

## **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 three classes (say I, II, and III) according to the required QoS. Whenever there is no eligible class-I and II traffic, class-III traffic which requires no delay bound guarantees is transmitted, while uplink transmissions are controlled with a reservation scheme. Class-I traffic which requires a bounded delay and guaranteed throughput is handled with the framing strategy [1] which consists of a smoothness traffic model and the stop-and-go queueing scheme. We also establish the admission test for adding new class-I connections. We present a modified framing strategy for class-II voice uplink transmissions which utilizes the wireless link efficiently at the cost of some packet losses. Finally, we present the performance (average delay and throughput) evaluation of the reservation scheme for class-III traffic using both analytical calculations and simulations.

*Index Terms* — Wireless LANs and MAC protocol, QoS guarantees, star topology, dynamic Time Division Duplexing (TDD), stop-and-go queueing, framing strategy, admission tests, reservation ALOHA.

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# 1 Introduction

Wireless local area networks (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 [4-8] and LANs [9,10].

In this paper, we consider how to provide QoS guarantees for heterogeneous traffic on a WLAN. The following three types of QoS for each connection are considered: (1) maximum packet delivery delay; (2) transmission throughput, defined as the long-term fraction of time the channel carries the connection's traffic; and (3) packet loss probability, which is the percentage of packet losses for the connection. Table 1 summarizes the classification of various traffic. Class-I traffic requires bounded delay and no packet loss. Real-time data and video from the high-priority sub-band are categorized into this class. Class-II traffic also requires bounded delay, but is tolerable of some packet losses with a certain probability. A packet which arrives at the destination after the delay bound is also considered lost. In those applications involved with this class traffic, the lost packets either do not degrade the performance of the application much or can be recovered at the receiving end. Voice and video from low-priority sub-bands are examples of this class. Finally, class-III traffic like the conventional data service does require zero loss, but no bounded delay. Class III can also be divided further into two subclasses [11]: (1) class III-A which is delay-sensitive like FTP and remote log-in; and (2) class III-B which is delay-tolerable like paging and e-mail. Class III-A has priority over class III-B.

	Bounded delay	Unbounded delay
Zero loss	Class I Real-time data Video(high-priority sub-band)	Class III Conventional data services File transfer, E-mail
Loss tolerable	Class II Voice Video(low-priority sub-band)	

Table 1: The classification of traffic

We adopt a reservation scheme which is similar to the reservation ALOHA [12,13] or packet reservation multiple access (PRMA) [14] for uplink class-III 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 methods.) This reservation scheme is a promising multiple access protocol for class-III traffic, as it provides higher throughput and smaller average delay than other collision-based random access protocols like ALOHA. Basically, class-III traffic is transmitted when there is no class-I and II traffic to be transmitted since it has a lower priority than class I and II. The reservation scheme is also used for establishing new class-I and II connections requested by the mobile clients.

The framing strategy [1,5], which was originally proposed as a framework for congestion management in integrated service packet networks, is, with some modifications, used to provide guaranteed delay bound and guaranteed throughput for class I. The framing strategy is composed of a smoothness traffic model and stop-and-go queueing, and provides packet-delay bound and guaranteed transmission throughput. To this end, 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. For this purpose, slots and control mini-slots alternate. We also present a modified framing strategy for class-II voice uplink transmissions, where some packet losses do not affect the voice quality much. We define the traffic model of speech, and describe a scheme that utilizes the wireless link efficiently by transmitting class-III packets while the voice transmitter is in the silent mode.

In a companion paper [15], we also proposed another scheme for supporting QoS in a WLAN using a token allocation and slotted ALOHA. Compared to this, the scheme proposed in this paper has several advantages and disadvantages: (1) instead of allocating tokens for uplink transmissions, stop-and-go queueing is adopted with the continuously alternating slot and mini-slot format (2) due to the alternating slot and mini-slot format, the maximum achievable throughput is expected to decrease while enabling us to schedule the uplink and downlink more dynamically and flexibly; (3) there are only a finite number of delay bound levels in contrast to the other scheme in [15] with virtually infinite delay bound levels; (4) using the framing strategy, the scheduling and admission test are much simpler; and (5) with the reservation scheme instead of slotted ALOHA, better delay and throughput performances of class-III traffic are attained.

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

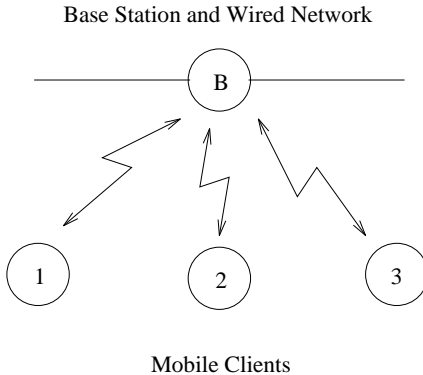


Figure 1: A centralized wireless network with a base station.

reservation scheme for class-III traffic. Section 4 considers the framing strategy with QoS guarantees for class I and defines the admission test for establishing a new connection. As part of the framing strategy, a smoothness traffic model and stop-and-go queueing are described. In Section 5, we present a modified framing strategy for voice transmissions, which achieves an efficient link utilization at the cost of some packet losses. Section 6 presents the analysis and simulation results of the average delay and throughput of the reservation scheme for class III. Finally, the paper concludes with Section 7.

## 2 Specifications and Assumptions of System

As shown in Figure 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 (e.g., ATM LAN) via a wired link. 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 [16]. As shown in Figure 2, the transmission ranges of mobile 1 and 2 do not allow them to hear each other, but can both be heard by the base station in between. The mobile 1 and 2 are hidden terminals to each other.

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

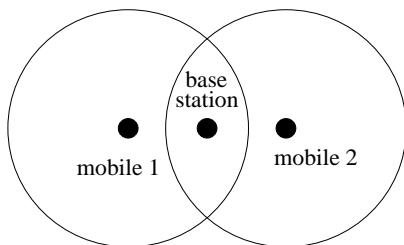


Figure 2: The hidden terminal situation.

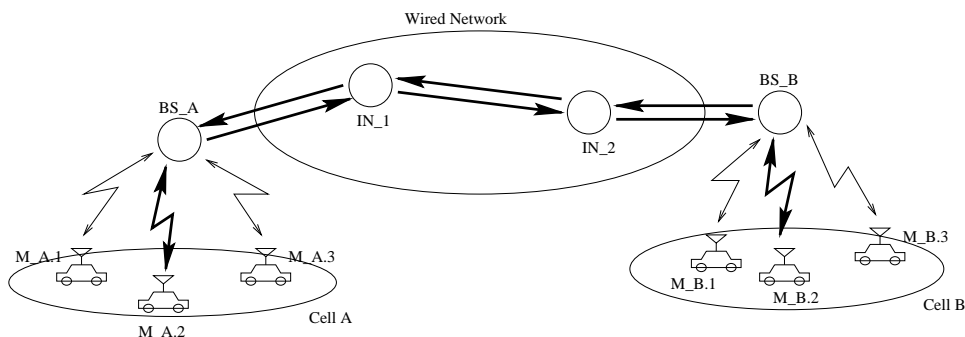


Figure 3: The end-to-end transmission from a sending cell to a receiving cell via a wired network.

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 as shown in Figure 3. 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 the uplink and the other part is for downlink transmissions. FDD, as in AMPS (FDMA), IS-54 (TDMA), and IS-95 (CDMA), 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 [17]. We assume all packets, like ATM

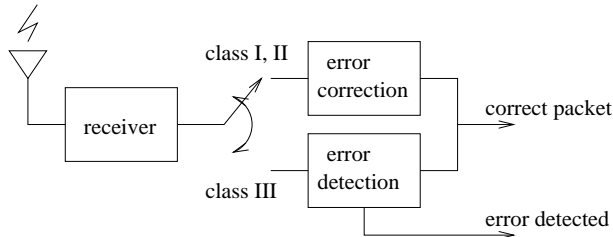


Figure 4: A dual-mode receiver equipped with both error correction (for FEC) and error detection (for ARQ) decoders.

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 a cell. (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]).

Since the wireless channel is inherently unreliable (due to noises, interferences, and multipath fading), 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 [18] 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 III to ensure virtually error-free transmission of data. But, it is ruled out for class I and II because of its difficulty in making delivery-delay guarantees. We use an FEC (Forward Error Correction) scheme for class I and II, instead.<sup>1</sup> To this end, the receiver is equipped with a dual-mode channel decoder (error correction and detection mode). The transceiver pair works as follows, assuming use of a specific channel code: (1) at the transmitter, a packet is channel-encoded, and then transmitted; and (2) at the receiver, the received packet is decoded by an error-correction decoder (if class I and II) or an error-detection decoder (if class III) as shown in Figure 4. The dual-mode receiver is expected to work well since using a channel code, the decoder can detect more errors than those correctable [19]. To ensure virtually error-free packet delivery, we assume that certain channel coding and diversity schemes with a dual-mode receiver (with an ARQ scheme) are used to combat noises, interferences, and multipath fading. We will not

<sup>1</sup>Although combined FEC and diversity seems to be the only way for error-protection of class I and II, 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 — the more capability, the more redundancy needed. So the proposed scheme here might not be applicable for reliability-critical real-time data traffic of class I.

consider error-combating techniques any more since they are not within the main scope of this paper, but we assume that a packet is received correctly unless that packet collides with one or more concurrent packets.

### 3 Protocol Description

When a mobile client wants to send a packet, regardless whether it is destined for another client in the same cell or for a remote cell, the client must send the packet to its base station first, which will then forward the packet to the final destination, sometimes via other base stations. Dynamic TDD transmission is used in the network under consideration, i.e., a wireless channel is time-shared for both downlink and uplink transmissions under the dynamic control of the base station. Based on the star topology, only the downlink channel is assumed to be the broadcast type. Thus, when the base station transmits packets, all but their destination mobiles in the cell ignore them. By contrast, a mobile cannot hear other mobiles' uplink transmissions, and only the base station can determine if a collision has occurred in the uplink channel.

For the time-being, we confine ourselves to class-I and III traffic. A modified version of class-I scheme will be considered later as the scheme for class-II voice communications. We adopt two different strategies for class I and III. 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 client) is identified by (1) which client the connection is for and (2) whether the connection is downlink or uplink transmissions. For the QoS provision, class-I traffic has priority over class-III traffic, where the transmission is controlled by the framing strategy (to be discussed in the next section). Class-III traffic doesn't need the concept of connection, but if there is a pending message (which consists of a number of fixed-size 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-III 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 Figure 5. In a slot of duration  $T_s$ , a (fixed-size) 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  $T_{m.s}$  is used to transmit a control packet. Control packets are used by the base station to announce to the mobiles information of the next slot, i.e., each control packet includes the information on the next slot: (1) for the downlink or uplink, (2) for class I, II, or III, and (3) for which client. These regularly alternating slots and mini-slots are expected

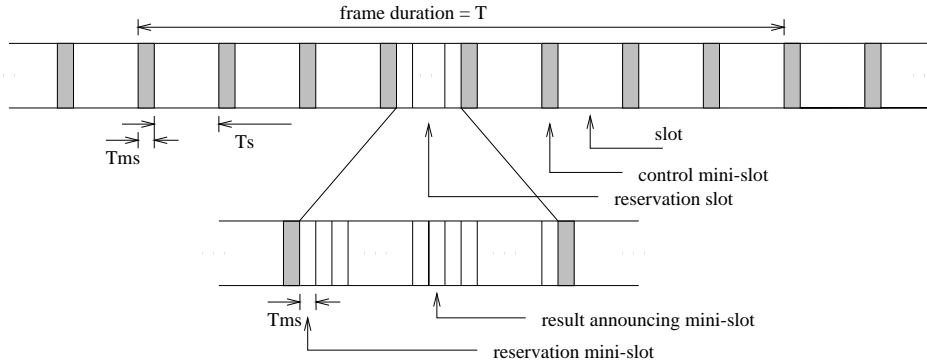


Figure 5: Dividing the time-axis into mini-slots, slots, and reservation slots. A frame includes a number of slots.

to help each mobile client 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 exists a slot, called the *reservation slot*, which is used for requesting an uplink class-I connection establishment or an uplink class-III 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 mobile clients 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 client 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 like the class of traffic it wants to transmit. If class-I traffic is requested, the mobile client needs to send the traffic characteristics (defined in the next section) as well. The result of each of  $K/2$  mini-slots can be one of success (of which mobile<sup>2</sup>), collision, and 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 continuously 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 mobile clients 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

<sup>2</sup>Due to the *capture effects* [3], a reservation request can be transmitted successfully even in the presence of concurrent reservation requests from other mobiles.



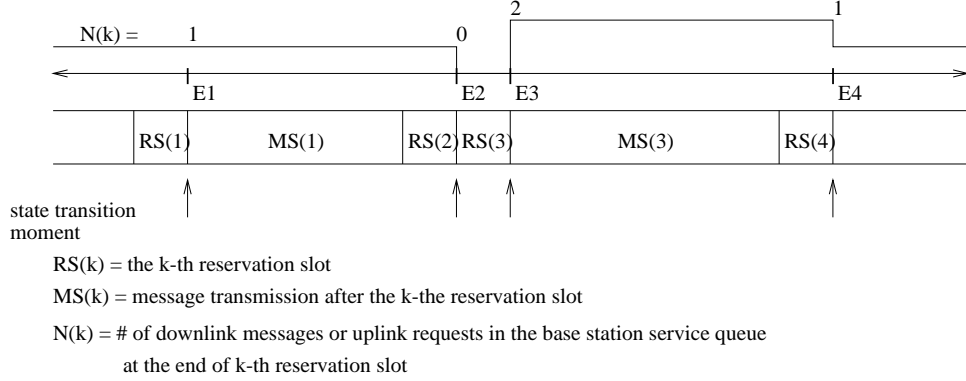


Figure 6: Timing diagram of class-III communications during the absence of class-I and II traffic

the probability  $q_{ret}$  until it is successful. (The retransmission probability  $q_{ret}$  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 client who made the request wants (1) to send a pending class-III message if it was for class III, or (2) to establish a class-I connection if it was for class I.

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-III message as shown in Figure 6, where only class-III traffic (without subclasses) exists. Two first-in-first-out (FIFO) base station class-III service queues are implemented for two subclasses of class III, in which both the uplink transmission requests (from mobile clients) and the downlink messages (from the wired network attached) are queued together. The contents of the queues are updated at the end of every reservation slot (marked with arrows in Figure 6, where only one base station class-III service queue exists): 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 transmission 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 III-B can be served whenever the first queue for class III-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 III 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* slots were issued since the reservation slot, the base station will serve more messages until the issued slots in total exceeds *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 Figure 5. 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. 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 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 and leaky-bucket [6] in the source end nodes and a flow/congestion control or packet scheduling scheme such as the framing strategy and Fair Queueing (or PGPS) [6] 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 smooth 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,  $T_1^f, T_2^f, \dots, T_G^f$ , and each frame size is multiple of smaller frame sizes, i.e.,

$$T_g^f = I_g \cdot T_{g+1}^f, \quad g = 1, 2, \dots, G - 1, \quad (4.1)$$

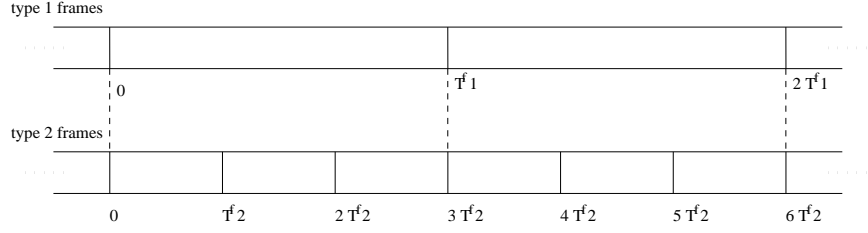


Figure 7: Two frames with duration  $T_1^f = 3T_2^f$ .

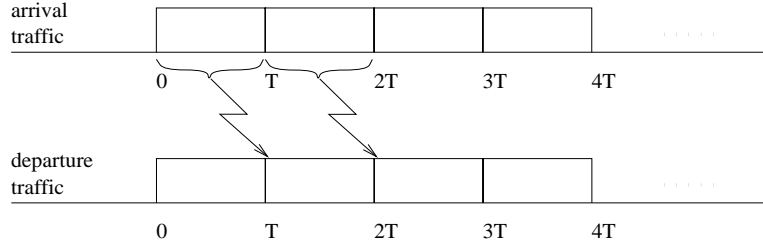


Figure 8: When packets arriving on each frame become eligible for transmission.

for some integer  $I_g$ . For all  $g$ ,

$$T_g^f = K_g \cdot (T_s + T_{m.s}), \quad (4.2)$$

for some integer  $K_g$ , i.e., there are a finite number of slots in each frame. Each frame of duration  $T_g^f$  is called a *type- $g$  frame*. For each connection  $i$ ,  $T_i = T_g^f$  for some  $g$ , and the connection is called the *type- $g$  connection*. Figure 7 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  $2T_g^f$ , implying the existence of  $G$  delivery delay bounds.

## 4.2 Stop-and-Go Queueing

**Downlink Transmissions:** The communication 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 as follows.

**Rule (a)** 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 (Figure 8). During a frame, the set of eligible packets of the corresponding type are transmitted.



marked by arrows. One can see that connection 1 has priority over connection 2 even though they are of the same frame size since connection 1 is downlink. Connection 3 has the lowest priority. Through the blank slots in the figure, class-III traffic, if any, will be transmitted by the rules presented in Section 3.  $\square$

### 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

$$\sum_{g=1}^G M_g^f \cdot (T_s + T_{ms})/T_g^f \leq 1, \quad (4.3)$$

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 Appendix A for a formal proof. Note that in Example 1, the summation in Eq. (4.3) 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-III traffic (say  $S$ ). In such a case, Eq. (4.3) should be modified by replacing one with  $1 - S(T_s + T_{ms})/T_s$ .

## 5 Modified Framing Strategy for Uplink Voice Transmissions

In this section, we modify the framing strategy for uplink voice (class-II) transmissions. Speech is encoded by the 8-bit PCM<sup>3</sup> at 8 kHz, and so, the bit rate is 64 kb/s [20]. Assume that the target delay bound is  $2T_g$  for some  $g$  for a voice connection  $i$ . Since the frame size

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<sup>3</sup>This voice coding is used in the current wired telephone communications. In the existing wireless communications, other voice coding schemes like ADPCM, QCELP, and VSELP are used to reduce the bit rate due to the wireless bandwidth limitation [2]. However, in this paper, we adopt the simple PCM scheme to show how the framing strategy can be modified to accommodate the voice communication more efficiently.

is  $T_g$ , up to  $N_f (= \lfloor T_g/T_{sample} \rfloor)$  samples are transmitted during a frame, where the sampling period  $T_{sample} = 1/(8 \times 10^3) = 1.25$  msec. If a packet can accommodate  $N_s$  samples,  $\lceil N_f/N_s \rceil$  packets are necessary during a frame to transmit the voice traffic within the delay bound, and so  $(M_i, T_i) = (\lceil N_f/N_s \rceil, T_g)$  traffic model is adopted for connection  $i$ .

It is well-known that voice traffic is modelled by an “on-off” model since a speech signal is in either *talking* or *silent* mode. So a voice client generates bursts of packets, corresponding to *talk-spurts*, while in talking mode, and no packets while in silent mode. From the  $(M_i, T_i)$  smoothness model, in each frame of duration  $T_i$ , there may or may not be packets to be transmitted. With this traffic model, we can use the above framing strategy for downlink, but it will not be efficient for uplink since the base station will issue at least one slot every  $T_i$ , and while in silent mode, there will be no packet transmission in that slot. We modify the previous framing strategy for uplink voice transmissions of connection  $i$  so that class-III traffic may be transmitted while in silent mode.

The admission test given in Eq. (4.3) is also used for a new voice connection. Assuming that connection  $i$  has passed the admission test, the base station issues at least one slot for connection  $i$  in each frame, and the corresponding mobile transmits voice packets via those slots. If an issued slot is empty within a frame, the base station assumes that the mobile is in silent mode, and will stay there for a while. To utilize the link more efficiently, the base station uses  $M_i$  slots in each frame for class-III traffic beginning at the next frame after the empty slot, i.e., by the scheme in Section 3, up to  $M_i$  downlink packets of class-III are transmitted or uplink slots are issued for class-III traffic in each frame, while notifying that these slots were originally for connection  $i$ . While the mobile stays in silent mode, class-III traffic transmissions will be successful. However, after this process, the mobile of connection  $i$  will transmit a packet in the issued slot when the mobile returns to the talking mode, and then there will be a collision between the packet and a class-III packet. Detecting this collision will lead the base station to assume that the mobile has now returned to talking mode. The collided class-III packet should be retransmitted later. The base station also tries to issue another slot in the same frame for the collided connection- $i$  packet. If all slots are reserved for class-I and II connections (i.e., the equality holds in Eq. (4.3)), it may be impossible to issue another slot in the same frame. But, this is not a serious problem since voice traffic is class II in which some packet losses are tolerable. After the frame in which the collision occurred, the base station starts to issue  $M_i$  slots in each frame exclusively for connection  $i$ .

## 6 Performance Analysis of Class-III Communications

This section analyzes the performance of the reservation scheme for class-III communications, where smaller average delay and larger throughput are desirable. Here, all traffic belongs to class III, where class III-A and III-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 or II traffic, the reservation scheme considered here is activated whenever there are no eligible class-I or II packets to be transmitted.

The analysis in this section is based on that of slotted ALOHA with TDD in [17] with some modifications. For uplink accesses using the reservation scheme, we use the model of  $K_u$  clients. Our analysis will be based on the following assumptions.

- A1.** The message length has a geometric distribution with parameter  $p_{len}$  measured in the number of fixed-size packets.
- A2.** Downlink messages arrive from the wired network (to which the base station is attached) according to a Poisson process. Let  $\lambda_d$  (messages/mini-slot) be the overall message arrival rate for all connections from the wired network to the mobile clients in the cell.
- A3.** Messages are generated for transmission at each of the  $K_u$  clients according to independent Poisson processes. Let  $\lambda_u/K_u$  be the generation rate at each client, so  $\lambda_u$  (messages/mini-slot) is the overall generation rate by all clients.
- A4.** A request is transmitted correctly unless it collides with other requests, i.e., error-free transmission of packets.
- A5.** Each request involved in an uplink collision must be retransmitted in a later reservation mini-slot until the request is successfully received. A client is said to be *backlogged* when it was notified by the base station to have a request that was not transmitted successfully and hence must be retransmitted.
- A6.** Closed-loop behavior of clients, i.e., each backlogged client will discard newly-generated messages until the successful transmission of the request.

Note that from **A1**, we consider the inter-cell communications only since downlink messages are assumed to arrive exclusively from the wired network, not from the mobile clients within the cell via uplink transmissions. We also make the following simplification 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  $q_{ret}$  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. Figure 6 shows a timing diagram of class-III 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  $q_{ret}$ . 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.  $T_{nrst}(k)$  ( $= L_{nrst}(k)(T_{ms} + T_s)$ ) is the time period from the  $k$ -th reservation slot to the  $(k+1)$ -th reservation slot, and so  $L_{nrst}(k)$  is the number of corresponding slots including the reservation slot during  $T_{nrst}(k)$  with the following conditional distribution:

$$q_{len}(l|n) \triangleq P_{L_{nrst}(k)|N(k)}(l|n) = \begin{cases} 1, & \text{if } l = 1, n = 0, \\ 0, & \text{if } l \neq 1, n = 0, \\ 0, & \text{if } l < 2, n \neq 0, \\ p_{len}(1 - p_{len})^{n-2}, & \text{if } n \geq 2, n \neq 0. \end{cases} \quad (6.1)$$

Note that when  $N(k) \neq 0$ ,  $L_{nrst}(k)$  is one plus the message length measured in the number of packets. The distribution of the number of the downlink message arrivals,  $N_{arr}(k)$ , from the end of the  $(k-1)$ -th reservation slot to the end of the  $k$ -th reservation slot given the condition  $L_{nrst}(k-1) = l$  is

$$q_{arr}^l(i) \triangleq P_{N_{arr}(k)|L_{nrst}(k-1)}(i|l) = e^{-\lambda_d l (T_{ms} + T_s)} \frac{(\lambda_d l (T_{ms} + T_s))^i}{i!}. \quad (6.2)$$

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

$$q_{gen}^l = 1 - e^{-\lambda_u l (T_{ms} + T_s) / K_u}. \quad (6.3)$$



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

$$\begin{aligned} Q_{gen}^l(i, m) &\triangleq P_{N_{gen}(k)|M(k-1), L_{rrst}(k-1)}(i|m, l) = \binom{K_u - m}{i} (1 - q_{gen}^l)^{K_u - m - i} (q_{gen}^l)^i, \\ Q_{ret}(i, m) &\triangleq P_{N_{ret}(k)|M(k-1)}(i|m) = \binom{m}{i} (1 - q_{ret})^{m - i} (q_{ret})^i. \end{aligned} \quad (6.4)$$

Now,  $N_{ret}(k) + N_{gen}(k)$  clients will transmit requests in the  $k$ -th reservation slot. Accordingly, we obtain the following state transition relationship:

$$\begin{aligned} N(k) &= \begin{cases} N(k-1) + N_{succ}(k) + N_{arr}(k), & \text{if } N(k-1) = 0 \\ N(k-1) + N_{succ}(k) + N_{arr}(k) - 1, & \text{if } N(k-1) \geq 1 \end{cases} \\ M(k) &= M(k-1) + N_{gen}(k) - N_{succ}(k), \end{aligned} \quad (6.5)$$

where  $N_{succ}(k)$  is the number of the successful requests during the  $k$ -th reservation slot.

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

$$P_u(\hat{j}, \hat{k}, \hat{l}) = \begin{cases} 0, & \text{if } \hat{j} > \hat{l} \text{ or } (\hat{j} = \hat{l} \text{ and } \hat{k} > \hat{l}), \\ \binom{\hat{k}}{\hat{j}} \frac{\hat{l}!}{(\hat{l} - \hat{j})!} A(\hat{k} - \hat{j}, \hat{l} - \hat{j}) / \hat{l}^{\hat{k}}, & \text{otherwise,} \end{cases} \quad (6.6)$$

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:

$$A(k', l') = \begin{cases} 1, & \text{if } k' = 0, \\ 0, & \text{if } k' = 1, \\ \sum_{g=1}^{\lfloor k'/2 \rfloor} \binom{l'}{g} \sum_{\mathbf{C}(\mathbf{n})} \binom{k'}{n_1 n_2 \dots n_g} \binom{g}{m_1 m_2 \dots m_{g'}}, & \text{if } k' \leq 2, \end{cases} \quad (6.7)$$

where  $\binom{k'}{n_1 n_2 \dots n_g}$  ( $= k'! / n_1! n_2! \dots n_g!$ ) is the  $g$ -th order multinomial coefficient, and the condition  $\mathbf{C}$  of the  $g$ -th order vector  $\mathbf{n} = \{n_1, n_2, \dots, n_g\}$  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  $\mathbf{m} = \{m_1, m_2, \dots, m_{g'}\}$ , 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 the condition  $(M(k), N(k), L_{nrst}(k)) = (m, n, l)$ :

$$P_{m,m+i}(m, n, l) = \sum_{x=0}^{K_u-m} \sum_{y=0}^m Q_{gen}^l(x, m) Q_{ret}(y, m) P_u(x - i, x + y, L_{ms}), \quad (6.8)$$

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

$$P_{n,n+j}(m, n, l) = \begin{cases} \sum_{x=0}^j q_{arr}^l(x) Q_{succ}(j - x|m, l), & \text{if } n = 0, \\ \sum_{x=0}^{j+1} q_{arr}^l(x) Q_{succ}(j + 1 - x|m, l), & \text{if } n > 0, \end{cases} \quad (6.9)$$

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

$$Q_{succ}(i|m, l) = \sum_{g=0}^{K_u-m} \sum_{r=0}^m P_u(i, r + g, L_{ms}) Q_{gen}^l(g, m) Q_{ret}(r, m). \quad (6.10)$$

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

$$P_{(m,n),(m+i,n+j)}^l = P_{m,m+i}(m, n, l) P_{n,n+j}(m, n, l). \quad (6.11)$$

Averaging the effect of the condition  $L_{nrst}(k)$ , we obtain the state transition probability of the 2-dimensional Markov chain  $(M(k), N(k))$ :

$$P_{(m,n),(m+i,n+j)} = \sum_{l=1}^{\infty} P_{(m,n),(m+i,n+j)}^l P_{L_{nrst}(k)|N(k)}(l|n). \quad (6.12)$$

Finally, we can obtain the steady-state probability of this Markov chain:

$$\pi_{m,n} = \lim_{k \rightarrow \infty} P(M(k) = m, N(k) = n). \quad (6.13)$$

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

$$R_u^{succ}(i, l) = \frac{i}{l(T_{ms} + T_s)}. \quad (6.14)$$

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

$$R'_{succ}(m, n) = \sum_{i=1}^{L_{ms}} \sum_{l=1}^{\infty} R_u^{succ}(i, l) Q_{succ}(i|m, l) q_{len}(l|n). \quad (6.15)$$

Note that  $R'_{succ}(m, n) = R'_{succ}(m, 1)$  for all  $n \neq 0$ .

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  $t$ . When the steady-state probability of this process is given by

$$\hat{\pi}_{m,n} = \lim_{t \rightarrow \infty} Pr(\hat{M}(t) = m, \hat{N}(t) = n), \quad (6.16)$$

we get

$$\hat{\pi}_{m,n} = \frac{\pi_{m,n} E[L_{nrst}^n]}{\pi_0^{bsq} E[L_{nrst}^0] + (1 - \pi_0^{bsq}) E[L_{nrst}^1]}, \quad (6.17)$$

where  $E[\cdot]$  is the expectation of a random variable,  $\pi_n^{bsq} = \lim_{k \rightarrow \infty} P(N(k) = n) = \sum_m \pi_{m,n}$ , and  $L_{nrst}^n$  is the number of the slots between two consecutive reservation slots,  $L_{nrst}(k)$ , given  $N(k) = n$ . It is easily shown to be  $E[L_{nrst}^0] = 1$  and  $E[L_{nrst}^1] = 1 + 1/p_{len}$ . For a given time  $t$ , if  $\hat{M}(t) = n$  and  $\hat{N}(t) = m$ , then the conditional request success rate is  $R_u^{succ}(m, n)$ . Thus, by averaging this over time, we get the average request success rate:

$$R_u^{succ} = \sum_m \sum_n R'_{succ}(m, n) \hat{\pi}_{m,n}. \quad (6.18)$$

**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  $N(k) = 0$ ,  $L_{nrst}(k) = 1$ . Then the generation time of a message — generated in  $[B_k, B_{k+1})$  for an arbitrary  $k$  — will be uniformly distributed in  $[B_k, B_{k+1}]$  [21], 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  $N(k) > 0$ ,  $L_{nrst}(k)$  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  $E[L_{nrst}^1 - 1](T_s + T_{ms})$ . Consequently, we obtain the mean value of  $V$  as:

$$E[V] \approx \frac{E[L_{nrst}^0]}{2} (T_s + T_{ms}) \hat{\pi}_0^{bsq} + E[L_{nrst}^1 - 1] (T_s + T_{ms}) (1 - \hat{\pi}_0^{bsq}). \quad (6.19)$$

Secondly, we need the mean value of  $\hat{M}(t)$  in the steady state:

$$E[\hat{M}] = \sum_m \sum_n m \hat{\pi}_{m,n}. \quad (6.20)$$

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  $E[\hat{M}]/G_{new}$ . Now, the average delay measured is given as:

$$D_u^{succ} = E[V] + T_s + \frac{E[\hat{M}]}{G_{new}} \quad (\text{mini-slots}), \quad (6.21)$$

where the first term corresponds to the time to the next reservation slot given in Eq. (6.19), 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,

$$G_{new} = R_u^{succ}. \quad (6.22)$$

Finally, we get the desired throughput-delay relation under the stable condition:

$$D_u^{succ} = E[V] + T_s + \frac{E[\hat{M}]}{R_u^{succ}} \quad (\text{mini-slots}). \quad (6.23)$$

**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

$$W_{total}^{max} = \frac{T_s}{T_s + T_{ms}} \frac{E[L_{nrst}^1 - 1]}{E[L_{nrst}^1]}. \quad (6.24)$$

Note that the actual total incoming rate (including both uplink and downlink) to the base station service queue is  $\lambda_{total} = \lambda_d + R_u^{succ}$ . Now, if  $\lambda_{total} E[L_{nrst}^1 - 1] T_s \leq W_{total}^{max}$ , 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

$$\begin{aligned} W_d &= \lambda_d E[L_{nrst}^1 - 1] T_s, \\ W_u &= R_u^{succ} E[L_{nrst}^1 - 1] T_s. \end{aligned} \quad (6.25)$$

Note that since all clients are statistically identical, the individual uplink throughput is given by dividing  $W_u$  by  $K_u$ .

**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

$$E[\hat{N} - 1 | \hat{N} > 1] = \sum_n (n - 1) \hat{\pi}_n^{bsq}, \quad (6.26)$$

where  $\hat{\pi}_n^{bsq} = \sum_m \hat{\pi}_{m,n}$ . Note  $(\hat{N}(t) - 1)$  corresponds to the number of downlink messages or queued requests in the base station service queue for  $[E_k, E_{k+1})$ . Now, the delay in the base station service queue is given using Little's theorem:

$$D_{queue} = E[\hat{N} - 1 | \hat{N} > 1] / G_{new}, \quad (6.27)$$

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

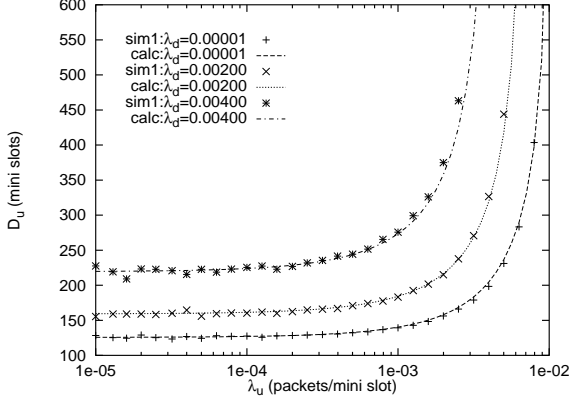
$$D_d = E[L_{nrst}^1 - 1](T_s + T_{ms}) + E[V] + D_{queue}, \quad (6.28)$$

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. (6.19), and the uplink delay is given by

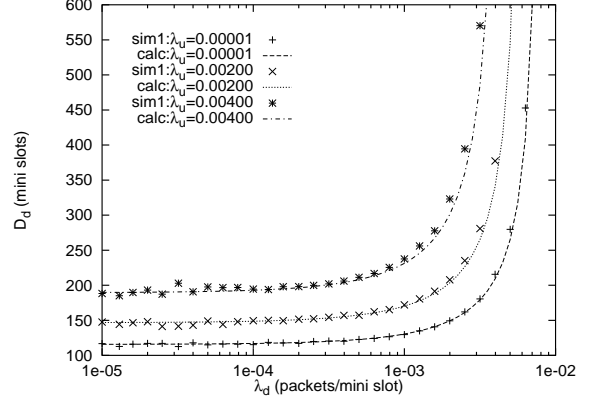
$$D_u = E[L_{nrst}^1 - 1](T_s + T_{ms}) + D_u^{succ} + D_{queue}. \quad (6.29)$$

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. Note that the difference between two delays come only from the difference between  $E[V]$  and  $D_u^{succ}$ , and so  $D_u - D_d = T_s + E[\hat{M}] / R_u^{succ}$ .

**Numerical and Simulation Results:** Now, 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 numerical/simulation results are based on  $p_{len} = 0.1$ ,  $q_{ret} = 1.$ ,  $K_u = 5$ ,  $K = T_s / T_{ms} = 10$  (and so  $L_{ms} = 5$ ). Figure 10 (a) plots the uplink delays  $D_u$  as  $\lambda_u$  increases for three different  $\lambda_d$  values, while Figure 10 (b) plots the downlink delays  $D_d$  as  $\lambda_d$  increases for three different  $\lambda_u$  values. We observe that the numeric calculations (with mark *calc*) and the simulations (with

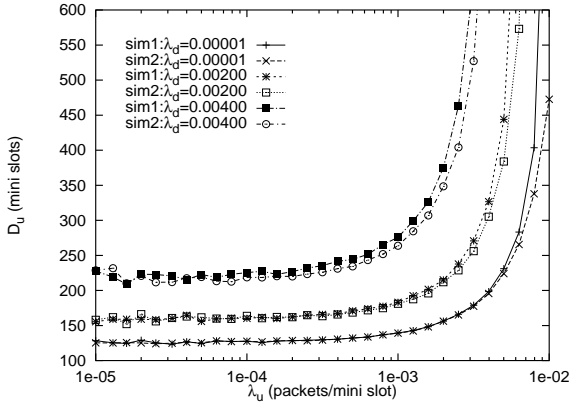


(a) Uplink;  $D_u$  vs.  $\lambda_u$

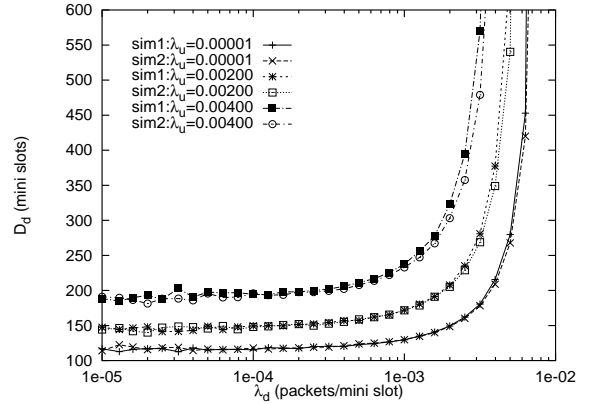


(b) Downlink;  $D_d$  vs.  $\lambda_d$

Figure 10: Comparison between analytical calculations (marked with *calc*) and simulations (marked with *sim1*);  $p_{len} = 0.1$ ,  $q_{ret} = 1.$ ,  $K_u = 5$ ,  $K = 10$ , and  $L_{ms} = 5$ .



(a) Uplink;  $D_u$  vs.  $\lambda_u$



(b) Downlink;  $D_d$  vs.  $\lambda_d$

Figure 11: Comparison between the simplified protocol (marked with *sim1*) and the actual protocol (marked with *sim2*);  $p_{len} = 0.1$ ,  $q_{ret} = 1.$ ,  $K_u = 5$ ,  $K = 10$ ,  $L_{ms} = 5$ , and  $MNRS L = 10$  (only for *sim2*).

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. (6.19). In both cases, delays are almost constant for small rates, but monotonically increase, and then go to infinity as the actual total incoming rate,  $\lambda_{total} = \lambda_d + R_u^{succ}$  goes to  $W_{total}^{max}/(E[L_{nrst}^1 - 1]T_s)$  ( $\approx 8.26e^{-3}$  in the results). Due to the closed-loop behavior of the clients,  $R_u^{succ} \leq \lambda_u$ . Hence, in the figures, the marginal rates (at which delays are infinite) look larger for uplink under the same parameters. Note that the uplink delays are larger than the downlink delays by as much as  $T_s + E[\hat{M}]/R_u^{succ}$  under the same condition.

Figure 10 (a) and (b) compare 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 continuously 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.

## 7 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 (delay bound and zero loss), class II (delay bound and some loss tolerable), and class III (no delay bound and zero loss). Class-III traffic is divided into two subclasses according to whether the traffic is delay-sensitive or not. The protocol is based on the framing strategy for class I, a modified framing strategy for voice (class II), and a reservation scheme for class III, where class I and II have priority over class III. 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. A modified scheme for voice transmissions was presented for efficient link utilization at the cost of some packet losses. When there is no eligible class-I and II traffic, class-III traffic is transmitted. Uplink class-III transmission reservation and uplink class-I and II connection establishment were requested using the reservation scheme. We finally analyzed the average delay and throughput of the reservation scheme for class-III traffic, and presented the numerical calculation and simulation results.

## A Proof of Eq. (4.3)

Including the new connection, if all of the required slots for class-I connections can be reserved, the new connection can be established. Since  $M_i$  slots need to be reserved for connection  $i$  of type  $g$ , within a type- $g$  frame, a total of  $M_g^f$  ( $= \sum_{\{T_i=T_g^f\}} M_i$ ) slots should be reserved. Note that  $T_G^f$  is the least common multiple (LCM) of  $\{T_1^f, T_2^f, \dots, T_G^f\}$ . Since each frame repeats itself, it is enough to consider only one type- $G$  frame of duration  $T_G^f$ . There are  $M_G^{max}$  ( $= T_G^f / (T_s + T_{m.s})$ ) slots within one type- $G$  frame. There are also  $N_g$  ( $= T_G^f / T_g^f$ ) type- $g$  frames in one type- $G$  frame for all  $g$ . From  $g = 1$ , we start to reserve  $M_1^f$  slots in each of  $N_1$  type-1 frames. In each frame, slots are reserved from the beginning part. Next, for  $g = 2$ , we reserve  $M_2^f$  slots in each of  $N_2$  type-2 frames among  $M_G^{max} - N_1 \cdot M_1^f$  available slots. With the same pattern, we reserve up to type- $G$  connections, i.e., reserve  $M_g^f$  slots in each of  $N_g$  type- $g$  frames among  $M_G^{max} - \sum_{g'=1}^{g-1} N_{g'} \cdot M_{g'}^f$  available slots. To reserve all of the required slots, we get the condition  $\sum_{g=1}^G N_g \cdot M_g^f \leq M_G^{max}$ , which is equivalent to Eq. (4.3).  $\square$

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