message to a successful request for its transmission. The first term in the delay is the average time V from the message generation to the beginning of next reservation slot. When $N(k) = 0$, $L_{nr}(k) = 1$. Then the generation time of a message — generated in $[B_k, B_{k+1})$ for an arbitrary k — will be uniformly distributed in $[B_k, B_{k+1}]$ [11], where $B_{k+1} - B_k = T_s + T_{ms}$, since messages are generated according to a Poisson process, and so $E[V|N(k) = 0] = (T_s + T_{ms})/2$. When $N(k) > 0$, $L_{nr}(k)$ has a geometric distribution plus one. Since the geometric distribution is memoryless, when a message was generated, $E[V|N(k) > 0]$ is approximated to be

$E[L_{nr}^1-1](T_s+T_{ms})$. Consequently, we obtain the mean value of V as:

$$E[V] \approx \frac{E[L_{nr}^0]}{2}(T_s+T_{ms})\hat{\pi}_0^{bsq} + E[L_{nr}^1-1] \cdot (T_s+T_{ms})(1-\hat{\pi}_0^{bsq}). \qquad (21)$$

Secondly, according to Little's theorem, the average time spent in the backlog is the ratio of the average of backlogged clients to the average message generation rate G_{new} or $E[\hat{M}]/G_{new}$, where the average of backlogged clients $E[\hat{M}] = \sum_m \sum_n m\hat{\pi}_{m,n}$. Now, the average delay measured is given as:

$$D_u^s = E[V] + T_s + \frac{E[\hat{M}]}{G_{new}} \quad \text{(mini-slots)}, \qquad (22)$$

where the first term corresponds to the time to the next reservation slot, the second term to a reservation slot time, and the third term to the average backlog delay. For the whole system to be stable, the average rate of new message generation must equal the average message transmission request success rate, i.e., $G_{new} = R_u^s$. Finally, we get the desired throughput-delay relation under the stable condition:

$$D_u^s = E[V] + T_s + \frac{E[\hat{M}]}{R_u^s} \quad \text{(mini-slots)}. \qquad (23)$$

Throughput Analysis: Due to the existence of control mini-slots and reservation slots, the maximum achievable throughput W_{total}^{max} is less than one, and is dependent on the message length distribution. Assuming that for all k, $N(k) > 0$, a reservation slot and a message transmission will alternate continuously, thus achieving the maximum possible throughput which is given by

$$W_{total}^{max} = \frac{T_s}{T_s+T_{ms}} \frac{E[L_{nr}^1-1]}{E[L_{nr}^1]}. \qquad (24)$$

Note that the actual total incoming rate (including both uplink and downlink) to the base station service queue is $\lambda_{total} = \lambda_d + R_u^s$. Now, if $\lambda_{total} E[L_{nr}^1-1]T_s \leq W_{total}^{max}$, i.e., if the base station service queue is in the stable condition, the downlink throughput W_d and uplink throughput W_u would be

$$\begin{aligned} W_d &= \lambda_d E[L_{nr}^1-1]T_s, \\ W_u &= R_u^s E[L_{nr}^1-1]T_s. \end{aligned} \qquad (25)$$

Average Delay: First of all, we need the queueing delay in the base station service queue, i.e., the average delay from the entrance of a downlink message or an uplink request into the service queue to the start of its transmission. We first obtain the average number of the queued downlink messages or uplink requests in the base station service queue which is given by

$$E[\hat{N}-1|\hat{N}>1] = \sum_n (n-1)\hat{\pi}_n^{bsq}, \qquad (26)$$

because $\hat{N}(t)-1$ corresponds to the number of downlink messages or queued requests in the base station service queue for $[E_k, E_{k+1})$. Now, the queueing delay is given using Little's theorem:

$$D_q = E[\hat{N}-1|\hat{N}>1]/G'_{new}, \qquad (27)$$

where $G'_{new} = \lambda_{total} = \lambda_d + R_u^s$ for the system to be stable. Now, the downlink delay is given by

$$D_d = E[L_{nr}^1-1](T_s+T_{ms}) + E[V] + D_q, \qquad (28)$$

where $E[V]$ is the average time from a downlink message arrival to the end of the next reservation slot, which is approximated to the value from Eq. (21), and the uplink delay is given by

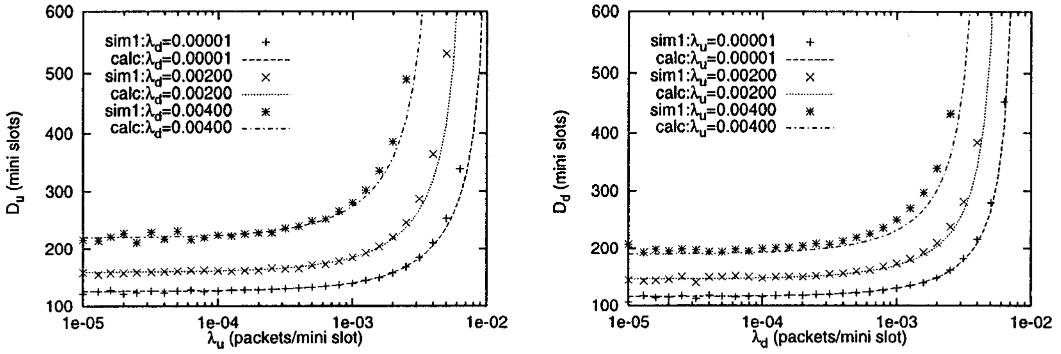
$$D_u = E[L_{nr}^1-1](T_s+T_{ms}) + D_u^s + D_q. \qquad (29)$$

In both equations, the first terms stand for the message transmission delays, the second terms for the delays from the arrival/generation of a message to the entrance into the base station service queue, and the third for the queueing delays in the service queue.

Numerical and Simulation Results: We show some analytical calculation results using the equations above, and compare them with the simulation results. For the simulations, we generated Poisson traffic, and followed the assumptions given at the beginning of this section. The results are based on $p_l = 0.1$, $q_r = 1.$, $K_u = 5$, $K = T_s/T_{ms} = 10$ (and so $L_{ms} = 5$).

Fig. 6 (a) plots the uplink delays D_u as λ_u increases for three different λ_d values, while Fig. 6 (b) plots the downlink delays D_d as λ_d increases. We observe that the numeric calculations (with mark *calc*) and the simulations (with mark *sim1*) are very close to each other for the same parameters. Note that we did rarely use approximations for our analysis except for in Eq. (21). In both cases, delays are almost constant for small rates, but monotonically increase, and then go to infinity as the actual total incoming rate λ_{total} $(= \lambda_d + R_u^s)$ goes to $W_{total}^{max}/(E[L_{nr}^1-1]T_s)$ ($\approx 8.26e^{-3}$ in the results). Due to the closed-loop behavior of the clients, $R_u^s \leq \lambda_u$. Hence, in the figures, the marginal rates (at which delays become infinite) appear larger for uplink under the same parameters. Note that the uplink delays are larger than the downlink delays by as much as $T_s + E[\hat{M}]/R_u^s$ under the same condition from Eqs. (28) and (28).

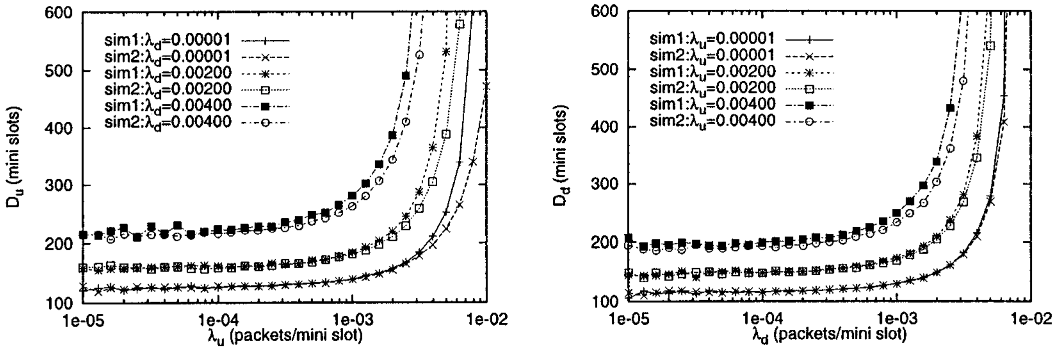
Fig. 7 compares the simplified protocol with the simplifications **S1** and **S2** (marked with *sim1*) and the actual protocol without S1 and S2 (marked with *sim2*) using the simulation results of the delay-versus-rate relationship. For the actual protocol, *MNRSL* = 10 was used. In both graphs, we observe that delays are smaller for the actual protocol, especially for large rates, since the messages can be transmitted consecutively without the appearance of a reservation if the first message has less than *MNRSL* packets. Consequently, the marginal rates are larger for the actual protocol, implying that the maximum achievable throughput be larger for the actual protocol.



(a) Uplink; D_u vs. λ_u

(b) Downlink; D_d vs. λ_d

Figure 6: Comparison between analytical calculations (marked with *calc*) and simulations (marked with *sim1*).



(a) Uplink; D_u vs. λ_u

(b) Downlink; D_d vs. λ_d

Figure 7: Comparison between the simplified protocol (marked with *sim1*) and the actual protocol (marked with *sim2*).

6 Concluding Remarks

In this paper, we have considered a WLAN providing QoS guarantees for heterogeneous traffic in a cell. According to the required QoS, traffic is categorized into class I (real-time) and class II (non-real-time). The protocol is based on the framing strategy for class I and a reservation scheme for class II, where class I has priority over class II. When each class-I connection follows a smoothness model, it was shown to be possible to guarantee the delay bound and throughput using the stop-and-go queueing. The admission test for a new class-I connection was also defined. When there is no eligible class-I traffic, class-II traffic is transmitted. Uplink class-II transmission reservation and uplink class-I connection establishment were requested using the reservation scheme. We finally analyzed the average delay and throughput of the reservation scheme for class-II traffic, and presented the numerical calculation and simulation results.

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