The Performance of a New Wireless MAC Protocol

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Abstract

In wireless communication we have to share the bandwidth among many physically dispersed device. The Medium Access Control (MAC) protocol plays a crucial role in the network performance. In this paper, we present a new MAC protocol for cellular based system that uses a variable frame length that shrinks and expands according to the system load. Our protocol also uses a per-upper-level-data-packet reservation request, where the reservation is made when a higher level data unit arrives and it uses a variable number of reservation minislots that varies according to the number of stations requesting reservation. Our simulation indicates that the proposed protocol has a very good performance with a mixture of voice and data traffic.

1 Introduction

Wireless communication is going through a huge transformation that is paralleled only by the huge growth in wired communication in the past decade. Wireless communication today doesn't mean only cellular phones but it is also Third generation multimedia personal communication systems [12] Wireless LANs, Disk Top Network and the Bluetooth vision [8], second and third generation cellular data systems [3].

Medium access control plays a very crucial role in the performance of wireless networks. In wireless networks the limited bandwidth channel is shared among all the users and we must have a good medium access control in order to fairly share the channel among all the users, and in the same time maximizing the channel utilization and minimizing the delay. The first attempt to do so was in the ground breaking ALOHA network [1] built at the University of Hawaii and used distributed terminal to access a central computer. However, in today's networks, different types of traffic, with different quality of service require a completely different MAC protocols.

In this paper, we assume our network to be a wireless access loop with a base station that is connected to a wired network. We also assume that the mobile devices transmit 53-bytes cells. This type of cells makes it compatible with wireless ATM networks and is suitable for wireless networks since a larger data transfer unit (such as TCP/IP frame) increases the probability of errors. Even though the application program may send and receive larger size packets, each packet is divided into smaller cells that will be transmitted in the slots in each frame. We assume that the slot size is 60 bytes. That is suitable to carry an ATM cell and the medium access control overhead.

2 Related Work

In this section, we present a brief preview of medium access control (MAC) protocols for cellular based networks. There is a huge volume of literature on that topic and we can not cover it in the space allocated to this paper, however the interested party may refer to [11]. MACs can be divided into fixed assignment methods and non-fixed assignments methods. Fixed assignment methods are suitable only in rare cases where we have a stream of data from a specific number of nodes, and there will not be any changes for a long period of time, that is analogous to circuit switching which is not very useful in bursty computer data or even for variable rate multimedia traffic. Non-fixed assignment methods are random access (ALOHA and CSMA), or demand assignment where the bandwidth is allocated based on demand that usually changes during the course of the connection. Due to the limited space here we will, in few sentences, explain few of the previously proposed protocols.

In [10] the authors proposed a protocol called Distributed Queue Request Update Multiple Access (DQRUMA). In their protocol they used the access request on a per slot basis and used two different request access protocols. We believe that request and acknowledgement on a per slot basis is computationally demanding on the base station and is not very effective especially when the data to be transmitted will usually take more than one slot..

A hybrid TDMA/CDMA protocol is described in [6] where multiple CDMA codes per slot are used together with a priority for real time traffic to control the access to the uplink. A token is used for scheduling in SWAN protocol [2]. Many other protocols have been proposed such as [5,13,4].

3 Our Protocol

In this paper, we assume a wireless local loop where a number of mobile users are roaming in a small geographical area covered by a single base station. The base station is connected to a wired network. The users could be a cellular phone users, or data users using a handheld device to access the Internet or send a receive messages through the wired network.

We assume an FDM transmission where the uplink and down link uses two different frequencies. Scheduling on the down link is done by the base station and is easy to implement since the base station have all the information and can schedule transmission in any way it decides. In this paper, we concentrate on the uplink where mobile nodes have to compete in order to be able to successfully transmit to the base station.

The mobile nodes could be transmitting regular data or voice signals. Of course the requirement and QoS for voice and data are quite different. Our proposed protocol can adjust the B.W. allocated to each group. Also, it can control the time to acquire the medium by changing the number of reservation minislots according to the number of nodes requesting reservation. Our protocol is a variation of the protocol proposed in [9] in order to accommodate the different requirements of voice and data transmission.

MAC protocols that depend on competition for nodes to acquire the right to transmit are divided into two main categories. Either the competition is done on a per connection basis, or on a per cell basis. In a per connection basis the mobile nodes requests the right to transmit and once granted that right, it will be repeated in every frame until the connection is terminated. On a per cell basis, the mobile node requests transmission every time it receives a cell to transmit. In our protocol, the node request permission to transmit on a per higher-level-data-unit basis. If a node receives a message or a talk spurt starts, the node will compete to acquire the right to transmit. Once the right is granted, the node can continue to transmit, and increase or decrease the B.W. allocated to it without competition until the end of the talk spurt or burst of data.

This is very important point since our model deals with voice and data. Voice traffic is not a typical VBR traffic. Statistically speaking, voice conversation consists of talk spurts with an average of 1 sec. followed by silence periods with an average of 1.35 seconds [7]. If a slot will be reserved permanently for every voice user with the required bandwidth, that would waste more than 50% of the allocated bandwidth. If we compete to acquire the cannel for each slot, that resulted in the major bottleneck being acquiring the medium for the talk spurt. Our proposed protocol deals efficiently with this point resulting in a reasonable delay and loss probability.

The uplink channel is divided into slots; each slot is 60 bytes long. The reason for 60 bytes is to be able to carry the 53-byte ATM cell and 7 bytes for MAC overhead and error correcting codes. Slots are grouped into frames such that each frame starts with a number of contention minislots (We assume the length of the minislot to be 15 bytes), followed by a variable number of regular slots. Nodes request the right to transmit by sending their ID in one of the contention minislots. Nodes compete for the minislots in a regular S-ALOHA fashion without binary exponential backoff. We are not assuming any capture effect. In reality capture effect might result in one of the colliding requests surviving the collision, so it might improve the performance and our work here is considered a lower bound. Also we did not deal with errors. We are assuming that there is enough room in the MAC header to incorporate some error detection and/or correction capabilities, in order to concentrate on the performance of the protocol itself.

Frames are separated by a time that is equal to a one regular slot. During that time the base station will be able to respond to the requesting nodes and let them know the position and the number of the slots that is allocated to them in the next frame.

The operation of the protocol is as follows.

1. The base station sends on the down link the number of the competition minislots in the next frame, the length of the next frame, and the slot allocation to each mobile in the next frame.
2. If a node wants to start transmission, and currently this node is idle, it sends its ID in one of the completion minislots.
3. If only one node sends a request in any competition slot, the base station makes a reservation in the next frame and update the frame length.
4. After that, the slots starts, each node can send a cell in one of the slots that is allocated to it. If the number of cells in the mobile buffers increases past a threshold, it sets a bit in the cell trailer to indicate that it wants one more cell in the next frame (increase its allotted bandwidth). If the number of cells in the buffer falls below the number of cells allocated in the current frame, it sets another bit in the cell trailer to indicate to the base station that it doesn't want this cell in the next frame.
5. Then the base station adjust the number of cells allocated to the currently active terminals by reducing the number of cells allocated to each node that requested a reduction, and increasing he number of cells for nodes that requested increase (bounded by a maximum number of cells for each node in a frame). Then go to step 1.

For the number of minislots in a frame, we considered two variations of the protocol.

- In the first method, mixed competition (MC), we consider a scenario where the base station dynamically sets the number of competition minislots according to the load. In this method, if the number of collisions in the previous frame is greater than the number of idle slots, the base station doubles the number of minislots in the next frame. If the number of idle minislots is more than the number of collisions, the base station reduces the number of minislots in the next frame by half.
- In the second method, separate competition (SC), we separated the competition minislots for voice and data. We assumed a fixed number of competition minislots for voice (we considered this number to be 10, 20, 30, and 40% of the number of voice nodes), while the data minislots are adjusted similar to method 1 (however the increase is by 50% and the decrease is by 33% instead of doubling and halving).

4 Simulation Results

We simulated the above protocol for a variable number of mobile nodes as we explain later. A maximum number of 200 nodes and a transmission rate of 4Mbps are considered. Every mobile node is equipped with a buffer to store the packets until transmission. The buffer size is B_{max} .and is set to 100 cells. The nodes are divided into two groups, voice nodes and data nodes.

The voice nodes are transmitting digitized voice signal. We assume a model where the speaker alternate between talk spurts and silence. The talk spurts and silence are assumed to be exponentially distributed with average of 1.0 and 1.35 sec.[7] During the talk spurt, we assume that digitized voice at 32Kbps is being generated. The mobile node collects sampled voice and arranges it in 48 bytes cells, add the header (both ATM and MAC) and store it in the buffer until it is transmitted. Since voice cannot tolerate excessive delay, we assume that a packet will be discarded if it was not transmitted in a specific amount of time (TD_{max}). We assumed two values for TD_{max} , 32 msec, and 100 msec. If a packet is discarded, it is called lost (probability of loss is a good indication of the quality of the transmitted voice signal [7]).

For data sources, we assume that packets arrive at each data node with exponential distribution with average of 1.5 packets per second. The packet length (in cells) is exponentially distributed with an average of 30 cells. Using these values, the average load generated from a data node is equivalent to a voice node.

Figures 1 and 2 show the results of using separate competition minislots (SC), with , TD_{max}=32 msec, and the number of minislots set to 20%,30%, and 40% of the voice nodes. respectively, and mixed competition (MC) where the voice and data nodes compete for the same reservation minislots, and the number of minislots vary according to the system load. We can see that by having a 40% reservation minislots or MC mode, we can support up to 130 voice calls (utilization of 45%) with 1% probability of loss, and an average delay of 6 msec. We can also see that although 40% reservation minislots or MC give the best loss performance, it is not the best from the delay point of view. Figure 3 and Figure 4 show the same results for TD_{max}=100 msec. From these figures we can see that when TD_{max}=100 We can support up to 200 voice calls (a utilization of about 70%) with a delay of 35 msec. .

Figure 5 shows the average delay in μsec for both voice and data for the two methods, and Figure 6 shows the probability of loss for TD_{max}=100msec. The data collected for these two figures assume that there are 100 data nodes in the background (as an added load to the system). In this case, it is clear that having separate competition slots produce the best results for both the probability of loss and delay and we can support up to 100 voice calls. Figure 7 and Figure 8 show the same results for TD_{max}=32 msec. Again, we find that for TD_{max}=32 msec, setting the number of competition minislots to 40% of the number of voice nodes gives the best performance, and we can support up to 50 calls. These figures also show that the delay and loss probability for voice is better than data mainly because of the preferential treatment of the voice over data in the competition minislots.

5 Conclusion and Future Work

In this paper we introduced a new medium access control protocol for cellular based systems that can be used efficiently where the nodes are sending a mixture of digitized voice (phone calls), and regular data communication. Our simulation results indicate that with a B.W. of 4 Mbps our proposed protocol can afford a wide mix of voice and data transmission. For future work, we are concerned with dividing the frame (as well as the competition minislots) between voice and data in order to guarantee a specific QoS of voice call and the associated call admission control, we are also interested in using TDD n order to minimize the delay and probability of loss.

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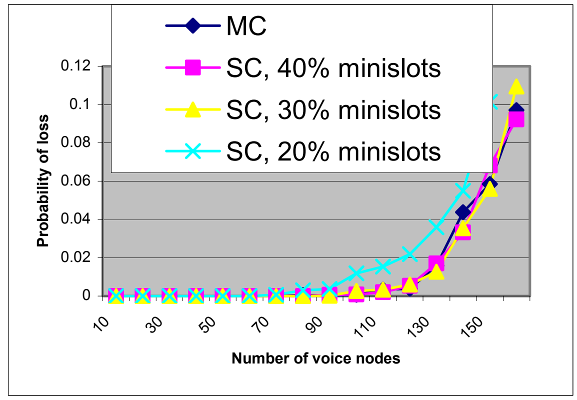

Figure 1: Number of voice nodes vs. average probability of loss for TD_{max}=32 msec.

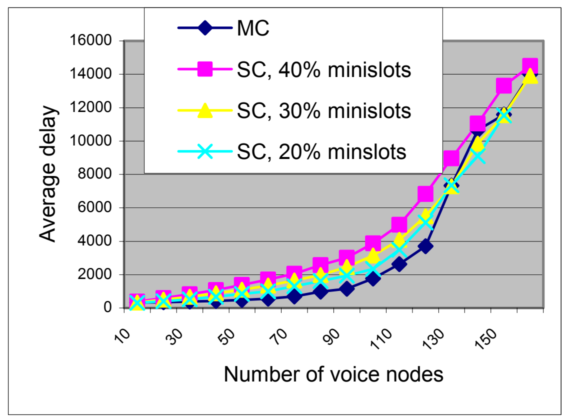

Figure 2: Number of voice nodes vs. average delay for TD_{max}=32 msec. (delay in µsec)

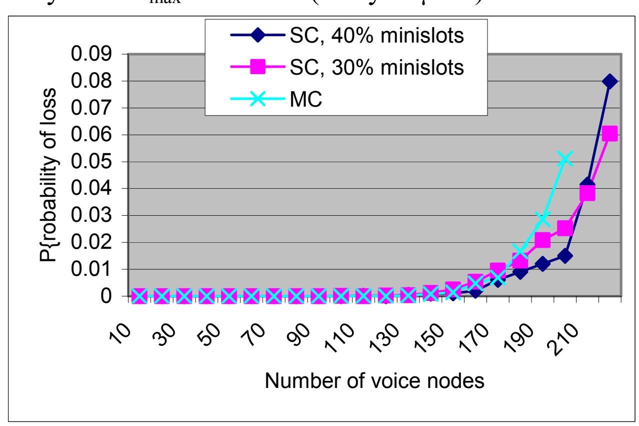

Figure 3: Number of voice nodes vs. average probability of loss for TD_{max}=100 msec.

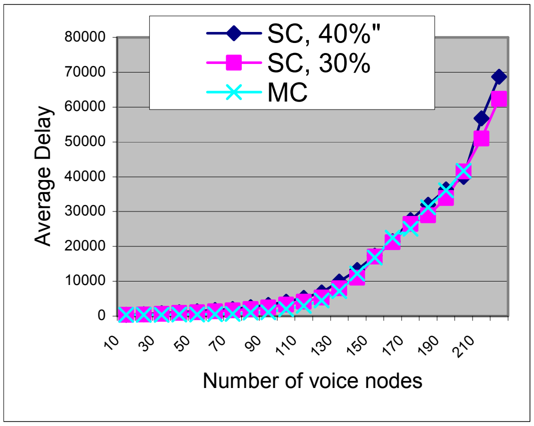

Figure 4: Number of voice nodes vs. average delay for TD_{max}=100 msec.(delay in µsec)

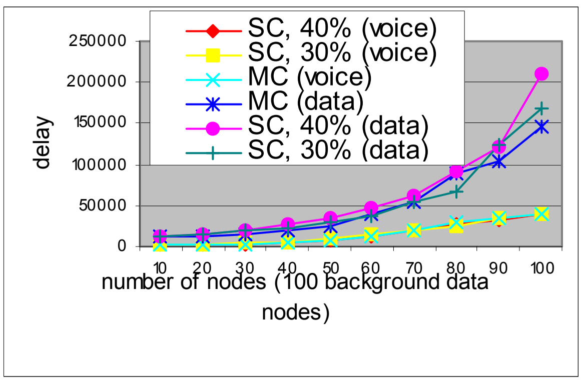

Figure 5: Number of voice nodes vs. average delay assuming 100 data nodes as a background traffic and TD_{max}=100 msec. (delay in µsec.).

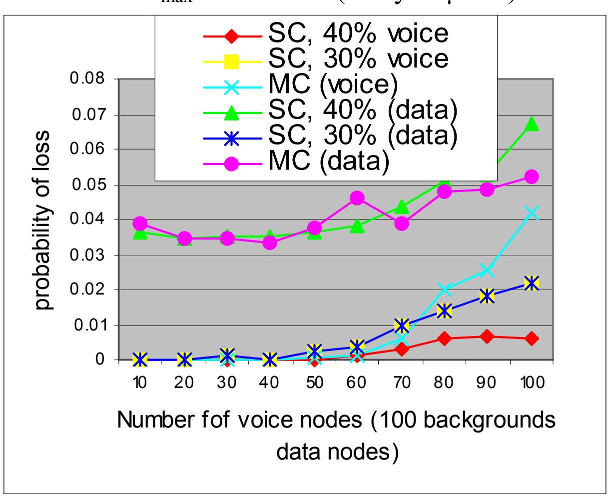

Figure 6: Number of voice nodes vs. probability of loss assuming 100 data nodes as a background traffic and TD_{max}=100 msec.

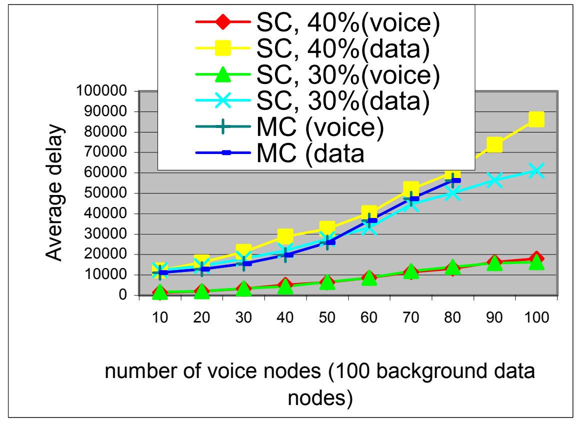

Figure 7. Number of voice nodes vs. average delay for TD_{max}=32 msec. (assuming 100 background data nodes)

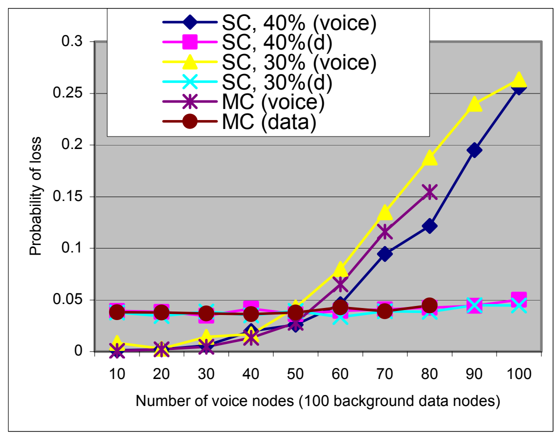

Figure 8. Number of voice nodes vs. probability of loss for TD_{max}=32 msec. (assuming 100 background data nodes)