QoS Support of Real-Time Multimedia Traffic in Wireless IP Networks

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ABSTRACT

Supporting real-time multimedia applications over the packet wireless network poses a huge challenge due to the stringent QoS requirements and the time-varying and location-varying characteristics of the wireless channel. In this paper, we study a downlink packet scheduling algorithm designed to support real-time applications with hard deadlines over the wireless network. The scheduling depends on the channel characteristics as well as the deadline of each packet. A hybrid TDMA/CDMA physical layer is used to assess the channel condition and to support the target BER of different users. RTP packet headers are used to calculate the deadline of each packet. A modified form of EDF is used to schedule packets in order of urgency and channel condition. Our results indicate that by jointly considering the channel information and the application-level information, a significant performance gain can be achieved. In addition, a more fair service is achieved even though channel conditions of different users vary greatly due to propagation losses and fading effects of the wireless channel.

Keywords: Wireless, cellular, IP, CDMA, QoS, packet scheduler

1. INTRODUCTION

Voice has been the primary driver of the tremendous growth in cellular wireless networks. However, there are increasing demands for a larger variety of multimedia services over the wireless network. Unlike in the voice-only system, the future networks must be able to handle heterogeneous traffic with varying characteristics and quality of service (QoS) requirements. Some of the traffic will be delay-sensitive (e.g. voice calls, video conferencing) while others will be less sensitive to delay but more sensitive to errors (e.g. email, file transfers). Circuit-switched connections currently employed in wireless networks will not be able to provide sufficient flexibility and efficiency required to support such heterogeneous traffic. Thus, packet-switched networks that encompass the wireless link are seriously being considered[1]. Because of the popularity and ubiquity of Internet Protocol (IP) networks and related applications, the most promising candidate for the future wireless network is IP-based. Although there are several open issues with building wireless IP networks (e.g. efficiency, QoS support), the benefits of having a consistent network infrastructure is significant not only in terms of managing complexity, but also in taking advantage of the ever decreasing costs and increasing performance of Internet equipment. Such a network may also simplify the development and deployment of new features and enhancements throughout the entire network.

The use of applications that require better than best effort service provided by the traditional Internet has been increasing over the years, and the mechanisms needed to provide the necessary QoS guarantees have been an active area of research.[2] Providing guarantees on various QoS measures such as bandwidth, delay, delay jitter, and packet loss in the packet wireless network has been more of a challenge due to physical characteristics unique to the wireless channel including limited bandwidth and high and bursty bit error rates (BER). Supporting multimedia services over the wireless network has been the subject of numerous studies. However, in these works, the focus was on the performance of the wireless link alone rather than on the end-to-end QoS perceived by the underlying applications and the end users. Thus, they fail to take advantage of the network-level and the application-level requirements and information.

While the third generation mobile wireless network developments such as WCDMA and CDMA2000 have increased the available capacity of the wireless network over the current system[3,4], the rapidly growing demand for bandwidth will continue tax the capacity. The increased burstiness and different QoS requirements of multimedia traffic further the case for packet-switched wireless network over the traditional circuit-switched one. In this paper we present a new packet scheduling algorithm for the wireless link that is designed to support real-time applications in an end-to-end wireless IP network. The main goal of the scheduler is to allocate the limited amount of resources to contending sessions while satisfying the QoS needs of each session, and this should be done in the manner that makes the most efficient use of the available resources. Only the downlink (from the base station to the mobile station) is considered in this paper. The motivation behind studying the downlink first is that since the base station (BS) is aware of the states of the packet queue and the channel, an intelligent decision can be made utilizing the information. Also, despite the initial belief the uplink was the bottleneck, there are some evidence that the downlink may be true bottleneck[5]. This problem may worsen as more asynchronous services are carried over the wireless network. The presented scheduling algorithm relies on information available from different layers of the protocol stack, namely the CDMA-based physical layer and Real Time Protocol (RTP)[6] layer, which is widely used in transporting real-time traffic over the Internet. By taking advantage of a multitude of layers of information available in each packet, better performance can be achieved. Admission control plays a critical role in providing the desired QoS by allowing only those sessions that can effectively be serviced by the related packet scheduler. Given the performance results of the packet scheduling policy, a suitable admission control can be developed.

2. NETWORK MODEL

A wireless IP network with the cellular architecture of the type shown in Figure 1 is assumed in this paper. One or both endpoints of the network may be attached over the wireless link. The standard IP network infrastructure is brought as close as possible to the wireless link. The BS also functions as an IP router in this architecture. In this architecture only the BS must be specially modified for the wireless link; all other parts of the network infrastructure are independent of the wireless link and may remain unmodified. To ensure adequate performance of a given application, not only must the wireless link provide adequate QoS, the wired IP network (i.e. the backbone network) must also provide adequate QoS. It is well known that different types of traffic have different QoS requirements. Some traffic, such as Transmission Control Protocol (TCP) traffic, tends to be more sensitive to errors and less sensitive to delay. Other traffic, such as real-time traffic (i.e. RTP traffic), tends to be more sensitive to delay and less sensitive to errors or packet losses. Among many proposals, the QoS architecture most likely to be deployed on the Internet in the near future seems to be the Differentiated Services (DiffServ) model[7]. It uses the Type of Service (TOS) field of the IP header to classify each packet into different aggregate, and each aggregate is then serviced using a particular per-hop behavior (PHB) based on the TOS classification. Of the PHB's that have currently been standardized by the IETF, only the services provided by expedited forwarding (EF)[8] seem suitable for real-time applications. A preliminary work by Nadas et al.[9] and our own work indicate that a service utilizing the EF PHB can provide the small delay and delay-jitter requirements of real-time applications.

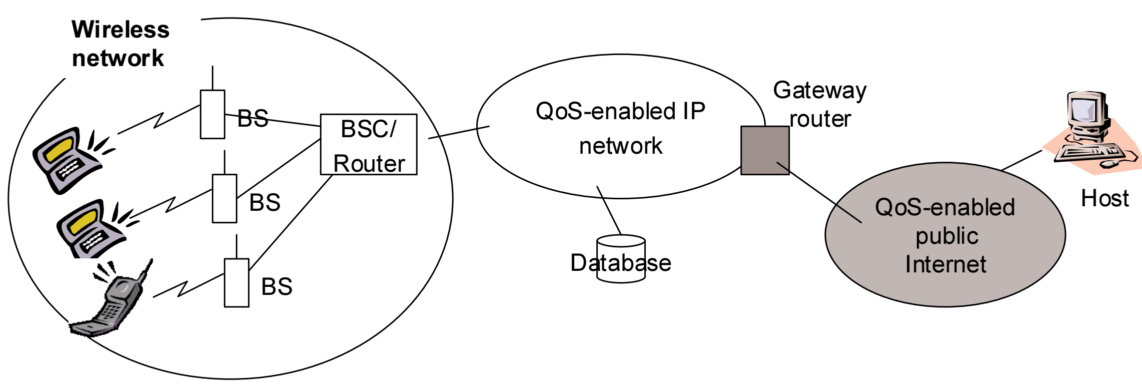

Figure 1. Simplified architecture of an all-IP wireless network

All mobile stations (MS's) share a common RF channel for downlink communication using a slotted direct-sequence code division multiple access (DS-CDMA) protocol. The advantages of DS-CDMA for cellular voice communication have become well known in recent years[10]. Some of these advantages have been exploited to support multimedia services over the wireless channel[11,12]. In general, the capacity of a DS-CDMA system is interference-limited, i.e. the interference from other users, both in the same cell and the neighboring cells, limit both the quantity and the quality of on-going connections. Thus, in the downlink, the total transmit power emitted by the BS must be carefully controlled. Otherwise, the signal-to-interference ratios (SIR's) of on-going connections will degrade and result in unacceptably high error probability. In the current CDMA system, an admission control designed for the worst (or near-worst) case is used to control the interference. This approach has been adequate for circuit-switched connections. However, in our packet-switched system, we employ a slotted model in which data may be sent only to a subset of MS's with on-going connection at any given time (i.e. in each time slot) in order to control the transmit power and thus the interference. Other MS's maintain synchronization contact with the BS using a low-rate channel. Acquisition delays typically associated with such a discontinuous transmission scheme can be decreased using this approach. This hybrid TDMA/CDMA scheme is similar to that proposed by Brand et al.[13] and Ojanperä[14]. Power control traditionally has not been an essential part of DS-CDMA operation in the downlink since the multiplexed signals of different users experience the same channel characteristics and thus do not suffer from the near-far problem of the uplink. However, power control can also be used to support different BER requirements of multimedia traffic by assigning different target power levels (i.e. the target SIR) to different traffic classes. In fact, in several IMT-2000 proposals, the benefits of fast transmit power control in the downlink have been recognized and are included in the proposal. To accommodate different bit rate requirements, both variable spreading gain and multi-code CDMA are utilized[15,16].

One of the problems with extending IP over the wireless network is the overhead associated with transmitting IP and other higher-level protocol headers with each data packet. The problem is more severe for real-time applications since multiple layers of protocol are typically used in the data plane. As is currently done on the Internet, our model uses IP, UDP, and RTP to ensure delivery and playback of real-time data. Together, these headers are at least 40 bytes long. In comparison, a G.729 vocoder using 20 millisecond frames generates a payload of 20 bytes during speech activity. The overhead problem has been addressed by an IP/UDP/RTP header compression scheme[17], which reduces the IP/UDP/RTP packet header to two bytes most of the time when UDP checksums are not used. However, compression and decompression must be performed on a link-by-link basis, requiring each end (e.g. a router) to maintain sate for each session. The need for per-flow state and packet processing raises the scalability concern in large networks. On the other hand, in a wireless network, the BS (which also functions as a router) must keep track of each active MS (i.e. maintain state), and the header compression scheme can readily be accommodated. The information gathered during packet header processing is also used by the proposed scheduler.

3. PACKET SCHEDULING ALGORITHM

The packet scheduling problem for conventional wired packet networks has been the subject of numerous studies over the years (for a comprehensive survey, see Zhang's paper[18]). There also have been numerous attempts to adapt the algorithms developed for the wired network to the wireless network[19]. Such adaptations are necessary due to the unique problems of the wireless channel not commonly encountered in the wired channel. The wireless channel is typically modeled using a two-state Markov chain. The two states correspond to the "good" and the "bad" channel states for each MS. When the channel is good, then relatively error-free transmission can take place. However, when the channel is bad, then the transmission would likely be error-prone. Thus, transmission is allowed only to those MS's that have the good channel state. Different schemes differ in the ways of keeping track and compensating those MS's that were not able to transmit because of the channel condition. The main premise behind this approach is that the channel goodput can be increased by swapping the channel usage between the error-prone (the bad state) and error-free (the good state) flows since transmitting to a MS with the bad channel state most likely would result in the packet being received with an error anyway. The threshold between the good and the bad channel states must be chosen carefully as it can have a large effect on the performance of the scheduling algorithm. Also, the threshold may have to be different for different MS's if the QoS requirements are different.

The fast transmit power control available in the TDMA/CDMA system under consideration can overcome some of the deficiencies associated with the two-state channel model. In a typical DS-CDMA system, all active MS's utilize the entire bandwidth concurrently. Using orthogonal or near-orthogonal codes allows each MS to select only its own signal and regard the signals intended for other users as background noise. Thus the transmit power allocated to each MS must be carefully controlled. Because more transmit power can be allocated to a MS experiencing poor error-prone channel condition, the fidelity of the channel can be increased to the desired level (i.e. the target BER). However, increasing the transmit power to only one MS degrades the performance of other users in the same cell. Thus the transmit power allocated to other users in the same cell must be increased by the same factor. Because the available transmit power at the BS is limited, transmitting over the poor channel may limit the number of simultaneous transmissions and thus the total throughput. Even if the available transmit power at the BS were unlimited, it cannot be increased without bound since the interference in neighboring cells will increase and cause degradation in the SIR's of the connections in these cells.

Clearly, to maximize the overall throughput of the channel, the transmission should take place only to those MS's with good channel conditions. This minimizes the transmit power required at the BS so that the maximum power available at the BS is not the throughput-limiting factor, and it also decreases the interference to neighboring cells. However, real-time applications require timely delivery of data packets, and allowing transmission only to those MS's with good channel conditions may cause a large fraction of packets to be queued at the BS and not be delivered in time for playback. This is the tradeoff that must be made by the scheduler. The scheduler must decide whether it is worthwhile to transmit a particular packet to the destined MS even when the MS is experiencing poor channel condition, thus decreasing the instantaneous channel throughput. Increasing the transmit power to compensate for the poor channel condition of one or more MS's may prevent the delivery of other packets, and, conversely, queuing the packet until the channel condition improves may result in the packet not being delivered on time. In this paper, perfect power control is assumed, i.e. the forward channel condition for each MS is accurately determined and fed back to the BS, and the transmit power is adjusted accordingly to meet the target SIR at each MS.

There are several criteria that can be used to design and to measure the performance of a packet scheduler, including throughput, delay, delay jitter, and packet loss rate. For real-time applications, perhaps the most important criterion is the packet loss rate, i.e. the percentage of packets that are not delivered correctly to the destination by the end of the packet's usefulness (i.e. the deadline). Packet loss can be caused by random bit errors that cannot be corrected at the receiver, buffer overflow at any of the routers in the packet's path, or the aggregate delay experienced by the packet being larger than the maximum tolerable delay of the application. The first factor is ignored in our simulation as the target SIR already accounts for an acceptable level of BER and packet error rate (PER). The second factor can be ignored if the backbone routers provide QoS guarantees appropriate for real-time applications. Thus, only the last factor is considered in our work. The object of the proposed scheduler, then, is to deliver each packet to the destination before its deadline with appropriate BER, and if this is not possible, to minimize the number of packets that violate the deadline. The deadline of the packet is upper-bounded by the tolerable end-to-end delay of the particular application and must accounts for such factors as coding/decoding delay, propagation delay, and queuing delay over the network that serves the BS. A traditional deadline-based scheduling algorithm uses a deadline that is calculated a priori, i.e. when a packet arrives at the scheduler, a pre-determined amount of time is added to the arrival time to calculate the deadline. The deadline, thus, has very little to do with the "real" deadline of the packet. In the proposed packet scheduling policy, the "real" deadline of each packet is assumed to be known or can be estimated with reasonable accuracy. Timestamps are commonly used in transporting real-time data, primarily for intra- and inter-media synchronization (e.g. RTP header has a field for timestamp). By synchronizing the clocks at the sender and the receiver (including the BS in this case) using e.g. GPS, the deadline of each real-time packet can be determined. Since the BS functions as the last hop router in the wireless IP network, the BS performs the IP/UDP/RTP header compression. Calculation of deadline for each packet requires only a small additional effort if header compression is performed.

The proposed scheduling algorithm is based on the earliest-deadline-first (EDF) policy. Because packet lengths are variable and different MS's can have different maximum transmission rates, the deadline for each packet is normalized by calculating a priority metric Φ. Φ of packet i at time t is defined in the following way:

$$\Phi_i(t) \equiv \frac{\lceil (L_i(t)/M_i)/\tau \rceil}{\lfloor R(t)_i/\tau \rfloor} \tag{1}$$

where L_i is the (remaining) length of packet i, M_i is the maximum transmission rate of the destined MS, R_i is the time remaining until the deadline of packet i, and τ is the length of a time slot. Longer packets may require several slots to transmit completely, and Φ corresponds to the fraction of remaining slots, before the deadline, in which the packet must be transmitted with rate R_i to meet the deadline. Note that if Φ is greater than 1, then the packet cannot be delivered before its deadline and thus is dropped at the BS rather than being delivered to the MS belatedly. If Φ equals 1, then the packet must be transmitted in every slot until the packet is completely transmitted in order to meet the deadline. In each slot, the scheduler calculates Φ of each packet and schedules packets in decreasing order of Φ. However, before a packet is allowed to be scheduled for transmission, both the intra- and inter-cell interference requirements must be met. The maximum transmit power available at the BS, and the need to prevent excessive inter-cell interference limit the maximum number of simultaneous transmissions in each slot. They can be quantified using Equations (2) and (3) below:

$$\left(\frac{E_b}{N_0}\right)_i \geq \frac{W}{R_i} \frac{hP_i}{\sum_{j \in S, j \neq i} hP_j + \eta_0 W} \tag{2}$$

$$\sum_{i \in S} P_i < I_0 \qquad S = set\ of\ all\ active\ MS's\ in\ the\ cell \tag{3}$$

Equation (2) must be satisfied for each MS in the cell for each connection to meet the desired level of SIR. W is the chip rate, R_i is the information rate, P_i is the transmit power allocated to MS i, and h is the minimum channel gain among MS's scheduled for transmission in the slot. Setting a limit on the total transmit power by the BS controls the amount of inter-cell interference (Equation (3)). The maximum aggregate transmit power, I_0, is determined a priori by the network operator to satisfy the BER requirements of a certain percentages of MS's in the cell (95% in our simulation). If condition (3) is violated, then excessive inter-cell interference will be generated, and the performance of MS's in neighboring cells will degrade. If the neighboring BS's are allowed to exchange information regarding the status of queued packets at each BS, then it is possible to dynamically adjust I_0 so that the limitation in Equation (3) is more relaxed. We will see later that this results in improved performance.

In contrast to previous works on packet scheduling over the wireless network, packets are allowed to be transmitted to MS's with poor channel conditions under certain conditions. Recall that transmitting to a MS with poor channel condition may reduce the overall throughput of the channel. Thus, if a packet is not considered urgent (i.e. Φ of the packet is less than some threshold), the scheduler may delay the transmission of the packet until the next epoch at which time the channel condition of the destined MS may have improved. In the mean time, by refraining from transmitting to the MS with poor channel condition, more packets may be transmitted utilizing the same amount of power. However, if the packet is considered urgent (i.e. Φ is greater than the threshold), the packet is transmitted with appropriately high transmit power despite the fact that it uses a disproportionately large amount of transmission resource. The threshold that determines the decision is denoted PRIO_THRESHOLD.

4. SIMULATION RESULTS

The performance of the proposed packet scheduler was evaluated using computer simulation. Only the scheduling of real-time packets (specifically voice and videoconferencing traffic) is studied in this paper. Other issues, such as fair use of unused bandwidth and admission control, will be addressed in future work. For the purposes of this study, the bandwidth and power allocated to the real-time traffic is assumed to be fixed. The following system parameters were used in the simulation:

Table 1. Parameters and their values used in the simulation

Item	Value
Carrier frequency	2 GHz
Chip rate	1.25, 3.75 Mcps
Frame length	10 ms
Slot length	0.625 ms
Distance loss exponent	4.0
σ of shadow fading	6 dB
Cell radius	1000 m
Convolutional code rate	1/2, 1/3, 1/4
Max transmission rate for voice terminal	156 kbps
Max transmission rate for video terminal	625 kbps
Avg bit rate of voice call	4 kbps
Avg bit rate of videoconferencing call	56.6 kbps

Both voice traffic and videoconferencing traffic were modeled as Markov-modulated stochastic processes. The voice traffic modeled the traffic generated from a G.729 vocoder with voice activity detection. The videoconferencing traffic was modeled after a variable bit rate (VBR) H.263 encoder. The packet arrival statistics at the BS were derived by simulating an IP Premium Service utilizing EF PHB (using a non-preemptive priority queue). IP/UDP/RTP header compression was used over the wireless link. The wireless channel model incorporated distance attenuation, shadow fading, and Rayleigh fading.

The performance of the proposed scheduler was simulated under moderately heavy traffic loads. The performance was initially compared to a first come first serve (FCFS) scheduler which approximates the common approach of aggregating all real-time packets into a single queue and servicing it in the order of arrival. The FCFS scheduler performed reasonably well compared to the proposed priority-based scheduler when real-time traffic consisted only of voice calls (Figure 2). However, as much burstier VBR videoconferencing traffic was introduced, the FCFS scheduler began to under perform (Figure 3). The PRIO_THRESHOLD for the above cases has been set to 0, i.e. packets are transmitted even over poor channel conditions.

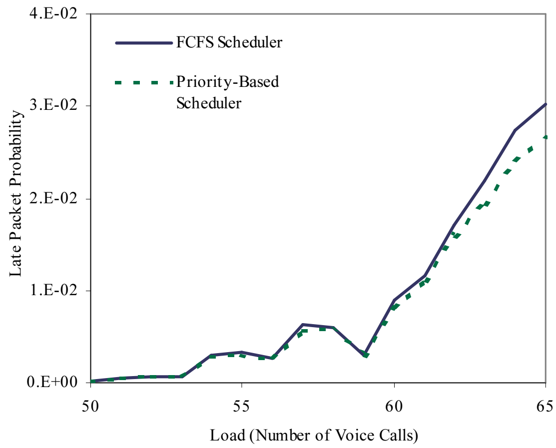

Figure 2. Late packet probability with real-time traffic consisting of voice calls only (MS speed = 5 m/s, W = 1.25)

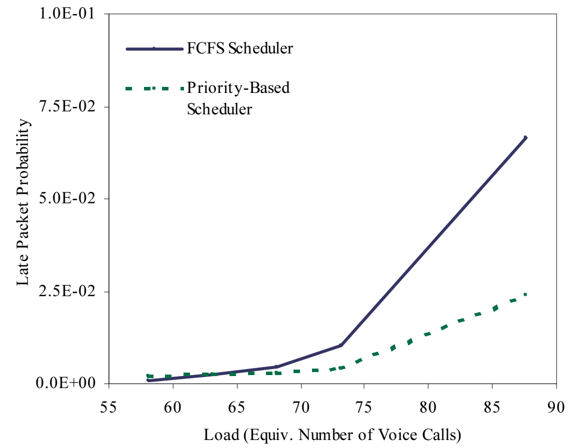

Figure 3. Late packet probability for mixed traffic (MS speed = 5 m/s, W = 1.25 MHz)

In Figure 4, the late packet probability is plotted as a function of PRIO_THRESHOLD with MS's moving at different speeds. If Φ of the packet was greater than PRIO_THRESHOLD, the packet was transmitted even over poor channel, decreasing the throughput of the channel as a consequence. From **Figure 4**, we see that the MS speed does not impact the performance as much. The late packet probability reached the minimum around the PRIO_THRESHOLD value of 0.08 for both speeds shown. The similarity may be due to the fact that the average fade durations for different MS speeds were shorter than the typical amount of time a packet spends in the BS queue. Thus, the fading characteristics appear statistically similar over the duration of the packet's existence at the BS. Simulations were performed with two different values of channel bandwidth (1.25 MHz and 3.75 MHz). Increased statistical multiplexing possibility introduced by the wider bandwidth (and correspondingly larger number of flows) did not result in improved performance for FCFS scheduler. However, similarly considerable performance improvement (i.e. reduced late packet probability) was observed when the proposed scheduling algorithm was used.

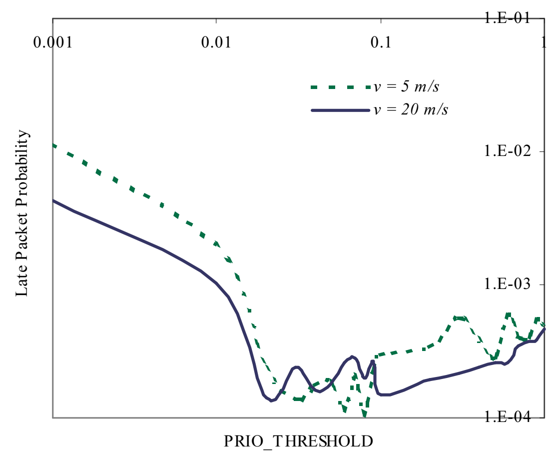

Figure 4. Late packet probability as a function of PRIO_THRESHOLD and MS speed

As previously noted, if neighboring BS's are allowed to exchange information regarding the packets queued at each BS, it is possible to dynamically adjust I_0 so that the limitation in Equation (3) is more relaxed. Thus, more urgent packets in one cell can be serviced by reducing the amount of traffic over the wireless channel in neighboring cells. This scenario was simulated using a 2-cell configuration. Figure 5 shows the improvement in capacity when such a scheme is used. Capacity here is defined as the maximum number of voice calls in which late packet probability is less than or equal to 10^{-3}. The improvement in capacity saturates rather quickly as the load in the neighboring cell decreases to 90% since at this point, the capacity is no longer limited by inter-cell interference but rather by intra-cell interference. This point is controlled by the maximum value of I_0 in Equation (3). If it is known that neighboring BS's can share information, then the maximum value of I_0 can be increased to take better advantage of lower loads in neighboring cells.

Although not explicitly considered in the design of the scheduler, another important performance metric is the fairness of service among the MS's. Because propagation losses can vary several orders of magnitude among the MS's in the same cell, a MS farther away from the BS is more likely to be in poor channel, i.e. more transmit power is needed to achieve the desired level of SIR. Figure 6 shows the cumulative distribution function of the late packet probability. It shows that with PRIO_THRESHOLD equal to 0.1, 90% of MS's experienced late packet probability less than 5.4×10^{-4} while with

PRIO_THRESHOLD equal to 0 (i.e. all packets are urgent and are transmitted even over poor channel conditions), the 90-percentile value is increased to 1.5×10^{-2}. Thus, much fairer service (i.e. more MS's experience an acceptable level of late packet probability) is achieved using the proposed scheduler.

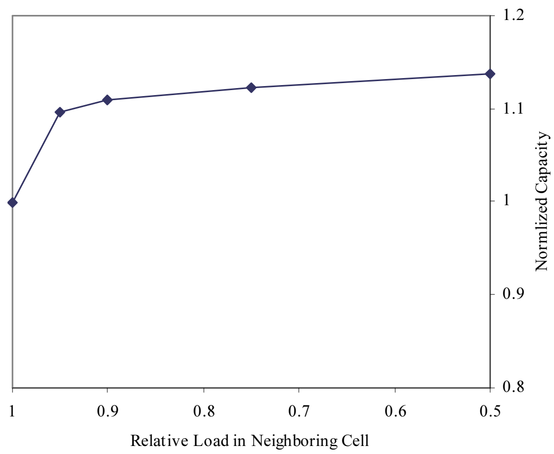

Figure 5. Change in capacity for different loads in the neighboring cell (MS speed = 5 m/s, PRIO_THRESHOLD = 0.08)

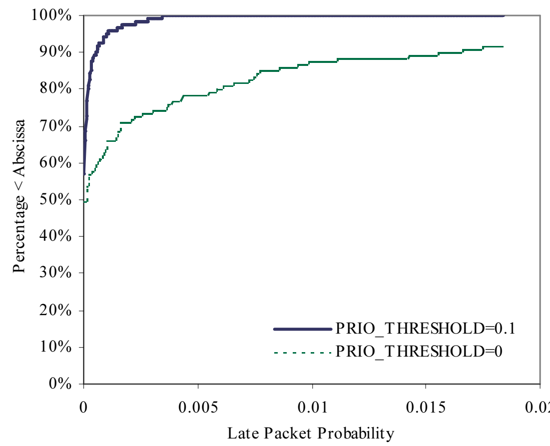

Figure 6. CDF of late packet probability of MS's for different PRIO_THRESHOLD values (MS speed = 20 m/s, W = 3.75 MHz)

5. CONCLUSION

A scheduling algorithm designed to support real-time traffic over the wireless link in a wireless IP cellular network was presented in this paper. The wireless link was assumed to be the last hop of an end-to-end IP network. The real-time applications were supported using RTP over UDP/IP. Fast transmit power control of a hybrid TDMA/CDMA physical/data link layers was used in conjunction with the time information available from RTP and RTCP layer to design a data link/network layer packet scheduling algorithm appropriate for real-time traffic. Exploiting information available from a multitude of protocol layers, a packet scheduling algorithm with much improved performance over FCFS scheduling was designed. Our scheduling algorithm is also easily able to take advantage of lower loads in neighboring cells using a simple metric. As the demand for multimedia services over mobile wireless networks continues to grow, it will be important to utilize algorithms that jointly consider and exploit all available information, even from different layers of the protocol stack, to more efficiently utilize the limited resources.

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