Dynamic Channel Adaptive MAC with Frame Length Prediction (DCAM/FLP)

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Abstract

Medium Access Protocol (MAC) for wireless networks plays a very crucial role in the network performance. Wireless links have a smaller bandwidth compared to wire-line links, they are also more susceptible to interference than wire-line links. In this paper, we propose a MAC protocol for cellular networks. We assume two different types of traffic (real-time audio and regular computer traffic) each with its own different QoS requirement. It also assumes that the link is error prone and alternate between periods of good and bad states. We used simulation to compare our protocol with existing protocols and we have shown a much better performance especially for real-time audio transmissions.

1 Introduction

Wireless communication today doesn't only mean cellular phones but it is also personal communication systems [14] [18] Wireless LANs, Disk Top Network, the Bluetooth vision [11], and second and third generation cellular data systems [3].

In wireless communication, we have to communicate over an unreliable communication link with a limited bandwidth where the signal is subjected to multipath interference, reflection, and diffraction which lead to a high bit error rate. 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 maximize the channel utilization and minimize the delay.

In this paper, we assume a cellular network where the area is divided into cells, and in each cell a base station is controlling the access of the mobile nodes in its cell. The nodes are either sending a digitized voice signal, or sending data packets. The audio nodes should transmit the audio packet within a pre-specified time limit, if not transmitted by that deadline, it will be dropped. On the other hand data packets could be delayed and the only limiting factor is the size of the buffer, if the buffer is full and a packet arrives ready for transmission, one of the packets in the buffer is dropped.

We assumed the Gilbert-Elliott channel model [8,9], in which the state of the channel alternate between *good* and *bad* states. The duration of the good period and bad period are exponentially distributed random variable with mean τ_g and τ_b respectively. In our simulation we used τ_g =0.1 sec. and τ_b

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=0.0333 sec. The probabilities of bit-error in each of the two states are p_g and p_b respectively. We used $p_g=10^{-6}$, and $p_b= 10^{-2}$ similar to [17].

We assume that the channel state is estimated at the receiver, and is feedback to the transmitter. If the channel state is *bad*, then some sort of error correction coding is used.

Our contribution in this paper is designing a protocol that is suitable for both data and real-time audio transmission. In this paper, we present the protocol and also present simulation results to compare the protocol to existing protocols. Our results show that our protocol does have a very good performance and compares favorably with the existing protocols.

The organization of this draft paper is as follows. In section 2, we present previous work, Section 3 presents our protocol, section 4 shows the simulation results. Section 5 is a conclusion.

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 cannot cover it in the space allocated to this paper, however the interested party may refer to [17]. In this section, we present only a small subset of the more relevant protocols.

In [13] 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 In [19], the idea of DQRUMA was extended fro CDMA systems. In [1] The authors proposed a MAC protocol for both real-time and regular data. However their work did not guarantee QoS for audio nodes.

Chen, -et al [5] have proposed a novel Medium Access Control protocol RAP for wireless data networks. RAP used a random selection process to reduce the number of simultaneous transmission attempts.

The fundamental idea of RAP is that the base stations only poll those active node (with packets ready to transmit) under their coverage and use random selection to permit requests.

Adaptive Modulation Reservation TDMA system (AM-R-TDMA) was proposed in [14]. The frame consists of reservation control sub-frame and the Traffic Channel (TCH) sub-frame. One frame consists of 20 slots with a frame length of 5ms. In each frame, the first five slots each of which is further divided into four mini slots are used for the channel reservation requesting, the following 15 slots are used for the packets transmission. The mobile uses the mini slots in the reservation control sub-frame to make reservation requests by slotted-ALOHA algorithm. The base station collects request and makes the allocation decision

A hybrid TDMA/CDMA protocol is described in [7] 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 [4, 6, 7, 12,18].

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 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.

In order to allocate the slots more efficiently, the base station keeps a “buffer status table”, which contains the buffer status of each active terminal which includes the length of the buffer and the time stamp of the (Head of Line) HOL cells. The “buffer status table” will be updated by the mini slot requests and piggyback updates (In every transmitted cell by a mobile node, the mobile nodes includes its new buffer status in the MAC header)..

When a mobile node requests a connection, it sends the number of cells in its buffer, and a time stamp for the HOL (Head of Line) cell in its buffer. Also, if the node has no more cells to send, in the last transmitted cell it indicates that there are 0 cells in its buffer, so the base station will not allocate any more cells to it until it successfully compete in one of the reservation minislots.

The base station receives successful reservations during the contention minislots. It updates its buffer status table. Then it calculates the estimated frame length (We used two different techniques to estimate the frame length. In the first, we used the deadline of the most urgent cell in the queues as an indication of the expected frame length, In this case, the predicted frame length is the most urgent cell time + a constant (determined experimentally). In the second we used the number of active terminals as a prediction of the frame length and we set the predicted frame length to be a constant (determined experimentally to be between 4-5)multiplied by the number of active nodes.

Then the base station starts scheduling the transmission by the different nodes. Starting with the most urgent cell to transmit in the audio nodes, we start scheduling as many cells from this node until the next cell has a deadline that is greater than the current time + the expected frame length (this cell can wait till the next frame without affecting its deadline), then, we got to the next audio node and so on.

After completing the scheduling of all audio nodes, if the actual frame length is less than the expected frame length, then we start scheduling cells from the data nodes. By dividing the number of remaining cells by the number of data nodes.

In our protocol, there are channel monitors in each terminal and in the base station. The channel monitor can estimate the channel state in several ways: by using a pilot symbols, by monitoring to the SNR (Signal to Noise Ratio), or by monitoring the channel bit error rate.

After estimating the state of the channel, an adaptive FEC scheme is applied if necessary to correct the channel errors. The FEC scheme in our protocol is the Reed-Solomon (R-S) FEC coding. A Reed-Solomon code is specified as RS(n, k) with s-bit symbols. This means that the encoder takes k data symbols of s bits each and adds parity symbols to make an n symbol codeword. There are n-k parity symbols of s bits each. A Reed-Solomon decoder can correct errors of up to $t=(n-k)/2$ symbols in n-symbols codeword. Using FEC will result in overhead and will reduce the number of data bits transmitted in a slot. (in our simulation we the FEC resulted in 30% overhead and produced an error rate that is similar to the good channel error rate).

The scheduling algorithm schedules the number of cells that should be transmitted in the current frame. In order to transmit these cells with FEC, we must compensate this connection with more slots to accommodate the overhead.

In our protocol, the base station will allocate more slots for a connection that suffers from a high error rate (bad channel condition

The protocol at the base station can be represented by the following pseudocode.

```
While(there are un-scheduled rt- connections{
    k = the node with min. HOL time-stamp
    for(j=0; j<Queue_length(k);j++) {
        deadline=HOL(k)+Td_max
        If(deadline<end_time) {
                    Schedule this cell
                HOL(k)=HOL(k)+T_S
        }
        Else {
        Compensate for the bad channel state
        Break
        }
    }
}
```

Ts is the time difference between generating 2 voice cells. Td_{max} is the maximum delay for a voice cell, if not transmitted within that time, it will be discarded., *end time* is the scheduled end time of the current frame.

For the non-real-time connections, we propose the following scheduler.

```
While (the frame length < the predicted frame length) {
    Find the terminal with the longest queue
    Allocate one slot for this terminal
}
```

4 Simulation Results

We simulated the above protocol, using CSIM [20], for a variable number of mobile nodes as we explain later. A maximum number of 200 nodes and a transmission rate of 4Mbps are assumed. 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 periods. The talk spurts and silence periods are assumed to be exponentially distributed with average of 1.0 and 1.35 sec. Respectively [10, 15] 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} = 100 msec.)..

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.

Figure 1 shows the cell loss rate (CLR) vs. the number of data nodes assuming an error free channel.. The figure compares our protocol with DQRUMA [13]. and the protocol presented in [1]. From this figure we can see that our protocol and the one in [1] outperforms DQRUMA. Our protocol can support up to 200 terminals while DQRUMA can support only up to 140 with a reasonable cell loss rate. For noisy channel, DQRUMA will be much worse than our protocol and will skew the graphs. We decided not to include it in the error-prone channels.

Figure 2 shows the CLR vs. the number of data mobiles for error-prone channels. Our protocol can support up to 200 nodes with reasonable cell loss rate. Figure 3 shows the average delay for data nodes vs. the number of nodes for error-prone channels. Up to 180 nodes can be supported with very little delay,. After that the delay grows exponentially with increasing the number of nodes.

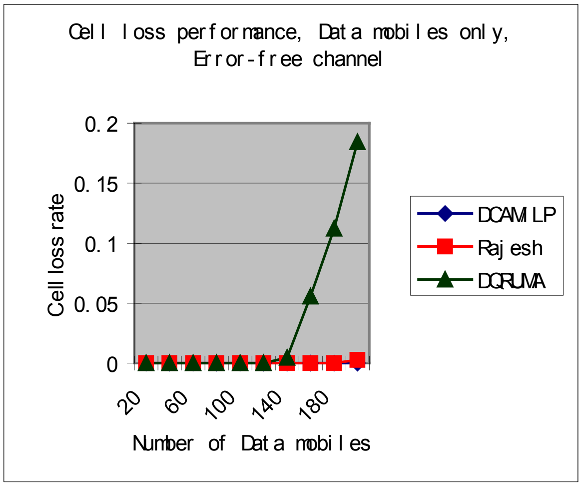

Figure 1: Cell loss rate vs. Number of nodes

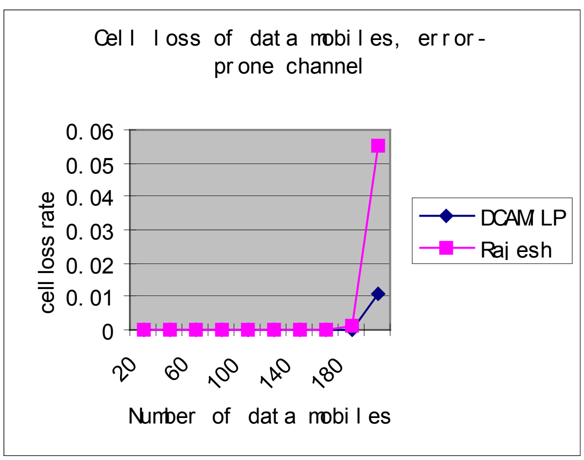

Figure 2: CLR vs. Number of data nodes

Figure 4. shows the cell loss rate vs. the number of voice mobiles. We show the results for our protocol using two different techniques to predict the frame length, and compared it with the one in [1]. Our results show that up to 190 nodes can be supported with a cell loss rate <1% which is considered a very good quality voice transmission.

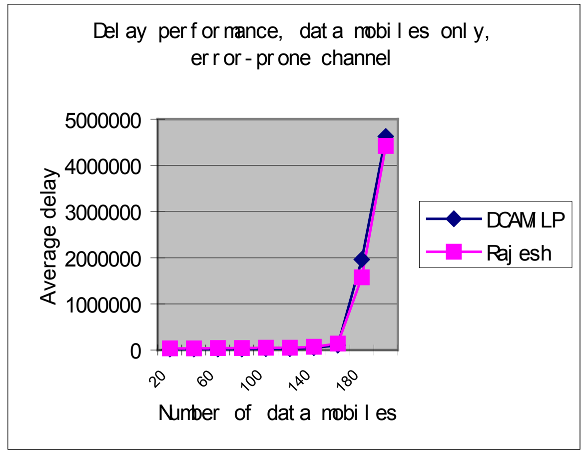

Figure 3: Delay vs. number of data nodes.

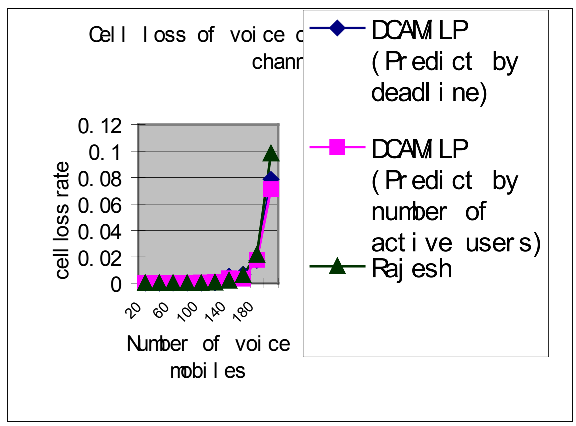

Figure 4: CLR vs. number of audio nodes.

In Figure 5 we show the cell loss rate for audio nodes vs. the number of audio nodes. In this case, we assume that there are 100 data node sharing the channel with the audio nodes. The Figure shows that using our protocol and by predicting the frame length based on the number of active voice nodes, we can support more than 100 voice nodes even when we have 100 data nodes

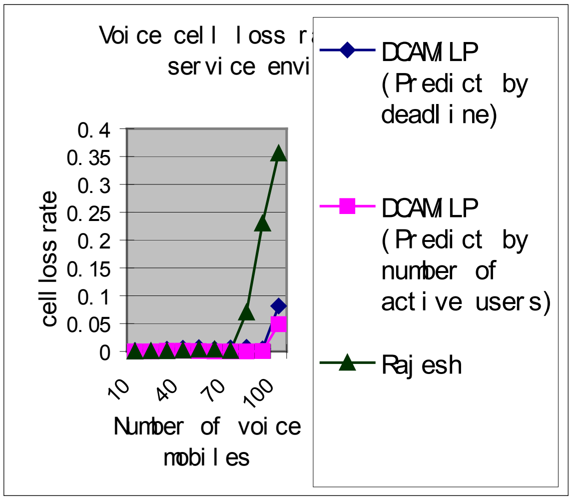

Figure 5: CLR for audio nodes vs. number of audio nodes (100 background data nodes).

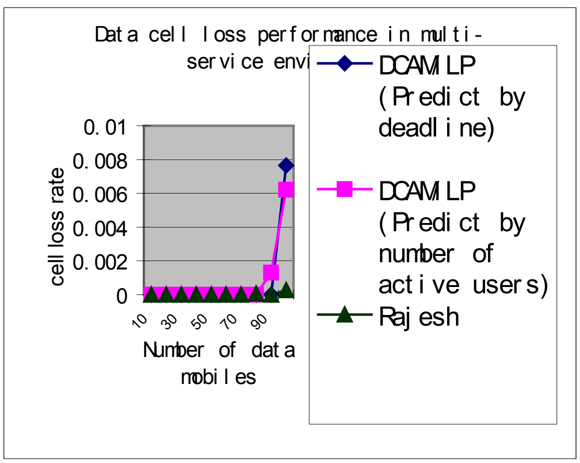

Figure 6: Data CLR vs. number of data nodes (100 background audio nodes).

Figure 6 shows the cell loss rate for data nodes vs. the number of data nodes. It is worth noting here that our proposed protocol behaves as good as the one in [1] when the number of data nodes is less than 80. After 80 nodes, the protocol in [1] is much better. That is because our protocol gives priority to real-time nodes on the expense of data nodes.

From the above discussion we can conclude that our protocol is superior to the other two protocols for real-time connection, where the delay of the packet may result in discarding it. The price we have to pay is higher cell loss rate for data mobiles. Although, we believe that could be remedied by using a larger buffer for data mobile. However, larger buffer will lead to a higher delay, although a delay of few seconds may not have a big effect if you are retrieving you mail, but for interactive applications might not be desirable.

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. We have simulated our protocol and showed that it compares favorably with previous protocols.

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