FAFS: A New MAC Protocol for Wireless ATM

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Certificate

Certified that the work contained in the report entitled
"A New MAC Protocol for Wireless ATM",
by *Sandeep Jain*, has been carried out under my supervision and
that this work has not been submitted elsewhere for a degree.

Dr. Dheeraj Sanghi

May, 1999

Preface

Wireless ATM (WATM) is being envisaged as the next generation wireless solution for meeting the demands of the users. Besides supporting high data transfer rates, it also provides QoS (Quality-of-Service) guarantees to the connections. Since the wireless channel is error-prone (Bit Error Rate $> 10^{-3}$) and bandwidth is scarce, it becomes an important design issue to efficiently allocate bandwidth among users. MAC (Medium Access Control) layer arbitrates the channel access to the wireless medium and is also responsible for bandwidth allocation to different users.

In this report a new MAC protocol (**FAFS - Fair Access Fair Scheduling)** is described, which provides a new contention scheme along with a novel scheduling algorithm that allocates data slots to the sources based on their packet queues, negotiated traffic requirements, signal intensity and the last time when the channel was granted to the connection (for best effort service only). Our results show that the delay probability of the real time packets is decreased using the techniques mentioned in the paper. We also provide improved service to the best-effort sources by avoiding large queues at the mobile terminal.

Acknowledgements

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Contents

Chapter 1

Introduction

A MAC protocol is a set of rules to control the access to a shared wireless communications network medium among various users. These users, in the context of the wireless communications network, are active users within the cell (local users), handoffs from neighboring cells, and new users requesting access within the cell.

1.1 Why Wireless ATM?

ATM with B-ISDN signalling is the technique that is becoming the common dominator for all types of services and networks. From the service point of view, ATM can combine both the transmission of speech and computer data into the wired networks, while scaling well into different network types as LANs, MANs and WANs.

The choice of wireless ATM is motivated by the fact that the future wired communication networks will be dominated by the broadband ATM network. Therefore, the provisioning of transparent transmission of packets between the wireless and wired networks, in which the packet processing time is kept minimum, is highly desirable.

On of the fundamental ideas of ATM is to provide bandwidth on demand. Bandwidth has traditionally been expensive and scarce resource. This has affected the application development and even the suer applications. So far, application development has been constrained by the fact that data transmission pipes cannot support various QoS parameters and the maximum data transmission bandwidth that the applications have to interface with is relatively small. Finally, ATM has removed these constraints. Bandwidth has become truly cheap and there is a good support for various traffic classes.

The benefits of a wireless ATM access technology should be observed by the user as improved service and improved accessibility. By preserving the characteristics of ATM transmission, wireless ATM offers the promise of improved performance and Quality of Service (QoS), not attainable by other communication systems like cellular systems, cordless networks or wireless LANs.

1.2 Problems with Wireless ATM

ATM was designed for networks with *high bandwidth* and *low bit error rate* (BER $< 10^{-9}$). But for the wireless networks bandwidth is scarce (limited by the radio spectrum) and bit error rate is high (BER $> 10^{-3}$). The ATM payload is relatively small; therefore, the addition of any single byte in the header will greatly affect wireless channel throughput efficiency.

The ATM header in the wireless medium may be compressed for the purpose of reducing the packet overhead. However, this header compression will add some complexity at the wireless/wired interface (where compression/decompression takes place) which might increase the total data transmission time.

1.3 Background on MAC Protocols

Multiple access schemes and protocols are classified according to the bandwidth allocation mechanism, which can be *static* or *dynamic*, and to the type of control mechanism being exercised. In general, multiple access schemes can be classified into following three main categories:

- Fixed Assignment Techniques.

- Random Access.

- Demand Assignment.

1.4 Fixed Assignment

These techniques reserve a constant-capacity subchannel to each connection for its whole duration, and they perform very well with constant bit rate connections in terms of both service quality and channel efficiency. It is theoretically possible to calculate the delays incurred in transmission. e.g. TDMA (Time Division Multiple Access) and FDMA (Frequency Division Multiple Access).

But they are inappropriate for wireless networks because of the inefficient radio channel spectrum utilization. Moreover, their performance decreases drastically when they are asked to support many infrequent users with variable rate connections.

1.5 Random Assignment

In these type of protocols, data is transferred "randomly". If a collision occurs the packet is retransmitted after some random delay. e.g., ALOHA. It is well known that these protocols are easy to implement and attain minimum delays under light load, they suffer from long delays and instability under heavy traffic loads. These protocols are generally implemented for ad-hoc networks [1].

1.6 Demand Assignment

Here the users are required to provide explicit or implicit information regarding their needs for bandwidth, which will be assigned on demand. The assignment can either be done either by a central authority (usually a master node called base station (BS)) in case of infrastructure networks [2] or by the communicating nodes themselves as in case of ad-hoc networks.

[1]These networks are created "ad-hoc" by several laptops/palm tops communicating simultaneously in a small area (e.g. conference room) and establish a peer-to-peer network amongst themselves without help from any infrastructure such as wired/wireless backbone

[2]In these networks there is a central authority usually BS which assigns channel slots to the mobile nodes

These protocols are split in two phases - *reservation* and *transmission*. Unlike fixed assignment, demand assignment schemes minimize wasted bandwidth by assigning bandwidth on demand. Also unlike random assignment, demand assignment protocols avoid bandwidth wasted due to collision by providing the connections with free contention bandwidth during active periods. Because of these features, MAC protocols based on demand assignment are the most appropriate for the requirements of integrated wireless networks.

Demand assignment protocols are usually more complex, but perform well under a wide range of conditions, although the reservation phase results in time and bandwidth consumption. The basic problem in these protocols is the delay that packets have to experience before they get the channel access (channel access delay). This problem becomes more severe if we are considering delay sensitive applications. Moreover even after channel has been allocated to a source, it is difficult to decide on the number of data slots to be allocated to it. This allocation must be based on the negotiated traffic parameters and for optimal performance they can also be based on the dynamic traffic characteristics of the source. The latter optimizes the bandwidth allocation but is a complex task and exact way of achieving it is still a research issue.

1.7 Typical MAC Frame

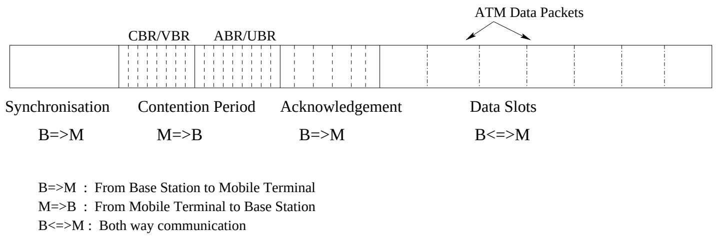


Figure 1: Typical MAC Frame

Figure 1 shows a typical MAC frame. *Synchronization period* is used to synchronize the mobile nodes with the BS. For variable boundaries MAC frame, it is also used to inform about these boundaries. *Contention period* is used by the mobile nodes to send their channel access request to the BS. The *Acknowledgement period* is used by the BS to inform the mobile nodes about the successful contention requests and their subsequent position in data slots for transmission/reception. The *data slot* is used for actual transmission of data.

In these protocols the channel bandwidth is time-slotted and represented by a single or several frame(s), depending on the channel rate and the type and rate of applications. Each frame is divided into uplink (UL) and downlink (DL) periods (channels), each of which is further divided into subperiods (given to individual connections). Downlink traffic can be transmitted in a separate channel (using a different frequency band - Frequency Division Duplex (FDD)), or it can time share a single frame with the UL traffic by using time division duplex (TDD). The latter provides the facility in controlling the available bandwidth by dynamically allocating the length of each period.

1.8 Requirements of MAC Protocol

The protocol should be able to handle different ATM QoS classes i.e. CBR (Constant Bit Rate), VBR (Variable Bit Rate), ABR(Available Bit Rate) and UBR (Unspecified Bit Rate) in a satisfactory manner. Additionally, the protocol must be able to cope with the complex radio environment where fading, multi-path propagation interference and burst-errors further complicate the situation. Since power is scarce at mobile nodes, MAC protocol can additionally implement power-saving feature in itself. This may be implemented by allowing less switching between UL and DL modes (which consumes a lot of power and channel time) and permitting a mobile node to enter in "sleep-mode" if and when the node detects less power. The power issues do not matter at the base station as it is not a mobile node. Hence it is beneficial to reduce the complexity at the remote terminals and moving it to BS. At the same time, the scarce and expensive radio resources must be effectively utilized. Network premises and system protocols need to be efficient and reliable.

Also a reduction in the amount of control signals exchanged between BS and the remote terminals is a must. This will reduce the probability of collision, improve network utilization, and help maintain QoS levels, in addition to mobile terminals' cost and power consumption. One of the basic requirements of MAC protocol is to show a reasonable performance even when the number of mobile node in a cell increases beyond a certain limit. The protocol should not should not show drastic loss of performance in such of overloading. Hence *adaptability* is one of the most desired features of the protocol.

A detailed overview on characteristics and requirements of MAC protocol can be found in [4]. The paper compares some existing MAC protocols and also presents a simulation model.

1.9 Existing MAC Protocols

Recent years have seen the deployment of a large number of MAC protocols. These protocols are mostly TDMA protocols which are either FDD based or TDD based. The MAC protocols belonging to FDD category include DQRUMA [12], PRMA/DA [8], DSA+ [6], DTDMA/PR etc. PRMA (Packet Reservation Multiple Access) was proposed by Rutgers WINLAB for packet-oriented voice transmission over a wireless channel. PRMA was proposed to improve bandwidth efficiency over fixed assignment TDMA. It uses slotted ALOHA schemes for time slot reservation and TDMA for data transmission. It is a simple protocol compared with some other demand assignment protocols; however, it cannot be used for WATM networks. This is because PRMA suffers from a serious problem of variable channel access delay (packet or burst) which does not suit to the requirements of some of the stringent ATM services.

Several other protocols have been proposed for enhancing the performance of PRMA; some of them are Aggressive (APRMA) [7], Data Steal over Voice (PRMA-DSV) [11], Minipackets [11], Centralized [8], and PRMA/Dynamic Allocation [8]. All these enhancement PRMA protocols dealt with improving channel efficiency more and prodding some kind of fairness for data applications. However, the variable channel access delay problem has not been solved.

DQRUMA (Distributed Queuing Request Update Multiple Access) does not make any distinction between VBR and ABR; it treats both as "bursty" traffic.

Consequently, it does not consider any priority handling mechanism. The advantage of this protocol is its use of minislot packets for access contention. In this was, the probability of receiving a packet in good shape is larger than when a longer packet is employed.

A contribution of this protocol is the consideration of two types of reservable subframes with movable boundary: the long-term subframe, meant to be used by CBR and VBR connections, and the short-term subframe (whose slots have to be released after a one-packet transmission., aimed at ABR connections. This may be used to provide more bandwidth to CBR and VBR connections according to their service needs.

Some protocols belonging to this category are MASCARA, PRMA/ATDD and DTDMA/TDD.

The MASCARA (Mobile Access Scheme Based on Contention and Reservation) [14] protocol was proposed as the MAC protocol for the WATM Network Demonstrator (WAND) project being developed with the support of the European Community (EC). The unique feature of the protocol is its use of *leaky bucket* mechanism to allocate data slots to mobile nodes and to monitor the data transmission rate of the mobile nodes. parameters. One drawback of the proposal is the lack of any contention procedure (they are investigating slotted ALOHA and Stack algorithm).

The MAC protocols differ mostly in their contention policy and the scheduling algorithm [1]. Some protocols do not consider priority at all [12] while others use leaky bucket [14], negotiated traffic descriptor [8] or number of waiting ATM packets [6] to allocate data slots. Our protocol uses a combination of these parameters along with fading information to allocate data slots to sources. This helps in improving the throughput for real-time sources and reduces the overall delay in transmission.

A detailed comparison of the protocols is given in [10].

[1] It decides about the number of slots and position of the slots in data frame given the traffic descriptor and other parameters of mobile nodes that have successfully contended for channel (uplink) or for BS (downlink)

1.10 Standards

IEEE is developing an international WLAN standard identified as IEEE 802.11. The project was initiated in 1990, and several draft standards have been published. The IEEE 802.11 draft standard describes mandatory support for a 1 Mbps WLAN with optional support for asynchronous data transfer. The purpose of the standard is two-fold

1. "To provide wireless connectivity to automatic machinery, equipment or stations that require rapid deployment, which may be portable, or hand-held or which may be mounted on moving vehicles within a local area"

2. "To offer a standard for use by regulatory bodies to standardize access to one or more frequency bands for the purpose of local area communication".

This standard is gradually emerging as a mature standard and the manufacturers and vendors are endorsing it.

ETSI (European Telecommunications Standards Institute), the European counterpart of FCC (Federal Communication Commission) is developing another standard for wireless networks - HIPERLAN1 (High Performance radio LAN version 1). HIPERLAN is primarily based on ad-hoc networking applications and support higher data rates. It allows inter-networking with 802 type wired LANs. The channel width is 23.5 MHz and the bit rate is 19 Mbps. Their interface is of CSMA (Carrier Sense with Multiple Access) type and it allows "best effort" with some user control over the useful life time of the packet. So far no complete hardware implementation has been done [17].

1.11 Research Issues

There are various design issues involving the design of the MAC protocol for WATM networks. Some of the key requirements of the protocols were discussed in earlier section. To meet the requirements of different users in a satisfactory manner, the protocol should be able to allocate data slots efficiently and effectively. The scheduling algorithm thus forms an important part of the protocol and consequently a hot topic for research. The performance of the protocol mostly depends on the allocation algorithm. Static traffic parameters are known for the mobile nodes, but efforts are on to model their *dynamic traffic parameters* so as to help in effective allocation of data slots.

Another issue which remains debated is the frame length. Fixed frame length helps to guarantee delays for the delay-sensitive applications. But on the contrary it is an inflexible approach as it is able The variable length frame introduces a difficulty in assigning capacity to MTs with CBR services. Assuming the case of a voice (64 kbps) call, if the frame length (in milliseconds) is less than the time to fill an ATM packet (6 ms), there may be frames where no slots need to be assigned. Otherwise, if the frame length is longer than 6 ms, it might be necessary to assign more than one slot in a frame for this call.

The boundaries of the reservation and allocation periods can be movable or static. Movable boundaries add flexibility to the protocol wherein reservation period can decrease/increase in length depending upon channel load. But this also means informing the mobile nodes about the boundaries in each successive MAC frame. Having fixed slot lengths saves this information transfer though at the cost of flexibility.

The *packet size* for data transmission has also been an important issue. Since the BER of the wireless channel is high, intuitively the use of small packet size seems justified. To improve the reliability of data often a technique called FEC (Forward Error Correction) is used. FEC uses error-correcting codes to combat bit errors (due to channel imperfections) by adding redundancy *(parity bits)*, to information packets before they are transmitted. This redundancy is used by the receiver to *detect* and *correct* errors. This issue is discussed in detail in [16]. The authors on basis of their calculations conclude that for wireless networks the largest possible codeword size for an application should be used. The exact size of the to be used depends on the application under consideration. For instance, if the application is packet voice, the packetization delay increases with an increase in codeword size. Since for packet voice, maximum end-to-end delay has to be bounded, the maximum size of the codeword is limited.

Chapter 2

FAFS

2.1 Introduction

FAFS is a protocol for the infrastructure wireless networks where there is a single base station (access point to the backbone network) and many mobile terminals. In other words, it is based on a client-server paradigm where the server (BS) is used to arbitrate the channel among different clients (MTs).

Ours is a TDMA (Time Division Multiple Access) kind of protocol in which only one carrier frequency is used for transmission. This has been chosen to allocate bandwidth for UL and DL connections according to their requirements. Giving one carrier frequency for each may result in inefficient utilization of bandwidth.

Our protocol uses a combination of parameters like leaky bucket, negotiated traffic descriptor and number of ATM data packets waiting for transmission for allocation of data slots. The protocol combines this information with the fading information to allocate data slots. This helps in improving the throughput of real-time sources and reduces overall delay in transmission. The protocol also uses a novel feature to avoid denial-of-service for the best effort class of services. A UBR source transmits a parameter to base station signifying its unsuccessful contention requests. The BS reserves some slots for the UBR source if the value of parameter exceeds some fixed threshold. FAFS is adaptive to changing network traffic. It has the provision for dynamically controlling the contention period length. The contention period slot actually places an upper limit on the number of mobile nodes that can establish a connection with the BS.

We use minislots for channel access which helps to decrease wastage of bandwidth in case of collision. Power constraints at MT are also taken into consideration wherein an MT notifies BS about start of sleeping mode. This allows BS not to consider that particular MT while assigning data slots in subsequent frames.

Fixed sized frames have been used for information transfer between MT and BS . Thus an upper bound can be obtained on the delays of packets that are to be sent from the BS to the MT and vice-versa.

2.2 Protocol Details

We consider a fixed length frame for transmission between BS and MT. This frame is shared by both UL and the DL connections. Four class of services are considered (as in the case of wired ATM) i.e CBR, VBR , ABR and UBR - considered in order of decreasing priority.

The MAC frame is divided into six fields: synchronization, contention type 1, contention type 2, acknowledgment and data. Field boundaries are variable in nature. The MAC frame is shown in Figure 2.

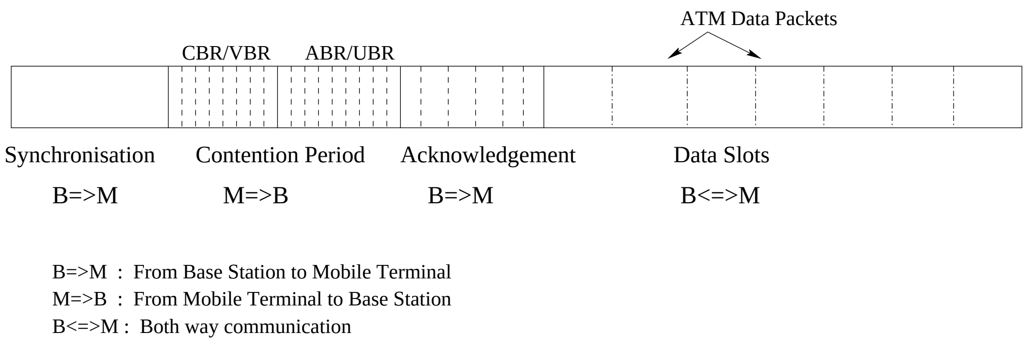


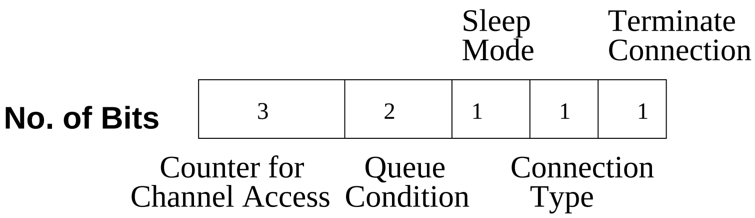
Figure 2: MAC frame

Synchronization field is used to transmit information about the field boundaries to the MTs and also to synchronize the MTs with the BS. Contention field is an important part of the frame as the MTs request for the channel in this slot. The acknowledgement period is used by the BS to transmit information about the allocated data slots to the respective MTs. The data slot is used for actual data transfer between the BS and the MTs.

The number of slots in the contention field poses an upper limit on the number of active connections that can be set up with the BS and consequently on the number of the MTs to which it can provide service. Hence the length of contention field is a critical parameter for design of the MAC frame. If we decrease it, we may be denying service to more MTs; if we increase it, we are consuming more of the frame space leaving lesser space for actual data transmission.

In order to provide QoS we use two different contention slots. First slot is used by VBR/CBR sources and the second slot (contention type 2) is used by UBR/ABR sources. This is done so that the higher priority sources (CBR/VBR) do not have to contend with the low priority services (UBR/ABR) for getting access to the channel. This helps in ensuring delay bounds for the CBR, VBR traffic. A contention field is made up of several *minislots*. Within their contention field, the sources request for data as follows. If a contention field has n minislots, any user requesting in that field picks one of the minislots with probability $1/n$ for sending its channel access requests. If two or more users send their requests in the same minislot, collision occurs. Since we use minislots, the resulting wastage due to collision is not significant. The QoS negotiation for the CBR, ABR and VBR sources takes place once their channel request has been successfully received at the BS. This information is transmitted by the MTs in the data slots. To insure this, BS allocates an UL slot in the data portion of the MAC frame to the new sources (CBR, VBR and ABR) to allow them to send the necessary information about the QoS desired (traffic descriptor). Since UBR sources come under the "best-effort transmission" class, no QoS is guaranteed to them and consequently no QoS negotiation with BS takes place. As the figure shows, the mini-slot for UBR/ABR contains a k-bit access id (to identify the MT) and a 8-bit field (to define the state of the connection) (Figure 3).

First three bits are used to specify the channel denial frequency. An active source might not get any data slots if there is a collision during contention or higher priority sources are alloted the channel. If the UBR source sends a request in the contention



Connection State for ABR/UBR

Figure 3: Connection State

slot and does not get back an acknowledgment it increases the 3-bit counter (the first three bits) by 1. This counter gives an indication to the BS about the number of times service-denial has taken place to the UBR source. The larger the value of this field, more is the time the source has been waiting to get access to the channel. If the counter reaches it maximum value it is not incremented further. It is reset to 0 only when it gets data slots for transmission. This information is used to differentiate among the various active UBR sources which have approximately the same queue length but different waiting time. We can increase the counter bit field to 4 bits obtaining a more precise measurement of the waiting time but at the same time compromising the bandwidth.

Next two bits are used for specifying the queue condition at the MT. There can be 4 possible states using 2 bits. These four states can be used to model the queue condition at MT viz. normal queue, building up of queue, worsening situation and abnormally large queue at MT.

Next bit being on indicates that the MT wants to enter the sleep mode. This information can be used by the BS while assigning data slots in the next frame i.e. the if the source was CBR/VBR or ABR the BS would not allocate any slots to it. The sleep mode can be terminated by the MT by sending a signal in the contention slot with the the next bit on. Next bit is used to specify the type of connection i.e. UBR/ABR or CBR/VBR. Last bit is used by MTs to terminate connection (for CBR, VBR and, ABR only).

For the CBR/VBR sources the mini-slot is same as that for UBR/ABR sources except that the initial 3-bit counter is absent. This is because an active CBR source will be allocated a fixed number of slots each frame hence there is no point in using the information about the time it last accessed the service. For the VBR source the queue length and the tokens in the leaky bucket give adequate information about the source traffic. If an active VBR source has not accessed the channel for an appreciable amount of time, its bucket will be full of tokens. Once a CBR source request has been successfully received at the BS and the connection is established, it does not need to contend for the channel again. When it wants to terminate the connection, it sends an end connection message in the contention slot. ABR sources also behave in the similar fashion. On the other hand, the VBR and UBR sources have to contend for the channel each time they want to send data UL.

We consider dynamic contention period boundaries. This helps FAFS to accommodate the changes in the network traffic. We use the following algorithm to change the contention period length. Initially both contention periods have equal length. This number can be based on the measurements of connect frequency of the MTs to the BS for a particular region in a given interval of time. The length of the contention period is decreased if the percentage utilization of the slots is below 50% and in the rest of slots BS receives no signal. On the other hand if BS receives a corrupted signal in rest of minislots (due to collision contention requests), the BS increases the contention period length. The increase/decrease takes place only if the situation continues for k-successive frames. The change in the length of the contention period is gradual (10-20%) so that the protocol adapts to the new situation slowly. This is desired as the network topology for wireless networks changes quite rapidly.

2.3 Connection Admission Control

A Connection Admission Control (CAC) policy is needed to determine if a new connection can be given channel access without violating the QoS requirements of the existing connections. Various models have been proposed for the admission control for ATM networks [9] [5]. The traffic descriptor for a connection is used to during the connection setup time to determine the traffic characteristics of the source.

For a new CBR source, the bandwidth requested is given by the Peak Cell Rate (PCR) parameter of the traffic descriptor. Using this value and the Cell Delay Variation Tolerance (CDVT), scheduling is performed. If this results in successful inclusion of the new source, then it is granted the channel otherwise it is denied channel access.

A VBR source uses the following parameters in the traffic descriptor: PCR, CDVT, Sustainable Cell Rate (SCR) and Maximum Burst Size (MBS). Given these parameters and a maximum delay of D sec the minimum bandwidth requirement for the VBR source can be found by using the function: min(max((1/PCR+CDVT-1)/(D(1/PCR-1)+CDVT), PCR) , max((1/SCR+CDVT+B-1)/(D(1/SCR-1)+ CDVT + BT), SCR)). For a new ABR source, the Mean Cell Rate (MCR) parameter is used to determine its bandwidth requirements. Again a temporary scheduling is performed and if this is successful, the source is granted the channel otherwise the call is blocked. For UBR sources there is no call admission as there is no bandwidth guarantee for this category of service.

2.4 Scheduling Algorithm

The scheduling algorithm decides the number of data slots a source is supposed to get after it has successfully contended for the channel. For optimal scheduling, it is necessary to satisfy the QoS of large number of users and minimize the cell loss due to ATM cell deadline violation. We use two basic parameters for determining this - the queue information at the MT and the leaky bucket at the BS. As packets are generated at the MT, queues start building up. The queue information is passed along with the channel request in the contention frame to the BS. The BS also maintains a leaky bucket for every connection except the UBR source. The leaky bucket is used to for policing as well as for deciding the number of data slots to be allocated to a particular source. Tokens are added to the bucket at a constant rate (equal to the MCR of the source). The size of the bucket is governed by the maximum burst size of the connection. Every time a slot is alloted to the source, a token is removed from the bucket. So at any time, the number of tokens in the bucket signifies the bandwidth that is available to the source as per the initial QoS negotiations. The information about queues at the MTs is maintained as a variable at the BS.

The scheduling algorithm works as follows. The DL requests are considered first as the volume of DL traffic is generally more than that of UL. The DL and the UL transmission takes place as a single chunk in the data slot. This is to avoid frequent shift between UL and DL transmissions which requires a *guard band*. CBR and VBR are first allocated data slots on the basis of packet deadlines and other information. Then ABR sources are considered. and given their minimum data slots. If there are more slots left, ABR additional bandwidth request and UBR sources are considered (in that order). In case there are no slots left and there are UBR sources with maximum counter value, we reserve a single slot to one of them (chosen randomly). The detailed algorithm is described below.

After contention for the channel has taken place, there are DL as well as UL requests for the channel. First DL connections are considered and then the UL ones. The deadlines of all the active real-time connections (CBR and VBR) are calculated first. Let these deadlines be $d_1, d_2, ...d_n$. These deadlines are then sorted in ascending order. The CBR source having the earliest deadline is chosen first for data slot allocation. Each CBR source is alloted a constant number of data slots per frame. This number is decided by the traffic descriptor of the source and is negotiated at the time the CBR source is assigned the channel. This is repeated for all active CBR sources. Then VBR sources are considered in an order defined by the function given below

$$f(x, y) = 0.7x + 0.3y$$

where
x = Number of tokens for that source
y = Queue length at the source

Priority among VBR sources is thus a function of queue lengths and number of tokens. This gives a fair access to the channel. The number of data slots allocated to it is calculated using the function: *min(no. of tokens, queue length)*. Using such a function allows BS to allocate an *optimal* number of slots to the VBR source. The ABR sources have to be given a minimum number of slots per frame. If after above allocation, no data slots remain or the remaining data slots are insufficient to meet the minimum data slots requirements of ABR sources, earlier alloted slots are *deallocated*. This is done by deallocating slots from the largest deadline and selectively removing the slots that were assigned to the VBR sources (CBR sources are not considered for deallocation because of the service guarantees). This is continued until until there are enough data slots to fulfill the minimum service requirement of the ABR users. This selective removal might result in gaps in the allocated slots. This space is filled by left shifting the slots that are allocated later. This shifting does not violate the deadline requirements of the ATM cells as this shift leads to an earlier scheduling.

Fot the UL allocation, the CBR and ABR sources get the required number of slots each MAC frame (PCR and MCR respectively). These slots are given in every frame unless the source decides to terminate the connection. VBR sources also get slots according to their deadline requirements and their queue condition. If some data slots remain after such an allocation, the scheduling algorithm considers the additional bandwidth requirements of the ABR connections and tries to allocate slots for them. If after such an allocation, slots are exhausted and there are some UBR sources waiting with the maximum counter value, a random selection of a UBR source from this group is done. This source is then allocated a single slot in the MAC frame from the ABR additional bandwidth slots. If there were no ABR additional bandwidth slots available, the UBR source is not allocated any data slot. The UBR sources are thus partially guaranteed some QoS. This selective transmission ensures that denial of service doesn't take place to the UBR sources.

The fading information is also used in scheduling algorithm. If the signal received from the MT at the BS is below a certain threshold or is detected to be of decreasing intensity, the BS assumes that the MT is going out of sight. This information is used for scheduling the DL packets. The downlink ATM packets meant for the sources (VBR,ABR and UBR) are delayed by few frames i.e these sources are not considered by the scheduling algorithm while considering allocation of the slots for the DL connection. CBR sources are assigned a constant number of slots each frame irrespective of the received signal intensity at the BS.

In order to improve the throughput (and hence overall performance) of the wireless system one should use forward error correction (FEC). Of course there is a question of which codes to use and how many bits to allocate. Various studies have been addressed to this issue [15] [13]. Any such appropriate solution can be used with our MAC algorithm to improve the performance.

Chapter 3

Simulation

This chapter presents the simulation results of the scheduling algorithm. We have simulated CBR, VBR, ABR and UBR sources for UL as well as DL.The simulator has been written in C language.

Channel and MAC Frame

The channel bandwidth is assumed to be 10 Mbps. The synchronization period is of 30 bits, the contention period length is 360 bits for real-time sources while for the non-real time sources it is 450 bits. The acknowledgement period is 1600 bits. Considering an average of 200 ATM data slots per MAC frame the total MAC frame time comes out be 10.3 msec (using bandwidth=10 Mbps).

Source Characteristics

The transmission rates for CBR, VBR, ABR and UBR are 64kbps, 128kbps, 128 kbps and 64 kbps. For CBR sources packets are generated at a constant rate given by its MCR. For simulating VBR and ABR sources, ON-OFF traffic source model has been used. It is has been modeled using a two-state Markov process. In ON state packets are generated at a constant rate while in the OFF state no packets are generated [9]. UBR sources have been simulated using Poisson traffic arrival model with mean arrival time 200 ms. We measure performance of the protocol in terms of cell loss probability (i.e. the ratio of number of dropped ATM cells to the total number of cells generated) and the mean delay experienced by the packets while in the queues at the MTs. For an efficient MAC protocol the cell loss probability and the mean cell delay should be low.

Fading

Fading has been implemented as a two-state Markov chain with a good state and a bad state (similar to the ON-OFF source model). Since the channel conditions may be different for different users we have used different transition matrices (giving probability of transition between the two states) for different users. For simulation purposes three different transition matrices have been considered which model three states of channel: good, bad and worse. At the time of inclusion of a source in the network, a transition matrix is picked at random which defines the channel state for the connection till the connection lasts.

Constraints

The simulation model does not take into account the handoffs that take place between adjacent cells. The effects of propagation delay are not considered in the simulation model. Assuming a cell radius of 500m the maximum back-to-back propagation time comes out to be 3.3 μsec which can be safely neglected considering the frame time to be 10.3 msec. The active mobile terminals always listen to the channel i.e. they do not enter "sleep mode". We have not considered this feature in the simulation for complexity reasons. Nevertheless this feature helps in conserving power at the MT and simultaneously saving data slots which otherwise would have been idle (in case the MT runs out of battery in the midst of a connection).

Results

The call (connection request) arrival process has been modeled as Poisson process. In our simulations we have considered a maximum of 40 real-time sources (CBR/VBR) and 30 non-real time sources (ABR/UBR). The mean arrival rate for the sources is 10msec. The connection of the sources with the BS is assumed to last till the simulation ends. We observed that if ABR and UBR sources were added to the network having CBR and VBR sources there was only marginal increase in the loss probability. We are considering loss due to violation of deadlines only. This may be attributed to the admission control policy being used and the different contention slots for CBR/VBR and ABR/UBR. VBR sources have to contend for the

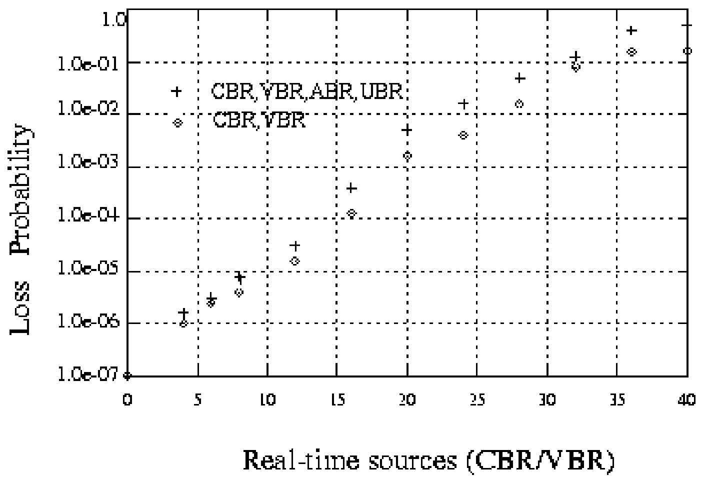


Figure 4: Loss probability of the real-time sources

channel every time they they need to send data to the BS. Similarly UBR sources also contend whenever there queue is not empty. By providing separate contention slots for these we decrease the probability of collision which has a considerable effect in improving the throughput of the system. Since the admission control algorithm is being used, the BS allows more connections only till the promised QoS of the existing connections is being satisfied.

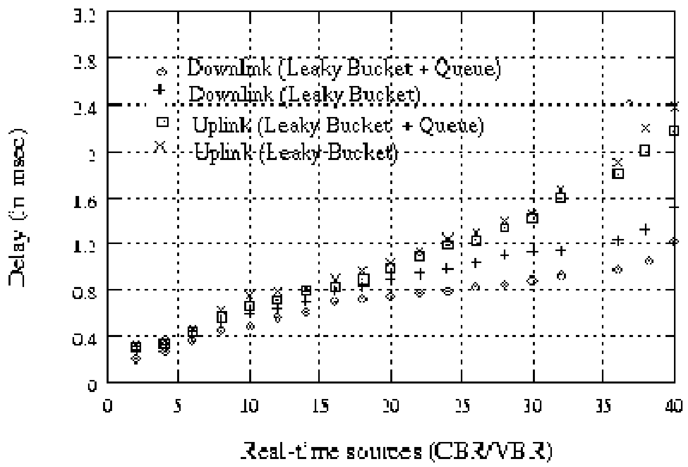


Figure 5: Overall delay of ATM packets

The mean cell delay is depicted in Figure 4. As we see the DL connections experience lesser delay than the UL packets. This is because the scheduling algorithm first considers the DL data as intuitively it would always be more than the UL. We also see that by using both the leaky bucket and queuing information we have slight improvement in the delays experienced by the cells. The queue information helps in considering those sources first that have pending queues at their end. This helps in decreasing the waiting time of the packets in the queue.

We also checked the loss probability with and without fading consideration. The results showed that the fading information helped to increase the total throughput of the system (around 10%) though it resulted in a slight increase in loss probability (around 2%). This can be explained by the conservative approach taken by the BS in attributing a weak signal to the loss of connection between the BS and the MT. Without fading information, BS would have allocated DL slots to the sources irrespective of their channel condition. This might have resulted in successful transmissions in some cases to the BS and thus deadlines of ATM cells for real-time connections would have been honored. But due to poor channel conditions this might result in unsuccessful transmission i.e. decreased throughput. By incorporating fading information, UBR and VBR sources gain most since they have no delay constraints on transmission (for them a *reliable* transmission is important rather than an *early* one).

Chapter 4

Summary

In this report we have described a MAC protocol for wireless ATM network. The unique feature of the protocol is the use of leaky bucket and queuing information to perform scheduling. It also uses a novel concept of having two separate contention fields - one for the real-time sources (CBR,VBR) and the other for the non-real time ones (ABR,UBR). This helps in improving the loss probability of ATM cells (due to violation of deadlines) by making the data slot requests of the VBR channel immune to the requests of ABR and UBR in contention frame. The use of minislots in the contention frame also helps in improving channel performance. The protocol also uses the fading information in scheduling which helps to increase the overall throughput of the system.

An admission control algorithm is also described which governs the entry of new connections to the network. Subsequently the scheduling algorithm is described which allocates slots to sources on basis of fading, leaky bucket, queue information and last channel access time. The simulations show that FAFS improves the channel throughput and decreases the loss probability by judiciously allocating slots to different connections.

Bibliography

[1] B. Walke, D. Petras, and D. Plassmann,"Wireless ATM: Limits, Challenges, and Proposals,"*IEEE Personal Communications*, Aug 1996, pp. 18-34.

[2] D. J. Goodman *et al.*,"Packet Reservation Multiple Access for Local Wireless Communications," *IEEE Transactions on Communications*, vol. 37, no. 8, Aug 89, pp. 885-90.

[3] D. Raychaudhari *et al.*,"WATMnet: A Prototype Wireless ATM System for Multimedia Personal Communication," *Proc. ICC'96i*, Dallas, TX, June 23-27, 1996, pp. 469-77.

[4] D. Sobirk *et al.*,"An Overview of Proposed MAC Algorithms for Wireless ATM," *The thirteenth Nordic teletraffic seminar NTS13*, ed by Emstad, Helvik and Myskja, NTNU, 1996.

[5] D. Yates,J. Kuroae,D. Towlset, and M.G.Hluchyj,"On Per-session End-to-End Delay Distributions and the CALL Admission Problem for Real-time Applications with QoS requirements,"*In Proceedings of acm sigcomm '93*, Mar 1993, pp. 227-236.

[6] Dietmar Petras *et al.*,"Performance of DSA++ MAC Protocol and SR/D-ARQ Protocol for wireless ATM under realistic traffic and channel models,"*Proceedings of wmATM'98*, Nov 1997, pp. 1-12.

[7] F. Khan and D. Zeghlache,"Analysis of Aggressive Reservation Multiple Access Schemes for Wireless PCS," *Proceedings ICC'96*, June 1996, Dallas, TX, pp. 1750-55.

[8] J. G. Kim and I. Widjaja,"PRMA/DA: A New Media Access Control Protocol for Wireless ATM," *Proc. ICC'96*, Dallas, TX, June 1996, pp. 240-44.

[9] J. Jangkok Bae and T. Uda,"Survey of traffic control schemes and protocol in ATM networks,"*Proceedings of IEEE*, 79(2):170-192, Feb 1992.

[10] J. Sanchez, R. Martinez, and M. W. Marcellin,"A Survey of MAC Protocols proposed for Wireless ATM," *IEEE Network*, Nov/Dec 1997, pp. 52-61.

[11] L. S. Koh and M. T. Liu,"A Wireless Multiple Access Control Protocol for Voice-Data Integration," *Proceedings Intl. Conference Parallel and Dist. Systems*, Tokyo, Japan, June 3-6, pp. 206-13.

[12] Mark J. Karol, ZHao Liu and Kai Y. Eng,"an Efficient Demand-Assignment Multiple Access Protocol for wireless packet (ATM) networks,",*Proc. International Conference on Communications*, pp. 1224-1231, June 1995.

[13] N. R. Figueira and J. Pasquale,"Remote-Queuing Multiple Access (RQMA): Providing Quality of Service for Wireless Communications,"*INFOCOMM'98*, pp. 307-314.

[14] Nikos Passas et. al.,"Quality-of-Service-Oriented Medium Access Control for Wireless ATM Networks," *IEEE Comm. MAG.*, Nov 1997, pp. 42-50.

[15] S. Choi and K. G. Shin,"A Cellular Wireless Local Area Network with QoS Guarantees for Heterogeneous traffic,"*INFOCOMM'97, pp. 1030-1037*

[16] S. Jiang and Danny H. K. Tsang,"On Architectures for Broadband Wireless Systems,"*IEEE Communications Magazine*, Oct 1997, pp. 132-140.

[17] The WLANA Resource Home Page http://www.wlana.com

[18] Wireless Medium Access Control and Physical Layer WG, IEEE Draft Standard p802.11,"Wireless LAN," IEEE Stds. Dept., D3, Jan 1996.