Media Access and Error Control Protocols for Wireless ATM

by

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ABSTRACT

ATM is a technology that is expected to unify the currently separate networks used for voice, video, and data applications while satisfying the desired quality of service (QoS) required by each traffic type. ATM was chosen as the technology for broadband integrated services digital networks (B-ISDN) backbone supporting all kinds of traffic types at high channel speed. On the other hand, wireless communications provide voice and data services for mobile and remote users. Thus, wireless ATM provides mobiles and remote users a reliable access to the fiber-optic based ATM network. In this work, we propose solutions for two main problems encountered in wireless channels, poor channel utilization and high bit error rate (BER). We propose an adaptive shared media access protocol that achieves higher channel utilization than conventional packet reservation multiple access (PRMA) protocol, and a hybrid error control protocol that achieves higher throughput than traditional automatic repeat request (ARQ) and forward error correction (FEC) protocols. The proposed protocols were numerically simulated and the results indicated that better channel utilization and higher channel throughput were achieved.

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List of Abbreviations

AAL	ATM Adaptation Layer
ABR	Available Bit Rate
ACK	ACKnowledgment
AFEC	Adaptive Forward Error Control
ARQ	Automatic Repeat reQuest
ATM	Asynchronous Transfer Mode
AWGN	Adaptive White Gaussian channel Noise
BER	Bit Error Rate
B-ISDN	Broadband Integrated System Digital Network
BS	Base Station
CCITT	Commite consultatif international telegraphique et telephonique
CDMA	Code Division Multiple Access
CLP	Cell Loss Priority
CODEC	COder/DECoder
CSMA/CD	Carrier Sense Multiple Access/Collision Detect
DId	Domain Identifier
DLC	Data Link Control
DLL	Data Link Layer
DLS	Domain Location Server
FDD	Frequency Division Duplexing
FDMA	Frequency Division Multiple Access
FEC	Forward Error Correction

GFC	Generic Flow Control
HEC	Header Error Control
HI	Handover Indicator
HR	Home Register
LAN	Local Area Network
LOS	Line Of Sight
MAC	Media Access Control
MId	Mobile Identifier
MR	Mobile Representative
MS	Mobile Station
MSC	Mobile Switching Center
MSP	Mobile Switching Point
NAK	Negative AcKnowledgment
NNI	Network Network Interface
PCS	Personal Communications Service
PCN	Personal Communications Network
PDU	Peripheral Data Unit
PRMA	Packet Reservation Multiple Access
PSN	Packet Sequence Number
PSTN	Public Switching Telephone Network
PT	Payload Type
PTI	Payload Type Identifier
QAM	Quadrature Amplitude Modulation
QoS	Quality of Service
QPSK	Quadrable Phase Shift Keying
RF	Radio Frequency
RSC	Reed-Solomon Codes
SAW	Stop And Wait
SRP	Selective Repeat Request

TDMA	Time Division Multiple Access
UNI	User Network Interface
VBR	Variable Bit Rate
VCI	Virtual Circuit Identifier
VPI	Virtual Path Identifier
WAN	Wide Area Network
WATM	Wireless ATM

Chapter 1

Introduction

Future telecommunications will provide a variety of services and mobile applications. The first feature is provided by asynchronous transfer mode (ATM) and the second feature is provided by wireless technology. ATM supports a variety of traffic types such as constant bit rate (CBR) like voice and video, variable bit rate (VBR) like internet applications, available bit rate (ABR) like file transfer and voice mail and unspecified bit rate (UBR) like mail messages. ATM also supports a wide range of data rates from 64 kilo bit per second (kbps) for digital telephone subscriber to one giga bit per second (Gbps) for fiber networks. ATM can be implemented in local area networks (LAN), wide area network (WAN) and broadband backbone networks. The next-generation of cellular network will support multimedia traffic with different quality of service (QoS) assigned to each traffic type. Wireless ATM is considered to enable the mobile or remote users to access the ATM backbone networks, and to improve the flexibility while providing end-to-end connection to mobile end users [1].

Wireless ATM is mainly motivated by the increasing number of mobile end users requiring the same guaranteed QoS supported by the ATM backbone broadband network. Integrating the high bit error rate (BER) and narrow bandwidth wireless networks into broadband networks with very low BER and bigger channel bandwidth, while maintaining the same QoS delivered to mobile users is a complicated task.

In this work, solutions are proposed regarding these two issues: The first is to enable more users to access the wireless ATM network while satisfying the QoS of each user by designing a wireless access control suitable for wireless channel in wireless ATM networks. The second is to reduce the cell loss ratio (CLR) in the high BER wireless channel due to fading and multipath effects, by designing an efficient error control protocol to maintain the QoS in the wireless ATM network.

1.1 ATM Technology

ATM is an emerging, cell-based technology that is expected to unify the currently separate networks used for voice, video, and data applications. Consolidation of these networks has the potential to provide significant economic benefits. The service is available in speeds ranging from 1.544 Mbps (T1 or DS-1) to 155 Mbps (Optical Carrier Level 3 or OC-3). Information transfer in ATM is connection oriented, i.e. a connection has to be established between the two end users before sending any information. ATM uses two connection concepts; the virtual channel (VC) and the virtual path (VP). A virtual channel provides a logical connection between end users. A virtual path defines a collection of virtual channels traversing the same path in the network [5].

ATM technology has the advantages of providing bandwidth on demand and QoS for different traffic types supporting circuit and packet switching traffic. In ATM networks, the data is divided into small, fixed length units called cells. An ATM cell is shown in Figure 1.1. A cell is 53 bytes long and contains 5 bytes as a header which comprise the identification, control priority and routing information for the User-Network Interface (UNI) and Network-Network Interface (NNI). The other 48 bytes of the cell are the payload.

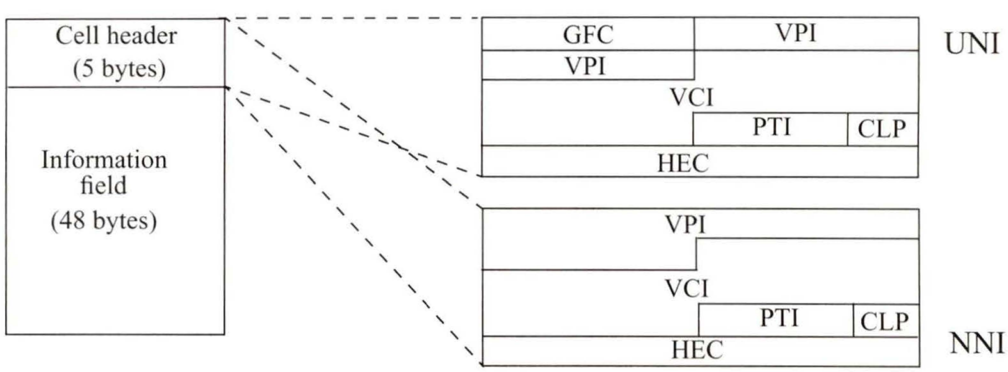


Figure 1.1 ATM cell header structure.

ATM does not provide any error detection operations on the user payload inside the cell and provides no re-transmission services.

The ATM cell header consists of the following fields:

1. Generic flow control (GFC) field is a 4-bit field defined at the UNI to assist the users in controlling their traffic flow according to a certain QoS. There is no GFC at the NNI.
2. Virtual channel identifier (VCI) is 12 to 16-bits at the UNI and 16 bits at the NNI. It identifies particular end-to-end switched connections and deals with switched functions of cells belonging to a certain logical connections. The value of VCI may change as the cell traverses the network.
3. Virtual path identifier (VPI) is 8 to 12-bits at the UNI and 12 bits at the NNI. It consists of a bundle of virtual channels, as illustrated in Figure 1.2, that are transported on the same physical media, from one end to another. It deals with the cross-connection functions of the cells. The VPI emulates the functions of the trunk concept in circuit switching, and it greatly enhances the concept of dynamic routing and resource management according to the required QoS for every VC grouped in the VPI. The VCI number is unique for VPI.

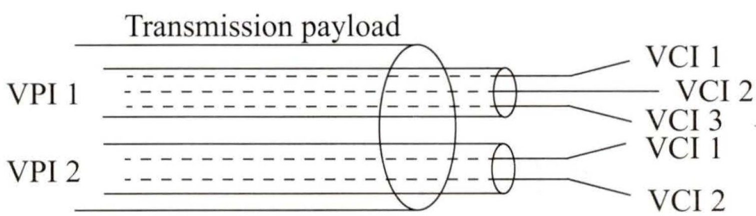


Figure 1.2 ATM connection identifiers.

4. Payload type (PT) is a 3-bit field and the payload type identifier (PTI) is used to distinguish network information from user information.
5. Cell loss priority (CLP) is a 1-bit field and is used to indicate the loss priority by the end point and to indicate selective cell discarding in network switches. Users can set the CLP bit to indicate a lower priority cell (CLP = 1), which may be subject to discarding depending on the network traffic conditions. If the CLP is not set (CLP = 0), the cell

has normal priority for traffic congestion control.

6. Header error control (HEC) is a one-byte field used for error detection and correction on the header.

Figure 1.3 shows the ATM layers in compliance with the international standards organization (ISO).

Upper Layers
ATM Adaptation Layer (AAL)
ATM Layer
Physical Layer

Figure 1.3 ATM layers.

The physical layer provides transmission of ATM cells over a physical medium that connects two ATM devices. The physical layer transforms the flow of cells into a steady flow of bits transmission over the physical medium

The ATM layer provides cell transfer capabilities and is common to all services. Figure 1.1 shows the ATM cell header structure at both the UNI and the NNI. The primary role of the header is to identify cells belonging to the same information stream.
The ATM layer is fully independent of the physical medium used to transport ATM cells. The main functions performed by ATM layer are

1. Multiplexing and de-multiplexing of cells into single stream on physical layer.
2. Translation of cell identifiers VCI and VPI separately or simultaneously.
3. Supporting QoS on the basis of CLP value.
4. Management functions like congestion control.
5. Extraction or addition of the cell header before or after the adaptation layer.
6. Implementation of flow control mechanism using GFC bits on the header.

The ATM adaptation layer (AAL) provides mechanisms for supporting different transport protocols such as TCP/IP over ATM cells. AAL1 and AAL2 have been defined by the Committee consultatif international telegraphique et telephonique (CCITT) for use in the wide area for support of constant and variable bit rate services (CBR and VBR), and AAL3/4 for connectionless data transport. The main functions of the AAL layer are

1. Segmentation of higher layer information into the ATM payload size and vice versa.
2. Supporting data transport over ATM.

ATM is a connection-oriented service, i.e. prior to receiving service, the user must request a connection to the intended receiver. Based on the set of traffic descriptors presented to the Admission Controller at the connection setup, the Admission Controller will attempt to find a route through the network such that, if the load represented by the traffic descriptors is presented to each switch encountered along that route, QoS guarantees will be met. If such a route can be found, a virtual connection number (VPI/VCI) is assigned for that call, and the routing tables of the intervening switches are provided with instructions for the routing of each ATM cell bearing that virtual connection number within its cell header. The user is now free to communicate over that newly established virtual connection. Once inside the network, all cells associated with a given virtual connection flow over the route are assigned to that connection and delivered in sequence. If no such route can be found, the user's connection request would be rejected. The AAL is responsible for converting a user's message into a sequence of ATM cells at the transmit end, and for reassembling ATM cells into complete messages at the receiving end. Here, a message may be an individual data packet, an image, or a continuous bit stream.

In a real circuit-switched network, each connection enjoys exclusive access to the resources reserved for that connection. Thus, once a user's connection has been admitted, it is not possible for other users to use the resources assigned to that connection. In a virtual-connection-oriented-network, however, resources are not assigned on an exclusive basis, but rather are statistically shared among multiple connections. Since each user does not continuously use all the assigned network resources, it is possible to share all the resources to the benefits of all users. Hence, it is possible for a virtual connection to unacceptably degrade the service quality enjoyed by other virtual connections. Accordingly, at the first switch or statistical concentrator, the network implements a policing procedure to ensure that cells not conforming to the service contract are either rejected or tagged as a violation cells candidates for deletion if they subsequently interfere with QoS guarantees.

1.2 Digital Wireless Technologies

Digital wireless systems are emerging as to accommodate the varying services provided, the increasing number of mobile and remote users, and to provide a high QoS. There are three main competing modulation schemes, time division multiple access, frequency division multiple access (TDMA/FDMA) and code division multiple access (CDMA).

The principles of TDMA and FDMA are basically simple. Traditionally, voice channels have been created by dividing the radio spectrum into (ever narrower) frequency RF carriers (channels), with one conversation occupying one duplex channel. This technique is known as FDMA. TDMA divides the radio carriers into an endlessly repeated sequence of small time slots. Each conversation occupies just one of these time slots. So, instead of just one conversation, each radio carrier carries a number of conversations as illustrated in Figure 1.4. The TDMA and FDMA, referred to later as fixed assignment protocols, are efficient in stream type applications, where the source generates data at a constant rate, and all the sources equally share the available bandwidth. An example of TDMA is the digital telephone system, whereas an example of FDMA is the first-generation analog mobile networks. The traditional narrow-band TDMA has been modified to support different traffic types over the same channel by changing channel access protocol [1].

In CDMA, several signals are simultaneously transmitted in the same portion of the spectrum by using orthogonal pseudorandom codes. CDMA was first established in analog systems, but with the motivation for digital cellular communications, it is established in digital systems as well [36]. The technique was initially proposed for military use, where the difficulties of detecting or jamming such a signal made it an attractive choice for covert communication. The occupied bandwidth is considerably greater than the information rate, depending on the number of orthogonal codes available for the transmitter.

Personal communication networks (PCNs) may use a shared wide-band CDMA band, and another digital cellular service may use TDMA[2]. There is a controversy among experts in the field regarding the relative merits of spread-spectrum CDMA and narrow-band TDMA/FDMA for private communication networks. The preferred technique may actually vary with the specific PCN application scenario.

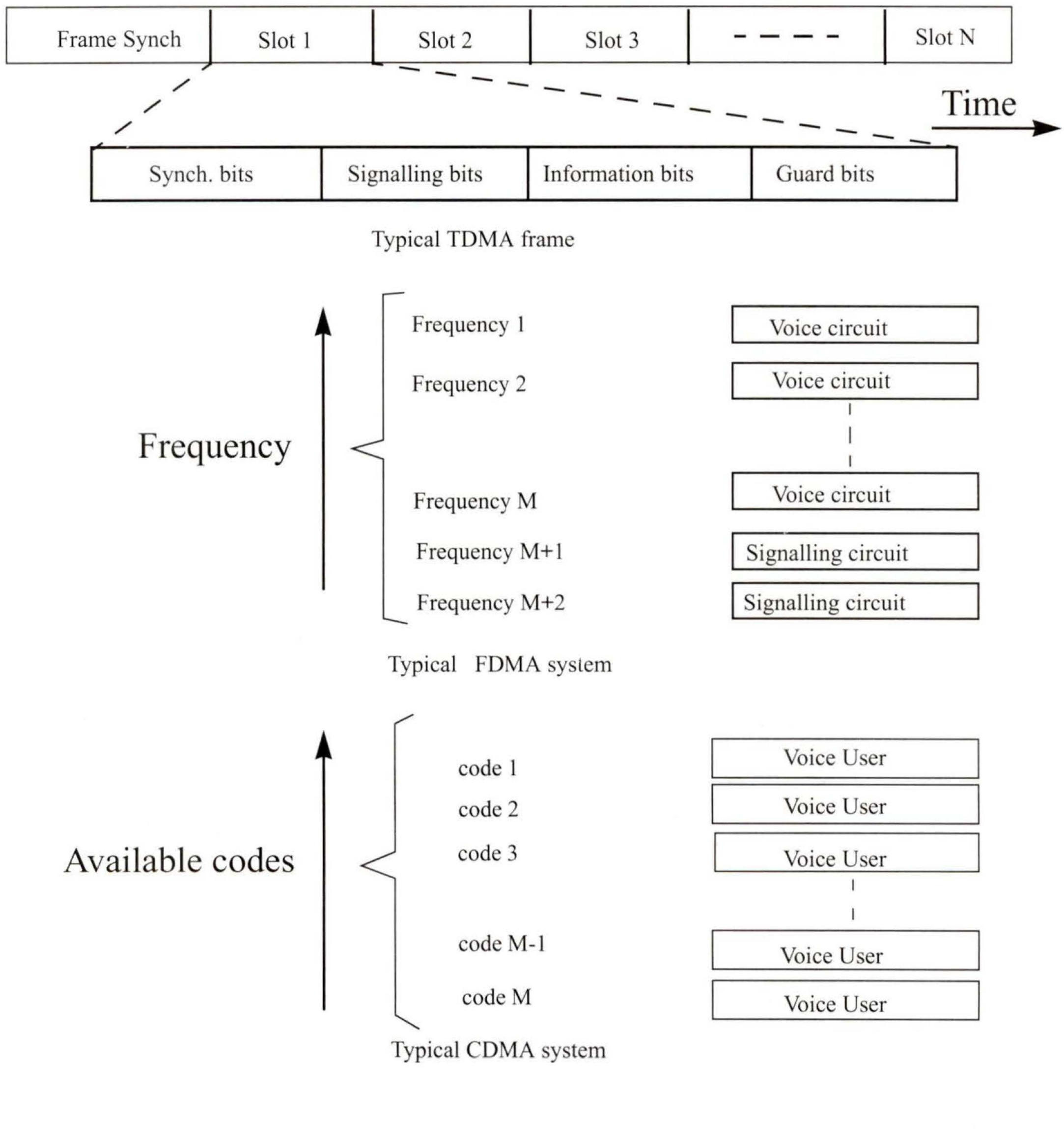


Figure 1.4 TDMA, FDMA and CDMA systems.

1.3 Cellular Communications

Modulation schemes and coding techniques by themselves are not enough to accommodate the increasing number of mobile end users. Space multiplexing is used to increase network capacity. PCN based on micro cellular technology is represented in Figure 1.5. A geographical region is divided into small cells. Each cell is served by a base-station and each group of base-stations are connected to a mobile switching centre (MSC). The diameter of each cell is assumed to be as small as few hundred meters, such that the round-trip propagation delay within a cell is of the order of a micro second.

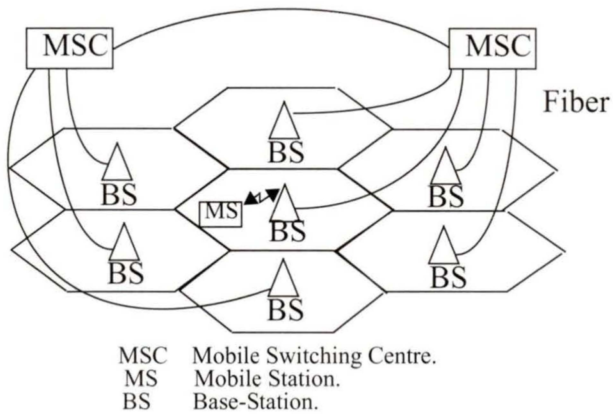


Figure 1.5 typical PCN system.

The adjacent cells will operate at different frequency ranges to avoid interference. The non adjacent cells can operate at the same frequency range for the purpose of frequency re-use. The wireless channel is divided into two parts. Up-link for information transfer from the mobiles to the base-station, and down-link from the base-station to the mobiles. as shown in Figure 1.6.

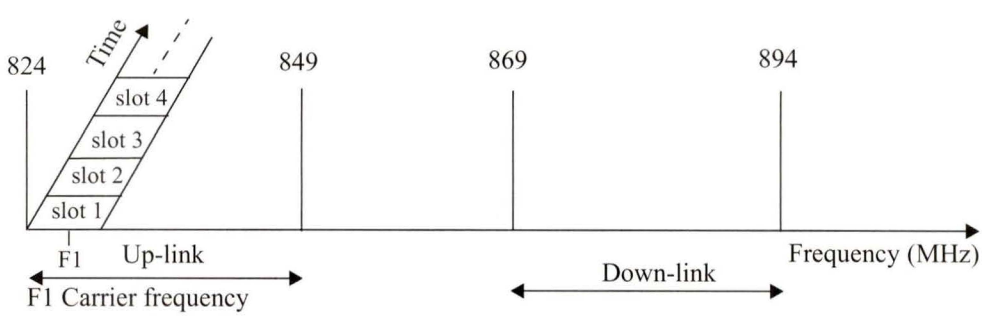


Figure 1.6 Frequency allocation for North American mobile system.

In TDMA/FDMA PCN system, the up-link and the down-links are divided into time slots of fixed lengths, forming a frame. Each time slot periodically repeated in each frame constitutes a physical channel. On the other hand, a logical channel is defined as any sequence of time slots, at most one per frame, not necessarily periodic. For each full-duplex connection, e.g. a voice connection, two logical channels are required, one up-link and another down-link.

1.4 Wireless ATM Overview

The design of a wireless ATM system has to be based on a vision of future wireless data communication requirements. The general aim of wireless ATM is to design integrated services wireless networks that provide connections to fiber-optic based ATM networks in a relatively transparent, seamless, and efficient manner. This means that the proposed systems should support a reasonable range of service classes, bit-rates, and QoS levels associated with ATM. It is recognized that there may be quantitative differences in achievable service characteristics due to fundamental limitations of the radio medium, but these are hoped to be kept to a minimum via innovative technical approaches [1]. Design requirements are shown in Figure 1.7.

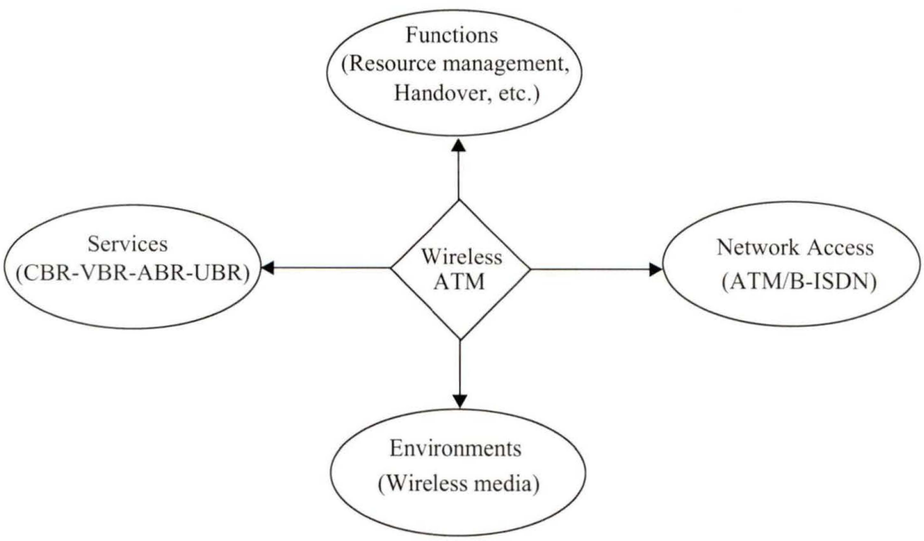


Figure 1.7 Wireless ATM design requirements.

The design requirements of wireless ATM are:

1. Network access: ATM/B-ISDN is the backbone network. The problem is simply how to extend the ATM and B-ISDN signalling to the wireless end-user over the unreliable radio link and how to include mobility into this wireless ATM network.

2. Environments: Wireless ATM should provide the same QoS of ATM networks for different traffic types. Suitable modulation techniques, error-correction protocols and shared media access control over the wireless channel should be implemented to maintain the QoS in the high BER wireless environment.

3. Services: Wireless ATM systems should support the same services in ATM, such as constant bit rate (CBR), variable bit rate (VBR), available bit rate (ABR) and unspecified bit rate (UBR) with the same guaranteed QoS.

4. Functions: The functions that are required to be implemented within the wireless ATM include mobility, such as supporting handover between adjacent cells and resource management, such as allocation of resources to guarantee the QoS guaranteed for each user.

Wireless ATM architecture and technologies are studied in details in chapter 2.

1.5 Dissertation Outline

The main two issues studied in this thesis are MAC and error control protocols for wireless ATM implementations. QoS of the provided multimedia applications to the mobile users must be sustained in wireless networks. The QoS includes available bandwidth, cell delay, cell loss ratio and other traffic parameters set when the mobile initializes a call. The multimedia applications include all traffic types where all services are provided with the associated QoS of each service type.

Chapter 2 introduces an overview of wireless ATM technologies. Overall structure and system architecture are presented and wireless ATM layers are discussed in detail. Location and connection establishment as well the handover are introduced. The chapter also provides an overall understanding of the wireless ATM system such that the motivation for the proposed MAC protocol and the proposed error control protocol introduced in chapters 3 and 4 respectively.

Chapter 3 introduces an overall view of existing MAC protocols for the wireless channel. Advantages and disadvantages of each MAC protocol are discussed. Packet reservation multiple access (PRMA) MAC protocol is discussed and a proposed adaptive MAC protocol achieving higher channel utilization and better QoS is studied. Simulation results of the proposed protocols are introduced and discussed in detail.

Chapter 4 introduces an adaptive error control protocol over the wireless channel. ARQ and FEC protocols are discussed. Advantages and disadvantages of each system are discussed and the motivation for the proposed error control protocol is explained. A hybrid protocol is introduced and simulation results of the proposed error control protocol are presented.

Chapter 5 summarizes thesis contribution and presents suggestions for the future work.

1.6 Thesis Contributions

The thesis addresses two main issues in the implementation of wireless ATM system. The first issue is the low channel utilization of the wireless channel in shared media access. The second issue is the high BER rate in the wireless medium which could result in QoS degradation when integrating the wireless network to ATM backbone.

To achieve high channel utilization over the wireless link, the media access control over the wireless channel should be able to support mixed traffic types, with different QoS. The proposed adaptive MAC protocol is based on assigning the bandwidth over the wireless channel on demand. The bandwidth is assigned to CBR users on circuit switching basis, and to VBR, ABR and UBR users on packet switching basis. Assigning the bandwidth based on the supported traffic sustains the QoS and results in better channel utilization. The proposed adaptive MAC protocol not only assigns the bandwidth on demand, but controls the bandwidth consumed by users requests, such that the wireless channel is better utilized than in traditional on demand assignments protocols.

While fiber-optic cables provide a very low BER transmission media for ATM applications, wireless channels suffer from very high BER. QoS can not be guaranteed over the wireless network unless a robust error control protocol is implemented on the wireless link. A proposed adaptive hybrid error control protocol is studied for the wireless channel. The adaptive protocol proved to be more efficient than the traditional forward error correction (FEC) and automatic repeat request (ARQ) protocols. The proposed adaptive hybrid FEC error protocol is implemented using Reed-Solomon (RS) codes and maintains high error correction capability for severe channel conditions.

Chapter 2

Review of Wireless ATM Technology

There are several factors that tend to favor the use of ATM cell transport for PCNs:

1. Flexible bandwidth allocation and service type selection for a range of applications.
2. Efficient multiplexing of traffic from bursty data/multimedia sources.
3. End-to-end provisioning of broadband services over wireless and fiber networks.
4. Making use of the available fast ATM backbone switching capabilities.
5. Improvement of service reliability with packet switching techniques.
6. Ease of interfacing with wired B-ISDN systems.

The basic idea of wireless ATM is to use a standard ATM cell for network-level functions, while adding a wireless header/trailer on the wireless link for the wireless channel specific protocol sublayers [1]. This chapter reviews the architecture of a wireless ATM network and outlines the basic principles of a wireless ATM system and design issues.

Section 1 reviews the wireless ATM system hierarchy and the elements of its structure and the wireless ATM cell format is studied.

Section 2 reviews the wireless ATM protocol stack such that the wireless ATM layers are fully compatible with the standard ATM layers. The physical layer, MAC and data link layer (DLL), through which the wireless ATM layers fit into the standard ATM layers are fully discussed.

Section 3 reviews the enhancements to support terminal mobility within fixed ATM network. These enhancements include location management, connection establishment, and handover procedure in wireless ATM system.

2.1 Wireless ATM Structure

A hierarchical wireless ATM structure is shown in Figure 2.1. The structure is based on integrating the wireless network into the ATM network. The base-station acts as a cell relay by removing the wireless header from the wireless ATM cell and replacing it with a standard ATM header. The output of several base stations are multiplexed into a single ATM switch port to simplify the switching.

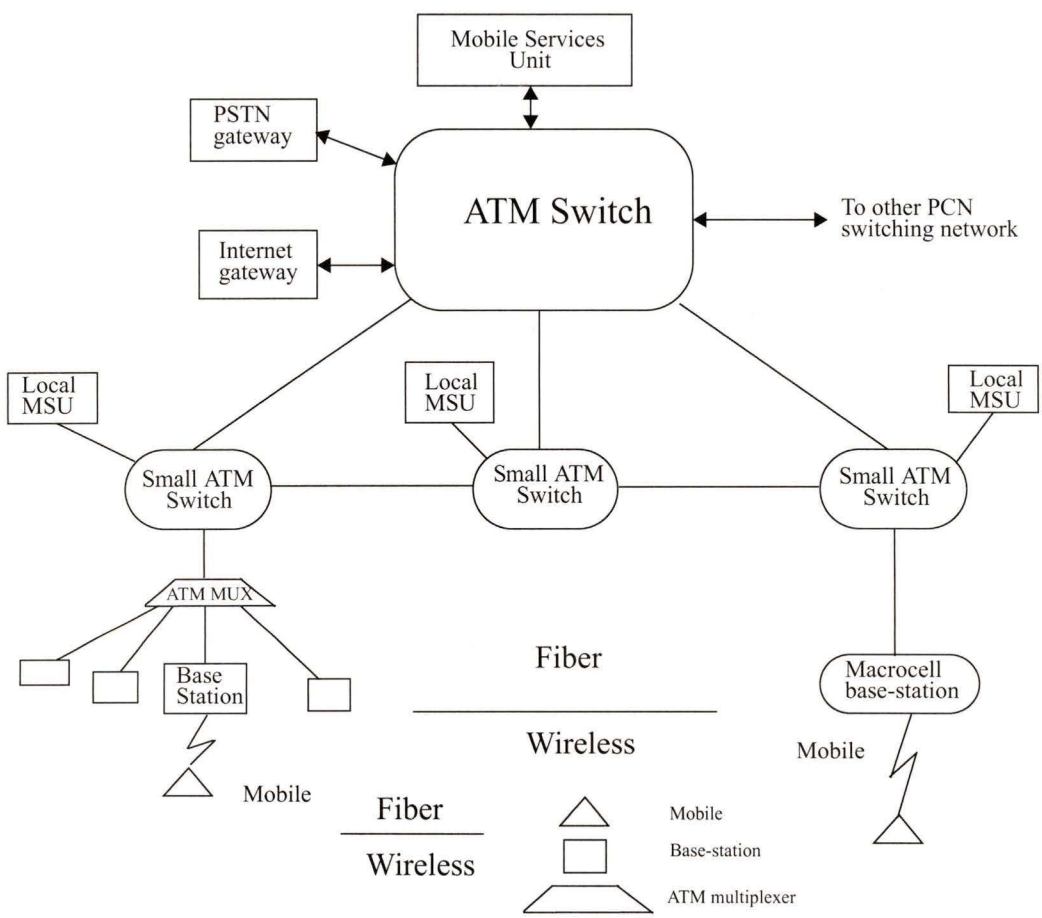


Figure 2.1 Wireless ATM on ATM backbone.

The wireless ATM hierarchy in Figure 2.2 shows that ATM virtual circuits with QoS control are supported on an end-to-end basis via standard ATM signaling functions which are terminated at the mobile unit.

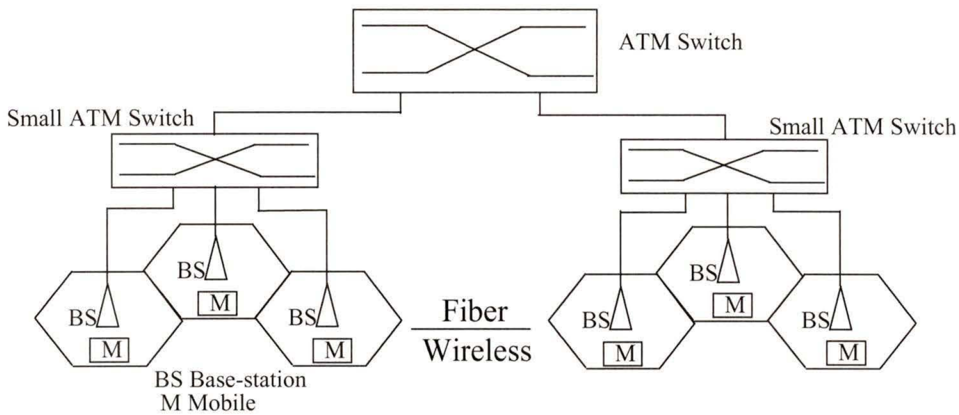


Fig 2.2 Wireless ATM hierarchy.

The role of the base-station in the wireless ATM as shown in Figure 2.3 is:

1. The base-station receives the wireless ATM cells from the mobiles.
2. The base-station performs cell format conversion by removing the wireless header and replacing the header with the standard ATM cell header (Cell relay).
3. The base-station delivers it to the fiber network via the multiplexer or directly to the ATM switch port depending on the base-station output traffic volume.

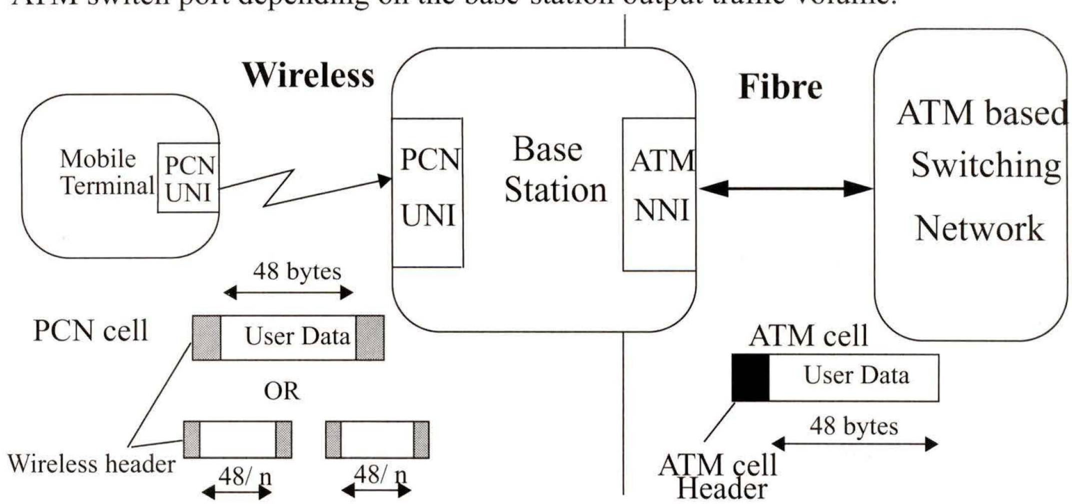
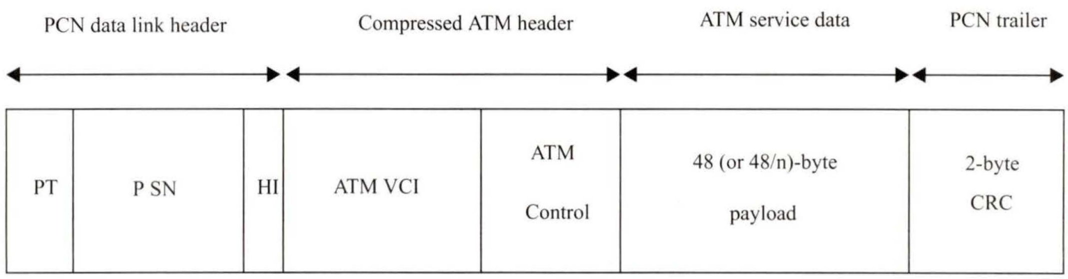


Fig 2.3 Wireless interface to ATM network.

The ATM cell size (53 bytes) was initially designed for channel rate of 64 kbps or higher, carried over fiber-optic. 53-bytes can represent a long packetization delay for low-rate applications (less than 64 kbps). Therefore, some wireless LANs may use 16 or 24-bytes payload in case of transmission rates less than 64 kbps [2]. The ATM header can also be compressed and expanded to standard ATM at the base-station [1]. An example of ATM header compression is to use two bytes containing 12-bits VCI and 4-bits control (payload type, cell loss priority etc.) [1].



PT Payload Type PSN Personal sequence number HI Handover Indicator

Figure 2.4 Proposed WATM cell format.

One of the cell formats proposed in [2] is to have a compatible pay-load size and addressing scheme, which could be different from the standard ATM cell format as in Figure 2.4. Mobility should be as transparent as possible to the end-points. Therefore, the VCIs used by the end-points should not change during the handover. The allocation of the VCI should remain valid as the mobile moves through different pico-cells within the same domain. This can be done by splitting the VCI space into a number of fields like Domain Identifier (DId), Mobile Identifier (MId) and Virtual Circuit number (VCn) for mobility management. A 16-bit CRC is also used to detect bit errors, due to high error rate of mobile networks. MId, DId and VCn, all together, form a unique field per mobile and this field will be used as a compressed VCI in the wireless media. In the mean time, WATM header should contain payload type (PT), packet sequence number (PSN), and handover indicator (HI) to form a radio-link control field in the WATM header as shown in Figure 2.4.

The format conversion from wireless ATM cell format to standard ATM cell format is shown in Figure 2.5 where the wireless control is removed and the compressed VCI field is expanded to the standard format.

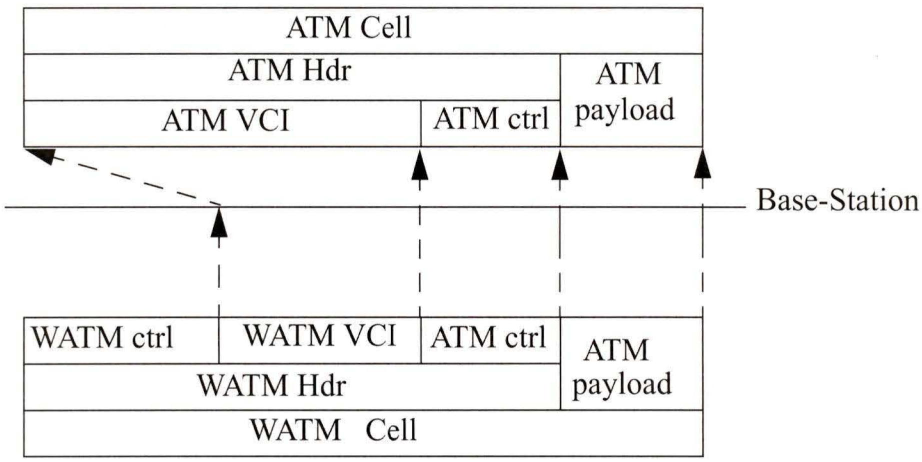


Figure 2.5 Cell relay. (WATM cell to standard ATM cell).

2.2 Wireless ATM Layers

The wireless ATM network has a protocol stack fully harmonized with that of standard ATM. The idea is to fully integrate the wireless channel-specific physical, MAC, data link control (DLC) and wireless network control sublayers into the ATM protocol stack as shown in Figure 2.6.

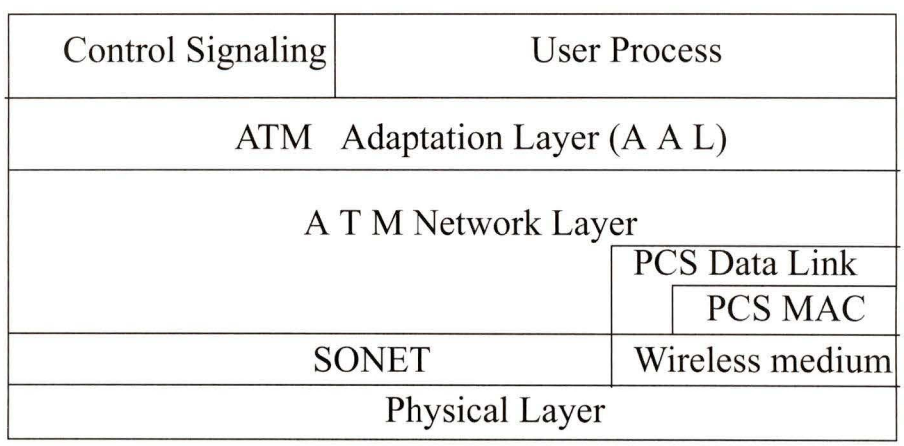


Fig 2.6 Wireless ATM Layers.

Since control functions are carried by the ATM processors in the fiber portion, the wireless ATM layers will carry the functions of physical layer and data link layer of ATM in the wireless portion of the network. The AAL layer functions are carried in the fiber portion as illustrated in Figure 2.4.

2.2.1 Physical layer

The basic design of the next-generation PCN is based on the selection of modulation techniques, and a set of bit rates. A bit rate in the range of 5-10 Mbps can be achieved using the existing wireless technologies in a pico-cellular environment. Thus, with the exception of HDTV, most other ATM applications can be supported (video services can be supported if some frames are carried on the wireless, and the rest of the frames are interleaved at the destination). The preferred technique may actually vary with the specific PCN application scenario to be addressed, so that it is likely that both TDMA and CDMA solutions will co-exist (Hybrid CDMA/TDMA) [1].

CDMA provides an efficient integrated solution for frequency reuse and multiple access and can typically achieve a net bandwidth efficiency 2-4 times that of comparable narrow-band approaches [1], [10]. However, a major weakness of CDMA for multi-service PCN is that for a given system bandwidth, spectrum spreading limits the peak user data rate to relatively low value [1] due to the reuse factor.

Narrow-band TDMA can be used to achieve higher bit rates, as the implementation is well understood. In a pico-cellular environment, we can achieve a bit rate in the range of 8-16 Mbps by using the narrow-band approach. Overall, with a good physical level design, it should be possible for macro (5-10 Km), micro (0.5 Km), and pico (100 m) cells to support baud rates of the order 0.1-0.25 Msym/s, 0.5-1.5 Msym/s and 2-4 Msym/s, respectively, where the symbol is the transmitted codeword over the wireless network and the baud rate is represented by sym/sec [1]. These rates should be sufficient enough to accommodate many of the broadband services as explained above except for HDTV which requires 100 Mbps.

2.2.2 Media access control (MAC)

One of the major problems of wireless ATM is to find a suitable channel sharing media access control technique at the data link layer. Shared media access leads to poor quantitative performance in wireless networks. The basic access schemes in the wireless media are CDMA, TDMA/FDMA, carrier sense multiple access / collision detect (CSMA/CD) and Polling. The main advantages and disadvantages of each scheme are listed as follows:

1. CDMA is the de-facto mode of spread spectrum operation [8]. Performance results from earlier studies show that packet CDMA can achieve good traffic multiplexing efficiency and performance for CBR, VBR and low-speed interactive data services [1]. CDMA manages more mobile users per cell, but limits the maximum data transmission rate of each mobile.
2. Narrow-band (TDMA/FDMA) can be used for higher data rates and maintains high channel utilization for CBR traffic. A dynamic TDMA MAC protocol can achieve higher channel utilization for VBR and ABR services on demand. A technique based on Slotted ALOHA is used for collision resolution of requests. The main disadvantage of the narrow-band access is the large number of collisions [9].
3. The CSMA/CD protocol is based on avoiding the collision of requests by sensing the carriers using a sniffer. CSMA/CD gives the required performance on coaxial cables, but is not suitable for wireless where all the mobiles in a cell are not in communication with each other. The main disadvantage of CSMA/CD is that it requires more bandwidth than other schemes since the mobile has to sense the wireless channel and listen for acknowledgment sent on the down-link before sending a request.
4. Polling. The polling protocol is based on assigning a bandwidth to the mobiles on demand. Each mobile has to be polled using a token signal each frame whether it has queued ATM cells or not. The mobile has to send an acknowledge whether it has data to send or not. The polling protocol is easy to implement since it has a simple control management. On the other hand, the bandwidth consumed by token signals and acknowledgments of mobiles having no data to send, leads to a poor channel utilization in the case of high channel load.

One of the problems of the contention-based methods (CDMA, TDMA and CSMA/CD) is that in mixed-media, applications access can not be prioritized. The main problem of Polling scheme is the low channel utilization in the case of high traffic volume.

The challenge in designing the MAC protocol for wireless ATM is to identify a wireless capable MAC providing a sufficient degree of efficiency and transparency for multimedia applications.

2.2.3 Data link layer (DLL)

Wireless ATM needs a data link layer protocol, and it should be as transparent as possible as shown in Fig 2.7. A custom data link protocol is needed due to the high error rate and different packet size of wireless ATM. Wireless ATM may use 16 or 24-bytes payload (in case of lower rate than 64 kbps) as 53 bytes may be too long for wireless ATM. The data link protocol may contain service type definition, error control, segmentation and reassembly for wireless ATM cells, and handover support.

<table>
<tr><td>WATM ctrl</td><td>WATM VCI</td><td>ATM ctrl</td><td rowspan="2">ATM payload</td><td rowspan="2">CRC</td></tr>
<tr><td colspan="3">WATM Hdr</td></tr>
<tr><td colspan="5">WATM Cell</td></tr>
</table>

Figure 2.7 Mapping ATM payload to WATM cell.

Payload type field is needed so as to indicate whether a packet is of type supervisory, control, CBR, VBR, ABR, etc. The PT field simplifies the base-station protocol processing. Wireless ATM should provide control over errors due to high noise interference and poor physical level characteristics of the wireless medium. This is achieved by using a PCN packet sequence number field in the header along with a standard 2-byte CRC frame check sequence trailer [1]. Since wireless ATM may use 16 or 24-byte cells, segmentation and reassembly is required which can be done using the packet sequence number (PSN) in the wireless ATM cell header.

Handover is an unavoidable feature of mobile cellular communications. Handover occurs when the mobile unit leaves the area of one cell and enters the area of another. There are two types of handover, soft handover and hard handover. In soft handover, the mobile does not have to change the carrier frequency when moving from one cell to another. In hard handover, the carrier frequencies of up-link and down-link will change when the mobile moves from one cell to another. Therefore soft handover without any data loss is important for any wireless network. This can be implemented by using bits in the header (PSN field) which indicate PDUs before and after the handover.

2.3 Wireless ATM Design Enhancements

In this section, design enhancements for wireless ATM are studied. The main design enhancements of interest are location management and connection establishment, and handover.

2.3.1 Location management and call establishment

In order to establish a connection between a mobile unit and the base-station, the mobile must first be located. Two techniques are needed to achieve this; searching and registration [2]. Searching involves a form of broadcast in which the whole network is queried. In registration, mobiles are responsible for their own registration at a well-known registration point. Subsequent enquiries about the mobiles are directed to this register using a static routing mechanism. The architecture explained in [2] uses a hierarchical registration scheme to simplify the searching process before connection establishment. The elements of the hierarchy as shown in Figure 2.8 are Mobiles, Base-stations, Mobile Representatives (MR), Mobile Switching Points (MSP), Domain Location Servers (DLS) and Home Registers (HR).

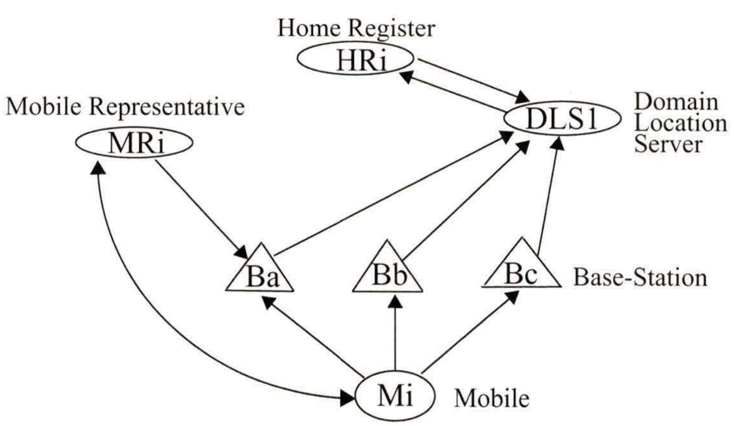


Figure 2.8 Registration, control and signalling in the proposed WATM structure.

The elements of the wireless ATM searching and registration are:

Mobile	A mobile user which is interacting with the network.
Base-station	The access point for a group of mobiles, acting as a gate between the wireless and the fiber/copper part.
MR	A software entity handling control and management functions for the associated mobile.
MSP	A switching point within the network which is responsible for connection establishment in the fiber/copper part of the network.
DLS	A server responsible for the mobiles management within the domain.
HR	A server responsible for storing the current locations of the mobiles.

Locating a mobile is presented in the following steps

1. When a mobile is within a domain, it is registered at the appropriate DLS and this registers the mobile at its HR.
2. The HR keeps a record of the mobile current DLS location. Each mobile has a statically bound home address which is mapped to the HR address as shown in Figure 2.7.
3. The mobiles should be allowed to roam freely from cell to cell. This process should not require any user intervention. Detection of movement in the total access communications system (TACS) is performed by having the base-station monitor the received signal strength from the mobile. If the signal falls below a preset limit, the switching centre looks for another base-station which is receiving a stronger signal and transfers the call to that station. An interval can be set aside in each frame during which newly arrived or activated mobiles attempt to inform the base-station of their presence.

Connection establishment is presented in the following steps

1. In connection establishment, the host generates a connection signal, specifying the network address of the two end-points. If the destination address is that of the mobile unit, then the local DLS is contacted.
2. If the local DLS has no knowledge of the unit, then the routing information is forwarded to the HR of the mobile. The HR returns the address of the remote DLS which

in turn returns the address of the mobile MR.

3. Once the address of the MR is known, then all the requests are directed to that particular MR. When the connection request arrives at the MR, it first consults the mobile, which in turns decides whether to accept the call or not.
4. If the mobile accepts the call, then it allocates a virtual circuit number and returns it to the MR. The MR then creates the virtual channel between the MSP and the remote end-point and adds the new virtual channel to the active and inactive virtual paths between the MSP and the base-stations close to the mobile.

2.3.2 Handover

The handover is based on the concept of latent virtual circuits. A latent virtual circuit is a virtual circuit established at the MSP that will be used in the handover process when the mobile moves to another cell. The MSP will establish a latent virtual circuit for all adjacent cells to the mobile cell and will replace the old VPI/VCI field with the new VPI/VCI of the latent virtual circuit in the ATM cell header in case of handover[2]. In order to perform handover between pico/micro cells (pico-cell radius is from 10 to 50 meters while micro-cell radius is from 100 to 500 meters), the incoming virtual circuits are mapped between the previously active virtual circuits to the latent virtual circuits simply by changing the virtual path. Although the base-station may not be relaying cells for a mobile, it may be receiving cells from the mobile and can monitor the quality of the link.

The information is forwarded directly to the MR which also has knowledge of the physical relationship between base-stations to make decisions about when the handover is wise. Handover becomes necessary when the quality of the wireless link falls below a usable level. However, the normal case is that a link to a new base-station will be better than the current one before the throughput falls, so handover will take place before there is any data loss. The physical topology of the network, the actual bit error rates at the receiver and the received signal strength indicator (RSSI) are used to make the handover decision.

The control messages from the MSP for handover between base-stations B_i and B_j

3. The MSP enables transmission between B_j and the mobile.

4. The MSP Informs mobile of the handover and cancels reception at B_i from the mobile.

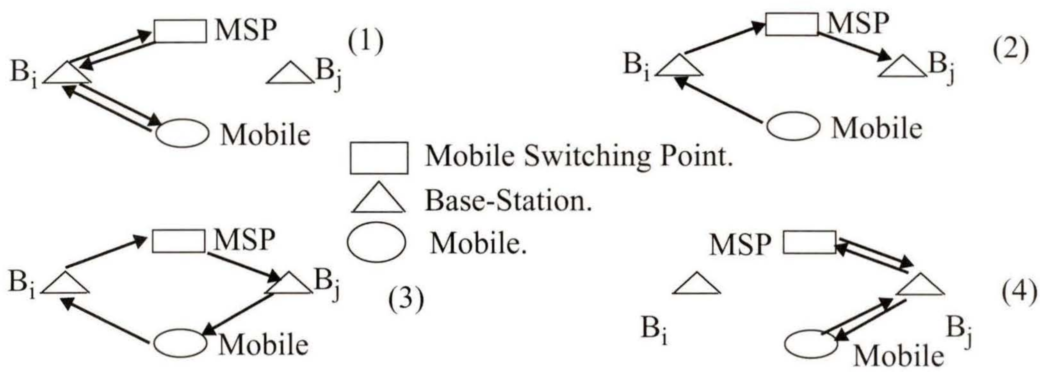


Figure 2.9 Handover procedure for pico / micro-cellular networks.

The protocol is based on the assumption that the mobile is always registered in the nearest base-station from which it receives the strongest signal. The handover algorithm is explained for the basic structure shown below in Figure 2.10.

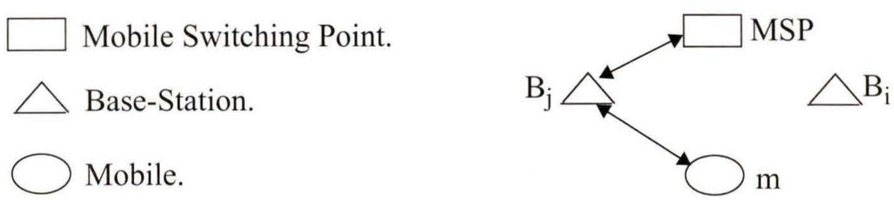


Figure 2.10 Basic Wireless ATM structure.

We begin by considering a packet destined for mobile m and received by the network layer of MSP from the fiber-optic network. The algorithm is based on a data link control (DLC) function at each node which manages the link between the node and its neighbors [9]. Figure 2.11 illustrates the handover algorithm between two base-station B_j and B_i. The handover algorithm can be explained in the following steps:

1. The network layer of MSP contains a routing table. The entry corresponds to the mobile m indicating that it has a connection to B_i.

2. When the mobile m moves from the cell of B_i to the cell of B_j, the mobile starts (JOIN) session with base-station B_j to establish a connection.
3. The base-station B_j starts (LEAVE) session with the base-station B_i informing it the mobile m is moving into B_j cell.
4. The base-station B_i starts (STOP) session with the MSP informing it to stop delivering ATM cells destined to mobile m since handover procedure is in process.
5. Base-station B_i delivers the queued ATM cells destined to mobile m in its buffer to B_j.
6. The MSP starts (LAST) session with the base-station B_i to remove it from the routing record of mobile m.
7. The base-station B_i starts (LAST) session with B_j informing it to confirm connection establishment with the MSP.
8. The base-station B_j starts (RESUME) session with the MSP such that the MSP will update the mobile m record in the routing table to indicate that it is connected to B_j.
9. The MSP starts delivering the ATM cells destined to the mobile m to base-station B_j.

The MSP will continue to deliver the assigned ATM cells for mobile m to base-station B_j until the mobile enters another cell controlled by another base-station.

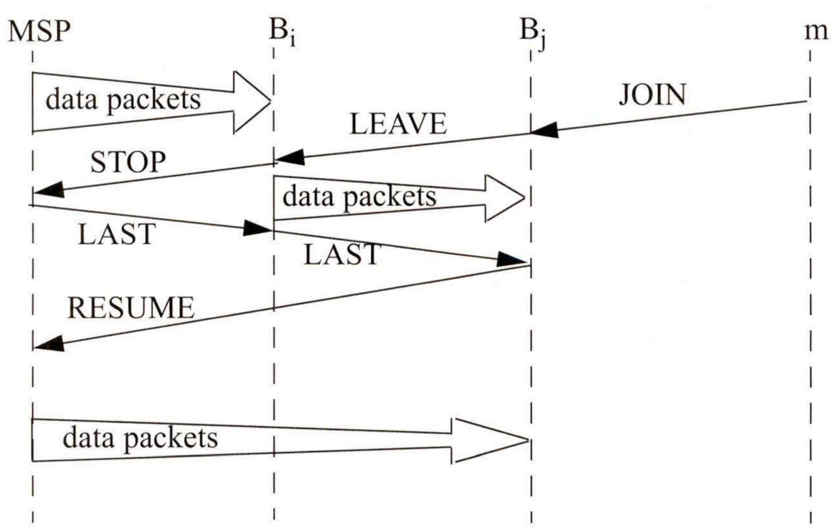


Figure 2.11 Handover Algorithm between two base-stations.

Since the physical layer, MAC and data link layer are the layers at which the wireless ATM layers fit into the ATM protocol stack, the design issues and enhancements in wireless ATM are mainly concerned with these layers. In chapter 3, a proposed adaptive MAC protocol enabling more voice and data users to the wireless network is introduced for wireless ATM implementation. In chapter 4, an adaptive error control protocol using a combination of AFEC at the physical layer and ARQ at the data link layer is introduced to improve wireless channel throughput.

Chapter 3

Media Access Protocols

A major technical issue related to PCN design supporting different types of services (multiservice PCN), is the selection of a suitable channel sharing media access control (MAC) technique at the data-link layer for mobile-to-base-station transmission discussed in section 2.2. The MAC technique used in PCN has a significant impact on system performance, system capacity, and remote terminal complexity. Next-generation PCN will be required to handle a more diverse mix of traffic types, including connection-oriented constant bit-rate (CBR) and variable bit-rate (VBR), as well as connectionless packet data and burst (file transfer) data. Thus, the adopted MAC technique must provide mechanisms to deal with each of these B-ISDN type services at reasonable QoS levels. It is recognized that the shared media access characteristics of wireless will lead to poorer quantitative performance than that achievable in fiber-optic networks. The challenge is to design a wireless multimedia-capable MAC protocol that provides a sufficient degree of transparency for many ATM applications and preserves the guaranteed QoS. Due to the limitation of the user peak data rate in CDMA systems, TDMA-based MAC protocols are studied in this thesis.

In this chapter, the basic MAC classes, their advantages and disadvantages to justify our choice of TDMA as the basis for the MAC protocol for wireless ATM are reviewed. TDMA MAC protocol based on polling, PRMA-based TDMA MAC protocol and a modified version of it are studied as protocols upon which the proposed MAC protocol for wireless ATM will be based. Proposed adaptive MAC protocol is introduced and numerical simulations are reported. Simulation results are studied at the end of the chapter. Channel utilization and call blocking probability of the proposed MAC are compared with those of conventional PRMA++ for the same traffic model.

3.1 MAC Classes

The main MAC design issue is to support VBR and packet data services with high channel utilization while maintaining a reasonable QoS for each service. When narrowband modulation is used at the physical level, some form of TDMA-based medium access control is typical for PCN systems supporting isochronous CBR traffic as in second generation digital cellular networks[20, 21]. When multiservice VBR and packet data requirements are also taken into account, the static TDMA approach must be extended to provide dynamic resource allocation. A notable early proposal for integration of voice and data in the wireless scenario is the Packet Reservation Multiple Access (PRMA) protocol proposed in [21]. Other variations of dynamic TDMA have also been proposed for multiservice PCS [13, 22].

Many channel access protocols have been proposed and analyzed in the past few decades [4-7, 36] for wireless, copper and fiber-optic network environments. Each scheme has its advantages and limitations, providing acceptable performance only in certain environments and with certain types of channel traffic. MAC protocols can be grouped into five classes [15]:

1. Fixed assignments.
2. Random access.
3. Centrally controlled demand assignments.
4. Distributed controlled demand assignments.
5. Adaptive strategies and mixed modes.

Classes 1,2,3 and 4 will be explained since they are the basis of the existing MAC protocols. Class 5 is still under research.

3.1.1 Fixed assignments protocols

Fixed assignments protocols, such as TDMA and FDMA, incorporate permanent sub-channel assignments in the time or frequency domain for individual users. These classical schemes perform well with stream-type traffic. The result is high utilization of the communication channel, and low channel access times. Fixed assignment techniques, however, are inefficient for bursty traffic applications. The fixed assignments protocols are not suitable for mixed traffic types with long idle durations.

3.1.2 Random access protocols

Bursty traffic is serviced more efficiently by a random access protocol. The ALOHA and carrier sense multiple access (CSMA) schemes are typical examples of random access protocols [22]. Random access protocols make the full channel capacity available to users based on random user requests. The random access protocols are packet oriented, whereas the fixed assignment protocols are slot oriented. The channel capacity is dynamically allocated on a per-packet basis. Random access protocols perform like fixed assignment protocols in case of light traffic volume, but the performance deteriorates as the traffic volume increases due to increased probability of requests collisions.

3.1.3 Centrally controlled demand assignments protocols (Polling)

Like the random access protocols explained above, demand assignment protocols provide the channel capacity to users on demand basis as requested by the mobiles. Unlike random access protocols, demand assigned protocols involve two stages

1. Reservation stage.
2. Transmission stage.

Demand-assigned protocols achieve high channel throughput by requiring users to reserve portion of the channel bandwidth. A portion of the channel capacity is required in this reservation stage. An example of demand assignment with central control is polling where each user is sequentially addressed by the base-station for transmission privileges.

3.1.4. Distributed controlled demand assignments protocols (contention)

In distributed control protocols, mobile users contend for bandwidth reservation. Requests are made on either a fixed-assignment basis, for the voice users, or contention basis for the data users. The distributed control demand assignments achieve high channel throughput for mixed traffic types but requires complex control management.

Based on the MAC protocols discussed in the previous sub-sections, fixed assignment protocols, were ruled out since the fixed slot assignment can not support different types of traffic such as VBR and ABR [22]. Random access protocols suffers from low channel utilization. Wireless ATM MAC must accommodate different traffic types while maintaining high channel utilization.

Demand assignments protocols are the basis of the MAC protocols for wireless ATM. In next section, centrally controlled demand assignments based on polling and distributed control demand assignments based on reservation are studied to justify our proposed MAC protocol.

3.2 Demand Assignments TDMA Based MAC protocols

Demand assignments TDMA based MAC protocols are chosen as the basis for wireless ATM MAC protocols for the following reasons:

1. There are no limitations on the data rate of the wireless ATM users, such that any number of the time slots per channel can be assigned for any user depending on data rate and QoS.
2. The demand assignments control in the base-station simplifies the access system schemes and the end users equipment design.
3. Better wireless channel monitoring and resources allocations are achieved in demand assignments MAC protocols by reducing the control traffic over the wireless network.

MAC protocols based on polling and PRMA are introduced in the following sub-sections.

3.2.1 MAC protocol based on polling

In MAC protocol based on polling, the base-station monitors the registered mobiles within its cell and polls them sequentially within a polling cycle, using tokens, to find out if any mobile has information to send. In this protocol, the base-station must have complete information about the registered mobiles, and the round trip delay is small compared to the time slot duration. The MAC protocol illustrated in Figure 3.1 is presented as follows:

1. A mobile cannot send ATM cells until it receives a token signal from the base-station.
2. When a mobile receives a token from the base-station and has information to send, it sends a number of ATM cells decided upon by the channel control management at the base-station.
3. If the mobile has nothing to send, it responds with a short pilot tone, to indicate that it has no data to transmit.
4. Within each polling cycle, the base-station sequentially sends a token to each mobile and immediately prepares for a reply from that remote.
5. After all the mobiles have been polled, the base-station can sequentially send ATM cells to the assigned mobiles on the down-link. The next polling cycle begins when either the time out interval has elapsed or the base-station has no cells to send, whichever occurs first.

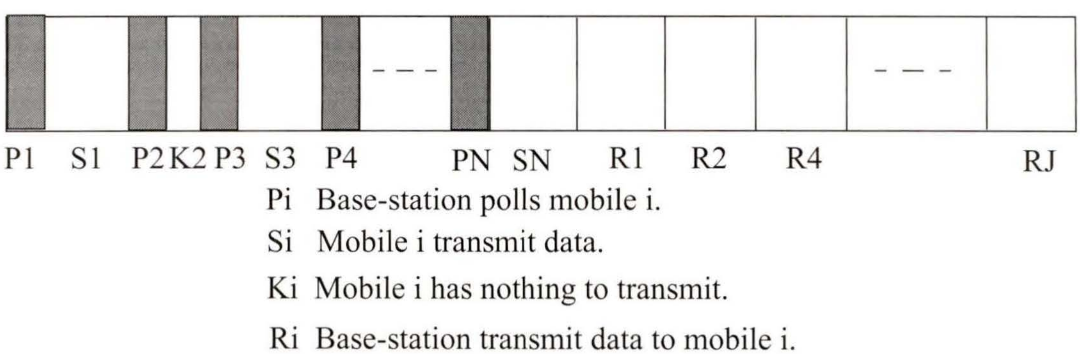


Figure 3.1 MAC protocol based on polling.

The protocol belongs to class 3, as mentioned before, and meets the following criteria:

1. The base-station controls the channel access and the mobiles send ATM cells only when they are polled.
2. The base-station keeps track of the mobile locations based on the information received from them, that is why a mobile has to send the token when polled, even though it has nothing to send.
3. Channel utilization depends on the number of registered mobiles in the cell, since the mobile is polled whether it has data to send or not.

Channel utilization of the MAC protocol is given by the following equation:

$$\text{Channel Utilization} = \frac{T-NP}{T} \tag{3.1}$$

where T is the frame period, N is the number of registered mobiles and P is the duration of single mobile polling signal. Channel utilization is illustrated in Figure 3.2. It is clear from Figure 3.2 that the same channel utilization efficiency for increasing number of users, requires more polling interval. This means that as the number of registered mobiles in a pico/micro cell increases, the channel utilization decreases since all the mobiles have to be polled to find out whether they have data to send or not.

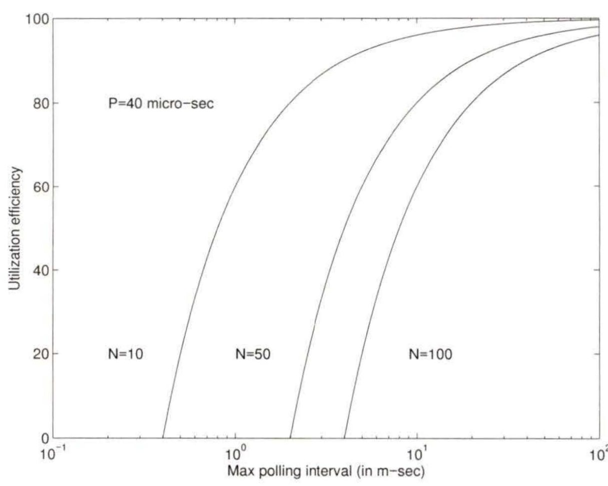


Figure 3.2 Channel utilization of the polling protocol.

Since the polling interval consumes more bandwidth as the number of registered mobiles in a cell increases, MAC protocol based on polling is not efficient for wireless ATM implementation. However, polling protocol can be efficient in wireless networks with fixed number of end users like wireless LANs.

3.2.2 MAC protocols based on Reservation

In order to support mixed traffic types, a MAC protocol must possess the following:

1. The ability to take advantage of the inactivity of the source, so as to more efficiently utilize the channel capacity by statistically multiplexing channels onto the time slots of the TDMA frame. Hence, the physical channel is only allocated to a logical channel when there is activity.
2. The ability to support the range of bit rates of the supported traffic types, by allowing multiple time slots per frame to be allocated to a logical channel on demand, and on a circuit switched as well as on a reservation access basis, depending on the service class requirements.

Packet reservation multiple access (PRMA) is a way of statistically multiplexing a number of voice terminals onto a TDMA carrier in short range communications [21]. In PRMA, as shown in Figure 3.3, the time slots on the up-link of the TDMA frame are considered to be identical such that mobiles can reserve any up-link time slot by transmitting information packets in these slots. However, with the original version of PRMA [21], the number of acknowledgment time slots on the Down-link is equal to the access attempt on the Up-link.

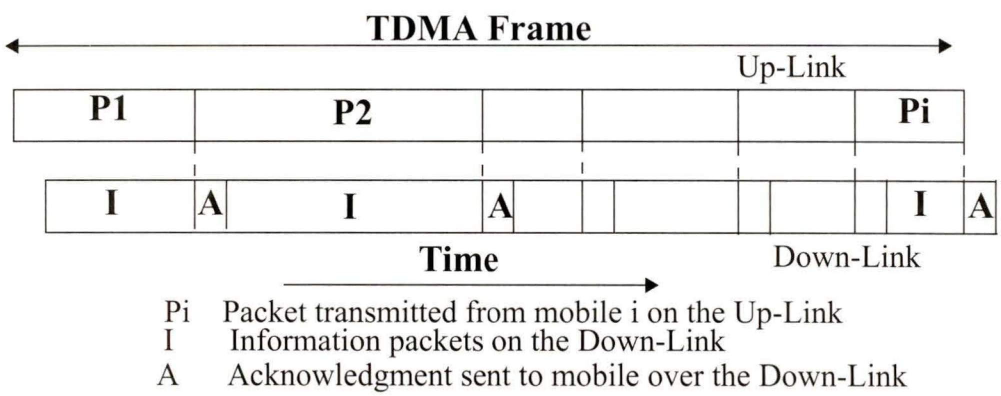
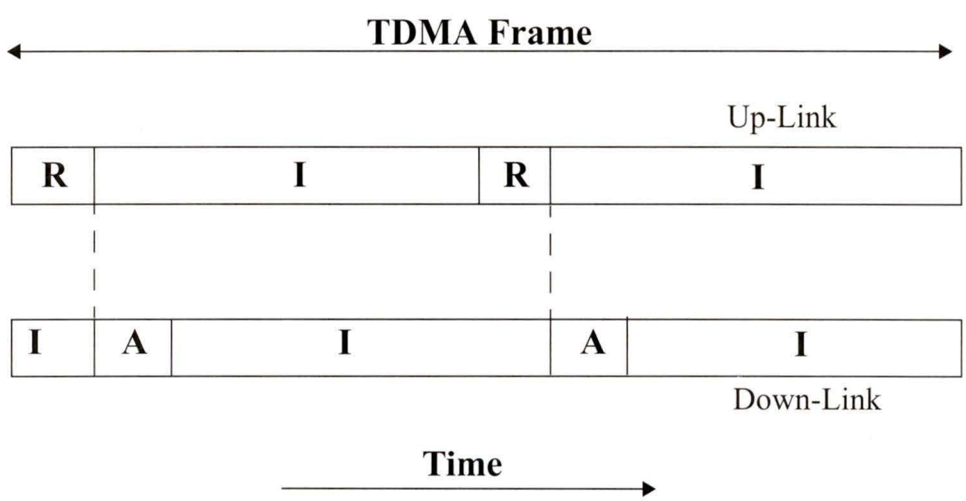


Figure 3.3 Frame structure in conventional PRMA.

The features of PRMA are presented as follows

1. Mobiles contend for slot positions in the frames by transmitting data packets.
2. If a successful contention of a slot occurs, the mobile will have access to these slots in the subsequent frames until the mobile completes transmission of the queued packets.
3. If more than one mobile contend for the same slots, a collision occurs and the mobiles will be asked to retransmit their packets according to a collision resolution protocol to avoid further collisions.
4. PRMA acts as slotted ALOHA during light traffic, and as fixed TDMA during heavy traffic.

In a variant of PRMA, PRMA++, the slots in the Up-link TDMA frames are not treated the same, but are split between slots where only contention-access takes place (R-slots), and slots where only the information packets are transmitted (I-slots) [9]. The PRMA ++ TDMA format is illustrated in Figure 3.4.



R Reservation slots available on the Up-Link
I Information slots available on the Up-Link or Down-Link
A Acknowledgments sent to mobiles over the Down-Link

Figure 3.4 Frame structure in PRMA++.

PRMA++ has two major advantages over conventional PRMA:

1. The limited number of slots in which contention-access takes place on the Up-link frame requires only this corresponding number of acknowledgment slots on the Down-link.
2. The separation of the physical channels in which access-contention and transmission take place, allows the transmission technique to be optimized for these two different traffic types.

The features of PRMA++ are identified as follows, where we consider that a mobile is associated with a single logical channel

1. The mobile gains access to the channel, by transmitting an Air-interface Channel Identifier (ACI) burst in an R-Slot on the Up-link frame. This contention access attempt is resolved through acknowledgment (A-slots) on the Down-link.
2. It is through the ACI that the mobile is able to bid for a slot or a number of slots per TDMA frame, in order to satisfy the instantaneous capacity requirements. The ACI transmission burst informs the base-station about mobile ID, Information size and type of reservation.
3. At the base-station, the kernel of the protocol contains a map of the available I-slots across all carrier frequencies. The base-station may thus allocate capacity across the two dimensions of time slot and carrier frequency, obtaining a corresponding increase in statistical multiplexing.
4. If the requested number of slots is not available, the mobile is told to wait until they become free (Signalled via the A-slots), or told to reduce the capacity allocation under control of some policing algorithm, depending on the service supported by the logical connection in question.
5. The mobile is kept in constant contact with the base-station, using a low rate signalling channel. This allows parameters associated with mobility such as timing advance and power control to be monitored and maintained.

The capacity required for the Up and Down-links is symmetrical.

3.3 Proposed Adaptive MAC Protocol

The proposed adaptive MAC protocol, shown in Figure 3.5, is of class 4 protocol (distributed control). It is based on TDMA and PRMA++, with adaptive number of reservation slots in the contention period. The voice (CBR) users and data (ABR/VBR) users have to contend for bandwidth on the reservation slots.

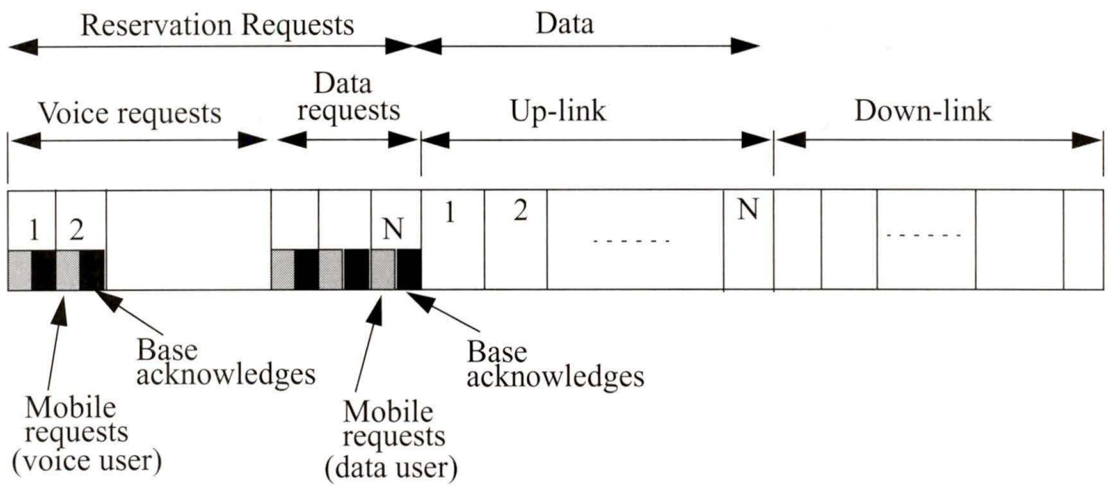


Figure 3.5 Proposed MAC protocol based on reservation.

The number of reservation slots depends on the traffic volume, taking into consideration that the voice users enjoy circuit switched-based service while bursty data users compete for the remaining available bandwidth of the contention period.

The proposed MAC features are presented as follows:

1. The voice user sends a request to reserve a slot in the contention period. Once the request is accepted, the base-station acknowledges the user, and a "circuit mode reservation" mode is used to reserve an access slot for the user to use until the end of the call. This slot is fixed for the user. The VBR / ABR user sends a request to reserve a slot in the contention period. Once the request is accepted, the base-station acknowledges the user and a "packet mode reservation" mode is used to reserve a slot for the user in the dynamic allocation part of the frame.
2. In case of collision, or if there are no available reservation slots, the voice user will send a request in the next frame, till the maximum voice access time is over. If the voice user

cannot reserve a slot, a voice call is blocked.

3. In case of collision, or no available reservation slots, the data users will send another request after a random time out period, in order to reduce the probability of another collision. If the message expiration time is exceeded, the message is lost.
4. The number of reservation slots will be variable depending on the traffic volume. A comparison between fixed access slot allocation and dynamic access slot assignments is shown in the simulation results.
5. The voice users are assigned fixed time slots on the information channel and the request slots are reserved for them from the call set-up until the end of the call on circuit switching basis. The advantage of this MAC is that it limits the contention period to only a portion of the channel bandwidth, achieving better utilization. On the other hand, the base-station design will be more complicated since it requires powerful processors to handle the problem of adapting the access bandwidth to the traffic volume and in the same time, handle the information slot assignments within this flexible environment.

3.4 Numerical Simulations

In this section, numerical simulations are performed using (C++) to compare the performance of the proposed adaptive MAC protocol to the performance of PRMA++.

The numerical simulation model is assumed to have the following considerations:

1. Handover: Handover is supported by the data link layer above the MAC sublayer. The handover is managed by the channel management system (CMS) which is responsible for channel resources allocation. Although the handover is not a parameter of the MAC protocol, but the more efficient the handover algorithm is, the less time the handover lasts. Less handover time results in less buffer size at the end users for ATM cell buffering and less dropped ATM cells during the handover. Handover algorithm is studied as system design enhancements in section 2.4.
2. Acknowledgments: The acknowledgments are sent to the users on frame by frame basis. Once the request is received, the base-station will acknowledge the mobile. The resourc-

es allocations management will try to allocate time slots on the upstream for the mobile.

3. Voice call setup: Call setup and establishment is performed once the voice users is granted one ATM slot on the upstream. Voice calls are circuit switched based, meaning that an ATM slot on the upstream will be assigned for the voice user until the end of the call.

The simulation is based on the following assumptions:

1. Traffic types: The channel is shared by CBR voice users operating at 32 Kbps, and packet users transmitting in available bit rate (ABR) mode with long file transfer messages averaging 5.12 Kb. The voice users will try to access the channel with a maximum call setup time of 5 seconds. If the voice user can not get access channel, the voice call is said to be blocked.
2. Channel speed: The channel speed is assumed to be 1,920 Kbps corresponding to the channel speed of micro cellular system in North America.
3. Data users will try to access the channel each time they have information packets to send. If the data users can not access the channel within a predefined expiration time, the packets will be dropped and the data packet are said to be blocked. The simulation results are shown for systems where the voice traffic share is 50% and 75%.
4. Request slots: Request slots were considered to occupy 7% of the channel bandwidth for the conventional PRMA++ and up to 8% in the proposed adaptive MAC.
5. Traffic model for voice users: The traffic source is a voice call random generator with average voice call generation of 5 calls for each user each 10,000 seconds. This means that for a system of 1000 voice users, the average voice call generation will be 5 calls each 10 seconds. The average duration of a voice call is 3 minutes for voice telephony.
6. Traffic model for data users: The traffic source is a random generator with Poisson distribution function. The data users are classified as critical users and normal users with ratio of 1 to 4. The critical users are those whose message expiration time less than of other data users. The message length is variable with an average length of 5.12 Kb. The average messages requests is one message in 10 seconds per user. This means that for 1000 data users, the average message generation is 100 data messages with normal distribution function.

7. If the expiration time of the message was longer than the access time and the allocation time, the message will not be delivered since it does not comply with the guaranteed QoS. The expiration time must be less than the access time and the allocation time to meet the QoS requirements of ATM.
8. Traffic volume: The simulation results were obtained for traffic volumes of 30, 40, 50, 60, 70 and 80 percent of the channel capacity to simulate the system performance for different traffic loads. This was done by estimating the number of voice calls and data messages within 45 minutes. The program was run for several hours (CPU time) for each traffic load. The program was run several times for each traffic load and the results were very much the same.

The simulation results compare the channel utilization and voice/data blocking probability for a fixed number of access slots system and adaptive number of access slots system. The numerical performance is considered for a steady number of voice and data users to simulate the MAC channel access efficiency where the handover is not an issue. Different traffic loads are assumed for obtaining the curves of channel utilization, voice and data blocking probability vs. channel load.

The TDMA frame duration length is 12 msec, based originally on 8 KHz sampling frequency of the voice. The reservation slot is 40 micro sec, with minimum of 20 reservation slots for fixed number-based system, and up to 25 reservation slots in the adaptive MAC protocol. The system is a symmetric Time Division Duplex (TDD). Voice users have channel speed of 32 Kbps, consuming 1 ATM cell per voice user in each TDMA frame. The average packet size of the data messages is 5.12 Kb, and average retransmission time is considered to be 8 msec. The flowchart of the MAC algorithm is illustrated in Figure 3.6. It is shown in the figure that the channel management system at the base-station is responsible for assigning time slots for voice and data users once the reservation request is successfully received at the base-station. This means that if the available time slots on the uplink are already assigned, the reservation request will be denied even if the reservation request did not suffer any collision with another request.

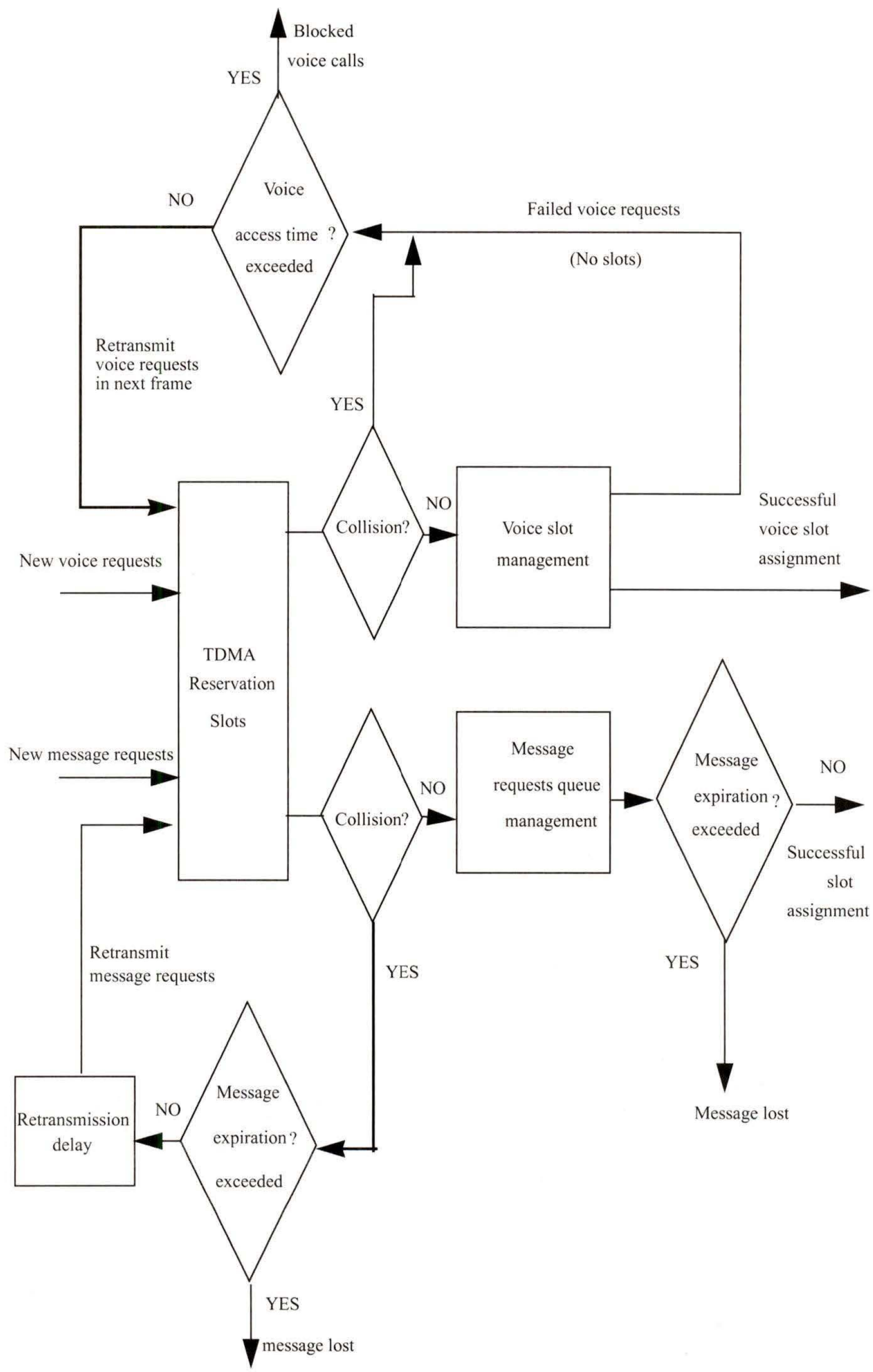


Figure 3.6 Flowchart of the simulated MAC protocol.

The simulation parameters of the PCN and their typical values are listed in Table 3.1.

Table 3.1 Simulation parameters

Parameter	Units	Value
TDMA Channel speed	Kbps	1,920
Frame length	msec	12
Request	slots / frame	20 -25
Wireless ATM payload size	Bytes	48
Average voice calls generated/sec/user	call/sec/user	$5*10^{-4}$
Average data message generated/sec/user	mes/sec/user	0.1
Average length of voice call	minute	3
Voice data rate	Kbps	32
Max. voice call set-up time	sec	5
Average request retransmission delay	msec	8
Average length of data message	Kb	5.12
Expiration time (critical, non critical data)	sec	0.05, 1
Ratio of critical / non critical data		1/4

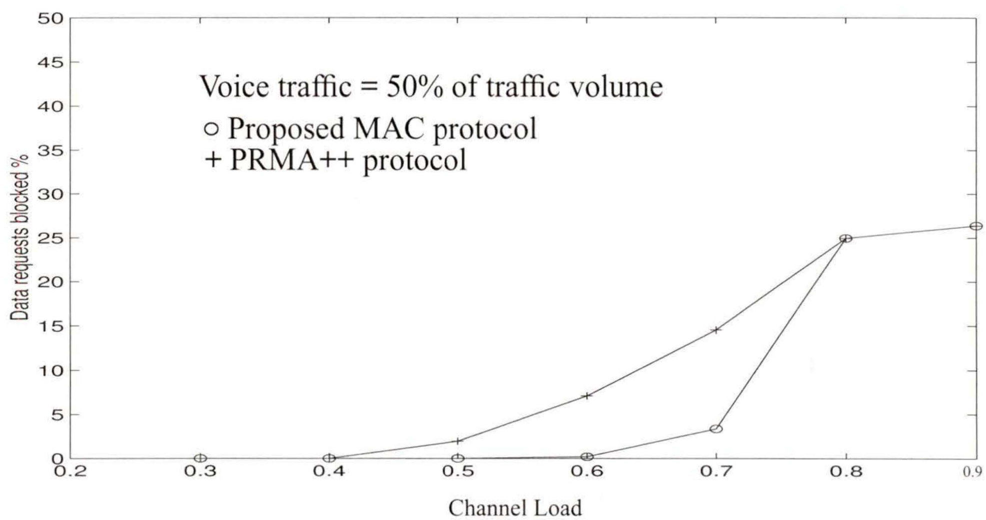
3.5 Simulation Results

The system described in the last section was simulated, once with a 50% voice traffic, and another with 75% voice traffic, since the voice traffic is still dominating the bandwidth in the second generation mobile networks. The simulation was carried using the parameters in Table 3.1 for a fixed number of reservation slots and variable reservation slots duration. Channel load is defined as the ratio between traffic volume to channel capacity.

The most important performance parameter is the voice calls blocking probability and data messages blocking probability. Blocking probability is the main factor that tends to favor one MAC protocol over another for the same traffic parameters. Blocking probability must be kept as minimum as possible for all traffic types to meet the QoS requirements of these traffic types. The voice traffic is still the dominant traffic type in the cellular systems. The proposed MAC protocol was proved to outperform the conventional PRMA++ protocol for both 50% and 75% voice traffic volume share, given that the base-station has the ability to monitor the traffic volume to change the number access time slots and informs the mobile users.

3.5.1 Data requests blocking probability

It can be inferred from Figure 3.7 that for a mild traffic volume (50-70% channel load), the proposed MAC reduces the possibility of data message blocking. The bandwidth of the channel can be better utilized by enabling more bandwidth to the reservation period. In the 75% voice traffic system, the proposed MAC is superior to the fixed reservation slot allocation, since the voice user always uses one ATM cell size slot on the information channel, reserving one more slot in the reservation period for an additional ATM cell over the information channel.



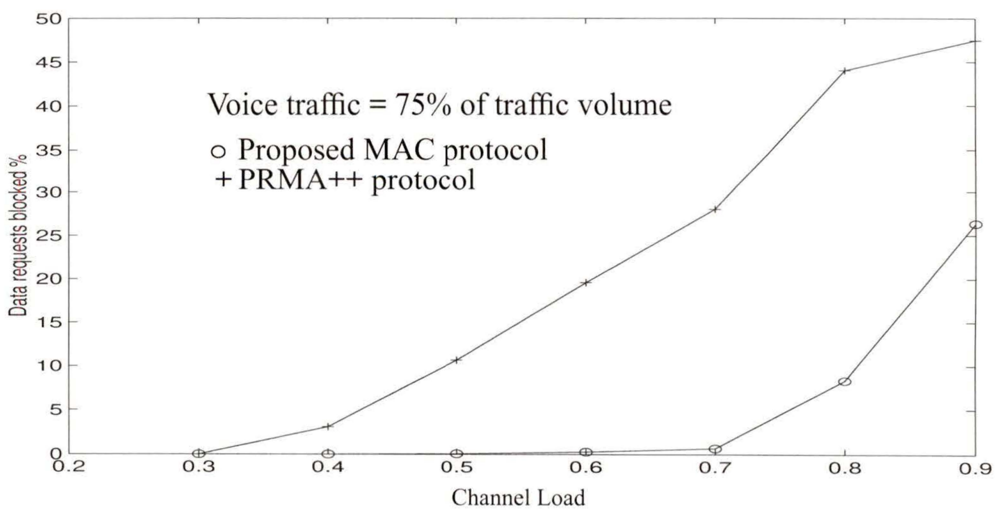


Figure 3.7 Data requests blocking probability for 50% and 75% voice traffic volumes.

For high traffic volume (80-90% channel load) with 50% voice traffic, the proposed MAC protocol acts like fixed conventional PRMA++ protocol, since any more reservation slots will reduce the available bandwidth for the information channel, resulting in even more blocking. Most of the mobile systems are designed for mild traffic, otherwise, the MSP will be overloaded.

3.5.2 Voice calls blocking probability

The voice call is said to be blocked if the user was not granted a reservation slot on the uplink within the call set-up max time. As shown in Figure 3.8, the proposed MAC maintains lower voice calls blocking over the entire range for voice dominated traffic (75% and more). In case of 50% voice load, the proposed MAC acts like conventional fixed PRMA++ protocol in high traffic volume.

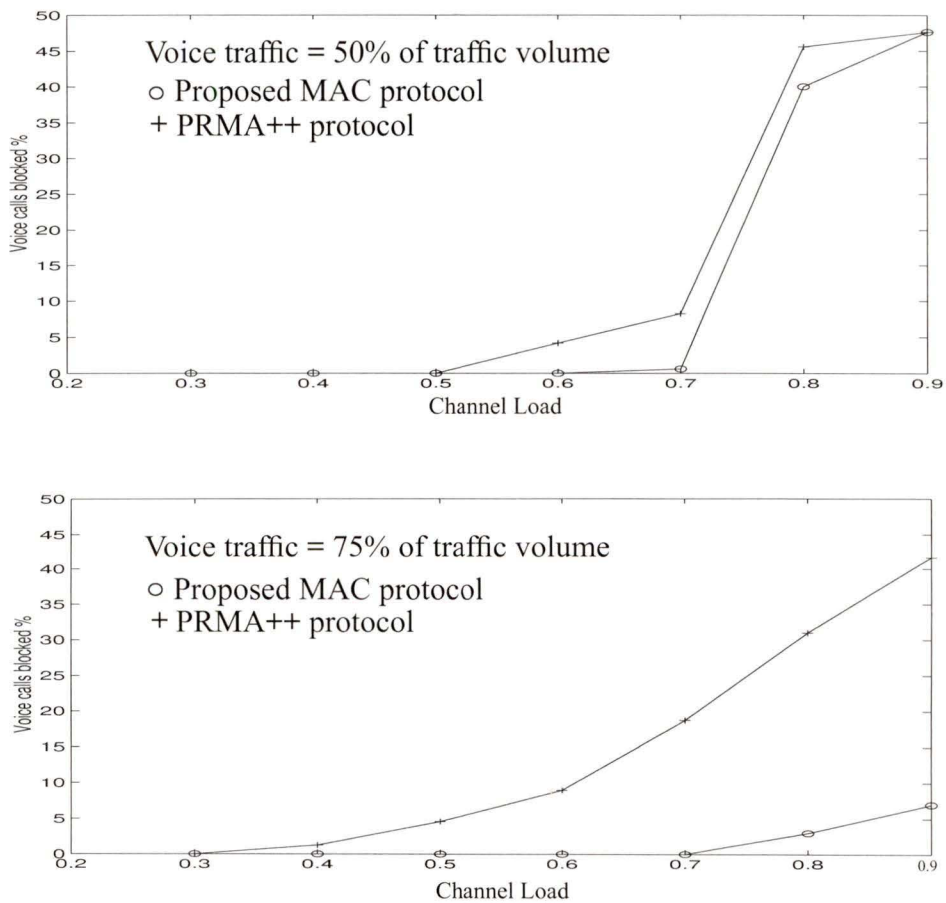
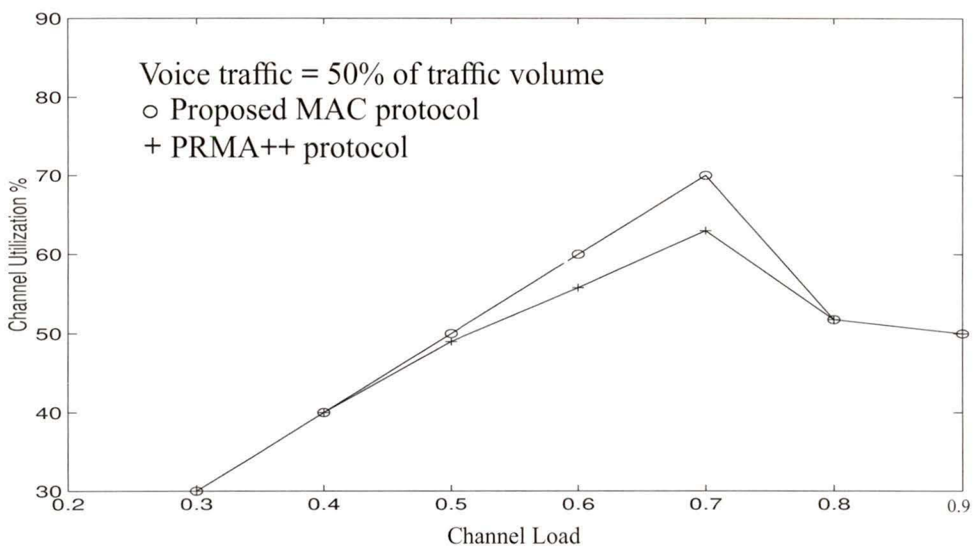


Figure 3.8 Voice calls blocking probability for 50% and 75% voice traffic volumes.

3.5.3 Channel utilization

Channel utilization is defined as the ratio between the bandwidth consumed by data and voice users and the available bandwidth. The channel utilization is calculated as the average number of times slots assigned to data and voice users to the available time slots on the uplink frame.



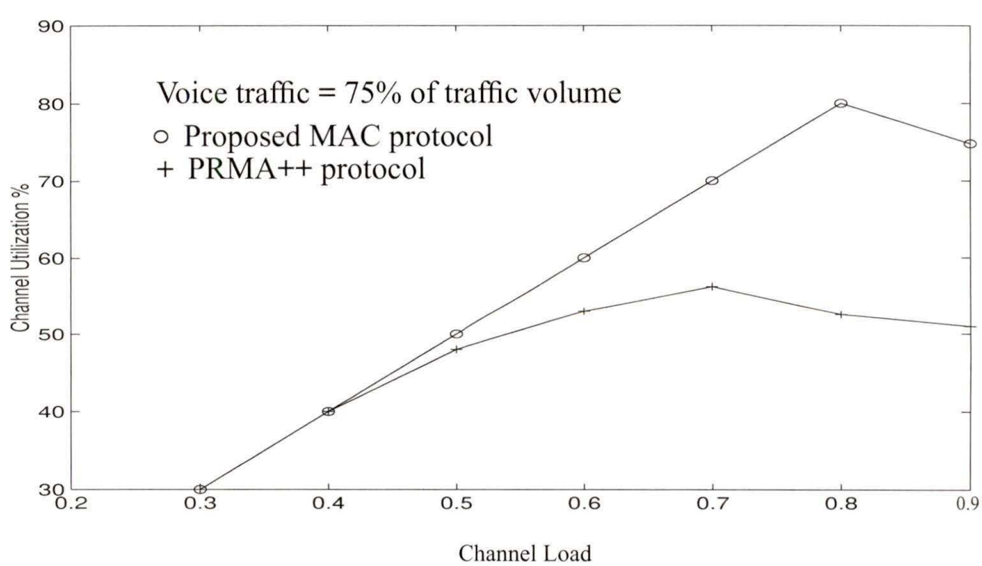


Figure 3.9 Channel utilization for 50% and 75% voice traffic volumes.

Based on the blocking probability for data requests and voice calls discussed in the previous two sections, and as illustrated in Figure 3.9, the proposed MAC protocol maintains higher channel utilization than the conventional PRMA++ protocol for the entire range of voice dominated traffic, and for mild traffic volume of 50% voice traffic volume.

3.6 Conclusion

An adaptive MAC protocol based on dynamic reservation access for the wireless channel in the contention period was introduced. The system is based on PRMA++ for PCN systems and designed to support wireless ATM traffic. the proposed system proved to be more efficient than the PRMA++ protocol for the mild traffic volume over the wireless channel, in case of 50% voice traffic. The system proved to be far better than the previous one for the whole range from light to high traffic volume in 75% voice traffic.

Data and voice blocking probability in the proposed MAC were found to be better than that of conventional PRMA++, for the mild traffic range (50-65%) since better channel utilization can be obtained by assigning more bandwidth to the reservation period given that the reserved information bandwidth is still large enough to accommodate more traffic. The decision of how many reservation slots will be added depends on the traffic statistics collected by the base-station over a pre-defined period of time. Reducing the blocking probability implies less retransmissions of voice and data requests, resulting in an overall less average access delay.

Since the voice traffic is still and for the next few years will dominate the traffic in the cellular networks, the proposed system is more suitable than traditional TDMA to support ATM services for the mobile end users.

Chapter 4

Error Control for Wireless ATM

Since transmission over almost noiseless optical fiber is assumed, no link-to-link error control of the information is performed. End-to-end error control can be carried out above the AAL layer at the user site [32]. Meanwhile, the requirements of integrated services introduced to mobile end users led to the development of a number of wireless networks [30]. The channel conditions in wireless personal communications are more severe than in twisted-pair copper wires or fiber-optic cables, with a mix of interference, multipath fading, Doppler shift and other transmission problems. The error control on the wireless link will be done at the physical layer, since the ATM cells have to be delivered free of errors to the broadbnad network. To cope with these severe conditions, two basic techniques are commonly used, automatic repeat request (ARQ) and forward error correction (FEC).

In ARQ the information is transmitted along with minimum overhead in a PDU. If the received PDU contains more errors than a certain maximum, the receiver will send a request for retransmission of the PDU to the sender. There are three main ARQ techniques: stop-and-wait (SAW), go-back-N (GBN), and selective repeat protocol (SRP). In a multi-access system these requirements can not always be accommodated. ARQ is mainly used in low-BER channels requiring a small number of retransmissions.

ARQ maintains high channel throughput since the packets are retransmitted only when they are received in error at the receiver. In the meantime, ARQ presents a delay since the packets will be retransmitted until they are correctly decoded. ARQ is suitable for ABR and UBR services where the data is sensitive to cell loss and insensitive to delay. An example of ABR services is data file transfer and e-mail messaging.

In FEC, the information is transmitted along with enough parity symbols that can detect and correct a pre-defined maximum number of erroneous symbols. In case of adaptive FEC, the number of overhead symbols used depends on the channel conditions. FEC is used when packet transmission time is small compared to the transmission delay. FEC is simple in terms of time management and does not require keeping copies of transmitted packets. It also does not require an immediate access to the reverse channel. It has operational complexity, but that can be handled in hardware. The main disadvantage of FEC is the bandwidth consumed by the parity symbols even in error-free channel. FEC is suitable for CBR and VBR real-time ATM services where the data is delay sensitive. An example of delay sensitive services is voice telephony and video conference services. However, since the packets can not be retransmitted if they can not be corrected, FEC is not efficient for loss sensitive services. FEC CODECs are based on Reed Solomon codes since the hardware implementation of the Reed Solomon codes is much easier than BCH codes.

Our objective is to find a means to counter the error expected in the wireless channel while being able to service both delay-sensitive and loss-sensitive services. Our technique is based on a mix of adaptive FEC and ARQ that will reduce the protocol complexity and minimize the bandwidth and delay, while satisfying the QoS requirements. From the above discussion, it appears that a hybrid protocol for error control on the wireless channel is the logical choice to minimize the delay bandwidth consumed by parity symbols for error correction.

As stated in the ARQ discussion, it is difficult to satisfy QoS requirements for delay sensitive applications by using ARQ. In the absence of channel errors, QoS can be satisfied by the MAC layer. The MAC layer is defined in wireless ATM to control the number of allocated ATM cells for mobiles depending on the channel conditions to guarantee the QoS. When channel errors are present, however, the MAC layer has to work in conjunction with the data link layer to satisfy the guaranteed QoS.

A realistic proposal is to implement ARQ on the fiber link where the BER is very low, and a hybrid AFEC on the wireless link to accommodate the guaranteed QoS of provided services on both the fiber and wireless channels.

Section 1 reviews the performance of ARQ in white Gaussian noise (WGN) channel and the disadvantages of ARQ protocol in wireless network. FEC presents a constant throughput regardless the channel conditions since the number of parity symbols are fixed and there are no retransmissions in the system.

Section 2 introduces an ARQ/Adaptive FEC (AFEC) protocol. The system methodology is introduced, and the combination between ARQ and AFEC is explained in details. The issues involved in the ARQ/AFEC protocol include BER estimation, adaptive correction capability, and the protocol performance.

Section 3 presents the cell loss ratio for the proposed protocol.

Section 4 reports the simulation results of the proposed system. The system performance is evaluated and analyzed to introduce the advantages of the proposed protocol over ARQ and FEC protocols.

4.1 ARQ Performance

This section studies the ARQ SRP and GBN performance in wireless environment to justify the motivations for the proposed error control protocol. The wireless channel is considered to be Additive White Gaussian Noise (AWGN) channel, for simplicity. The noise is assumed to have a constant power spectral density over the channel bandwidth, and a Gaussian amplitude probability density function (pdf). Although this type of channel may seem to be overly simplified, in fact it is practical. In micro-cells, it is possible to have a Line Of Sight (LOS) with essentially no multipath fading, giving a Gaussian channel [23].

Due to multipath fading and inter symbol interference (ISI), the BER will increase at the receiver side due to the channel conditions for the same SNR at the transmitter side. By using techniques to combat the multipath fading and inter symbol interference (ISI), such as diversity, equalization, channel coding, data interleaving, etc., we can get a BER very close to that of an ideal AWGN [23]. For the narrow-band TDMA, the approach is to use conventional QPSK or QAM. While QPSK utilizes the phase information to define the modulation level [24,25,26], QAM utilizes amplitude and phase information [27]. Figure 4.1 illustrates the channel throughput of GBN and SRP vs. BER. The packet size is considered to be equal to an ATM cell (53-bytes). At 32 Kbps voice transmission rate, voice cell packetization delay is 12 msec. However for voice traffic, maximum allowable end-to-end delay is about 50 msec.

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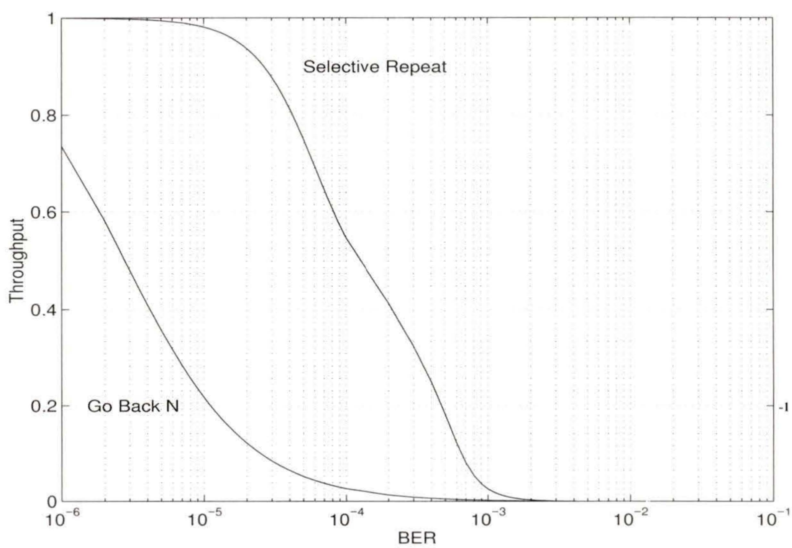

Figure 4.1 Throughput vs. BER for Go-Back-N and Selective Repeat ARQ.

Assuming worst case situation of mobile-to-mobile communication, and neglecting the round trip propagation delay and processing delay, we need an ARQ buffer size of only two ATM cells.

Channel throughput is defined as the ratio of information bit rate to channel bit rate. It is clear from Figure 4.1 that ARQ maintains high channel throughput for low BER channel, less than 10^{-6}, which is the case in fiber-optic cables and broadband networks. However, the channel throughput deteriorates as the BER increases which is the case in wireless channel conditions. The ARQ performance for the 32 Kbps voice user in wireless system where the normal range of BER is (10^{-2} to 10^{-4}) does not meet the QoS guaranteed for the voice user in wireless ATM implementation.

4.2 Proposed Error Control Protocol

The proposed error control protocol combines ARQ scheme with an adaptive FEC that uses Reed-Solomon code. Hybrid AFEC is based on the availability of a return channel on the down-link. When the return channel is available, an AFEC system which adaptively alters the error correcting capability according to channel state can be combined with an ARQ system [29]. The error control is performed at the ATM cell level. The ATM cell format is shown in Fig 4.2.

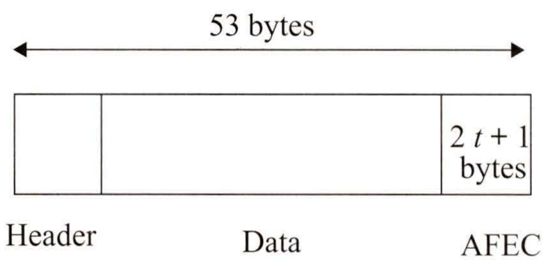

Fig 4.2 ATM cell format with AFEC.

The correction capability t is controlled by the base-station. The proposed error control protocol is based on the following assumptions:

1. The base-station receiver has two counters. Counter 1 is used for counting the incoming ATM cell from the mobile. This counter is set to zero after receiving M ATM cells. Counter 2 is used for counting the un-correctable ATM cells within M received cells. This counter is set to zero at the same time with the first counter.
2. The base-station sets $t = 0$ before it estimates the BER.
3. The error-correction capability (t) will be increased if the packet error rate is higher than a high threshold value, and will be decreased if the packet error rate is lower than a low threshold value.
4. The high/low threshold values are based on the BER associated with the error correction

capability.

5. The high and low threshold values associated with each t are stored in look-up table in the base-station and shall be used to alter the correction capability t.

The error control procedure will be as follows:

1. The base-station receives an ATM cell from the mobile.
2. The base-station increments the value in counter 1.
3. The ATM cell is decoded using AFEC decoder.
4. If the ATM cell can not be decoded (the number of errors are more than t), the receiver will send NAK to the mobile using ARQ protocol and increments the value in counter 2.
5. The base-station will keep decoding the retransmitted ATM cell using FEC and use the ARQ protocol to send NAK if the ATM cell is still un-correctable until the time-out expires.
6. After receiving M ATM cells, the number of un-correctable cells stored in counter 2 is compared with the high and low threshold values.
7. If the number of un-correctable ATM cells is between the high and the low threshold values, the correction capability would not be changed.
8. If the number of un-correctable ATM cells is higher than the high threshold value, the base-station will increment the correction capability t.
9. If the number of un-correctable ATM cells is lower than the low threshold value, the base-station will decrement the correction capability t.
10. The base station will send a control message to the mobile to change the correction capability to the new t.
11. The segmentation and re-assembly (SAR) of the mobile will adjust the data payload of the ATM cell by adding $(2t+1)$ bytes at the end of the ATM cell.

In order to implement one encoder and one decoder for different correction capabilities, Reed-Solomon code is used for hardware simplicity, where we can change the number of information symbols fed to the encoder and we need not change the hardware of the encoder when we want to change the error-correcting capability [29]. On decoding, we need to change the input parameters of the decoder with every parameter update, enabling adaptive error correction capability.

4.3 ATM Cell Loss Control

The channel BER is estimated based on the number of negative acknowledgments (NAK's) for M successively transmitted ATM cell. The rate of change of BER is assumed to be very small compared to the speed of the mobile in cellular systems. For a given ATM cell of length n bytes and error-correction capability t, the probability that a NAK is sent from the receiver to the transmitter depends on the channel BER. The symbol length is one byte long in case of $GF(2^8)$. The probability S of correctly decoding an ATM cell size n (53 bytes) is given by

$$S = \sum_{i=0}^{t} \binom{n}{i} P_b^i (1-P_b)^{n-i} \tag{4.1}$$

where P_b is the error probability in a symbol which can be expressed as

$$P_b = 1-(1-p)^m \tag{4.2}$$

where p is the BER and m is the number of bits per symbol.

From (4.1) and (4.2), the probability that an ATM cell contains an error that can not be corrected, Cell loss ratio(CLR), is given by

$$P_n = 1-S \tag{4.3}$$

The average number r_t of NAK's in M successively transmitted ATM cells can be expressed as

$$r_t = MP_n(t) \tag{4.4}$$

The choice of M depends on the service type since small M can lead to wrong BER estimation and large M can result in losing too many ATM cells before adjusting the error correction capability.

Cell loss ratio is a function of p, t and m, thus for a given p, we can vary t to get a corresponding CLR (P_n) stored in look-up tables at the base-station. If the number r_t exceeds a predefined threshold for certain M, the base-station will increase the error correction capability. The base-station is responsible for monitoring the channel conditions and informing the mobiles of the current channel BER.

The transmitter (mobile or base-station), changes the error-correction capability according to the channel conditions monitored by the base-station [28]. Selecting the appropriate t depends on desired cell loss ratio and the channel BER. Since the base-station is able to determine r_t, the BER can be obtained from look-up tables at the base-station based on equation (4.4). Once r_t exceeds the threshold values for the associated correction capability t, the base-station will increase t and informs the mobiles. Based on the control messages received from the base-station, the mobile alters the error-correcting capability of the Reed-Solomon (RS) code according to the estimated channel BER. The receiver decodes a received code-word, and then performs error control. If the decoded ATM cell is still erroneous, NAK is sent to the sender for retransmission.

4.4 Performance

This section studies the efficiency of the proposed hybrid ARQ/AFEC. The hybrid protocol is based on changing the number of redundancy symbols when the number on NAKs exceeds a pre-defined threshold value for certain number of packets.

We define the transmission efficiency η as the probability that the receiver can correctly decode a packet of size n sent from a transmitter with round-trip delay D, before a time-out T expires.

The base-station will decode the received ATM cells from the mobile using the AFEC decoder, and will use the ARQ protocol for retransmission if the ATM cell can not be decoded correctly using FEC. The ARQ will take place within the time-out duration until the mobile is ready to send another ATM cell. The parameters of the hybrid ARQ/AFEC are:

N is the normalized buffer size at the transmitter in ATM cells.

D is the normalized round-trip delay in ATM cell.

T is the normalized time-out before next ATM cell is ready for transmission.

S is the probability of successful packet transmission.

The efficiency η will be derived based on the following conditions:

1. Errors in the ATM cell are independent.

2.The reverse channel is available on both Up-link and Down-link.

The efficiency η is represented in [29] as

$$\eta = \frac{S}{(D+1)(1-S)S + 1 + (N+T)(1-S)^2 + TS(1-S)^N / \left[1-(1-S)^N\right]} \qquad (4.5)$$

when $N > D+1$, which applies to the wireless system we are studying.

Since the ATM cell contains $2t+1$ symbols for error detection and correction, therefore, the throughput η_t of the proposed hybrid ARQ/AFEC will be

$$\eta_t = \eta \frac{(n-2t-1)}{n} \qquad (4.6)$$

η_t is the channel throughput of the wireless channel to be evaluated in the following sub-sections.

4.4.1 System performance

Based on the model simulated in the previous chapter, the cellular system consists of several cells. The diameter of each cell is considered to be no more than a few hundred meters, so that the round-trip delay within the cell is of the order of a few micro seconds. Figure 4.3 illustrates the simulated micro-cellular model.

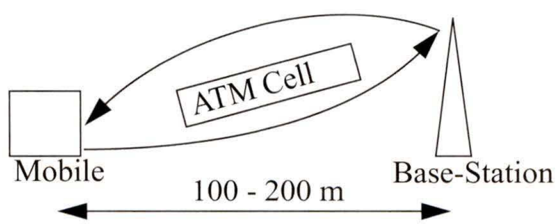

Figure 4.3 Round-trip delay in cellular networks.

The simulation is based on the following assumptions:

1. The mobile sends a packet with an ATM standard size (53 bytes) in every TDMA frame.

2. The data rate is 32 Kbps.

3. The error correction capability t is up to 8 bytes.
4. The threshold values are as listed in Table 4.1 and Table 4.2. In Table 4.1, error correction capability changes as the probability of an erroneous packet exceeds 10^{-2}, whereas, in Table 4.2, error correction capability changes as the probability of an erroneous packet exceeds 10^{-3}.
5. The time-out is considered to be 12 msec, the round trip-delay is 3 μsec, and the buffer size is at least 2 ATM cells.

Correction probability vs. BER and throughput vs. BER are studied. In some wireless ATM networks, the base-station has the capability of segmentation and reassembly. For these wireless networks, the ATM cells can be encapsulated in larger packets and the base-station has the capability of re-assembling the ATM cells from the received packets. For these wireless ATM implementations, average number of erroneous bytes per packet vs. packet size, and throughput vs. packet size are shown in the simulation results section based on Table 4.1 and Table 4.2. The values in these tables corresponds to the BER values for which the P_n exceeds the threshold value whether threshold value is 10^{-2} or 10^{-3}. The simulation of throughput and average number of erroneous bytes per packet is carried for BER values from 10^{-3} to 4×10^{-3} which are typical values for the BER of AWGN wireless channel.

Table 4.1 Threshold values when P_n exceeds 10^{-2}.

Correction Capability	BER
$t = 0$	$\sim 3\times10^{-6}$
$t = 1$	$3.1\times10^{-6} \sim 3\times10^{-4}$
$t = 2$	$3.1\times10^{-4} \sim 1\times10^{-3}$
$t = 3$	$1.1\times10^{-3} \sim 2\times10^{-3}$
$t = 4$	$2.1\times10^{-3} \sim 3\times10^{-3}$
$t = 5$	$3.1\times10^{-3} \sim 4\times10^{-3}$
$t = 6$	$4.1\times10^{-3} \sim 6\times10\text{-}^{3}$
$t = 7$	$6.1\times10^{-3} \sim 7\times10^{-3}$
$t = 8$	$7.1\times10^{-3} \sim 3\times10^{-2}$

Table 4.2 Threshold values when P_n exceeds 10^{-3}.

Correction Capability	BER
$t = 0$	~ 3×10^{-6}
$t = 1$	3.1×10^{-6} ~ 1×10^{-4}
$t = 2$	1.1×10^{-4} ~ 5×10^{-4}
$t = 3$	5.1×10^{-3} ~ 1×10^{-3}
$t = 4$	1.1×10^{-3} ~ 2×10^{-3}
$t = 5$	2.1×10^{-3} ~ 3×10^{-3}
$t = 6$	3.1×10^{-3} ~ $4\times10\text{-}^{3}$
$t = 7$	4.1×10^{-3} ~ 5×10^{-3}
$t = 8$	5.1×10^{-3} ~ 3×10^{-2}

4.4.2 Simulation results

Simulation results (using MATLAB) are illustrated for Tables 4.1 and 4.2. The results help us to control the packet size for minimum CLR in case of SAR at the base-station, and error correction capability within a micro-cell, in such a way to meet the QoS required by the mobile end user. The simulation is based on the fact the CODECs have the ability to adjust the number of parity symbols on receiving the control signal from the base-station to alter the error correction capability.

The error correction probability illustrated in Figure 4.4 for Table 4.1 is very close to unity for BER less than 10^{-2}.

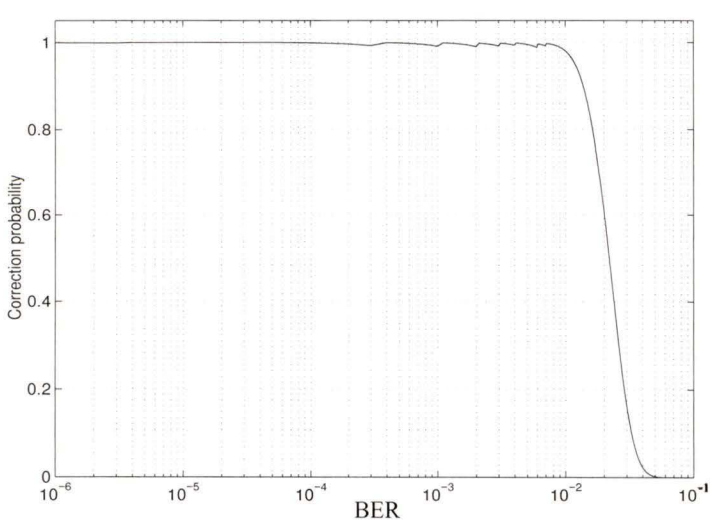

Figure 4.4 Correction probability based on Table 4.1.

Note that the error correction capability has the saw-tooth shape in the range between 10^{-3} and 10^{-2} due to altering error correction capability. The error correction probability is far higher than that of traditional GBN and SRP for the same BER range due to the increasing number of parity symbols. Correction capability is a function of the QoS guaranteed for the end user whether the application is loss sensitive or delay sensitive.

Figure 4.5 illustrates the channel throughput based on Table 4.1. The channel throughput is much higher than that of GBN and SRP for the same simulation parameters.

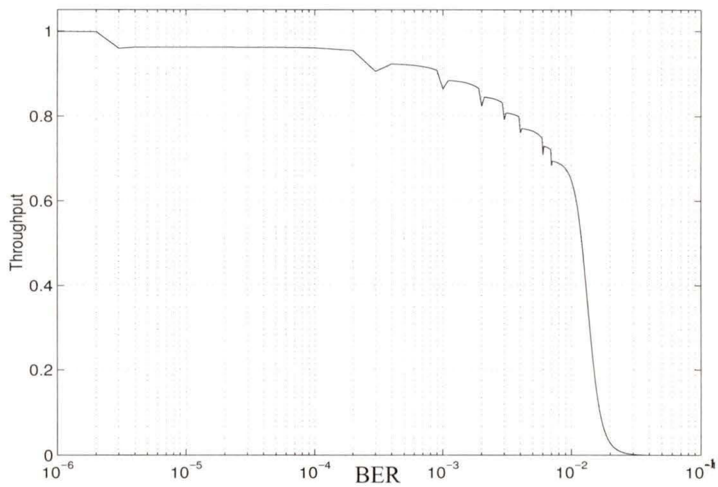

Figure 4.5 Channel Throughput based on Table 4.1.

Figure 4.6 illustrates the correction probability based on Table 4.2. The correction capability is very close to unity for the range of the range of BER less than 10^{-2}.

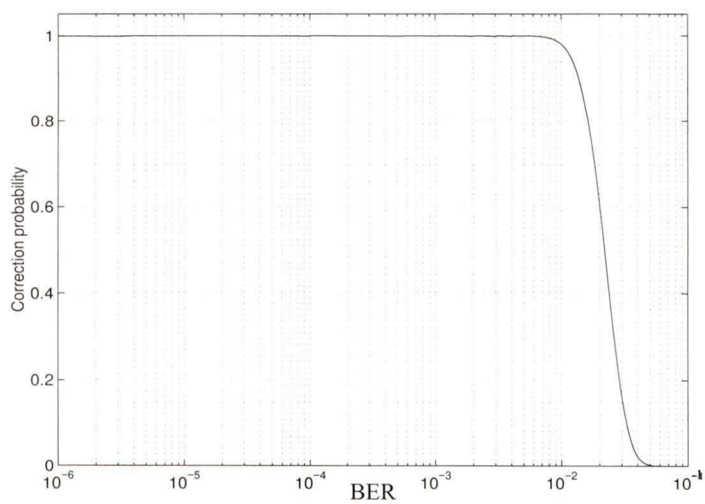

Figure 4.6 Correction probability based on Table 4.2.

The correction capability does not have the same saw-tooth shape as in Figure 4.4 since the thresholds value is much smaller than that of in Table 4.1.

Figure 4.7 illustrates the channel throughput for Table 4.2. It is very close to Figure 4.5 of Table 4.1. However, the throughput in Figure 4.5 is a bit better than of Figure 4.7 since the correction capability is altered faster in Table 4.2 than in Table 4.1. The choice of the threshold values depends on the guaranteed QoS.

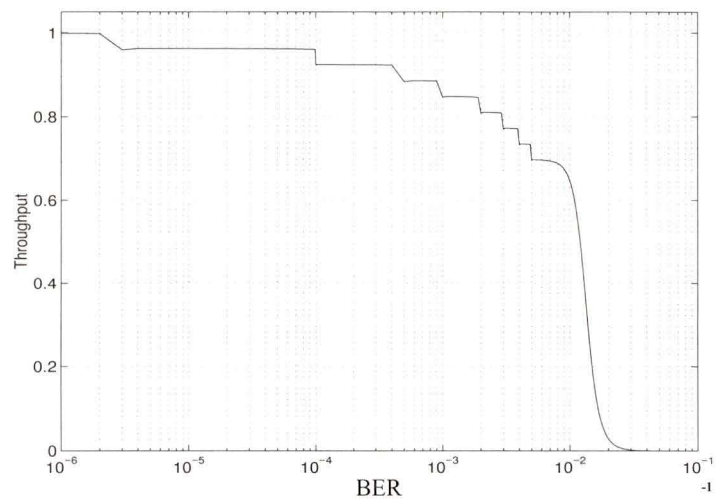

Figure 4.7 Channel Throughput based on Table 4.2.

Figure 4.8 illustrates the average number of erroneous bytes per packets vs. packet size for different BER values. This figure helps us to adjust the packet size as multiple of ATM cells depending on the required QoS for specific channel conditions.

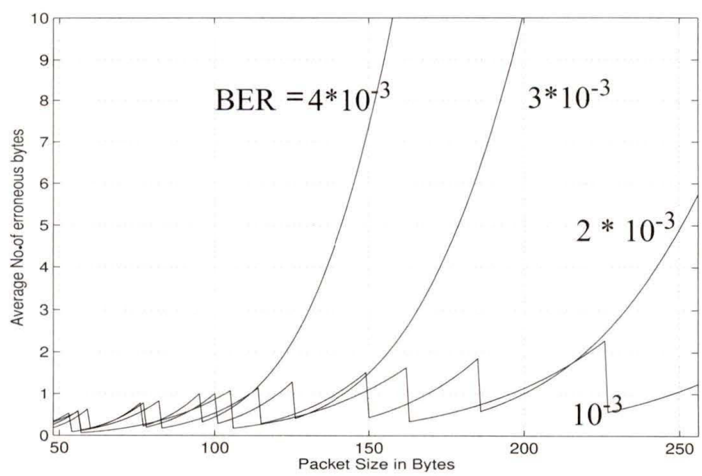

Figure 4.8 Average no. of erroneous bytes vs. Packet size as in Table 4.1.

Figure 4.9 illustrates the average number of erroneous bytes per packet vs. packet size for different BER values in case of SAR at the base-station. It is obvious that the average number of erroneous bytes is much less for Table 4.2 than for Table 4.1 due to the smaller threshold value for altering the correction capability in Table 4.2.

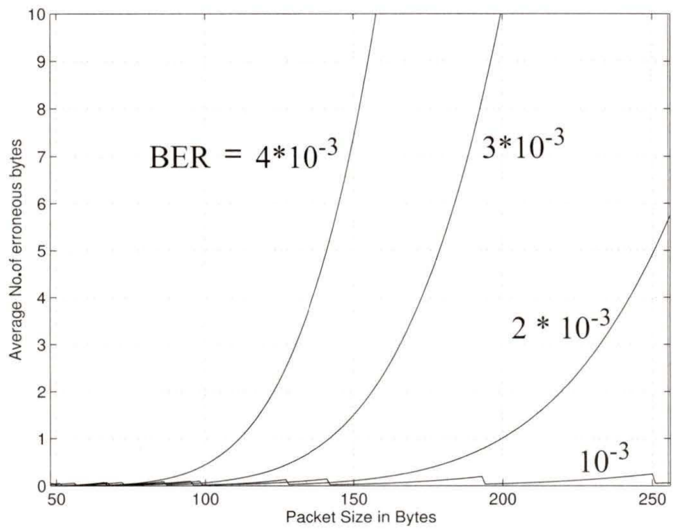

Figure 4.9 Average no. of erroneous bytes vs. Packet size as in Table 4.2.

Figure 4.10 illustrates the throughput vs. packet size. It is obvious from the figure that the bigger the packet size, the lower the throughput due to the increasing number of retransmission, since the probability of erroneous bytes per packet increases.

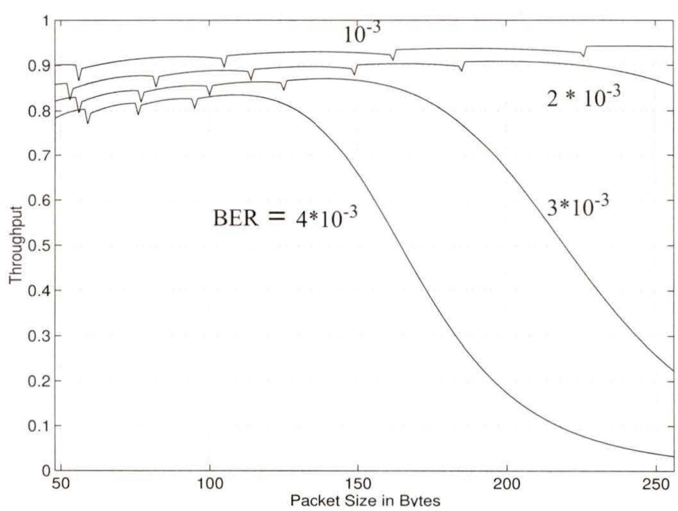

Figure 4.10 Throughput vs. packet size for different BER values as in Table 4.1.

Figure 4.11 illustrates the throughput vs. packet size for different BER values for Table 4.2. The figure is very close to Figure 4.10, however, the associated average number of erroneous bytes per packet is different for Table 4.2 compared to Table 4.1, which requires different number of parity symbols to achieve the guarantee the same QoS.

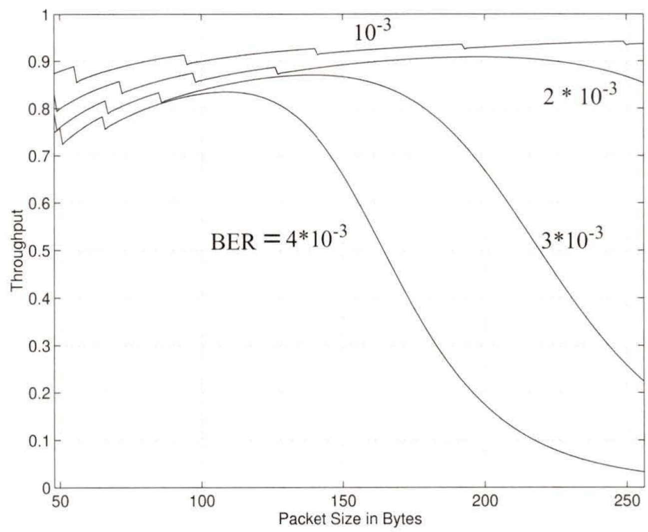

Figure 4.11 Throughput vs. packet size for different BER values as in Table 4.2.

It can be inferred from the simulation results of Figures 4.10 and 4.10, that the number of retransmissions and channel throughput depend on the threshold values for correction capability. The proposed system delivers good results for BER less than 2×10^{-3} in the whole range. The proposed system is designed for applications rates up to 64 kbps, including voice and data files, to be transmitted over the cellular networks.

4.5 Conclusion

A hybrid ARQ/AFEC was introduced for wireless ATM cellular network, supporting data rates up to 32 kbps which is the voice data rate in the cellular systems in PCN. The proposed scheme proved to be better than ARQ in high BER environment.

The system is based on Reed Solomon coding since the implementation of encoders and decoders are much easier than those of BCH codes. To support different services over the wireless channels, the error correction control will be performed on the ATM cell level, with larger symbol size to support correction capability in noisy environment. The proposed hybrid ARQ/FEC gives better results when the round trip delay is much less than the ATM packetization delay, which is the case in digital cellular systems.

Chapter 5

Conclusions and Suggestions for Future Work

The main objective of this work was to study the current wireless ATM technology for PCN, the problems associated with the implementation of such system, and to propose solutions for two main open issues in the WATM implementations. The first issue is the design of an appropriate MAC protocol to achieve high channel utilization in a mixed traffic environment. The second issue is the design of an error control protocol for the wireless channel between the mobiles and the base-stations to achieve the highest possible throughput and minimum delay.

It was shown from the results obtained that a better MAC protocol can be designed based on the current MAC protocols, with minor modifications to support various traffic types over the wireless channel. The proposed MAC protocol reduces blocking probability and increases channel utilization over a wide range of traffic volume. The proposed hybrid ARQ/AFEC, based on a combination between Go-Back-N and AFEC, proved to achieve higher throughout than the ARQ techniques in case of high BER, and in the same time, sustains the correction capability of FEC protocols depending on the channel conditions.

5.1 Main Contributions

This work provides an overview of wireless ATM network implementation and addresses the main issues involved in this implementation. The main contributions are in the areas of wireless channel access schemes and error control over the noisy channel in a wireless ATM network. The importance of this work is the fulfillment of the following two basic requirements in any foreseen wireless ATM implementation:

1. The MAC protocol used for wireless channel access satisfy the QoS of ATM traffic.
2. The error-control over wireless channel maintains high channel throughput in high BER conditions.

The design parameters of the proposed MAC and error-control protocols will depend on the wireless channel conditions, traffic volume and services provided over the wireless ATM network.

5.2 Suggestions for Future Work

In this work we presented a MAC protocol for mixed traffic type. The main idea is to control the number of available time slots to have better information channel utilization. The channel utilization can still be improved since the access time slots are assigned to voice users in circuit switched mode. This means that the access time slot is still reserved even if the user has nothing to send. Frequency and time slot hopping should be implemented to enable the data users to access the reserved time slots for voice users when they are idle.

Our proposed protocol uses slotted ALOHA as the access protocol in the contention period. A reservation protocol based on intelligent queuing algorithm in the base-station, though it is difficult to implement, can reduce the blocking probability and time delay.

The issues of handover, location management, registration, routing, etc. are still under investigation and the proposed solution will very much depend on the WATM environment and applications types to be supported.

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