ADAPTIVE RESERVATION TDMA PROTOCOL FOR WIRELESS MULTIMEDIA TRAFFIC

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ABSTRACT

An Adaptive Reservation Time Division Multiple Access (AR-TDMA) control protocol for Wireless Asynchronous Transfer Mode (WATM) networks is proposed in this paper. AR-TDMA combines the advantages of distributed access and centralised control for transporting Constant Bit Rate (CBR), Variable Bit Rate (VBR) and Available Bit Rate (ABR) traffic efficiently over a wireless channel. The contention slots access for reservation requests is governed by two protocols, the Adaptive Framed Pseudo-Bayesian Aloha with Adaptive Slot Assignment (AFPBA-ASA) protocol and the Framed Pseudo-Bayesian Aloha with Adaptively Prioritised Controlled Capture (FPBA-APCC) protocol. Both protocols provide different access priorities to the control packets in order to improve the Quality-of-Service (QoS) offered to time sensitive connections. AR-TDMA also features a novel integrated resource allocation algorithm that efficiently schedules terminals' reserved access to the wireless ATM channel by considering their requested bandwidth and QoS. Integration of CBR, voice, VBR, data and control traffic over the wireless ATM channel using the proposed AR-TDMA protocol is considered in the paper. The performance of the AR-TDMA in conjunction with the AFPBA-ASA protocol and FPBA-APCC protocol has been investigated and the simulation results are presented showing that the protocol satisfies the required QoS of each traffic category while providing a highly efficient utilisation of approximately 96% for the wireless ATM channel.

Keywords: ***R-TDMA, MAC, WATM, Priority***

1.0 INTRODUCTION

Asynchronous Transfer Mode (ATM) has been recommended by the International Telecommunications Union-Telecommunication (ITU-T) to be the transfer protocol of the emerged Broadband Integrated Services Digital Network (B-ISDN) [1] and the concept of wireless ATM (WATM) has been introduced to extend the capabilities of ATM over the wireless channel [2]. A major issue related to the realisation of WATM networks is the selection of a Medium Access Control (MAC) protocol that will efficiently and equitably allocate the scarce and valuable radio medium among the competing mobile stations while respecting the Quality-of-Service (QoS) requirements of each admitted connection. For compatibility with the wired ATM network, the WATM MAC protocol must support the standard ATM service classes: Constant Bit Rate (CBR), Variable Bit Rate (VBR) and Available Bit Rate (ABR) traffic.

In the past years, several researchers have developed MAC protocols for wireless ATM networks [3, 4, 5], and most of these protocols have been designed to support multimedia ATM traffic. A survey of wireless ATM MAC protocols can be found in [6] and [7]. Among the known MAC allocation algorithms for WATM, we can underline the protocols presented in [5, 8, 9, 10, 20] as being the most complete and efficient. Most of the previously proposed protocols suffer from pitfalls related to the random access protocols employed for initial access. They either limit the delay-throughput performance [5, 8, 9] or restrict the number of users [10]. A dynamic reservation-based MAC protocol for integrated multimedia traffic was introduced in [20] with emphasis on VBR source and allocation control algorithms. Generally, they do not offer access priorities between different service classes. In the MAC protocol presented in this paper, we adopt two new random access protocols that allow us to increase the throughput performance while maintaining the required QoS, by enabling appropriate control traffic access priorities for different service classes. We adopt the bandwidth allocation algorithms used in [20] but we focus on evaluating the new access schemes.

To improve the performance of the Adaptive Reservation TDMA (AR-TDMA) protocol, the protocol uses either one of the novel contention algorithms [11, 12] to manage the access to the control mini-cells such that contention can be minimised and access priorities can be provided to control packets in order to improve the QoS offered to connections with time sensitive traffic. Additionally, the new protocols adaptively adjust the number of uplink (mobile to base station) control slots as a function of the traffic load to optimise the channel utilisation. Piggybacking of requests is also used so that the protocol efficiency will not be limited to the performance of the random access schemes mentioned above. A key feature of our AR-TDMA protocol is the bandwidth allocation algorithms used to distribute the bandwidth among the CBR, voice, VBR, data and control traffic. A method to integrate the different allocation algorithms to provide ubiquitous wireless ATM services was also described. We present simulation results to illustrate the performance of the AR-TDMA protocol under a wide range of traffic conditions. These results show that the AR-TDMA protocol proposed in this paper performs well, under similar traffic conditions, compared to previous protocols. The performance improvement was achieved mainly due to the following features of our protocol; the dynamic adjustment of the number of uplink control slots; contention access priorities provided through the AFPBA-ASA protocol or the FPBA-APCC protocol to improve QoS of real-time connections by expediting the respective control packets; use of piggybacking concept in conjunction with the random access schemes; and use of both minimal in-band and out-of-band signaling to estimate the mobile traffic requirements in the scheduler.

The paper is organised as follows. Section 2 gives an overview of the AR-TDMA protocol. In Section 3, the source models are presented. In Section 4, we describe the resource allocation algorithms. An evaluation of AR-TDMA performance by simulations is presented in Section 5. Finally, Section 6 concludes the paper.

2.0 DYNAMIC RESERVATION TDMA PROTOCOL

2.1 Frame Structure

An AR-TDMA protocol is adopted to multiplex multimedia ATM connections over a single time division duplex (TDD) radio channel. Fig. 1 illustrates the frame structure employed by the AR-TDMA protocol. The modem preamble is used by physical layer functions in the radio receiver while the frame header identifies the beginning of a frame period. The fixed length AR-TDMA frame is time-duplexed into an uplink and downlink channel, for mobiles to communicate with the base station and vice versa, respectively. The boundary between these two parts is dynamically adjusted as a function of the traffic load. The downlink and uplink channels are further dynamically divided into control and data transmission periods, each consisting of an integer number of slots. The base station has absolute control to determine the number of each type of slots during a frame period and which mobile stations will receive or send information using the data slots.

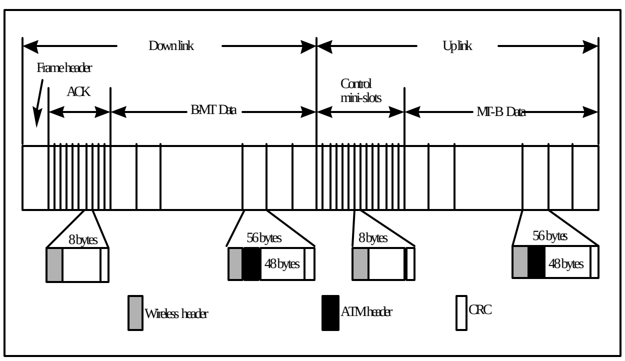
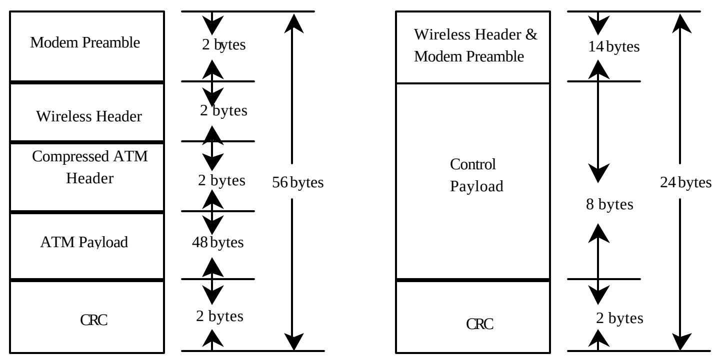


Fig. 1: AR-TDMA wireless ATM MAC protocol frame structure

The system architecture parameters adopted for the AR-TDMA protocol are adopted from [13]. Fig. 2(a) illustrates the format of uplink wireless ATM data packets. The cell format structure is harmonised with standard ATM cell format. The 2-byte wireless header incorporates an 8-bit cell sequence number to support DLC error recovery over moderately large bursts. This header also includes fields to enable other wireless network functions such as service type definition, handoff recovery, and cell segmentation. A 2-byte compressed ATM header is incorporated with 12 bits for VCI (virtual circuit identifiers) and 4 bits for ATM control. The header error control (HEC) function is not used on the radio link in view of a 2-byte CRC error detection provided by the DLC. A maximum of 20 cells can be acknowledged with a single WATM ACK message. This type of group ACK syntax reduces the number of short ACK packets requesting channel access from the MAC layer, thereby improving efficiency. Notice that additional 2 slots (14 bytes) of overhead is needed for the wireless data cell shown in Fig. 2 (b) in order for the transmission to happen between the mobile terminal and the base station. Therefore, the total number of slots needed is 9 (72 bytes).



(a) Wireless Data packet format (b) Wireless Control packet format

Fig. 2: Uplink wireless ATM cell format

Downlink transmissions are TDM multiplexed in a single burst from the base station to the mobile terminal, while uplink transmissions use dynamic TDMA. The downlink preamble provides frame synchronisation as well as training sequence for the purpose of equalisation. The base station uses the frame header to transmit its ID and information about the positions and sizes of the different sub-frames in the current frame. The downlink control messages, transmitted in the Base-to-Mobile Terminal (B-MT) control region, include allocation information, B-MT acknowledgement information, and other wireless network control messages. The B-MT data region contains downlink data cells. The MT-B control region is used by the MTs to transmit control messages to the base station in using one of the contention algorithms mentioned above.

The basic unit of the MAC allocation was designed to be a 8-byte slot. A 16-byte (2 slots) preamble was assumed for synchronisation and equalisation at the start of every burst. Data cells require 7 slots each (48-byte payload, 4-byte WATM header, and 2-byte CRC). Each data cell in the frame has the same length as 9 slots (i.e. 56 bytes). When a data cell is assigned for control purpose, it is divided into 3 control mini-cells (1 slot for control packet plus 2 slots for modem preamble) comprising of 3 slots per wireless control packet to be used in the MT-B control region. A 2-slot frame header and 2-slot modem preamble must be added to the overhead at the beginning of each B-MT or MT-B control region mobile transmission.

System parameters used in the evaluation are summarised in Table 1. As can be seen, the bandwidth allocated to the control regions of B-MT and MT-B is varied according to control traffic for one case and fixed at 20% at another case to facilitate comparison with the results in [13]. So the maximum data throughput (under ideal conditions, without overhead for the 20% fixed bandwidth case) is limited to 80%. Downlink packets (both data and control) are similar to uplink packets; however the wireless header can be reduced since the modem preamble at the beginning of the frame plays the same role. Slot allocations and cell acknowledgments for all mobiles are grouped in the same downlink control slots (i.e. each downlink control slot is not dedicated to a specific mobile). Table 1 summarises the AR-TDMA MAC frame parameters. It should be noted that the only parameters that directly influence the MAC protocol efficiency are: the frame duration, the number of slots per frame and the data to control slot size ratio.

2.2 Protocol Mechanisms

In the uplink channel, control slots provide a communication mechanism for a mobile station to send a reservation request during the contention phase of the connection, while the data slots supply it with contention-free bandwidth resources during the data transmission phase. An uplink control packet is sent when a mobile station needs to signal the base station and it does not have a data slot assigned to it in the current frame, such as when it needs to establish a new connection, has a new data message to send, begins a new talkspurt, etc. Uplink control packets provide the base station with the traffic characteristics and source status of the corresponding mobile stations' connections. Uplink control packets are sent in contention-mode according to the AFPBA-ASA and FPBA-APCC protocols described in Section 2.3. With these protocols, contention access priorities are assigned to mobile stations according to the required QoS of their connections. The contention delays of time sensitive control packets are thus reduced. Feedback for the uplink control packets is sent in the downlink control packets in the next frame. Mobile stations whose control packets have experienced a collision will repeat the contention process in subsequent frames until they are successful. When there is no uplink traffic from a mobile station but downlink packets are sent to it, the mobile station may send uplink control packets to acknowledge receipt of downlink packets and to enable the base station to adjust anti-fading parameters for this mobile. These uplink control mini-cells are directly assigned to respective mobile stations by the base station and are therefore accessed without any contention. After completing the contention procedure, the mobile station can use the data slots assigned by the base station without undergoing further contentions, until its data buffer is empty. While a mobile station has access to reserved data slots, traffic parameters and status information, e.g., specifying requirements for additional data slots, can be sent piggybacking on the data packets. The base station will use these traffic parameters to allocate uplink data slots to each reserving station according to the allocation algorithms presented in Section 4. When a connection has successfully sent its request, it enters the data transmission phase and monitors downlink control slots in each subsequent frame to receive its slot assignments, which identify the uplink data slots in the current frame in which the mobile stations should send its packets.

Table 1: AR-TDMA MAC frame parameters

Channel bit rate	8 Mbps
Frame duration	2 ms
Slot size	8 bytes
Slots per frame	250
Preamble size	2 slots
Frame header	2 slots
Data cell	7 slots
Control packet	1 slot
Number of mini-slots (wireless Control packet) per data cell	3 mini slots
Case One:	
B-MT & MT-B control	varied length
Case Two:	
B-MT control	8% (20 slots)
MT-B control	12% (30 slots)

2.3 Contention Access Schemes

In a reservation MAC protocol such as AR-TDMA, before obtaining contention free access to the channel, a mobile must wait for its request packet to be successfully sent to the base station, in contention with control packets from other connections. Therefore, the contention phase that a connection goes through before obtaining contention-free packet transmissions has a major effect on the delay QoS of a reservation MAC protocol. As the QoS of some classes of traffic is more sensitive to the contention delay than others, this calls for the design of a contention access scheme that provides a lower contention delay to time sensitive control packets while being adapted to the structure of AR-TDMA. The FPBA-ASA and the FPBA-APCC protocols described in detail in [11, 12] have been designed to satisfy these requirements, and are employed by mobile stations to access uplink control mini-cells. These protocols are based on the assumption that the probability distribution of the number of control packets waiting for transmission is Poisson [14]. In addition to the standard Aloha algorithm, the FPBA-ASA and FPBA-APCC protocols provide multiple levels of access priorities and are adapted to the frame structure of AR-TDMA. Before we summarise the contention algorithm, we must state that these two protocols (especially the FPBA-APCC) take advantage of the wireless channel and capitalise on the capture phenomenon. Therefore, we will assume a wireless channel effect for the contention period of the frame. We shall not consider the retransmission of corrupted data packets as that would overlap with the function of the DLC layer. We must also state the wireless channel assumptions for each contention protocol so that we can better evaluate its performance.

The characteristics and assumptions of the wireless channel are as follows:

- Mobile terminals of the p different priority classes are distributed according to a Poisson point process over a circular area whose radius is normalised to unity. As a result, each of them is equally distributed within a circle with a random distance r from the centre of the circular area.
- The propagation model includes attenuation due to the distance r, proportional to r^{-h}, where h is the power loss exponent and is typically equal to 4 in land mobile radio environment.
- The propagation model also includes shadowing, described by means of a log-normal random variable.
- Rayleigh fading, which causes the instantaneous envelope of the received signal to be an exponentially distributed random variable.
- Capture ratio b is taken as 2 dB for all cases which represents the carrier to noise ratio of digital systems.
- Background noise is negligible and does not count in the interference calculation.
- Interference from adjacent cells is not considered in the propagation model considered.
- The wireless channel effect is only considered with the contention algorithms. Its effect of the TDMA data slots of the AR-TDMA protocol is not considered.

Further assumptions specific to the AFPBA/ASA algorithm is that:

- The received power from a mobile terminal j, of the same priority class i at distance r can be expressed as:

$$P_{r_m} = R^2 e^{x_m} G r_m^{-h} P_T \tag{1}$$

where R is a Rayleigh distributed r.v. with unit power, e^x accounts for the shadowing ξ is Gaussian with mean zero and variance σ^2), Gr^{-h} is the deterministic loss law, and P_T is the transmitted power. G, h and P_T will be assumed to be the same for all packets whereas R and x will be assumed independent from one packet to another and identically distributed. There is no consideration for power control in this model and therefore, transmission power P_T is equal to unity. The log-normal shadowing standard deviation s_{dB} (known as *shadowing spread* or *dB Spread*), is given by:

$$s = (0.1 \log_e 10)\ s_{dB}. \tag{2}$$

Further assumptions for the FPBA/APCC contention algorithm are:

- The received power from a mobile terminal j, of the same priority class i at distance r can be expressed as:

$$P_r = R^2 e^x\ G r^{-h} v_i \tag{3}$$

where v_i is the uniformly chosen random mobile terminal power level computed according to the sub-optimal model as follows:

$$\underline{v_i} = (1, g^{1/(V-1)}, g^{2/(V-1)}, \ldots, g^{(V-2)/(V-1)}, g) \tag{4}$$

and

$$1 \le V \le [\log(g)/\log(b)] + 1 \tag{5}$$

where b is the capture ratio. Now the AFPBA/ASA and FPBA/APCC algorithms will be described in the context of the AR-TDMA WATM MAC protocol.

2.3.1 AFPBA-ASA Protocol

Suppose that K^t control mini-slots are available in frame t and there are p different priority classes with arrival processes of intensities $l_1, \ldots, l_p$. l_i is the average number of control packet arrivals of class i per frame in all mobiles. Let the number of contention slots assigned to each priority class i, (1£ i £ p); where a lower index corresponds to a higher priority packet class, be k_i, where $\sum_{i=1}^{p} k_i = K$. Let $\frac{k_i}{K}$ be the priority parameter of each control traffic class i. In order to maintain the priority order we must have $\frac{k_1}{K} \ge \frac{k_2}{K} \ge \ldots \ge \frac{k_{p-1}}{K} \ge \frac{k_p}{K}$ and the parameters satisfy the relation $\sum_1^p \frac{k_i}{K} = 1$. Each backlogged packet is independently transmitted in a frame t with probability $q_i = \min(1, k_i/\hat{n}_i)$ over a slot independently chosen from a set of k_i assigned slots for the traffic class i, with a uniform probability (each slot has a probability $1/k_i$ of being chosen).

The algorithm operates by maintaining for each priority class i an estimate $\hat{n}_i^t$ of the total number of backlogged control packets n_i^t at the beginning of each frame t. At the beginning of each frame t, for each priority class i, $\hat{n}_i^t$ is updated from $\hat{n}_i^{t-1}$, $\frac{k_i}{K^{t-1}}$, K^{t-1} and the feedback for frame $t-1$ according to:

$$\hat{n}_i^t = l_i + k_{nc} \max\left(0, \frac{\hat{n}_i^{t-1}}{K^{t-1}} - \frac{k_i^t}{K^{t-1}}\right) + k_c\left(\frac{\hat{n}_i^{t-1}}{K^{t-1}} + \frac{\frac{k_i^t}{K^{-1}}}{e-2}\right) \tag{6}$$

where k_{nc} is the number of idle or success slots and k_c the number of collision slots in frame $t-1$. In the presence of the wireless channel the slot outcome {success, idle, collision}, and i.e., k_{nc} and k_c in equation (8), is further determined by the following formula. The packet success probability P_{succ} for each mobile terminal is therefore expressed as:

$$P_{succ} = p\left[\frac{P_{r_1}}{\sum_{n=2}^{N} P_{r_n}} > b\right] \tag{7}$$

b is the threshold signal to interference ratio (or the capture ratio), n is the number of packets involved in the collision and P_{rn} is given by equation (7). The number of assigned slots for each priority class is given by the ASA algorithm as follows:

for (each priority class i in order of increasing priority)

$$\left.\begin{array}{l}
1 - \left\{Ceil\left(\left(\frac{l_i}{\sum_{i=1}^{p} l_i} * (K^t - \sum_{j=1}^{i-1} k_j)\right)\right)\right\} \le k_i \le \left(K^t - 1\right) \\
2 - 1 \le k_p \le \left(K^t - \sum_{i=1}^{p-1} k_i\right) \\
3 - K^t = \sum_{i=1}^{p} k_i \quad , \quad K^t >> p \quad (i.e., K^t \ge 5p)
\end{array}\right\}$$

2.3.2 FPBA-APCC Protocol

Suppose that K^t control mini-slots are available in frame t and there are p different priority classes with arrival processes of intensities $l_1, \ldots, l_p$. l_i is the average number of control packet arrivals of class i per frame in all mobiles. The algorithm operates by maintaining for each priority class i an estimate $\hat{n}_i^t$ of the total number of backlogged control packets n_i^t at the beginning of each frame t. A new control packet arriving during frame t is immediately regarded as backlogged and will attempt transmission in a slot k ($1 \le k \le K$) in each subsequent frame after its arrival until success. Each backlogged packet is independently transmitted in a frame t with probability $q_i = \min(1, K^t / \hat{n}_i^t)$ over a slot independently chosen with a uniform probability (each slot has a probability $1/k_i$ of being chosen).

At the beginning of each frame t, for each priority class i, $\hat{n}_i^t$ is updated from $\hat{n}_i^{t-1}$, K^{t-1} and the feedback for frame $t-1$ according to equation (6.1) and repeated here:

$$\hat{n}^t = l + k_{nc} \max\left(0, \frac{\hat{n}^{t-1}}{K} - 1\right) + k_c\left(\frac{\hat{n}^{t-1}}{K} + \frac{1}{e-2}\right) \tag{8}$$

where k_{nc} is the number of idle or success slots and k_c the number of collision slots in frame $t-1$. In the presence of the wireless channel the slot outcome {success, idle, collision}, and therefore k_{nc} and k_c in equation (8), is further determined by the following formula. The packet success probability P_{succ} for each mobile terminal is therefore expressed as:

$$P_{succ} = p\left[\frac{P_{r_1}}{\sum_{n=2}^{N} P_{r_n}} > b\right] \tag{9}$$

b is the threshold signal to interference ratio (or the capture ratio), n is the number of packets involved in the collision and P_{rn} is the received power given by equation (9). The number of logarithmically equi-spaced random transmitter power levels (LESRTPLs) assigned for each priority traffic class is calculated using the APCC algorithm as follows:

for (each priority class i in order of increasing priority)

A* - *If $i = 1$ calculate m_i using : $ceil\left(\frac{\hat{n}_i^t}{\sum_{i=1}^{p}\hat{n}_i^t}(V)\right) \le m_i \le (V-1)$

then, assign m_1 power levels to the highest priority class 1 starting from γ downwards calculated using eq. (4).

B* - *If $m_1 = V$ *then* $m_1 = (m_1 - 1)$; *and* $m_i = v_{\min} = 1$ *(for* $i = i+1,\ldots,p$*).*

(i.e., if the number of assigned power levels to priority traffic class 1 m_1 is equal to the total number of power levels, then reduce m_1 by one level and assign v_{min} to the rest of the priority traffic classes).

C* - *If $i > 1$ calculate m_i using : $1 \le m_i \le ceil\left(\frac{\hat{n}_i^t}{\sum_{i=1}^{p}\hat{n}_i^t}\left(V - \sum_{i=1}^{i} m_i\right)\right)$

then, assign priority class i, m_i power levels starting from (V-m_{i-1}) downwards calculated using eq.(4).

D* - *If $\left(V - \sum_{i=1}^{i} m_i\right) = 0$ *then* $m_i = v_{\min} = 1$ *(for* $i = i, i+1,\ldots,p$*)*

(i.e., if the balance of power levels available is zero, then reduce m_1 by one level and assign v_{min} to the rest of the priority traffic classes).

3.0 SOURCE MODELS

3.1 CBR Source Model

The CBR voice source generates a signal that follows a pattern of talkspurt and silent gaps, and a speech activity detector can be used to detect this pattern. Data transmission can thus be stopped during periods of voice inactivity to reduce the traffic and increase the statistical multiplexing. Therefore, a CBR voice source, can be described by an ON/OFF model. An ON/OFF source alternates between two states: the ON state where the source generates packets at rate R_v and the OFF state where no packets are generated. ON and OFF state durations are modeled by exponential distributions with means t_b and t_i, respectively. This model is similar to the "slow" speech activity detector model described in [15] and we have selected the same parameter values for t_b and t_i. A CBR voice source is also characterised by the encoder bit rate R_v and the packetisation delay to assemble a packet payload T_v. In our case, we have a payload of 48 bytes, we thus have $T_v = 384/R_v$. The voice activity ratio of a voice source is defined as $t_b/(t_b+t_i)$ and represents the average number of packet transmitted by the source per period T_v. Our CBR voice source model has a one packet buffer, i.e., when a new voice packet is generated, if the previous packet has not been transmitted, the buffered packet is discarded. The maximum cell transfer delay (MTD) is thus equal to the packetisation delay T_v. Finally, CBR voice request packets are assigned a high priority for the contention protocol. Table 2 summarises the voice source parameters.

3.2 ABR Source Model

ABR data sources are represented by a model where groups of packets arrive in the buffer at a certain rate. This model is in accordance with the AAL5 layer that is used for ABR and Unspecified Bit Rate (UBR) traffic. When AAL5 receives a packet from an upper layer, it segments the packet in ATM cell of 48 bytes (or wireless ATM cells in our case). It is thus reasonable to assume that these cells arrive in the wireless ATM MAC data buffer at approximately the same time since the packetisation rate will be much faster than the channel rate. ABR burst size L (i.e., number of packets per burst) is geometrically distributed with mean value $E(L)$. The inter-arrival time between two groups of packets (or the period between burst arrivals) on a virtual connection (VC) is exponentially distributed with mean t_d. Even if ABR data are non-real time traffic, we also assign a maximum transfer delay to the packets. This delay is chosen long enough such that it would not cause any loss under normal operation, but when the load is too heavy, this allows the old packets to be discarded. In fact, this is similar to having a finite buffer length. Finally, data request packets are assigned a low priority for the contention protocol.

Table 2: CBR voice source model parameters

t_b	1.00 s
t_i	1.35 s
Activity ratio	0.426
R_v	32 Kbps
T_v	12 ms
MTD	12 ms
Priority	high

3.3 VBR Source Model

ON/OFF sources have been successfully used to characterise VBR traffic [16, 17]. Taking the teleconferencing-type of traffic as an example, there are two typical kinds of scenes, the active one and the inactive one [18]. The system can be designed such that when it is inactive, no packet is generated, which corresponds to the OFF period. When the user is active, the number of packets generated in each frame is variable, which distinguishes it from the CBR source. This variable can be described by its probability distribution function, denoted as $Q^v = \{q^v_1, q^v_2, \ldots, q^v_i\}$, where q^v_i is the probability of creating i packets in a frame. The maximum number of packets generated by a VBR source in each frame is denoted as p^v. It is assumed that there is at least one packet generated in each frame when the user is active. The VBR user can be in the silence state (SS), the reservation state (RS), or the contention state (CS). Fig. 3 illustrates the transitions among SS, RS, and CS states. In this figure, q^v_s is the probability that an ON period of VBR source ends in a frame ($q_{\text{ON-OFF}}$); q^v_a is the probability that an ON period of VBR source is generated in a frame ($q_{\text{OFF-ON}}$); and P^v_{CR} is the probability that a VBR source obtains a reservation in the current frame. We assume here that the probability that a VBR user returns to the silence state before it obtains a reservation is zero. Then,

$$q^v_a = q_{\text{OFF-ON}} = 1 - \exp(-T/t^v_s) \tag{10}$$

and

$$q^v_s = q_{\text{ON-OFF}} = 1 - \exp(-T/t^v_o) \tag{11}$$

P^v_{CR} also depends on the system parameters. The length of the ON period is exponentially distributed with average duration t^v_o and the length of the OFF period is exponentially distributed with average duration t^v_s. The frame duration is denoted as T. Note that a more sophisticated source model can be used to describe the VBR traffic. However, no matter what source model is exploited, the mechanism of AR-TDMA protocol remains unchanged. ON and OFF state durations are modeled by exponential distributions with means t^v_o and t^v_s, respectively. The mathematical model used in this paper cannot represent every type of VBR connections but is good enough to show the protocol efficiency and properties. Finally, VBR request packets are assigned a medium priority for the contention protocol. Table 3 summarises the VBR source parameters and Table 4 illustrates the probability distribution of the packet generation rate of VBR source.

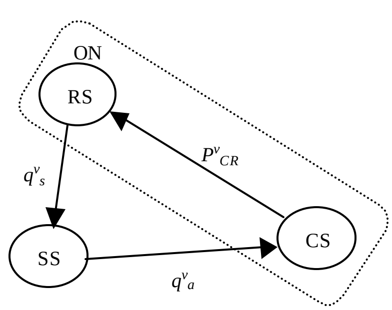


Fig. 3: VBR traffic model

Table 3: VBR source model parameters

t^v_o	2.00 s
t^v_i	2.70 s
Activity ratio	0.425
Average cell rate	384 Kbps
Peak cell rate	768 Kbps
MTD	50 ms
Priority	Medium

Table 4: Probability distribution of the packet generation rate of a VBR source

I (packets/frame/user)	1	2	3	4
q^v_i	0.3	0.5	0.1	0.1

4.0 BANDWIDTH ALLOCATION ALGORITHMS

The main objective of a WATM bandwidth allocation algorithm is to efficiently exploit statistical multiplexing while providing the negotiated QoS to each admitted connection in the network. As the intention of this paper is to focus on the performance of the AR-TDMA MAC protocol in conjunction with the new access schemes, the bandwidth allocation algorithms used in [20] were adopted with minor modifications. In this work, we only consider the bandwidth allocation for uplink transmissions since downlink transmissions can be scheduled in the same manner as in a wired ATM switch. For our AR-TDMA protocol, the base station scheduler decides, based on its current knowledge of the buffer states in the mobile stations and the connection traffic parameters, how to assign data slots to each connection and the number of slots to allocate for the control traffic. In this section, only the basic principles of the allocation algorithms are presented.

4.1 Slot Allocation Algorithm for CBR Traffic

CBR Virtual Circuits (VCs) are assigned slots periodically according to their bit rate. A group of M $\geq$ 1 frames (e.g., M=6) may constitute a MAC superframe which is used to obtain a reasonable granularity for bit rate assignments (e.g., Nx32Kbps) [13]. The positions of the assigned slots within a MAC superframe are maintained relatively static in order to facilitate operation of low complexity telephone terminals and also reduce the control signaling load on the wireless channel. In the case of CBR voice connections the slot allocation algorithm for CBR voice traffic uses a Time-To-Expiry (TTE) approach where the connection for which the CBR packet will expire and be discarded first receive the slot allocation. When a new CBR voice call is established, the mobile sends a request packet in the contention period using the either the AFPBA/ASA or FPBA/APCC protocol, with high priority until success. This control packet contains the voice connection encoder bit rate from which the cell inter-arrival time can be computed in the base station. At the beginning of a new talkspurt, the mobile also sends a high priority control packet in the contention period until success or the talkspurt ends. The control packet contains the time of arrival of the last voice packet. A connection is considered active when the base station receives a control packet indicating the beginning of a talkspurt.

For each active connection, the base station records the time of the last voice packet arrival. The scheduler is thus able to find the waiting time experienced by a packet and when it will be lost. It can also predict the arrival time of the next packet for each active voice connection. When the voice activity detector detects the end of a talkspurt it will set a bit in the last talkspurt packet to indicate to the scheduler the beginning of a silent gap. The scheduler then removes the connection from the active set after the reception of the last talkspurt packet. If this packet is lost, a control packet is sent to the base station, either in contention or in the next slot allocated by the base station to this CBR voice connection. For each active connection, the base station keeps the time of the last packet arrival if it has not yet been transmitted or, otherwise, the time of the next arrival. Active CBR voice connections are sorted in a list in increasing order of TTE where TTE = MTD. The scheduler then allocates slots to the connections that have waiting packets in order of increasing TTE. The allocation algorithm can be summarised as follows (in the algorithm the TTE always refers to the TTE of the first connection in the list) [20]:

while (L > 0 and TTE ≤ MTD)
/* TTE ≤ MTD ensures that the packet has arrived in the mobile */
    if (TTE ≥ 0)
        Allocate a slot to the first connection in the list
    L = L - 1
    if (This packet is the last of a talkspurt)
        Remove the connection from the active list
    else
        Enter this connection in the list with the TTE corresponding to the time of the next arrival
    else
        Packet loss
        Enter this connection in the list with the TTE corresponding to the time of the next arrival

4.2 Slot Allocation Algorithm for ABR Traffic

For the ABR traffic slot allocation algorithm, ABR VCs are handled on a burst-by-burst basis with dynamic reservation of ABR slots. The base station keeps the buffer length status of active connections (an active data connection is defined as a connection with a non-zero buffer length) and allocates slots to the connections using a combination of round-robin and fair bandwidth allocation algorithm. The first ABR VC is picked in a random fashion to prevent long bursts from overloading the channel and increasing delays for the other VCs. ABR slots assignment are demand-based in contrast to the CBR slot assignment which is either demand-based or periodic. When a batch of packets arrives at a mobile during frame t, the mobile MAC controller sends a piggybacked request or a control packet depending on the mobile's queue state at the beginning of frame $t + 1$. This gated mode of operation is used because we assume that the wireless packets are formed at the beginning of the frame and cannot be modified just before transmission to piggyback the new information. This is a realistic assumption since we do not know the processing time that this modification will require and it is also a worst case assumption.

If the queue is empty at the beginning of frame $t + 1$, the mobile sends a low priority control request packet in the following contention periods using the either one of the contention protocols until success. Otherwise, the request will be piggybacked to the next information packet transmitted to the base station. The request contains the information on the number of packets in the new batch arrival. This allows the base station to keep the exact buffer length of each active connection. Since slot allocation is announced at the beginning of the frame, allocation for this new request can only start during the next frame. Thus, for a batch which arrived in frame t and for which the request has been successfully received in frame $t + 1$, the slot allocation for these packets can only begin in frame $t + 2$. There is thus a minimum waiting time of $2T$, where T is the frame length, for data packets transmission. Based on the current buffer length status of all active connections, the base station scheduler allocates the L available slots to connections according to the fair bandwidth allocation algorithm. The algorithm, adapted to the slotted environment (i.e. we must assign an integer number of slots instead of a continuous bandwidth), can be described as follows [20]:

if (The sum of individual demands ≤ L)
    Allocate requested demand to all connections
    Update connections buffer length
else
    while (L > 0)
    /* C is the number of ABR active connections */
    Pick the first one of the C ABR VCs in a random fashion
    Let Fair = [L/C]
    if [Fair = 0]
        Fair = 1
    for (All active connections)
    if [L = 0]
        Stop
if [Connection buffer length ≤ Fair]
        Allocate buffer length
        Update buffer length
        Remove connection from active set and C = C +1

When a mobile receives its slot allocation, it can serve packets that are queued in its buffer in the order that it prefers. In this work, a FIFO queuing discipline for mobile's buffer has been implemented. Finally, at the beginning of each frame, the base station scheduler will update the buffer length status of connections for which it has received a request in the last frame (either a control packet or a piggybacked request) and if the connection was not in the active connection set it will be added to the list.

4.3 Slot Allocation Algorithm for VBR Traffic

The VBR source model generates packets using two rates, a maximum cell rate and average cell rate. Both are controlled by an internal transition probability between the two states ON and OFF, the activity ratio of the VBR source and the frame size. Each VBR connection is characterised by its basic cell rate, peak cell rate (PCR), maximum cell transfer delay (MTD), …, etc. Our allocation algorithm assumes one VBR VC per MT and will assign one data slot to carry one ATM cell per frame per MT, where the MT has to contend for the channel using one of our contention algorithms at the beginning of ON period for a slot. The MT will keep the slot for the duration of the ON period. If there are extra packets generated during the frame per MT, the MT will have to piggyback their requests with one of the data slots it uses on one of the active VCs. Extra packets that have piled up at the MT buffer and reached its MTD will have to be dropped. Our allocation strategy provides both guaranteed and best effort services for traffic parameters conforming and non-conforming cells, respectively. Guaranteed packets are served using a TTE approach while best effort cells are scheduled according to the fair bandwidth allocation algorithm. When a new VBR connection is established, the mobile sends a medium priority request packet in the contention period using the AFPBA/ASA or the FPBA/APCC protocol. This packet contains the VBR traffic parameters such as the source basic rate, the PCR, the MTD, …etc. These parameters can be re-negotiated during the course of the connection. The request packet also contains the initial cell rate, the first cell arrival time, and the initial number of packets in the guaranteed buffer. The status of the VBR connection buffer is predicted in the base station using information sent from the MT to the base station piggybacked on data packet or in control packet transmitted in the uplink control period using either one of the contention protocols. Each arriving packet is assigned a sequence number that is required for the data link layer protocol and it will also be used by the MAC protocol to reorder the packets. When the connections' status information has been updated in the base station, the scheduler allocates slots to the VBR connections according to their buffer status and QoS of each connection. The slot allocation algorithm is divided into two parts: guaranteed allocation and extra (best-effort) allocation. First, the scheduler allocates the available slots to the VBR connections that have waiting guaranteed packets using the Time-to-Expiry algorithm presented in Section 4.1. Then, if there are still available slots, the scheduler allocates the remaining slots to the VBR connections with best-effort packets in their virtual queue according to the fair bandwidth allocation algorithm described in Section 4.2.

The connection PCR is determined by the VBR source. If the VBR arrival flow conforms to the guaranteed traffic parameter (that is the number of assigned data slots on the frame), the generated packets will receive a high QoS. Otherwise, extra cells will receive a low QoS. When a queued packet exceeds its MTD, it is removed from the buffer. We have assumed that the guaranteed and best-effort queues have an infinite length. However, since packets are discarded when they exceed their MTD, overflow will be avoided if the queues' length are greater than MTD x PCR.

A control packet is transmitted to the base station during a connection if either the connection wants to re-negotiate its parameters, or the cell rate was equal to zero and packet arrival resumes or there is not enough space to include piggybacked information. The rate of change of information piggybacked on data packets allows the base station to know the source cell arrival rate after the packet arrival in the mobile. The base station can therefore predict when the subsequent packet arrivals occurred and future arrivals time. Furthermore, the scheduler can determine if the cells are guaranteed or best-effort packets and how many data slots are needed. When the connections' status information has been updated in the base station, slots are allocated to VBR connections according to their buffer status and connections QoS. The slot allocation algorithm is divided into two parts: guaranteed allocation and best-effort allocation. For the guaranteed service, VBR connections with packets in the virtual guaranteed queue are sorted in a list of increasing order of TTE where TTE = MTD – (Time – Arrival time of the first packet in the guaranteed queue). Assume that L slots are available for VBR transmissions and each VC is assigned one out of L ABR slots. The scheduler then allocates slots to the connections that have waiting packets in order of increasing TTE. The allocation algorithm can be summarised as follows [20]:

While (L > 0)
Allocate a slot to the first connection in the list
Remove the first packet in the guaranteed queue from the connection
L = L –1
Remove the connection from the list
if (There is a packet in the guaranteed queue of the removed connection)
Enter this connection in the list with the new TTE

Then, if L > 0, we allocate slots to the VBR connections with best-effort virtual buffer length of all active VBR connections (an active VBR connection for the best-effort allocation algorithm is defined as a VBR connection with best-effort packets in the buffer). The base station scheduler allocates the L slots that are still available to the connections according to the round-robin/fair-bandwidth allocation algorithm presented in Section 4.2. When the allocation for guaranteed and best-effort packets is over, the base station announces the slot allocation to MTs. However, the base station only indicates the VBR connections for which the allocation is made and not if the allocated slots are for guaranteed or best-effort packets. Each VBR connection uses its allocat ed slots to serve first all its queued guaranteed packets and then the best-effort packets.

4.4 Allocation Algorithm for Control Traffic

The number of slots allocated to control traffic is computed using the estimate of the number of backlogged control packets determined by one of the contention algorithms. Let CR be the number of control mini-cells per slot and CSmax the maximum number of slots that can be allocated for control purposes. The goal of the allocation algorithm for control traffic is to estimate the minimum number of control slots, smaller than the maximum allowed by CSmax, such that every control packet waiting for transmission will be sent during the next frame. This allows a dynamic adjustment of the number of control slots to obtain a compromise between a small number of control slots and a low transmission delay. The number of uplink control slots allocated is therefore determined as follows [20]:

Total = $\sum_1^p \hat{n}_i^t$
if = (Total ≥ 1)
Request = min ([Total/CR], CSmax)
else
Request = 0

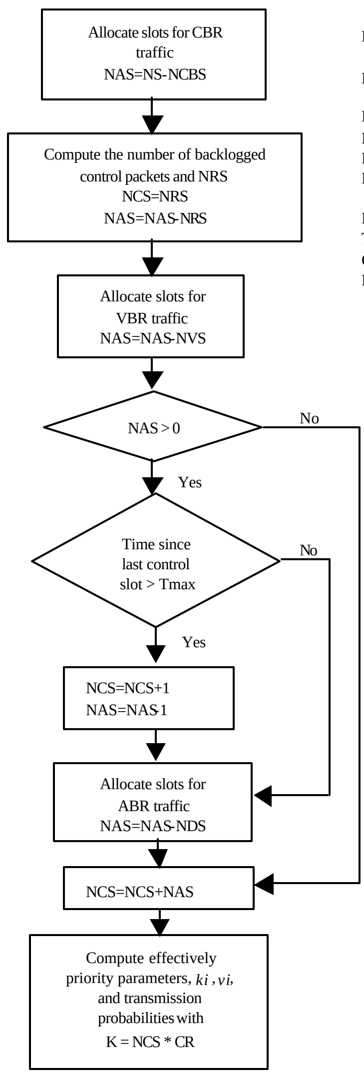
Furthermore, before allocating slots for best-effort traffic, if the time since the last allocated control slot exceeds a parameter Tmax, then a control slot is allocated for the next frame. This process ensures that a contention period will be regularly available to allow potentially time sensitive control packets to be transmitted. The parameter Tmax is therefore set such that delayed control packets do not result in loss of time sensitive packets. In our case, Tmax is set smaller than the MTD of voice packets. After slot allocations for user traffic have been done, all unused slots are allocated to the contention period for control packets.

4.5 Traffic Integration

We have described above how slot allocations are independently managed for each type of traffic. However, since the different WATM services share the same resources, an effective interaction between the allocation algorithms is needed to maximise the utilisation efficiency of the shared resources. Fig. 4 shows the algorithm used to integrate the different types of traffic for uplink transmissions using the AR-TDMA protocol. The available slots are distributed first to CBR traffic, second to control traffic, then VBR traffic and finally to ABR traffic. If there are any slots leftover, they will be assigned to control slots. This order of allocation is based on the different priorities and QoS requirements of the ATM services. The allocations for VBR traffic are made jointly by maintaining a single list of the active VBR connections with queued guaranteed packets in increasing order of TTE. Connections are then allocated slots in order of increasing TTE. Similarly, the algorithm maintains a single list containing the active ABR connections, and the fair bandwidth allocation algorithm is executed with this list with the first connection picked up randomly in a round-robin fashion. For control slots, the number of backlogged control packets at the beginning of the frame, and accordingly the number of required uplink control slots, is computed using the feedback from the previous frame's control slots. After the allocations for CBR, control, VBR and ABR traffic have been made, the number of control slots available in the current frame is known and the transmission probabilities can then be computed. Finally, the slot allocations and the contention parameters (number of control slots and transmission probabilities) are announced in downlink control slots.

5.0 PERFORMANCE EVALUATIONS

A *C* simulator has been written to evaluate the performance of the proposed wireless ATM MAC protocol. The program was run on *Microsoft Visual C++* (*MSVC++*) complier Version 6.0 and was linked with m*atlab* Version 5.2 to work in conjunction with the AFPBA/ASA and FPBA/APCC *matlab* simulators. As we focus on the performance of the proposed adaptive contention algorithms on the MAC protocol, we have assumed a wireless channel that consists of Rayleigh fader, slow fader and distance path loss for the reservation section (MT-B control region) of the protocol. For the rest of the frame, we have assumed a perfect radio channel without errors and fading.



NRS:	Number of requested slots for control purpose
NCS:	Number of allocated slots for control purpose
NS:	Number of slots in frame
NAS:	Number of available slots
NCBS:	Number of slots allocated to CBR traffic
NVS:	Number of slots allocated to guaranteed and best-effort VBR traffic
NDS:	Number of slots allocated to ABR traffic
Tmax:	Maximum time between control slots
CR:	Number of control mini-slots per slot
K:	Number of control mini-slots in the frame

Fig. 4: Traffic Integration Flow Chart [20]

The reason is that we do not include a Data Link Layer (DLC) retransmission algorithm for the corrupted packets transmitted in the data slots. Therefore, a perfect error-free channel is assumed there. The simulations were run for a minimum simulation time of 5000 seconds in order to assure accurate results. We have simulated the protocol with the system parameters given in Table 1 where the parameters for this channel are given. For the AFPBA/ASA and FPBA/APCC access protocols, the number of priority levels *p* is equal to 3. CBR control packets are assigned priority 1(high), VBR control packets are assigned priority 2 (medium) and ABR control packets priority 3 (low). The length of the window used for the moving time-average of the number of successful control packet transmissions for each traffic class is set to 100 frames. For most of the simulations, CSmax has been set to 2 and Tmax to 12 msec. We have assumed an infinite buffer model where cell losses are only due to exceeding the MTD.

We evaluate the efficiency of the protocol transmissions on both the uplink and downlink. We assume for Case One that the number of slots consumed by the MT-B control region on the uplink is equal to the number of slots consumed by the B-MT control region. The throughput is defined as the ratio of the average number of slots used for data packet transmissions per frame (excluding control packets) to the total number of slots available per frame. The offered load and achieved throughput thus include cell headers but do not include control packets. Therefore, control traffic contributes to a throughput reduction. We also did not consider preamble traffic sent at the connection admission, which will further reduce the achievable throughput. Furthermore, in the simulator we have not considered the exact position of allocated slots inside a given frame. This just marginally affects the delay experienced by a packet. In a real implementation, slot positions will be chosen such that packets that will exceed their MTD earliest will be allocated the first slot in the frame. In the simulation model, we have considered that all packet transmissions take place at the end of the frame. This decreases the simulation time and barely affects the results. Moreover, this approximation is reasonable since we do not know exactly the processing time and delay tolerance values.

5.1 Performance with Integrated Traffic

In this section, we present simulation results to show that the integrated allocation algorithm respects the required QoS of each traffic category while providing an efficient utilisation of the wireless channel. In order to completely investigate the performance of the integrated scheduling algorithm and the impact of each traffic and system parameter on the traffic QoS and protocol efficiency, a huge number of simulations is necessary. A first set of simulations have been run with 100 CBR connections, 100 ABR connections and VBR connections, all characterised with parameters in Table 5. The number of VBR connections is a varying parameter to illustrate the performance of the integrated system. Fig. 5 presents the voice and VBR traffic cell loss rates as a function of the number of VBR connections. The first thing that we can observe is that the integrated allocation algorithm meets the VBR and CBR QoS (i.e. 1% cell loss rate) for all the traffic conditions. Even for a throughput of 99%, as indicated in Fig. 6, the QoS of CBR and VBR traffic are maintained at the expense of ABR QoS. We note that the cell loss rate of VBR traffic is higher than for CBR traffic. This stemmed from the fact that the VBR traffic receives a medium priority (less than CBR traffic) service. The CBR cell loss rate plateau that we observed between 8 and 9 VBR connections might seem odd but similar results have been reported for similar traffic conditions [19].

Table 5: CBR, VBR and ABR connection parameters

Channel Speed	Traffic Type	No. of Connections	Average Connection Bit Rate	ON Duration	OFF Duration
8 Mbps	CBR	96	24 Kbps	1.00 s	1.35 s
	VBR	Variable	384 Kbps	2.00 s	2.70 s
	ABR	100	19.2 Kbps	$E(L)$ = 10 cells/burst	

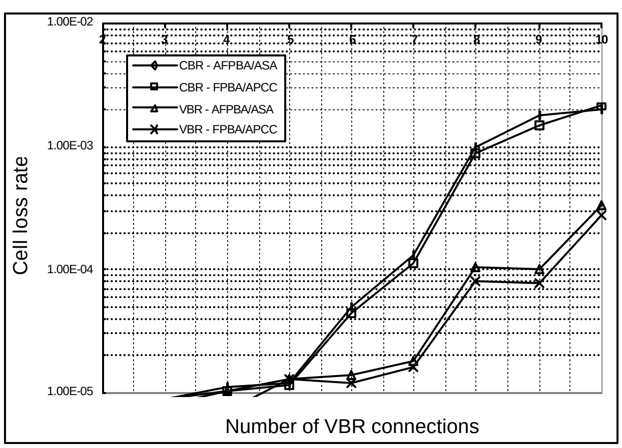


Fig. 5: Cell loss rate as a function of the number of VBR connections for the integrated system

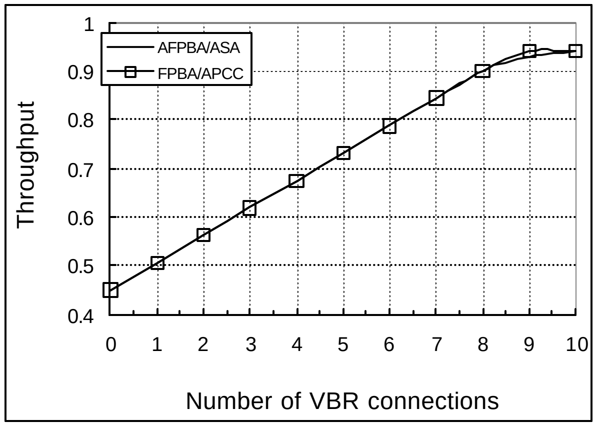


Fig. 6: Throughput as a function of the number of VBR connections for the integrated system

As explained before, the performance of CBR traffic is mainly determined by contention traffic which, in its turn, is influenced by CBR control traffic, ABR control traffic and the number of control slots available. When the number of VBR connections increases, the number of available control slots decreases, causing an increase in contention traffic which explains the increase of cell loss rate before the plateau (see Fig. 5). Before the plateau, the CBR and control traffic are almost constant (the number of CBR and ABR connections is constant), but when we reach 8 VBR connections, we see from the delay performance results in Fig. 7 the ABR delay increases dramatically. At this point, the ABR control traffic decreases because more requests are sent to the base station piggybacked on ABR packets. Therefore, two effects (i.e., the decreasing number of control slots and the decreasing data control traffic) nullify each other which creates the plateau. Then after, the decreasing number of available control slots becomes predominant which causes the increase in CBR cell loss rate. Fig. 7 shows that the VBR traffic delay approximately remains at its minimum value of 2 ms for all throughput values. On the other hand, ABR traffic experiences a minimum delay of 4 ms (as explained in Section 4.2) while the throughput is below 80%. When the offered load increases, the allocation algorithm can maintain a reasonable ABR traffic delay below 100 ms, up to a throughput of 96%. For higher offered loads, the ABR QoS rapidly decreases.

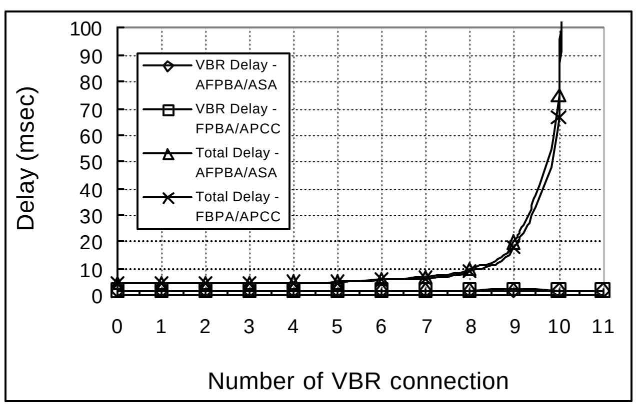


Fig. 7: Delay as a function of the number of VBR connections for the integrated system

In general, the FPBA/APCC protocol gives better performance compared to the AFPBA/ASA protocol in the integrated environment with different priority classes.

5.2 Performance with CBR and ABR Traffic

In this section, we discuss the system performance when CBR voice and ABR data traffic are integrated together on the wireless ATM channel. We have first evaluated the proposed system with the 8 Mbps channel parameters presented in Table 1. Table 6 gives the CBR voice parameters and the ABR source model and contention algorithm parameters used in the simulations. We have set an infinite limit on the maximum transfer delay for ABR packets. For each set of parameters, we have simulated the AR-TDMA with the AFPBA/ASA system and the FPBA/APCC system where CBR and ABR control packets are assigned a high and low contention priority, respectively.

Table 6: Parameters for data and voice integrated system simulations

Channel Speed	CBR Voice Connections	CBR Bit Rate	ABR Connections	$E(L)$	CR	CSmax
8 Mbps	80	24 Kbps	30	10	3	2

Therefore, we can evaluate the impact of the proposed contention protocols on the AR-TDMA WATM MAC protocol. We have also simulated the AR-TDMA MAC protocol with Case Two whereby the frame structure is similar to the NEC's model in [13] in order to investigate the impact of our proposed models on the performance of the NEC's model. Our simulation examines how the presence of CBR VCs operating across the wireless link affects ABR throughput and cell delay and how that, in reference, affect the CBR voice loss rate. The system was initialised with one base station and 15 MTs. Each MT has its own ABR VC. One ABR VC was chosen per MT for simplicity and to maximise frame overhead, while only 15 ABR VC pairs were chosen to increase the simulation speed. The mean ABR burst size was set at 10 cells. The CBR voice VC pairs were added in patches of 16 users at a time (8 users for the uplink and 8 users for the downlink). This is equivalent to transmitting one ATM cell (7 slots) per frame in each direction (one on the uplink and one on the downlink) and requires a modem preamble on the uplink. The CBR VC pairs were primarily chosen to examine the effects of reduced bandwidth availability on ABR VCs. Each patch of 16 CBR VC pairs required 6.4% of the total bandwidth of the wireless link.

In Fig. 8, we examined the priority schemes against a non-priority system that considers no priority between CBR and ABR traffic streams. From the cell loss results in Fig. 8, we observe that our priority schemes significantly reduce the voice cell loss rate, whereas it becomes too high without the priority protocols, and the FPBA/APCC contention algorithm performs slightly better than the FPBA/ASA contention algorithm. It has to be noted that for certain set of parameters, the voice loss rate can be low with the non-priority system and therefore, the impact of the AFPBA/ASA and FPBA/APCC protocols is less significant since there is not much to improve. Still, the results show that if the cell loss rate would become high enough to degrade the CBR voice connections performance, the AFPBA/ASA and the FPBA/APCC protocols keep it to an acceptable level where the cell loss rate is not a limiting factor of the AR-TDMA MAC protocol. Furthermore, the voice packet loss rate performance improvement due to the AFPBA/ASA or the FPBA/APCC protocols barely affects the ABR cell delay (Fig. 9 and Fig. 10) and throughput performance (Fig. 10 and Fig. 12).

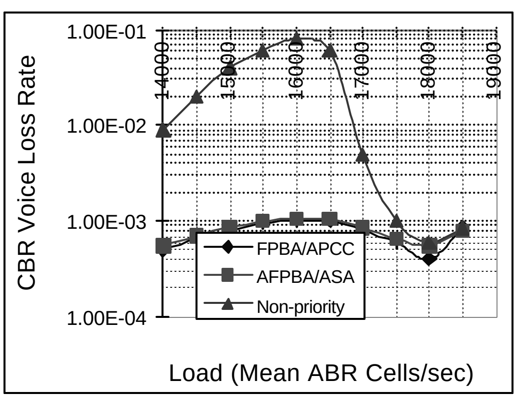


Fig. 8: Voice loss rate as a function of the load (mean ABR cells/sec) with 80 CBR voice connections (R = 8 Mbps)

The bell shape of the voice cell loss rate function can be explained by the effect of two different phenomena. When we increase the number of data connections, the traffic load and the number of control packets increase. There is, thus less control slots available to carry the higher control traffic. The contention traffic thus increases causing a longer waiting time for control packets (for both ABR data and CBR voice connections) and therefore increasing the CBR voice cell loss rate. However, while we increase the number of data connections, the data waiting time increases. Thus, a larger number of ABR requests are piggybacked on data packets. The control traffic then declines which explains the decrease in the CBR voice cell loss rate after a certain point. From the throughput results (Fig. 9, Fig. 10 and Fig. 11), we can see that the maximum sustainable throughput is quite high for the 8 Mbps AR-TDMA system utilising the AFPBA/ASA and FPBA/APCC contention algorithms. In Fig. 11, a comparison of the NEC's model and our AR-TDMA with a fixed MT-B and B-MT control regions was presented and it would be seen that the throughput achieved is quite high as well. Depending on the simulation parameters in Case One (Fig. 9 and Fig. 10) and Case Two (Fig. 11), the throughput varies between 96% and 98%. This means that only 7 to 10 slots per frame are used for control purpose or unutilised. The proposed AR-TDMA wireless ATM MAC protocol (utilising either of the contention algorithms) can support a high offered load (near the maximum that can be served by a perfect multiplexer) while maintaining the voice quality and the data waiting time at acceptable levels.

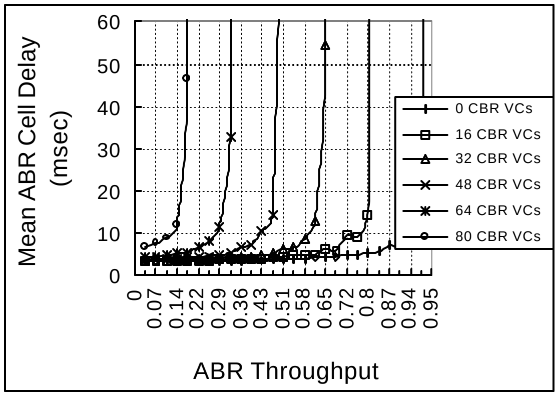


Fig. 9: Mean ABR VC Transmission /delay vs. Throughput for different numbers of CBR VCs. (AFPBA/ASA algorithm)

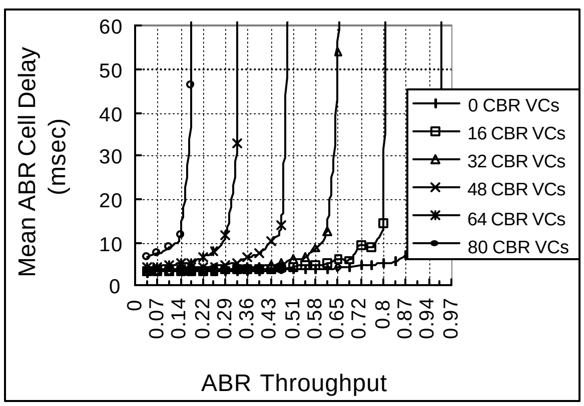


Fig. 10: Mean ABR VC Transmission /delay vs. Throughput for different numbers of CBR VCs. (FPBA/APCC algorithm)

In Fig. 11, we compared the AR-TDMA MAC protocol results (with the FPBA/APCC contention algorithm) with the performance of the NEC system presented in [13]. In the NEC MAC protocol, 12% of the slots are assigned for the uplink control period and 8% of the slots are assigned for the downlink control period (the maximum achievable throughput is thus limited to 80%). In our results, we have considered that the uplink and down link take the same transmission bandwidth as in [13] respectively, and therefore, our maximum achievable throughput is 80%.

The AR-TDMA throughput is reduced from 98.5% to 97.5%, while for the NEC protocol the maximum achievable throughput is 75%. The performance of the allocation algorithms for voice and data traffic are similar (i.e. achievable throughput versus maximum throughput), but the dynamic nature of the AR-TDMA uplink control period as well as the AFPBA/ASA and FPBA/APCC algorithms allow the AR-TDMA protocol to require less uplink control slots and achieve a much higher throughput than the NEC protocol.

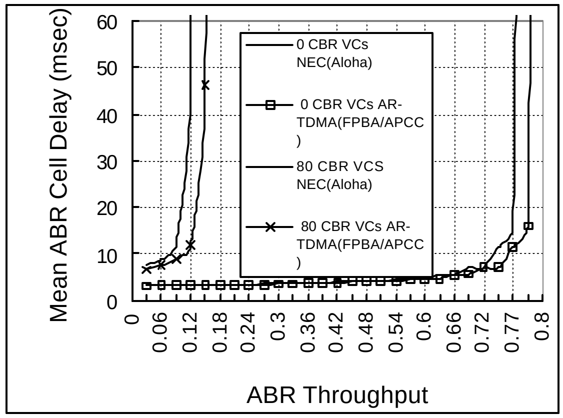


Fig. 11: Mean ABR VC Transmission Delay vs. Throughput for different numbers of CBR VCs. Comparison between AR-TDMA (FPBA/APCC) protocol and NEC (Aloha) protocol

Finally, we can highlight that our AR-TDMA MAC protocol with CBR voice and ABR data integration algorithm is quite efficient in keeping the CBR loss rate at low levels and maintaining near optimum throughput. We can also observe that the AR-TDMA protocol provides better QoS for voice connections compared to systems when priority is not implemented. However, the priority schemes cause a service deterioration for ABR data traffic but due to the use of piggybacked request transmission, the AR-TDMA protocol offers good QoS to ABR data while providing a better QoS to voice traffic. The AR-TDMA protocol can take advantage of the increasing statistical multiplexing gain to improve its performance and is not affected by the heavy contention traffic. Note that the CBR voice was always below 1% loss in all the simulation trials.

6.0 CONCLUSIONS

In this paper, we have presented a MAC protocol that efficiently integrates multiple B-ISDN traffic classes over a wireless ATM link. The motivation behind the development of this protocol was to obtain a high throughput efficiency while respecting the QoS requirements of the ATM traffic classes. We have introduced the application of the FPBA/ASA and FPBA/APCC protocols to manage the control slot access so as to improve the QoS of time-sensitive connections. Finally, an integrated resource allocation algorithm that provides slot allocation priorities to the different services has been described. We have presented simulation results that illustrate the performance of the AR-TDMA protocol under a wide variety of traffic conditions. The results indicate that AR-TDMA can provide throughput as high as 96% while limiting cell loss rates to less than 1% for both voice and VBR traffic, and data delay to less than 100 ms. We have also shown that the FPBP protocol can maintain the voice cell loss rate at under 1% regardless of the data traffic loads, whereas the voice cell loss rate would become unacceptable without contention access priority.

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