A Slotted-CDMA based Wireless-ATM Link Layer

Guaranteeing QoS over a Wireless Link

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Preface

The author carried out the work described in this thesis during the period February 1999 to December 2000 under the supervision of Professor Peplow in the School of Electrical and Electronic Engineering, University of Natal, Durban.

The author certifies that this thesis represents his own original work, except where specifically indicated to the contrary in the text, and that it has not been submitted to any other university for degree purposes.

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Abstract

Future wireless networks will have to handle varying combinations of multimedia traffic that present the network with numerous quality of service (QoS) requirements. The continuously growing demand for mobile phones has resulted in radio spectrum becoming a precious resource that cannot be wasted. The current second-generation mobile networks are designed for voice communication and, even with the enhancements being implemented to accommodate data, they cannot efficiently handle the multimedia traffic demands that will be introduced in the near future. This thesis begins with a survey of existing wireless ATM (WATM) protocols, followed by an examination of some medium access control (MAC) protocols, supporting multimedia traffic, and based on code division multiple access (CDMA) physical layers. A WATM link layer protocol based on a CDMA physical layer, and incorporating techniques from some of the surveyed protocols, is then proposed. The MAC protocol supports a wide range of service requirements by utilising a flexible scheduling algorithm that takes advantage of the graceful degradation of CDMA with increasing user interference to schedule cells for transmission according to their maximum bit error rate (BER) requirements. The data link control (DLC) accommodates the various traffic types by allowing virtual channels (VCs) to make use of forward error correction (FEC) or retransmission techniques. The proposed link layer protocol has been implemented on a Blue Wave Systems DSP board that forms part of Alcatel Altech Telecoms' software radio platform. The details and practicality of the implementation are presented. A simulation model for the protocol has been developed using MIL3's Opnet Modeler. Hence, both simulated and measured performance results are presented before the thesis concludes with suggestions for improvements and future work.

Contents

List of Figures

List of Tables

List of Abbreviations

ACK	Acknowledgement
ADC	Analogue-to-Digital Conversion
AP	Access Point
ATM	Asynchronous Transfer Mode
ARIB	(Japanese) Association of Radio Industries and Businesses
ARQ	Automatic Repeat Request
ASIC	Application Specific Integrated Circuit
BER	Bit Error Rate
BS	Base Station
CDMA	Code Division Multiple Access
DLC	Data Link Control
DMA	Direct Memory Access
DSP	Digital Signal Processing
EMIF	External Memory Interface
ETSI	European Telecommunications Standards Institute
FEC	Forward Error Correction
FIFO	First In First Out
FPGA	Field Programmable Gate Array
GPS	Global Positioning System
GSM	Global Systems Mobile
HPI	Host Port Interface
IC	Integrated Circuit
ISM	Industry, Science and Medicine
ISR	Interrupt Service Routine
MAC	Medium Access Control
McBSP	Multi-channel Buffered Serial Port
MC-CDMA	Multi-Carrier CDMA
MIPS	Million Instructions Per Second
MT	Mobile Terminal
OFDM	Orthogonal Frequency Division Multiplexing
PCI	Peripheral Component Interconnect
PRMA	Packet Reservation Multiple Access
RACE	Research into Advanced Communications for Europe
QoS	Quality of Service
RAM	Random Access Memory
SBSRAM	Synchronous Burst Static RAM
SDRAM	Synchronous Dynamic RAM
UMTS	Universal Mobile Telecommunications System
WATM	Wireless-ATM

Chapter 1 Introduction

The once dreamt of mobile office is now, in many ways, a reality. Laptops, mobile phones, PDAs (personal digital assistants) and other portable devices have made this possible. However, wireless networks still don't provide more than voice communication and very limited data services. In order to access true multimedia services it is still necessary to use static, wired connections. Furthermore, it is currently impossible to use mobile devices globally. While it is commonly accepted that GSM (global systems mobile) has become the dominant protocol for mobile communications there are still many major regions of the world where GSM is not used and even greater regions where there is no wireless coverage at all. The rapid rollout of enhancements to GSM, such as WAP (wireless access protocol) and GPRS (general packet radio service), promise to bring Internet services to the mobile phone and improved data rates for wireless connectivity respectively. However, in reality, these improvements are limited by the fact that GSM was originally designed only for voice communication, and hence it cannot efficiently provide service quality to multimedia applications.

The vision of the ultimate mobile device will continue to fuel the development of new techniques and technologies that may some day make that device a reality. But what is the ultimate mobile device? It would have to provide far more than the voice communications of an ordinary GSM phone. Different people have different ideas on what such a device should be able to do. It may have to incorporate a videophone, FM radio, MP3 player, all the features of a modern PDA (address book, diary, email, memos), as well as location aware services providing local information and maps. All these features would have to be contained in a small, lightweight device with a long battery life. A high-resolution display, large enough to watch television on, is a necessity. Of utmost importance is how the user interfaces with the device – perhaps a combination of speech-to-text technologies and stylus input on a touch sensitive screen. The emerging concept of the software radio may add another dimension to the capabilities of mobile devices, providing the platform for automatic protocol fine-tuning and feature upgrades. An ideal software radio would extend the digital domain all the way to the antenna, so that all the front-end processing that has traditionally been performed in the analogue domain is now done digitally with the potential to be "softly" reconfigured. This means that the radio is capable of communicating with multiple protocols over the wireless interface. This leads to the idea of mobile devices that adapt themselves to their local environment, offering whatever services the local network is capable of providing, and automatically upgrading themselves, when necessary, to support these services. For example, imagine if the current second-generation GSM phones could automatically upgrade themselves, over the wireless interface, to support WAP and GPRS.

There is no doubt that mobile phones have had a huge impact on the everyday lives of millions of people throughout the world. Finland made headlines at the end of 1998 when it became the first country with more than 50% of its population as mobile cellular subscribers [1]. By the end of 1999, nine countries had more mobile subscribers than fixed-line subscribers, and it is not only first world countries that have reached this milestone. In July 1999, Uganda, one of the least developed countries, became the first African country with more mobile subscribers than fixed-line subscribers. There are two main reasons for this rapid growth. The first is that wireless networks can be installed more quickly than fixed-line networks. The second is the use of prepaid services, as the vast majority of Ugandans would not meet the financial criteria for a subscription-based service. In fact, these two reasons apply to all developing countries. Economies of scale and technological advancements have changed the perception that mobile communications is a service for the wealthy. Mobile tariffs are coming down and today's mobile networks are generally cheaper to install than fixed-line networks. The result is that mobile is now considered to be more attractive than fixed-line for extending telecommunications access in developing countries. Although some developing countries like Uganda have more mobile than fixed-line subscribers, the percentage of their populations that have any telecommunications access at all, fixed or mobile, is still very low. Besides telecommunications access, Internet access is becoming increasingly important, with the potential to level the playing field for developing nations. Third-generation (3G) mobile networks promise high-speed Internet access, offering developing countries the opportunity to take a technological and commercial jump to ensure widespread availability of telecommunications and Internet access [2].

With the ever-increasing demands for wireless devices, radio spectrum has become a very valuable resource with governments auctioning frequency bands to telecommunications companies at staggering prices. For example, bidding for 3G/UMTS licenses began in Germany on 31 July 2000. Twelve blocks of 2 x 5 MHz paired bandwidths were up for sale, and competitors needed to secure a minimum of two blocks for a license. The Germans raised a massive $37 billion – the largest in Europe thus far [3]. This is after the equivalent British auction raised a staggering $35 billion – more than seven times initial forecasts. Investments of this nature pay tribute to the significant progress and developments being made in mobile communications. These exceptionally high licence costs do, however, serve to keep mobile communication prices high for the unfortunate consumers.

1.1 Problem Overview

The 3G mobile networks promise to extend the current voice and limited data services to true multimedia services along with high speed Internet access. While the standardisation of 3G, physical layer, wireless interfaces are now well on their way, appropriate link layer protocols that function above the physical layer are still being investigated. The link layer will be required to support a range of quality of service (QoS) levels in order to cater for the differing requirements of the various multimedia traffic types. When one contemplates guaranteeing QoS over wireless links, difficulties arise due to the unpredictable nature of such links. This is because of the relatively high and time-varying bit error rates associated with wireless channels. Future cellular wireless networks will be expected to accommodate varying combinations of voice, video, audio and data traffic, each presenting the network with different traffic characteristics and QoS requirements. The network will be required to provide performance guarantees in terms of throughput, end-to-end delays and packet loss ratios. Wireless links between mobile terminals (MTs) and base stations (BSs) will, in all likelihood, be the most unpredictable links in the network. Hence, this is where the greatest difficulty lies in providing any QoS guarantees. What is required is a flexible medium access control (MAC) protocol that can adapt to the array of traffic characteristics presented to it, along with a data link control (DLC) protocol capable of maintaining the quality of the wireless link under varying channel conditions.

The 3G wireless networks are being designed around code division multiple access (CDMA) technology. CDMA has recently gained widespread international acceptance as an upgrade that will increase system capacity and service quality. Unfortunately the dominant regions of the world have still not managed to agree on a common standard with the CDMA2000 standard coming into effect in North America, while Europe and Japan opt for their own wideband CDMA standard commonly referred to as WCDMA. These 3G standards allow for significant bandwidth improvements but do not directly address the issue of QoS guarantees.

This thesis proposes a wireless link layer protocol that addresses this problem of supporting multimedia traffic with different QoS requirements. Different MAC and DLC techniques are investigated and combined to produce a flexible protocol capable of handling a wide range of traffic characteristics. The protocol is based around the asynchronous transfer mode (ATM) philosophy due to its inherent QoS based support for multimedia traffic. The aim is to combine the advantages of a CDMA wireless link with the flexible features of ATM. The MAC protocol uses a demand-based assignment technique to allocate the available bandwidth to users. The scheduling algorithm that performs this bandwidth allocation includes support for multiple QoS levels by scheduling according to each user's bit error rate requirement. The DLC incorporates a retransmission protocol as well as supporting the use of application specific forward error correction techniques. Also of importance is an example of an implementation of the link layer protocol. This has been achieved using Alcatel Altech Telecoms' software radio platform and is described in the thesis. The software radio concept and what it promises in the future, as well as the hardware infrastructure used in the implementation are also discussed.

1.2 Background and History

Obviously there are numerous bodies throughout the world looking to standardise and promote their networks and protocols as the worldwide standard. Without some co-operation between the three major regions (North America, Europe and Asia) and, indeed, within these regions, the already somewhat different technology routes taken shall branch even further apart. A number of research programs are taking steps towards standards that will enable global mobility.

RACE (Research into Advanced Communications for Europe, 1985 – 1995) is a European research programme that developed the concept of UMTS (Universal Mobile Telecommunications System). UMTS aims at the *"provision of mobile multimedia services, extension of broadband services becoming available on the fixed networks, internet access and access to services anywhere and at any time based on a personalised user profile"* [4]. ACTS (Advanced Communications Technology and Services) is another European Community research programme whose projects build on the work of the earlier RACE programmes, adding further initiatives to UMTS and, in particular, third-generation (3G) mobile systems development. At the international level, International Mobile Telecommunications 2000 (IMT-2000) is an International Telecommunications Union (ITU) initiative to develop recommendations for 3G mobile systems. The vision of IMT-2000 is similar and closely related to the UMTS vision. IMT-2000 and UMTS aim to provide direction to many related technological developments and to assist in bringing together what are essentially competing wireless access technologies.

As previously mentioned, one of the strengths of ATM has always been its ability to provide QoS guarantees. But, wireless-ATM (WATM) networks have tended to be aimed at the pico-cell, wireless-LAN environment rather than public mobile networks as such networks have not provided sufficient bandwidth. In recent years there have been a number of ACTS projects involved in WATM development. These include AWACS [5], MEDIAN [6], SAMBA [7] and WAND [8]. The details of some of these projects are of interest in this thesis and shall be discussed in the next chapter. Because of its inherent QoS based support for multimedia traffic, the link layer protocol proposed in this thesis is based around ATM and uses a CDMA physical layer as this is what the 3G networks are based around. The following section discusses the global boom in mobile telecommunications, and is followed by an introduction to the ATM philosophy and the concept of CDMA.

1.2.1 The Mobile Cellular Boom

The ITU publishes a number of reports on world telecommunication developments and indicators. The following information and statistics have been obtained from the ITU's Telecommunications Indicators Update for April-May-June 2000 [1] and an executive summary of the World Telecommunication Development Report for Mobile Cellular [2] released by the ITU in 1999.

Between 1990 and 1999 the world market for telecommunications (services and equipment) doubled. In 1999 the telecommunications market was worth over USD 1000 billion. Mobile communications is driving this growth and now accounts for 24 percent of the total service revenues, as opposed to just 3 percent in 1990. Although revenues from the fixed telephone network continue to dominate the total market, its share has fallen significantly, from 90 per cent in 1990 to 61 per cent in 1999. At current trends, the value of mobile revenue will overtake total fixed-line revenue in about the year 2004. In fact, revenues from the fixed-line service have been declining globally since around 1996. Without the revenues from mobile, the telecommunications sector would be shrinking rather than growing.

The global boom in mobile cellular communications has been remarkable. At the end of 1990, there were approximately 11 million mobile subscribers around the world. By the end of 1999, there were more than 450 million subscribers. That means that the number of mobile subscribers worldwide has been doubling every 20 months since the beginning of the decade. Figure 1.1 shows this astounding growth in mobile subscribers during the 1990's.

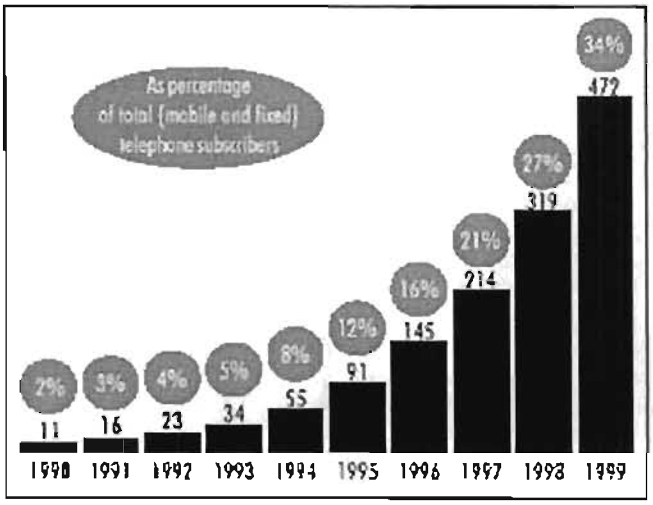


Figure 1.1 Worldwide mobile cellular subscribers (millions). [1]

A common way to gauge mobile growth is to compare the number of mobile subscribers to the number of fixed-line subscribers. Figure 1.1 provides such a comparison by showing the number of mobile subscribers as a percentage of the total (fixed and mobile) telephone subscribers. At the end of 1998, Finland and Cambodia were the only two countries in the world with more mobile subscribers than fixed-line subscribers. Seven more countries joined the club at the end of 1999, and membership continues to grow during 2000. It is no longer a question of *if* mobile cellular subscribers will overtake fixed-line, but *when*. The ITU forecasts that by the middle of this decade, there will be more cellular subscribers worldwide than fixed-line subscribers. Finland made headlines in 1998 when it became the first nation to top 50 per cent mobile penetration, and the mobile boom could be far from over. For example, in Nordic nations, the cellular penetration rate is forecast to reach 100 per cent. It is likely that other developed countries will follow this pattern. Meanwhile, in emerging economies, mobile is a perfect fit for much needed telecommunications infrastructure, since wireless networks can be quickly installed and pre-paid provides access for those who would not normally qualify for a subscription service.

Cellular networks have evolved considerably, from the so-called first-generation analogue networks to today's second-generation digital systems. But so far there has not been a unique global standard. The existing cellular landscape consists of a mix of analogue and digital systems with different networks often coexisting in the same country. One of the roles of the ITU is to define global telecommunications standards, but it did not issue technical recommendations for first or second-generation mobile systems. The ITU has, however, become actively involved in developing 3G standards. The ITU goal is to achieve a global 3G standard through the IMT-2000 initiative. IMT-2000 aims to provide seamless global roaming, higher transmission rates and standard service delivery.

The growth of the mobile sector is often overshadowed by the success of the Internet. While the growth prospects for the Internet should not be underestimated, the fact is that the mobile market is much bigger. The long-term future of mobile is likely to be closely linked to that of the Internet. With 3G enabling Internet access at high speeds and the potentially huge demand for wireless access to data services, 3G will create a virtually new industry.

1.2.2 Asynchronous Transfer Mode (ATM)

Modern networks have to provide performance guarantees for varying combinations of multimedia traffic. Such multi-service broadband networks are required to support a variety of service classes, each with its own traffic characteristics and quality-of-service (QoS) requirements. Two of the most prominent networking protocols are the Internet protocol (IP) and ATM (asynchronous transfer mode).

IP was designed to only provide a best-effort service, offering no performance guarantees and this is how today's Internet functions. However, the Internet Engineering Task Force (IETF) has recognised the new network requirements and the next generation Internet protocol, designated IPv6, incorporates many improvements to the protocol [9]. For example, IPv6 increases the 32-bit IP address size to 128-bits. The IPv6 base header has been simplified to contain only seven fields as opposed to the thirteen in IPv4. This reduces the processing cost of packet handling in routers. IPv4 provides no means for identifying individual data streams or flows. IPv6 introduces a flow label field in the base header, which allows packets belonging to the same flow to be identified. However, IPv6 alone cannot guarantee any real end-to-end QoS. Additional protocols are necessary to enable a specific QoS to be requested by applications.

The swift emergence of two IP standards in recent years verifies the strong interest in QoS with support for real-time as well as the current best-effort service. The resource reservation protocol

(RSVP) integrated services approach attempts to provide per-flow QoS assurances, whereas the differentiated services (DiffServ) approach aims at traffic aggregates that may not correspond to individual flows [10].

Many of the features being incorporated into IP are already supported in ATM. In fact, one of the strengths of ATM has always been its ability to provide a wide range of applications with different traffic characteristics and QoS requirements, while making efficient use of network resources. ATM switches also offer higher throughput performance than IP routers due to the small fixed size of ATM cells.

In some ways there is huge competition between the ATM enthusiasts, headed by the ATM Forum, and the Internet community. Both are pushing for their preferred architecture to dominate. However, both the IETF and the ATM Forum are doing a lot to combine the flexibility of IP routing with the speed of ATM switching. Multi-protocol over ATM (MPOA) was defined by the ATM Forum to support transparent transfer of network layer protocols such as IP and IPX over ATM networks. The IETF is defining Multi-protocol Label Switching (MPLS) to integrate IP routers and ATM switches.

While IP has dominated the LAN environment, ATM has traditionally been used for backbone links. This means that many of ATM's advantages have not been experienced by the network users because ATM has not been extended to their desktops. So, while the next generation networks may be viewed as a merger between IP and ATM, it is probably IP, being so entrenched, that will be tweaked to support the QoS features of ATM. However, there is always the worry that the backward compatibility requirements of the original best-effort IP protocol will lead to an inefficient and cumbersome next generation IP.

This thesis is concerned with extending QoS provisions to the wireless domain. The proposed protocol is based on ATM because of its firmly established QoS features. To this end the features of the ATM protocol are discussed further.

What is ATM?

ATM is a high-bandwidth, low-delay, packet like switching and multiplexing technique that supports both bursty traffic and continuous traffic through circuit-mode emulation. ATM's switching and multiplexing technology combines the benefits of circuit switching (constant transmission delay and guaranteed capacity) with those of packet switching (flexibility and efficiency for intermittent traffic). [11]

ATM is the network protocol chosen by the International Telecommunication Union Telecommunication Standardisation Sector (ITU-TSS) as the basis of the *broadband integrated services digital network* (B-ISDN). B-ISDN uses digital techniques capable of concurrently handling data, voice and video transmission. This makes good use of ATM's support, or any packet switching protocol's support, for multiple logical connections on a single physical circuit. B-ISDN's variety of traffic sources leads to a variety of traffic types i.e. from constant bit rate traffic to bursty traffic. ATM's ability to simultaneously support such varying traffic types is an important feature emphasising its suitability to a multimedia environment. Another very important feature of ATM is its ability to guarantee a quality-of-service (QoS) for each connection. A traffic contract is specified for each connection a user establishes with the ATM network. This contract specifies parameters such as peak cell rate (PCR), sustainable cell rate (SCR) and maximum burst size

(MBS) which the network must adhere to if it accepts the call. While this provides very flexible features, it does have the disadvantage that the required traffic profile needs to be known before the connection can be established,

In contrast to the asynchronous mode of ATM, synchronous transfer mode methods use time division multiplexing (TDM) to multiplex users onto a single channel with each user assigned a timeslot. Here, a user with a lot of information to send can only transmit when it's timeslot comes up even if none of the other timeslots are being used. Similarly, if a user doesn't have information to transmit when its timeslot comes up the slot is wasted. Because ATM is asynchronous, timeslots are assigned on demand with the information identifying the source and destination contained in the header of the ATM cell. Unlike most packet transmission protocols, ATM's packets are fixed at 53 bytes (5-byte header + 48-byte payload), and are referred to as cells. The small size of the cell is suitable for delay sensitive voice and video traffic that can't wait for large packets to fill up before being sent.

The 5-byte ATM cell header can be one of two formats: User-Network Interface (UNI) or Network-Network Interface (NNI). UNI cells are used between ATM switches and ATM end points, whereas NNI cells are used between ATM switches within the network. However, the structure of UNI cells and NNI cells is almost identical. Appendix A.1 provides details on the fields included in the ATM cell formats. Two of the most important fields are the VPI (virtual path identifier) and VCI (virtual channel identifier) fields. Together they are used to specify the next destination of a cell as it progresses, from switch to switch, through an ATM network. Unlike many other packet switching protocols, such as IP and IPX, ATM is connection oriented with the VPI/VCI fields in the cell header identifying a connection or virtual channel (VC). The VC must be set up across the ATM network before any data transfer takes place. ATM switches then route ATM cells through the network by mapping each received cell's VPI, VCI and input port combination to a new VPI, VCI and output port combination.

ATM was designed to take advantage of the high reliability of modem high-speed digital networks (especially fibre) and hence contains little overhead for error detection and correction. The 5-byte cell header includes a header error correction (HEC) field that provides an 8-bit cyclic redundancy check (CRC) on the cell header. No error checking is provided for the 48-byte cell payload. This is obviously a drawback when implementing ATM over unpredictable radio links that require sophisticated error detection and correction. Hence, this issue has to be carefully addressed in a WATM implementation that extends ATM into the wireless domain. As previously mentioned, ATM has become popular on site backbones, but has yet to gain popularity as a network standard for LANs, where the IP/Ethernet combination is most common.

Why Wireless-ATM?

The WATM standard is presently a hot topic amongst ATM enthusiasts. With numerous wireless access protocols in existence one may ask what need there is for yet another. In [12], Jouni Mikkonen outlined the main justifications for WATM development:

1. users will require wireless access to ATM/B-ISDN networks,
2. multimedia applications need a wireless platform with multimedia support, and
3. UMTS and wireless-LANs cannot meet all future data user needs.

Sunil Jagannath explained further the advantages of a WATM protocol in his thesis entitled "An Adaptive Data Link Layer Protocol for Wireless ATM Networks" [16]. One of the strengths of ATM has always been its QoS based support for multimedia traffic, therefore it seems natural not to terminate the ATM link at the base station (BS), but rather to extend it to the mobile terminal (MT) leading to a homogeneous end-to-end network, and simplifying the architecture. The extension of ATM's virtual channels with linked QoS parameters to the wireless portion of the network can provide many advantages. For example, VC identifiers can be used to suitably allocate shared wireless resources at the medium access layer, and error control schemes may be adapted according to VC parameters at the data link layer. The small size of ATM cells also has some advantages in that it leads to lower packet error rates and is suited to the slower speed of wireless links since it results in lower queuing delays.

Although the wireless access protocol presented in this thesis is based around the ATM philosophy, many of the ATM concepts are being incorporated into other protocols such as IP, so the same techniques used to provide QoS over wireless links here may be used to extend other protocols to the wireless domain.

1.2.3 Wireless Access Techniques

The demand for wireless access to networks is rapidly increasing as more and more people around the world find uses for and become aware of the advantages provided by mobility. The flexibility provided by mobile phones has been a part of everyday life for some time, but now many people want to extend this widespread flexibility, beyond a basic voice connection, into a variety of multimedia connections. The following extract from an article on wireless trends released by Lucent Technologies in 1997 [17] strongly emphasises this point:

> *"Some cellular customers regularly send and receive e-mail, files, and faxes – typically at rates under 10kbps – over voice circuits or via cellular packet systems that share facilities and spectrum with mobile phone systems. Corporations, universities, and hospitals have installed thousands of wireless local area networks – which use standard LAN protocols and transmit on unlicensed spectrum – to cut data terminals and their users loose from in-building wiring. Yet people want to move more data over the air, faster, ASAP. Demand is growing for wireless data applications addressing a range of markets, including mobile professionals, field sales and service, transportation and public safety, hospitals and campuses, banking and point-of-sale transactions, and telemetry." [17]*

There is currently a proliferation of wireless data transmission systems. The types of systems relevant to this thesis are those with a cellular architecture whereby MTs transmit traffic on a shared medium to a centralised BS with support for some form of roaming. Such systems are implemented using a variety of multiple access schemes to divide the resources into accessible sections that may be shared amongst MTs. Multiple access methods include frequency division multiple access (FDMA), time division multiple access (TDMA), code division multiple access (CDMA) and spatial division multiple access (SDMA), as well as variations or hybrids of such techniques. In fact, a spectrally efficient CDMA system may make use of all these schemes to some degree.

While multiple access schemes divide the medium into accessible sections, access protocols still require some method of sharing the resources between all the users. The various resource sharing methods can be grouped into *random access*, *dedicated assignment*, and *demand-based assignment* techniques [18], as shown in Figure 1.2. These three techniques are summarised below:

- *Random access* channels allow users to contend for the channel by transmitting as soon as they have a packet to send. Such protocols are suitable for bursty traffic but do not perform well with delay sensitive traffic. Examples of such protocols include ALOHA and slotted-ALOHA.
- *Dedicated assignment* channels assign each user a predetermined and fixed portion of bandwidth that is suitable for continuous traffic but can be wasteful for bursty traffic. GSM is a good example of a dedicated assignment protocol.
- *Demand-based assignment* schemes assign bandwidth according to reservation requests received from users. However, some form of dedicated or random access channel is still necessary for users to transmit these requests on. Demand-based protocols work well with variable rate traffic as well as multimedia traffic containing a variety of traffic types. The drawback is the additional overhead and delay caused by the reservation process.

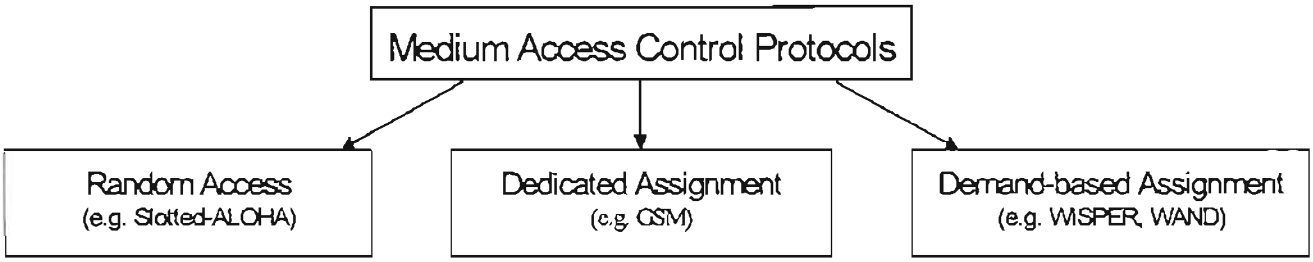


Figure 1.2 Medium Access Control protocols: resource sharing methods.

A 1996 paper by Jouni Mikkonen [12] summarised the 'state of the art' wireless access systems available at the time to include those shown in Table 1-I. Significant protocols not included in Mikkonen's table or that have emerged since then include IS-95, Bluetooth, IEEE 802.11b, and the proposed 3G standards: CDMA2000/WCDMA. A point worth noting is that all of these recently emerged protocols make use of spread spectrum multiple access schemes, whether direct sequence (like CDMA2000/WCDMA) or frequency hopped. The next section provides some details on the concept of spread spectrum.

Table 1-I Network and wireless access techniques. [12]

Wireless Data Access	Achieved Data Rates	Backbone Network
DECT	< 1 Mbps (per user)	ISDN
GSM data (GPRS, HSCSD), CDPD	< 144 kbps (per user)	Internet
ISM band WLANs	300 kbit/s-2 Mbps (shared)	LAN
HIPERLAN (draft ETS in Public Enquiry)	< 20 Mbps (shared)	LAN

The success of the IS-95 standard is in part due to the formation of the CDMA Development Group (CDG) with membership now including over 100 of the world's largest equipment manufacturers and service providers. The CDG is committed to leading the rapid deployment, development and evolution of IS-95 systems to meet the needs of customers around the world. The advantages of CDMA, demonstrated through the success of IS-95, have paved the way for the CDMA based 3G systems with the CDG now leading the transition to CDMA2000. Both WCDMA and CDMA2000 have the goal of providing data rates that meet the IMT-2000 performance requirements. The objectives for the IMT-2000 air interface can be summarised as follows [13]:

- Full coverage and mobility for 144kbps, preferably 384kbps
- Limited coverage and mobility for 2Mbps
- High spectrum efficiency compared to existing systems
- High flexibility to introduce new services.

The nominal bandwidth for all 3G proposals is 5MHz. The main reasons behind the selection of this bandwidth include the fact that 144kbps and 384kbps bit rates are achievable within 5MHz bandwidth with reasonable capacity. Also, the lack of available spectrum calls for reasonably small frequency bands, especially if the 3G systems are to operate in the same frequency bands allocated to 2G systems. However, the large 5MHz bandwidth can resolve more multi-paths than a narrower bandwidth, thus increasing diversity and improving performance. Despite the common goal for 3G laid down by IMT-2000, the different approaches taken by the standardisation bodies has resulted in the two 3G air interfaces. WCDMA has been developed as a joint effort between ETSI (European Telecommunications Standards Institute) and Japan's ARIB (Association of Radio Industries and Businesses). One of the criteria affecting the WCDMA standard has been the requirement for interoperability between WCDMA and GSM, thus allowing handovers between the two systems. CDMA2000, on the other hand, has been designed to be backward compatible with IS-95. The United States' TIA (Telecommunications Industry Association), which was responsible for the standardisation of IS-95, is responsible for the CDMA2000 framework. Despite the failure to achieve a single 3G standard, CDMA has established itself as the multiple access technique of choice. In the words of Perry LaForge, the executive director of the CDG:

> *"CDMA has experienced the most rapid deployment and rise in subscriber numbers of any wireless technology. The record for the next three years should be equally strong, even as CDMA technology evolves to meet the challenges of the 21st century."*

The reasons for CDMA's popularity are explained in the following section.

1.2.4 Code Division Multiple Access (CDMA)

This section introduces the concept of code division multiple access (CDMA), briefly explaining how it divides the bandwidth into separately accessible portions. Also included are the advantages of CDMA technology that have lead to its popularity.

What is CDMA?

CDMA technology makes use of a spread spectrum technique known as direct sequence spread spectrum (DS-SS). The term *spread spectrum* comes from the fact that, before transmission, the information contained in the data signal is spread to occupy a greater bandwidth than occupied by the original signal. A number of users' spread signals are then transmitted simultaneously in the same wide frequency band. The frequency spectrum of the original data signal is spread by multiplying it by a unique, higher rate, binary pseudo noise (PN) code that is uncorrelated with the data signal. The codes themselves are unique to each user or CDMA channel. So, instead of separating channels in time (TDMA) or frequency (FDMA), it is the unique PN code that enables a user to distinguish its data signal from the multitude of signals transmitted simultaneously over the same bandwidth. A second multiplication by a synchronised replica of the same PN code in the receiver allows the original signal to be recovered. Figure 1.3 illustrates the effect of spreading a signal. Although the spread signal occupies a greater bandwidth, the power contained in the spread signal remains the same as the power in the original narrowband signal.

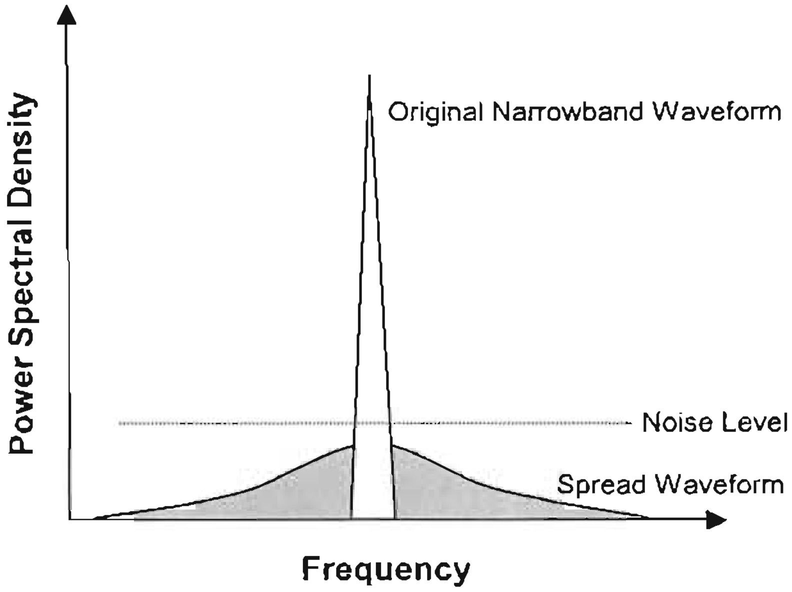


Figure 1.3 The effect of spreading on power spectral density.

For many years spread spectrum techniques have been used in military applications because of the difficulty in jamming or even detecting spread spectrum signals. There are other spread spectrum techniques besides DS-SS, the most common being frequency hopping (FH). A recently popular variation that has emerged is multi-carrier CDMA (MC-CDMA). Claude Shannon and Robert Pierce introduced the basic concepts of CDMA in 1949 by describing the interference averaging effect and the graceful degradation of CDMA, but it was not until forty years later that the use of CDMA in commercial applications was put into practice [13]. This commercialisation came about in the early 1980s as a result of Qualcomm's investigation into DS-SS techniques. Commercial applications became possible because of the availability of low cost, high-density digital integrated circuits, reducing the size and weight of mobiles to acceptable levels. Another hurdle that was overcome is the so called near-far effect whereby MTs transmitting with the same power but at different distances from the BS result in the signal from the MT closest to the BS 'swamping' the signals from MTs much further away because the signal from the closer MT arrives with a far greater amplitude. The solution to this problem followed from the realisation that optimal multiple access communications requires that all MTs regulate their transmitter output powers to the lowest strength possible to still maintain adequate signal quality. This means that power control techniques are a fundamental part of CDMA technology.

The most important parameter in spread spectrum systems is the *processing gain* (G_p): the ratio of the transmission bandwidth (BW_t) to the information (data) bandwidth (BW_d) as given by Eq. 1-1. This is also known as the spreading factor or spreading gain. The processing gain determines the number of users that can be supported by the system, the amount of multi-path effect reduction, the difficulty in jamming or detecting a signal, etc.

$$G_p = \frac{BW_t}{BW_d}$$ Eq. 1-1

As previously mentioned, spreading is achieved by multiplying the data signal by a pseudo random noise code (PN code). A PN code is a sequence of binary "chips". Figure 1.4 provides an example of how a data signal is direct sequence spread. The example uses a processing gain of five; i.e. five chips per symbol. Hence, the bandwidth of the spread signal is five times that of the original data signal. At the receiver a similar multiplication of the PN code by the spread signal allows the original data signal to be retrieved. This second PN code multiplication at the receiver must be synchronised with the multiplication at the transmitter. For the multi-user case, a number of different multi-user detection schemes are available [14], but the details of such schemes are beyond the scope of this thesis.

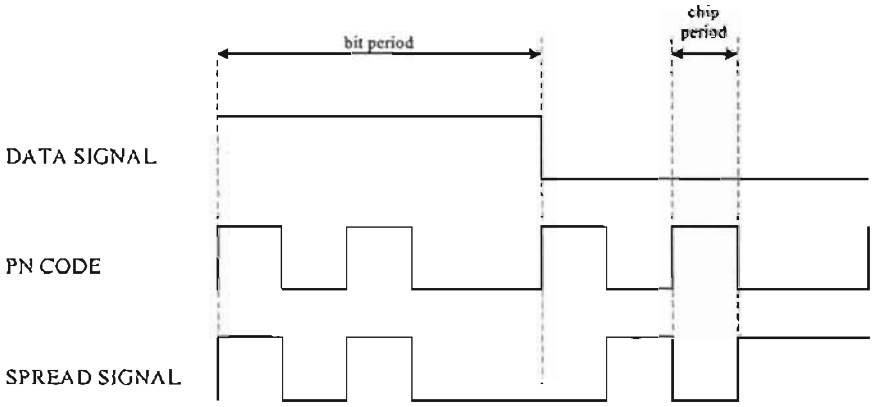


Figure 1.4 Direct Sequence spreading.

There are various families of PN codes, e.g. Walsh-Hadamard codes, M-sequences, Gold codes and Kasami codes. These can be divided into two main categories: orthogonal codes and non-orthogonal codes. Walsh-Hadamard codes are orthogonal whereas the so-called shift register sequences (M-sequences, Gold codes, etc) are non-orthogonal. The generation of PN codes is relatively easy using a number of shift registers. This allows for long code sequences and large processing gains. Using orthogonal codes has the advantage that transmitting multiple spread signals in the same frequency band results in zero multi-user interference at the receiver so long as the spread signals are perfectly synchronised. Such synchronisation is often difficult to achieve, in which case non-orthogonal codes are preferred. By replacing the requirement that the codes be orthogonal with the requirement that their cross-correlations (and hence their mutual interference) are sufficiently low, transmission over different code channels may be asynchronous. Reliability now depends on the number of simultaneous users rather than the number of potential users and, hence, it is possible to trade off reception quality for increased capacity [14]. Figure 1.5 demonstrates transmission in a multi-user CDMA system containing two users. The effect of multi-user interference is shown in Figure 1.5(d). Obviously the more codes in use the greater this interference will be.

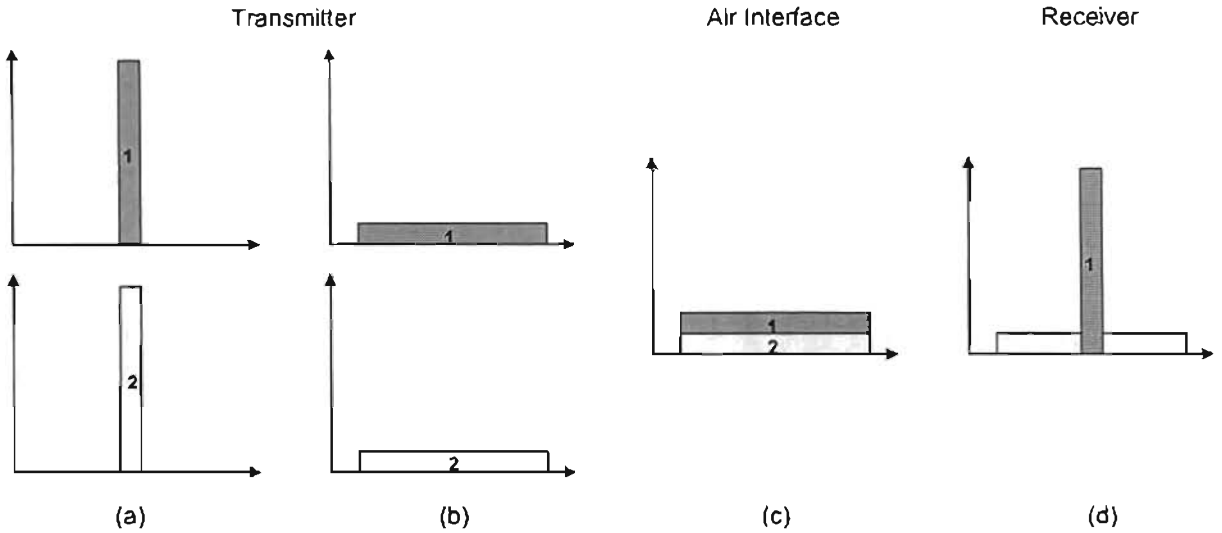


Figure 1.5 Transmission in a multi-user CDMA system: (a) Power spectral density of two narrowband signals before spreading, (b) the two signals after spreading, (c) the two spread signals overlapping in time and frequency over the air interface, and (d) the first signal de-spread with interference from the second spread signal.

The Benefits of CDMA

CDMA offers a number of benefits over other multiple access techniques explaining its current popularity. These benefits are explained in a paper from Motorola Cellular Infrastructure Group entitled "CDMA Technology & Benefits" [15], and are summarised below. The first advantage of CDMA is its so-called soft capacity. Because each CDMA code channel effectively sees the other CDMA code channels as noise, the number of code channels in use determines the multi-user or multi-code interference. This interference is, of course, over and above the multi-path noise (the effect of which is greatly reduced by spreading). So, CDMA has a soft capacity in that more codes may be used to increase the available bandwidth at the expense of reduced channel quality. One may take advantage of this trade-off to provide some control over the quality of the radio interface.

Another benefit is the patented soft hand-off technique used by CDMA systems. The fundamental advantage being that, during a hand-off, a MT may connect to more than one BS, reducing the possibility of dropped calls. The power control schemes that are such a critical component of optimal CDMA systems lead to increased MT power efficiency as the average mobile transmitting power is decreased. This results in a longer battery life for mobile devices. CDMA also optimises the use of radio spectrum that is becoming increasingly scarce worldwide. This is because CDMA has the capability to re-use a single cell frequency in adjacent cells, which also simplifies RF planning and implementation. Finally, the fact that spread spectrum signals are difficult to detect and to jam means that there is also enhanced privacy & security inherent in CDMA systems.

1.3 Roadmap of the Thesis

This chapter has presented some background history on mobile communications, the future of mobile – where it is heading – and the contribution of this thesis. An introduction to ATM, its impact on telecommunications networks and its future were also discussed. The chapter ended with an explanation of the concept of CDMA, its influence on current and future wireless systems, and the reasons for its popularity.

Chapter 2 summarises the relevant literature available on WATM networks as well as other CDMA based wireless protocols that have been proposed. The chapter begins with an overview of the WATM protocol stack as proposed by the ATM Forum, and the functionality required by each layer. This is followed by the evolution of WATM protocols, starting with the first WATM network that was built by Olivetti Research Limited in the early 1990s. The focus is on the link layer techniques used i.e. the medium access control (MAC) and data link control (DLC) protocols, and the mechanisms included to cater for different traffic types and service qualities. The merits and shortcomings of the various link layer methods are also discussed. A number of CDMA based MAC protocols are also presented with the focus the same as for the WATM protocols. Next is a summary of the fundamental design choices and their influence on the overall system. The chapter concludes by highlighting how CDMA and WATM may be combined to provide better QoS guarantees.

Chapter 3 proposes a WATM MAC protocol designed to take advantage of a hybrid TDMA/CDMA physical layer. The protocol combines some of the techniques mentioned in Chapter 2 and attempts to support as many different traffic characteristics as possible while maintaining suitable qualities of service. The structure of the WATM cell, frame header and some other specialised cells are discussed, followed by an overview of the scheduling algorithm used. The protocol was designed to be implemented using Alcatel Altech Telecoms' (AAT's) software radio architecture.

Chapter 4 presents the proposed DLC protocol. The chapter begins with a discussion of error control schemes, and how to maintain QoS while handling transmission errors. An overview of the supported retransmission scheme is followed an explanation of how application specific forward error correction (FEC) techniques are supported in preference to applying a fixed FEC scheme. The chapter closes with a brief discussion on the call admission control, and the role it plays in guaranteeing QoS.

Chapter 5 begins with a discussion of the software radio concept. The chapter provides an overview of AAT's software radio architecture, describing the hardware infrastructure used to implement the WATM protocol. The chapter then goes on to explain how the BS and MT link layers were implemented in software. The implementation of each block in the system is discussed. This includes a description of the interface with the physical layer, the WATM signalling protocol used, and the software running on host PCs that is used to generate multimedia traffic in order to test the protocol.

Chapter 6 introduces a number of traffic models used to generate multimedia traffic in order to test the protocol using a simulation model developed in Mil3's Opnet Modeler, as well as using the actual implementation. The parameters used in the simulation of the protocol are presented along with some results from the simulation. Some results from traffic measurements taken by applying the traffic models to the actual implementation are also included and compared to the simulated results.

Finally, the thesis is concluded in Chapter 7 with a discussion of the contribution made and some suggestions for possible improvements to the proposed protocol.

Chapter 2 Literature Survey

This chapter introduces some existing research on WATM and CDMA systems. It begins with an overview of the WATM protocol stack proposed by the ATM Forum and then discusses selected WATM projects providing an indication of how WATM research has evolved. A brief description of some CDMA based MAC protocols is presented followed by a summary of some of the fundamental design choices. The chapter concludes with some proposed MAC and DLC techniques to guarantee QoS in the wireless domain.

2.1 The WATM Protocol Stack

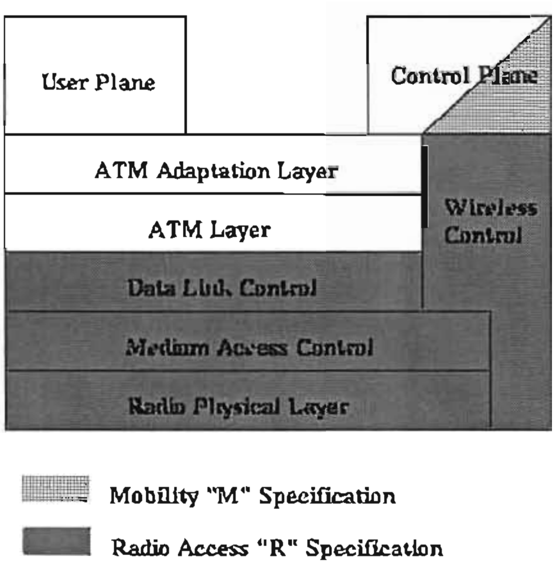


Figure 2.1 The WATM protocol architecture.

Figure 2.1 shows the WATM protocol architecture as proposed by the ATM Forum. It is divided into two distinct parts:

1. The *Mobility Specification* residing in the control plane deals with the higher layer control/signalling functions required to support mobility. These include functions to support mobile terminal hand-offs, location management, routing, and addressing. Such mobile ATM extensions are not within the scope of this thesis, although they are of great importance to the overall WATM protocol development.

2. The *Radio Access Specification* is responsible for the radio link protocols for WATM access. Radio Access Layers consists of a physical (PHY) layer, medium access control (MAC), data link control (DLC), and wireless control (for radio resource management).

The MAC and DLC layers are the main concern under the scope of this thesis. However, all the layers of the radio access specification are closely related and impact on one another. There are numerous issues that need to be dealt with and choices that have to be made in adapting ATM to a wireless link. Some of the more prominent issues, as related to the various access layers, are summarised below [19] [20] [21]...

2.1.1 Physical (PHY) Layer

While on a fixed ATM network a single user may enjoy connections of 25Mbps up to 155Mbps, a 25Mbps link is currently difficult to implement in a wireless environment. Nevertheless, if WATM is to provide a transparent extension of ATM into the wireless domain, then this is what PHY must strive for. The ATM Forum has proposed 25Mbps to be a standard bit rate in the 5GHz frequency band with a coverage radius of 80m. The frequency bands used need to be available and appropriate for the target environment e.g. at higher frequencies signals fade more quickly resulting in smaller coverage areas for a given transmission power. The WATM test bed proposed for this project will be required to have a cell radius of up to 1km. Hence, frequency bands below 5Ghz will probably be used in order to increase coverage. It is no good increasing transmission power to increase coverage as this will lead to the batteries of the mobile devices discharging too quickly.

ATM has been designed to run over very reliable fibre-optic links with bit error rates (BERs) of 10^{-9} and less. A radio environment may provide BERs of around 10^{-2} or 10^{-3}. Some sophisticated techniques (antenna diversity and equalisation) may be used to reduce multi-path effects, but these will not be sufficient to bring BERs to levels required to support tolerable Cell Loss Ratios (CLRs) for standard ATM adaptation layers. Hence, this problem will have to be countered throughout the radio access layer.

Support for links to moving MTs shall have to be provided by PHY. Generally W-LANs support MTs moving at walking pace, but systems with wider coverage areas, such as GSM, try support mobiles moving at higher speeds e.g. in cars or trains.

2.1.2 Medium Access Control (MAC) Layer

The WATM MAC must support functional point-to-point links for the higher layer protocols to use. Each MT will need to associate/register itself with an access point or BS, using its full WATM MAC address that may then be replaced with a much shorter address that only has local significance. Some form of BS initiated slotted-ALOHA contention period seems most popular to achieve this.

It is commonly agreed that in a shared (broadcast) environment some control over the usage of the medium is required i.e. The MAC layer must organise the access to the radio link such that data can be sent on each connection without collisions. A fair and efficient method for a frequently changing population of mobiles is required. The WATM requirements are extremely stringent, as QoS guarantees for a variety of traffic types have to be met. A flexible and dynamic allocation of available bandwidth will be required in order to maximise the utilisation of the channel.

One primary advantage of ATM is its support for QoS guarantees. If WATM is to be a truly transparent extension to a fixed ATM network then all the traffic management service categories of fixed ATM need to be supported. That includes constant bit rate (CBR), real time variable bit rate (rt-VBR), non real time variable bit rate (nrt-VBR), available bit rate, and unspecified bit rate. As MTs move between BSs their QoS arrangements may need to be renegotiated in order to maintain the level of service for the MTs already in the cell.

Some MAC protocols combine multiple ATM cells into single packets to minimise overhead, and increase efficiency. There is, however, another school of thought that believes it is better to support the transmission of single ATM cells in order to minimise latency and jitter.

Many mobile devices will be battery powered. Therefore, power saving techniques will need to be considered to allow MTs to power down during periods of inactivity, and while waiting for predetermined events. Control techniques for such power conservation must be incorporated into the MAC protocol.

2.1.3 Data Link Control (DLC) Layer

The DLC layer is responsible for providing service to the ATM layer. Errors in the radio channel need to be dealt with in this layer before passing cells on to the ATM layer. The aim must be to provide cell loss ratios (CLRs) approaching those of wired ATM. This link layer error control can consist of a combination of automatic repeat request (ARQ) and forward error correction (FEC) techniques. The technique chosen for each connection should depend on the connection's traffic service category and the parameters chosen. The DLC therefore requires traffic contract and QoS information on a per connection basis, meaning that close ties between the MAC and DLC layers are required.

The effectiveness of FEC will have to be determined for the radio channel's environment. The ARQ approach will have to be carefully designed so as not to break a connection's traffic contract, especially with regards to the cell delay variation (CDV) of CBR and rt-VBR traffic.

Another important issue that must be dealt with is cell sequencing. Traditionally, in a wired ATM network, the ATM standards state that the underlying protocols will not reorder cells, although in practice the various AALs detect mis-ordered cells and discard the affected PDUs. In a mobile environment, network re-configuration occurs much more frequently than in wired networks (due to hand-offs) and hence the problem of mis-ordering of cells is important. The hand-over mechanism used should minimise cell mis-ordering in order to reduce the number of cells discarded for this reason.

2.1.4 Wireless Control/Radio Resource Control (RRC)

The RRC is required to support the control plane functions related to the radio access layer. It should include control and management functions for PHY, MAC and DLC. The design issues will include control/management syntax for the PHY, MAC and DLC layers, and the interface to the ATM control layer.

2.2 The Evolution of WATM

Research into WATM networks has been going on for a number of years now. The concept of WATM first appeared in the literature in 1992 [22]. As early as 1994, Olivetti Research Limited had RATM, an experimental wireless ATM network, up and running. The ATM Forum's well established WATM working group is currently developing a set of specifications intended to facilitate the use of ATM technology for a broad range of wireless network access scenarios. These include both the "mobile ATM" extensions for mobility support within an ATM network, as well as the "radio access layer" for ATM-based wireless access [4]. As of August 2000 the ATM Forum's WATM specifications are still works in progress with the forecast for the final specification release still to be decided [23].

WATM research has generally been aimed more at the W-LAN environment based around pico-cells with radii of 100m or less, rather than the public mobile networks that have larger cell radii up to several kilometres. This is because ATM is traditionally used for broadband networking with its statistical multiplexing of many virtual channels into a single 'pipe' performing better with higher bandwidth 'pipes'. Obviously smaller pico-cells allow for higher bandwidths and smaller delays over the air interface. But, in the future, users of such high performance W-LANs will want these wireless multimedia capabilities extended to the public mobile networks so that they don't have to be in an office environment containing a wireless LAN to use them.

2.2.1 Olivetti Research Limited's RATM (1994)

John Porter and Andy Hopper of Olivetti Research Limited (ORL) present one of the earliest WATM implementations in a paper entitled "An ATM based protocol for Wireless LANs" [24]. The protocol is named Radio ATM (RATM) and an experimental version of this system had been built as early as 1994. Designed for indoor W-LAN use, the system consists of a large number of small pico-cells (with radii as small as 10m) each served by a base station. Reducing the size of the cells solves many problems due to multi-path effects and lack of line-of-sight, but increases the hand-over frequency. All cells operate on the same frequency and hence no hand-over is required at the physical layer. Instead, the system uses a form of soft hand-off similar to that used in CDMA systems. The unit of transmission over the wireless link is still the ATM cell, simplifying the base station requirements. Base stations are simple cell-relays that translate the cell header formats between the fixed and wireless networks.

The MAC protocol used is slotted-ALOHA with exponential back off. The slot size is the same as that of the RATM cell, and the slot structure is maintained by re-synchronising the mobile to the base station on every reception. The multiple overlapping pico-cell approach results in potential interference between adjacent cells. ALOHA is robust under such circumstances, with the state of the receiver being most important.

Slotted-ALOHA has a better delay performance than a fixed allocation scheme at low utilisation. Assuming a low utilisation may be a valid assumption when there are many small pico-cells and the chances of many mobiles being used in one pico-cell at the same time are fairly small. However, when large areas need to be covered it becomes economically unfeasible to cover the entire area with small pico-cells. The use of larger cells (with radii > 100m) means that each base station's utilisation will increase as more mobiles are likely to be used in a cell at any one time. Hence, ALOHA won't be a suitable protocol – under high utilisation ALOHA falls into a state of low throughput in which most transmission attempts are repeats.

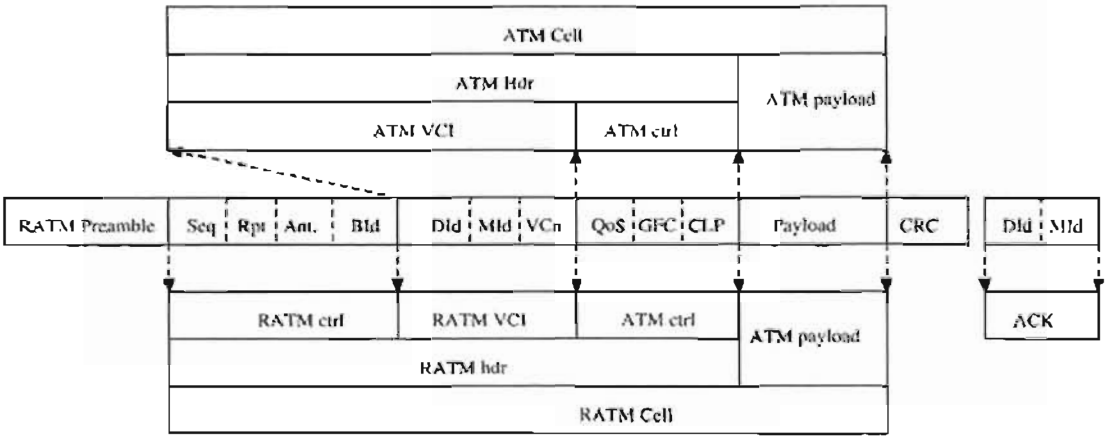


Figure 2.2 The 1994 RATM cell format. [24]

Figure 2.2 shows the relation between the ATM cell format and the RATM cell format. The RATM cell header has a number of extra fields, but condenses the ATM VCI/VPI into fewer bits and divides it into:

1. Did (Domain Identifier) – used to 'colour' cells in order to distinguish adjacent network management domains.
2. Mid (Mobile Identifier) – uniquely identifies the mobile within the domain.
3. VCn (Virtual Circuit number) – uniquely identifies a virtual circuit within the domain when combined with Mid.

The Bid field in the RATM cell header is a Base station Identifier used by mobiles to select a particular base station for reception. A 16-bit CRC is used to detect errors, and MAC acknowledgements are generated when a cell is correctly received.

The sequence number (Seq) is used to recover from missing acknowledgements: If a cell is correctly received but for some reason its acknowledgement is lost, the transmitter will re-send the cell. The receiving interface will detect that it has already received the cell with this sequence number and will send an acknowledgement but discard the cell. After every successful cell transmission the source will increase its sequence number. This process prevents duplicate cells from being passed on to the wired part of the network where they would only be picked up in the AAL and may result in the entire PDU block being discarded.

The technique of altering the ATM header over the wireless link by reducing the VCI/VPI fields to fewer bits and adding extra fields such as a CRC for error detection is commonly applied in most WATM protocols. This shall become apparent in the following sections. This initial RATM protocol provided limited mechanisms for controlling the QoS provided to each virtual circuit. This is mainly due to the hit and miss nature of ALOHA. However, RATM did avoid handing over an MT to a base station that couldn't support the MT's current QoS requirements. If, however, the mobile had to be handed over to such a station, the QoS parameters would determine those VCs that maintained their current service, and those that had to renegotiate. This highlights one of the disadvantages of small pico-cells in that they result in more frequent hand-offs and hence a greater chance of a VC's current QoS being renegotiated due to hand-off. Also, by utilising slot allocation schemes instead of ALOHA techniques, the more recent WATM protocols have provided mechanisms to adhere to traffic contracts across the wireless links, even under high loads.

2.2.2 Olivetti Research Limited's RATM (1996)

By 1996 the WATM research performed at Olivetti Research Limited in Cambridge had changed course somewhat. The single frequency ALOHA system described above had evolved into a frame based reservation protocol with adjacent cells coloured by frequency [25]. The multiple access technique had switched to TDMA. However, the cells continue to be small pico-cells.

Colouring is the allocation of individual channels to adjacent BS cells in order to prevent interference between them. Colouring techniques include separating adjacent BS cells in frequency or time or CDMA codes, depending on the multiple access scheme in use. Some form of colouring is necessary in order that QoS guarantees may be successfully satisfied. In the previous system all BSs operated on the same frequency, and there was no real colouring between them. This was made possible by the use of ALOHA. The new RATM system uses frequency division as its cell colouring technique. This was chosen because TDMA requires the modem to operate at a multiple of the available bit rate and requires synchronisation between base stations; and the large bandwidth required by CDMA was not considered to be practical. Each base station operates on a single frequency with adjacent base stations operating on different frequency channels.

The MAC protocol is a frame based reservation protocol, optimised to support a small number of mobiles – all that is necessary for a pico-cell environment. The ABR, VBR and CBR traffic classes are supported by the protocol. A variable number of cells may be transmitted in a MAC frame, with the length of the frame determined by the number of outstanding reservation requests and the recent contention history. The allocation of bandwidth to the upstream and downstream traffic is completely flexible due to the variable frame structure. This is significant, as it may be difficult to predict the actual proportion of uplink and downlink traffic.

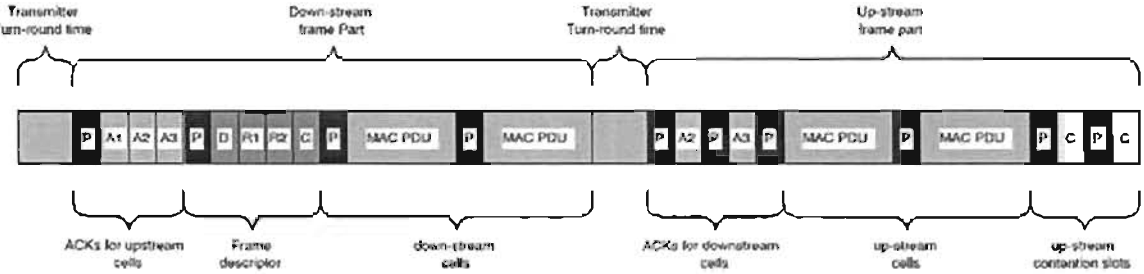


Figure 2.3 The 1996 RATM MAC frame. [25]

The MAC frame, shown in Figure 2.3, consists of a downstream part and an upstream part. Each element that can be received independently has a preamble (P). A1, A2 and A3 are acknowledgements (ACKs) for the previous upstream cells and reservation requests. The frame descriptor field consists of the number of downstream cells (D), VPIs for upstream reservation indications (R1, R2) and the number of contention slots I. This is followed by downstream MAC PDUs consisting of an ATM cell and MAC protocol wrapper (Figure 2.4). A2 and A3 are ACKs for the downstream cells in the same MAC frame. Notice that each of these ACKs is preceded by a preamble, whereas the upstream ACKs are grouped together with just a single preamble. This is because the upstream ACKs are all transmitted by a single radio interface (i.e. the BSs), whereas various different mobiles transmit the downstream ACKs. Following the downstream ACKs are upstream MAC PDUs (Figure 2.4). The frame also contains at least one contention interval, which is used in a slotted-ALOHA mode to make a reservation in the subsequent frame. Reservation requests may also be piggybacked onto a cell transmission (Figure 2.4).

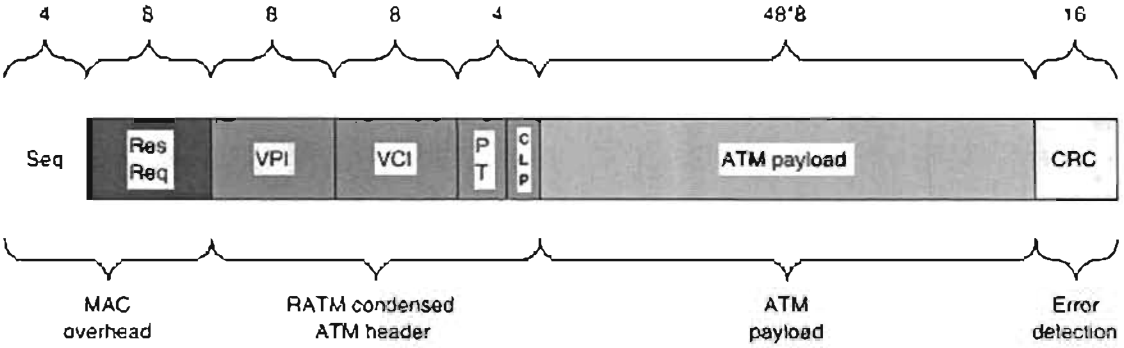


Figure 2.4 The 1996 RATM MAC PDU. [25]

The MAC PDU, as shown in Figure 2.4, contains a sequence number, reservation request, ATM cell including a condensed header, and a 16-bit CRC. The ATM header is compressed by reducing the 28-bit VPI/VCI of fixed ATM cells to 16-bits over the wireless channel, as well as dropping the 8-bit CRC normally included in the fixed ATM cell header. Instead, the 16-bit CRC at the end is used to detect errors in the entire PDU. Cells that are received correctly are acknowledged in the subsequent acknowledgement interval. An unacknowledged cell is repeated a number of times before being discarded. The number of repeats being dependent on the traffic class for the cell. If a cell is not correctly received, the reservation is carried over into the next frame.

The number of upstream and downstream cells in a MAC frame is limited to 10 in order to place an upper bound on the cell delay variation for CBR traffic. Each cell transmission is independent such that in an error burst only the effected cells are lost. Hence, each cell has its own preamble and CRC. This helps to retain the advantages of transmitting short PDUs in a high BER environment.

As the required bandwidth of a single traffic source approaches the available bandwidth, the number of cells being queued in the RATM interface will increase. This will result in outstanding reservation requests dominating each frame. Thus, in [25], the MAC protocol is described as behaving like slotted-ALOHA for low load and approaching TDMA as the load is increased. This description can be misleading though because if the number of traffic sources is increased rather than increasing the bandwidth of a single source, the slotted-ALOHA contention part of the protocol in which reservation requests are transmitted will become congested. So, although the protocol uses small pico-cells, having numerous MTs in a single cell will result in the MAC protocol featuring slotted-ALOHA type congestion rather than performing like a TDMA protocol.

Overall, the 1996 RATM protocol described in [25] is a significant improvement over the original RATM protocol of 1994. The fundamental improvement being the switch from a slotted-ALOHA protocol to a frame based reservation protocol, allowing the traffic classes defined by the ATM Forum at that time (ABR, VBR, CBR) to be directly supported.

2.2.3 WATMnet

WATMnet is an experimental wireless ATM network prototype developed at NEC USA's C&C Research Laboratories in Princeton, NJ, during 1994 and 1995. The prototype micro-cellular WATM network, described in [26], utilises a dynamic TDMA protocol with time division duplexing (TDD) to provide integrated multimedia communication services to remote terminals. The MAC protocol incorporates all three channel resource sharing methods: dedicated, random and demand assignment [18]. Figure 2.5(a) shows the MAC frame format used. The downlink transmission is a single burst containing modem preamble, frame header, control information, cell acknowledgements and data in the form of wireless link data cells [Figure 2.5(b)]. The 56-byte wireless link data cells consist of a 48-byte payload, 4-byte compressed ATM header, 2-byte wireless header and 2-byte CRC. Both the wireless control packets and acknowledgement packets are 8-byte mini-packets. The uplink portion of the frame contains a contention access control sub-frame for reservations, followed by ABR/UBR, VBR and CBR data slots respectively.

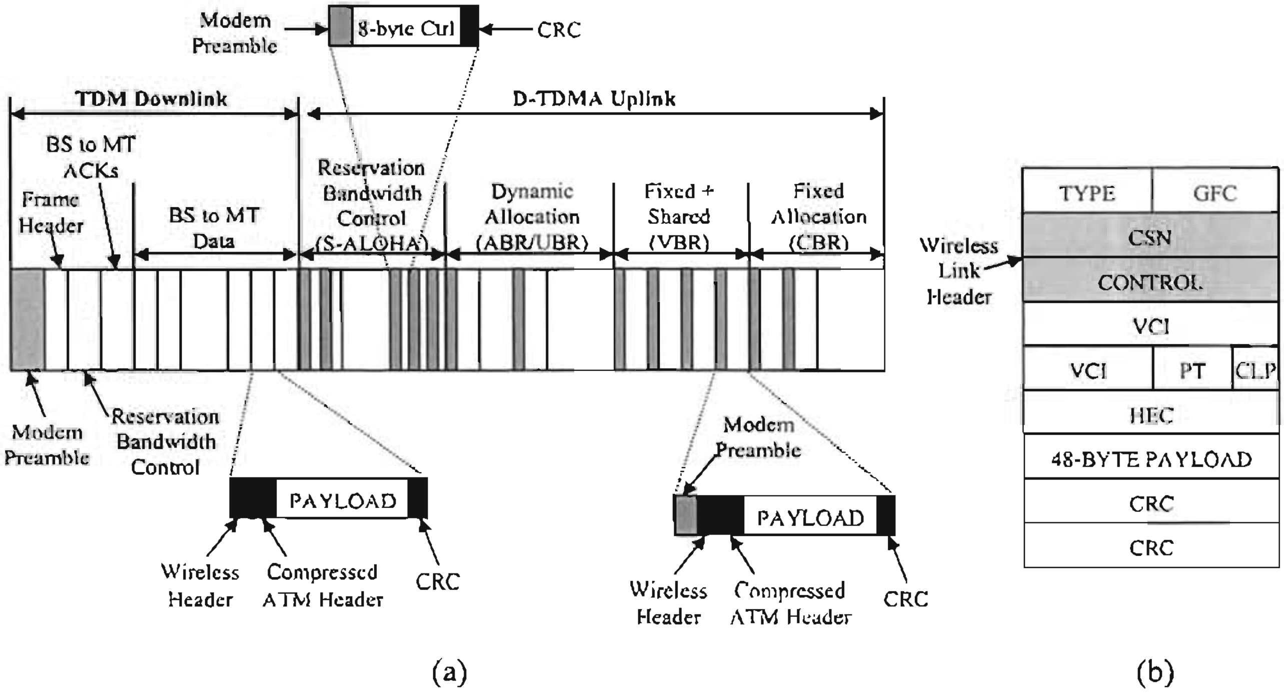


Figure 2.5 WATMnet's (a) TDMA/TDD frame format, and (b) data cell format.

MTs send transmission requests to the BS in the dedicated reservation slots using slotted ALOHA random access. At the BS a supervisory MAC processes the requests and builds a schedule table for both the uplink and the downlink. For CBR traffic, slot allocation is performed once during call set-up with a fixed allocation of slots assigned according to the VC's bit rate requirement. For this purpose a group of K MAC frames (e.g. K = 6) constitutes a MAC super-frame that is used to obtain a reasonable granularity for bit rate assignments. VBR slots are assigned with the aid of a usage parameter control (UPC) based statistical multiplexing algorithm. As with CBR VCs, VBR VCs have fixed slot allocations, however the unused slots are shared with other traffic classes. Arriving CBR and VBR calls are blocked when slots are not available. Finally, ABR/UBR slot allocation is handled on a burst-by burst basis with dynamic reservation of ABR/UBR slots as well as unused CBR and VBR slots in each frame. ABR/UBR calls are always accepted and assigned contiguous slot allocations whenever possible.

When [26] was written the initial WATMnet prototype implemented a simplified MAC layer only supporting ABR VCs, and hence the DLC only supported the ABR service. The DLC uses the 2-byte wireless header shown in Figure 2.5(b) for sequence numbering of WATM cells, and the 8-byte ACK packets for group acknowledgements of up to 20 cells. A zero cell loss data-link policy is implemented for ABR traffic with no time limit for error recovery imposed. A selective retransmission technique is used, so correctly received cells are buffered at the receiving end while the transmitter re-transmits the erroneously received cells until all cells in the group are correctly acknowledged. The receiver needs to buffer the cells in order to ensure sequential delivery to the ATM layer. This DLC process is applied on a per VC basis and was to be extended to include VBR and CBR traffic types in the next phase of the work.

2.2.4 Rapidly Deployable Radio Networks

The Rapidly Deployable Radio Networks (RDRN) project investigates WATM technology for high performance mobile radio networks in the framework of the Global Mobile Information Systems (GloMo) program initiated by DARPA [27]. The conceptual view is different to the traditional approach, with the overall goal being the implementation of a wireless network architecture that is self-configuring and rapidly deployable in an army battlefield situation or a civilian disaster relief operation. The overall system features the combination of two overlaid wireless network technologies: (1) a low-speed, low power, omni-directional packet radio network intended for location dissemination, topology configuration, and link set up management between RDRN nodes; and (2) a high-speed, multi-directional, beam-forming network over which point-to-point wireless ATM links are established. The radio networks are able to operate over distances as far as 10km.

The RDRN system is made up of two types of nodes. Remote Nodes (RNs) provide WATM access to end-users. Edge Nodes (ENs) serve as radio access points to enable switching and connectivity among RN users and the ATM LAN. When initially deployed, each RDRN node retrieves its location from a GPS receiver. The location is used to steer antennae beams towards nearby nodes. ENs are capable of forming multiple beams in the direction of other ENs (at 10 Mbps) or towards RNs in the vicinity (at 1 Mbps). Using a TDMA structure, as many as 64 users (RNs) can compete for the available bandwidth in a particular beam. Unlike other WATM implementations found in the literature, wireless access in RDRN is not restricted to the last hop. This makes the RDRN system unique by enabling multi-hop wireless topologies.

Packets transmitted over the WATM network are encapsulated using the WATM frame format shown in Figure 2.6. There are 3 types of WATM packets: delay sensitive data packets, loss sensitive data packets, and control packets. Only the two data packet types contain ATM cells. The frame contains a W-PHY layer header for channel equalisation and timing at the physical layer on the receiver. The W-MAC appends a header containing a link-level address, frame type (delay sensitive data, loss sensitive data, or control), encoding scheme (not yet implemented when [27] was written), and number of ATM cells (for data type frame) / control type (for control type frame).

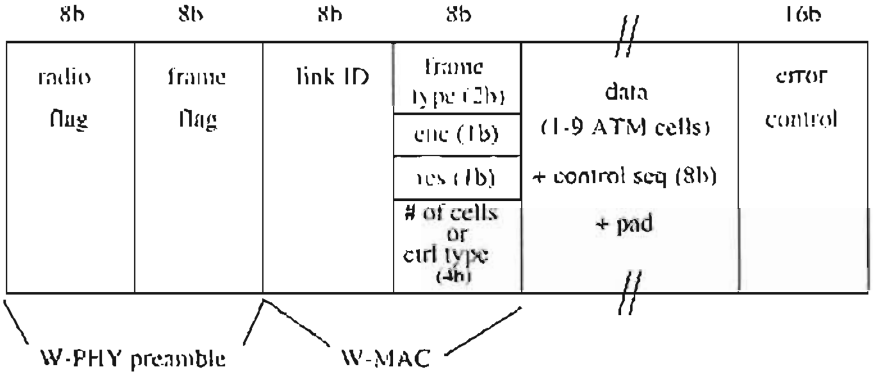


Figure 2.6 RDRN WATM frame format. [27]

An interesting feature of RDRN is the approach taken in the wireless DLC. The DLC protocol is where the resilience to the unreliable radio link is implemented such that the QoS requested by individual virtual channels may be supported. RDRN's adaptive DLC, as described in [28], identifies two fundamental traffic types: delay sensitive and loss sensitive. All VCs are categorised as one or the other. When a WATM frame carrying delay sensitive cells is received with errors, the frame is dropped. Frames carrying loss sensitive cells are retransmitted when lost or received with errors. Retransmissions are implemented using a sliding window and go-back-n ARQ scheme that guarantees the delivery of the ATM cells to the upper ATM layer in the correct sequence.

The novel feature of the wireless DLC protocol is the adaptive control it provides. A two-state channel model is used to treat the link as either a high bit error rate (HBER) link or a low bit error rate (LBER) link. This requires an estimate of the wireless channel conditions at frequent intervals. The estimation method is based on a ratio of the total number of frames received without error to those received with error. Since the channel is slowly varying, the ratio is computed at an interval corresponding to an estimate of the interval over which the channel is assumed to remain unchanged. Using two threshold values to prevent oscillation between the two states, the ratio is used to determine what state the wireless channel is in: HBER or LBER.

The channel's state is then used to adjust the length of the RDRN frame. For LBER the frame length may be increased to increase data throughput. However, increasing the frame length will naturally increase the frame error rate for a given BER. Hence, there must be an optimal frame length for a given BER. Optimal frame lengths have been determined by experimentation [28]. Similarly, for HBER the frame length is decreased to reduce the likelihood of errors in each frame. Another mechanism used for loss sensitive traffic, in the HBER state, is pre-emptive retransmission or the *N-copy* scheme. The basic idea is to send multiple copies of a frame with each transmission / retransmission with the hope that at least one of the frames will be received error free, reducing the total number of retransmission requests.

2.2.5 Wireless ATM Network Demonstrator

Magic WAND (Wireless ATM Network Demonstrator) is a joint European project (ACTS Project AC085) to specify and implement a wireless access system for ATM that extends the service characteristics and benefits of ATM technology to mobile users [29]. The project actively promotes the standardisation, notably in ETSI and the ATM Forum, of wireless ATM access as developed in WAND. WAND terminals are designed for a micro-cell environment with frequent hand-offs (much like ORL's RATM). Communication between mobile terminals and the access points takes place in the 5 GHz frequency range at a nominal transmission speed of 20 Mbps and a range of up to 50 metres.

The WAND MAC layer multiplexes different virtual connections using a dynamic TDMA scheme, as shown in Figure 2.7. The WAND frame has a variable length with time division duplexing (TDD) used to separate the uplink and downlink. Hence, there are periods for uplink traffic and downlink traffic. The frame consists of a downlink broadcast period (FH), a period for reservation based downlink and uplink, as well as a contention-based period used for unpredictable uplink traffic such as control information to be exchanged for the registration of a new MT at an access point (AP). The structure and content of each frame will depend on the currently active ATM connections.

Thus, based on the above, the MAC scheme employed in WAND has been called Mobile Access Scheme based on Contention And Reservation for ATM (MASCARA). MASCARA has been very well documented in [30] including extensive SDL descriptions of the architecture.

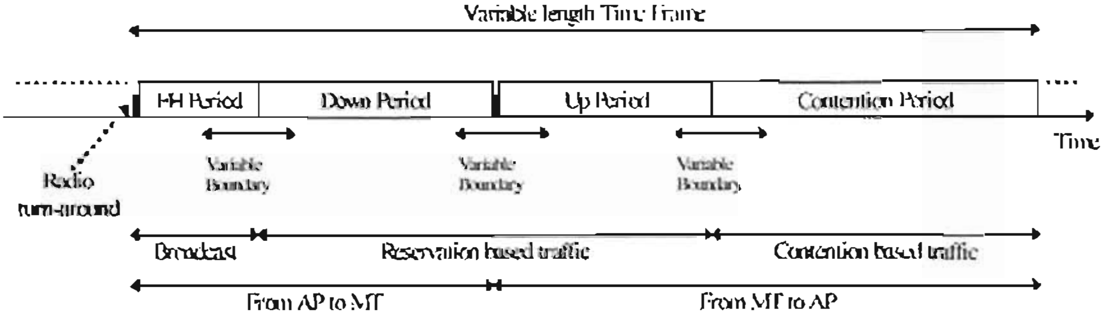


Figure 2.7 General structure of the MASCARA time frame. [30]

Each frame is divided into fixed timeslots comprising 54 bytes. Each slot can accommodate one ATM cell payload (48 bytes) along with a modified ATM header (4 bytes) and some error control information (2 bytes). Another reason for the size of the timeslot is that wand uses orthogonal frequency division multiplexing (OFDM) with 16 sub-carriers as its modulation technique. The slot size is an integer multiple of an OFDM symbol.

An AP allocates slots to an ATM connection according to the traffic requirements it has at every specific moment. An AP can determine the downlink traffic requirements from the arriving downlink ATM cells. Uplink needs are either expressed via reservation requests that are piggybacked onto uplink MASCARA protocol data units (MPDUs), or derived from each ATM connection's traffic and QoS parameters. A time frame always begins with the frame header (FH), containing a descriptor of the current time frame. A frame ends with the contention period that is used to deal with the traffic that can't be anticipated. A slotted-ALOHA based technique is used to gain access to the medium in the contention period.

Because of the physical layer overhead in the wireless link, efficient transmission can only be achieved if the length of the transmitted data packets is not too small. On the other hand, the high BER that is characteristic of wireless links dictates that smaller packet lengths are preferable to keep the packet error rate below acceptable values. This is the issue dealt with by RDRN's DLC protocol. A single ATM cell is considered too small to be transmitted as a single MPDU. Hence MASCARA defines the concept of a 'cell train' which is a sequence of ATM cells transmitted as the payload of an MPDU. So each MPDU comprises a MPDU header followed by a payload of

ATM cells all generated by or destined for the same MT. The time required by the physical layer to initiate a MPDU (physical header), plus the MPDU header transmission time, is equal to one timeslot. Figure 2.8 summarises the frame structure. In many ways each MPDU is equivalent to the entire RDRN frame, but many MPDUs make up one MASCARA frame.

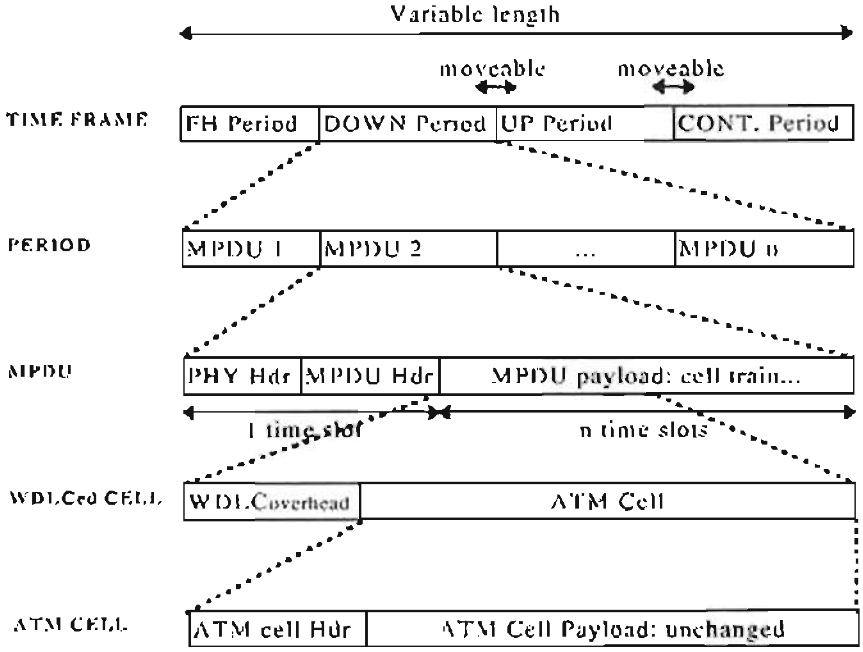


Figure 2.8 Summary of the MASCARA TDMA structure. [30]

The MPDU header contains control information such as source and destination addresses and reservation requests. In some situations when only control information (e.g. reservation requests) needs to be conveyed an MPDU may only contain the header and no payload. The 20 byte fixed ATM addresses are reduced to 2 byte MASCARA addresses across the wireless link. These shortened addresses are used in the MPDU as opposed to the 20-byte addresses, making for more efficient use of the radio link. The only additional operation required is a mapping between the MASCARA address and the real ATM address at the AP. Similarly, as in ORL's RATM, the VCI/VPI fields of the ATM header are shortened to a single byte to make room for additional fields to be used, for example, for flow control. This means that up to 256 connections may come from a single MT and up to 65536 MTs may be associated with a single AP.

In order to provide each VC with the service promised to it, MASCARA incorporates a scheduling function that has to cope with the ATM connections and traffic contracts established with the ATM network during call set up. A priority is introduced for each connection based on its service class. CBR connections are assigned priority 5, then rt-VBR, nrt-VBR, ABR and finally UBR with priority 1. Additionally, a token pool, located at the AP, is assigned to each connection. Tokens are generated at a fixed rate equal to the mean cell rate specified in the connection's traffic parameters. For every slot allocated to the connection, a token is removed from the pool. The size of the pool (maximum number of tokens allowed) is equal to the "burst size" of the connection. The burst size depends on the characteristics of the connection, and is the maximum number of cells that may be transmitted at a rate above the specified mean cell rate. Hence, at any instant the state of the token pool gives an indication of the bandwidth utilised by the connection – the reasoning being that the connection with the most tokens has consumed proportionally the least bandwidth compared to its declared one. So, the MASCARA scheduler allocates slots to connections with the most tokens first.

In order to maximise the number of ATM cells transmitted before their deadlines, each cell is initially allocated a slot as close to its deadline as possible. To obtain high channel utilisation the algorithm is work conserving such that the channel never stays idle as long as there are cells to be transmitted. Consequently the scheduler re-arranges the ATM cells to obtain final cell transmission times as early as possible given the initial ordering. The final MAC report for the Magic WAND project [30] provides a more detailed description of MASCARA's scheduling algorithm with examples.

Transmission errors are handled by the WAND DLC protocol, described in a paper by Meierhofer [31]. The WAND DLC implements a go-back-n (GBN) ARQ strategy using a sliding window method to deal with transmission errors. To apply this strategy the modified ATM cell header includes a sequence number (SN), and a 2-byte CRC trailer for the error detection at the receiver. The receiver only accepts cells that arrive error free in the correct order. Every cell is acknowledged by sending a feedback with the corresponding SN to the transmitter. This feedback may be piggybacked on a dedicated field of the modified ATM cell header in the reverse direction or, if there is no traffic in the reverse direction, the receiver generates a special acknowledgement cell. If a cell is incorrectly received then the SN of the successfully received cell as repeatedly acknowledged until the expected SN arrives. Thus, it is possible for the transmitter to detect a cell loss and then resume transmission from this cell. The sliding window's length determines the maximum number of cells that the transmitter may send without getting any feedback i.e. it basically specifies how far ahead of the ACKs the cell transmissions can get.

This GBN ARQ with sliding window is the same method used for loss sensitive data in the RDRN project, but unlike RDRN that simply dropped incorrectly received cells for delay sensitive data, WAND uses this GBN ARQ method for all traffic types. However, a selective cell discarding mechanism is implemented in order to deal with the high cell delays for delay sensitive data. Every cell has a limited lifetime during which retransmission is possible. When this time expires the transmitter discards the cell. Obviously, reducing the cell lifetime reduces the maximum cell transfer delay, but increases the cell loss ratio. Thus, by choosing appropriate cell lifetimes, different service classes are supported in a manner that is more flexible than that of the RDRN project. In [31] Meierhofer evaluates the DLC performance to determine optimal values for the window length, cell lifetime, etc.

2.3 CDMA MAC Protocols

None of the WATM protocols described above used CDMA as their multiple access technique. However, as discussed in the previous chapter, CDMA has become firm favourite for next generation air interfaces. CDMA technology presents new challenges to multimedia MAC protocols as they not only have to efficiently share resources, but also guarantee the stability of the system by controlling the multi-user interference level during peak bursty periods. In addition, channel users with different types of traffic may be affected by fluctuating interference levels and, hence, fluctuating channel qualities. The following section describes some MAC protocols designed to use CDMA physical layers.

2.3.1 Multidimensional PRMA with Prioritised Bayesian Broadcast

Multidimensional, packet reservation, multiple access with prioritised Bayesian broadcast (MD PRMA BB) is a proposed MAC strategy for the uplink channel of the UMTS terrestrial radio access (UTRA). Brand and Hamid Aghvami contributed to research efforts surrounding improvements to the basic packet reservation multiple access (PRMA) protocol by designing a joint CDMA/PRMA protocol suitable for hybrid CDMA/TDMA air interfaces [32].

In conventional PRMA fixed length timeslots are grouped into MAC frames. These slots are either contention slots (*C* slots) or reserved for a particular MTs information transfer (*I* slots) as indicated by the BS. When an MT has a packet spurt to transmit, it switches from an idle state to a contention state. When in the contention state the MT contends for a *C* slot according to some permission probability p_x. This contention process is repeated on a frame-by-frame basis. Once the first packet of the spurt is successfully transmitted in a *C* slot, the BS responds with an acknowledgement that implies a reservation of the same slot, converting it to an *I* slot for the remainder of the spurt. The MT now switches to a reservation state and has uncontested access to the channel to complete the transmission of its packet spurt. This is referred to as implicit resource allocation. While a MT is in the contention state, delay sensitive packets are dropped when exceeding some delay threshold, and the contention process is repeated with the next packet in the spurt. The state diagram for MTs is depicted in Figure 2.9. Note that it is possible for an MT to return to the idle state from the contention state without passing through the reservation state. This only happens in exceptional cases when an entire delay sensitive packet spurt is dropped before a successful transmission in a *C* slot.

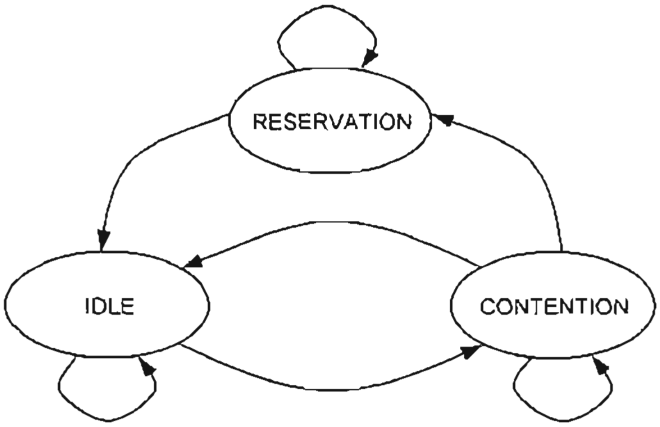


Figure 2.9 PRMA MT state diagram.

MD PRMA extends this by defining timeslots not only in the time domain, but also in an additional dimension e.g. the frequency domain, or in the case considered here, in the code domain. As with PRMA, the channel parameters are adapted to the bit rate of the standard service such that during a packet spurt with this service one packet per frame needs to be transmitted. The fundamental extension required to PRMA in MD PRMA is that an MT in contention mode selects any one of multiple *C* slots in the same timeslot with equal probability.

The transmission permission probability p_x with which an MT in contention mode may access a *C* slot is service class and timeslot dependant. It was found in [33] that the best delay/throughput performances are achieved by methods that use deferred first transmission (whereby a new packet spurt is immediately considered to be backlogged) and that estimate the number of backlogged terminals (terminals with a packet to transmit but no reservation) currently in the system to calculate p_x. MD PRMA BB uses a pseudo-Bayesian broadcast method to calculate p_x. These probabilities are also related to a load-based access control to ensure control of the CDMA multi-user interference level and, hence, maintain the stability of the system. The probabilities for each service-class and each timeslot of the next uplink frame are broadcast by the BS in the downlink frame. This broadcast will also include acknowledgements for successful contention transmissions in the previous frame.

In PRMA a MT in reservation mode implicitly maintains control of its timeslot until it has completed transmission of its packet spurt, however, for certain services such as data services, MD PRMA BB places an upper limit on the number of frames for which a MT may reserve a timeslot.

So, by controlling contention access probabilities (p_x) and allocation (number of frames), the protocol is able to track delay requirements and dropping probabilities for different services. One of the main shortcomings of MD PRMA BB is its inability to support high bit rate services requiring multiple slot allocations in the same frame. Another drawback is that it allows different services with different BER requirements to share the same timeslot so the capacity of the slots is limited by the most demanding service (with respect to BER).

2.3.2 Majoor's WATM Over CDMA

In [34] Majoor and Takawira propose a MAC protocol for ATM over CDMA channels. The protocol supports CBR, VBR and ABR traffic. Time is divided into MAC frames consisting of a fixed number of slots. Each slot may contain one ATM cell plus additional overhead. An uplink frame is further divided into four sections, as depicted in Figure 2.10. The first section contains two mini-slots for CBR and VBR reservations. A mini-slot is half the length of a full slot. The following three sections are reserved for CBR, VBR and ABR traffic respectively, with a moveable boundary between the CBR and VBR sections.

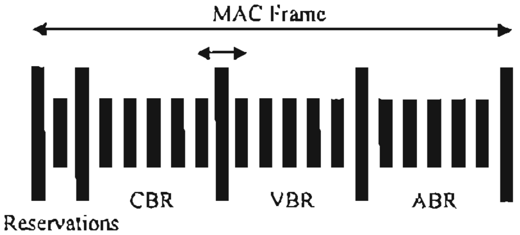


Figure 2.10 Majoor's uplink frame structure.

A MT with cells to transmit selects a spreading code from a finite pool at random and transmits a reservation packet in a reservation mini-slot. A connection admission algorithm is performed at the BS to determine whether the connection may be accommodated in the relevant data section. Mobiles with successful reservation requests are informed via the downlink and their allocated bandwidth is guaranteed until the end of the call. MTs with ABR traffic do not make reservations, but transmit their data in the ABR section in contention with other users in a slotted spread ALOHA fashion. At the end of each frame the BS broadcasts the slot allocations for all MTs. The codes to be used are also broadcast so no code clashes occur in the CBR and VBR sections. MTs transmitting at different data rates are accommodated by a combination of assigning different amounts of slots to users as well as using multi-code CDMA; i.e. one MT may transmit several packets in the same timeslot by using different CDMA codes.

Obviously VBR traffic may alter its data rate from frame to frame, thus the number of slots required changes accordingly. A piggyback concept is used to inform the BS of how many cells need to be transmitted in the following frame. However, there is a limit to how many cells a VBR source may transmit in the VBR section of any one frame. The excess cells have to be transmitted in the ABR data section. When a MT with CBR or VBR traffic is unsuccessful in a reservation request, the MT remains in a contention state until the termination of the call or a reservation is achieved. In this state the MT transmits in the ABR frame section along with regular ABR traffic.

The protocol offers different service qualities via different BER classes. This is achieved by assigning higher powers to users with more stringent BER requirements; i.e. the traditional CDMA power control technique is extended so that all users with the same BER requirement are received at the same power level. The technique used to determine the maximum number of MTs allowed to transmit simultaneously in each BER class while still maintaining suitable signal to noise ratios (SNRs) is explained in [34]. It is up to the connection admission algorithm to ensure that SNRs better than the minimum are maintained. This novel power control technique accommodates stochastic traffic variations to guarantee BERs to users.

2.3.3 WISPER

Wireless multimedia access control protocol with BER scheduling [35], or WISPER, is another protocol designed to take advantage of CDMA to maintain the BER of a user's transmission channel below a given specification, but using a different technique. WISPER may be classified as a slotted, demand based assignment protocol using frequency division duplexing (FDD) to separate the uplink and downlink. Figure 2.11 shows how time is divided into frames for both the uplink and downlink, with the length of a frame corresponding to the packet arrival rate of the most abundant traffic class (usually voice). For the uplink each frame is divided into packet slots and one request slot. A packet slot can carry any traffic class while the request slot may be used by new MTs to place admission requests to be admitted to the wireless network, or by registered MTs to place transmission requests.

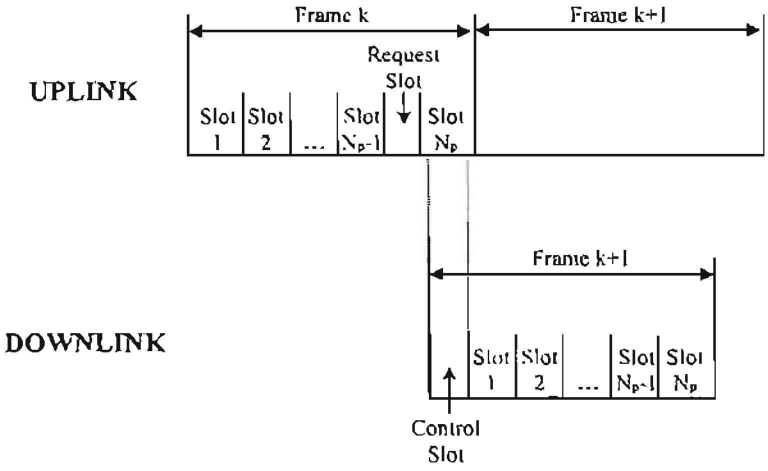


Figure 2.11 WISPER uplink and downlink channels: timing diagram.

A new MT admitted to the system is assigned a unique primary pseudo-noise (PN) code C^{PN}. In order to transmit at rates higher than the basic rate, a MT derives different spreading codes from the primary PN code by

$$C^{(i)} = C^{PN} \times D_i, \qquad D_i \perp D_j, \qquad i \neq j. \qquad \textbf{Eq. 2-1}$$

where D_i are from a set of orthogonal codes such as Walsh codes. So, using its own unique set of orthogonal codes, a MT may transmit more than one packet in a timeslot to achieve transmission rates above the basic rate. Obviously this also allows different MTs to support different transmission rates by supporting different numbers of codes.

Whenever a MT has new packets ready for transmission it sends a transmission request to the BS indicating the number of packets in the batch and their corresponding timeout value. The transmission request is either piggybacked onto another packet being transmitted, or sent via the uplink request slot using the MTs primary PN code. The BS receiving the request keeps track of the associated batch in a data structure containing information such as the number of packets in the batch, the packets' timeout value and the MT that owns the batch. This information is kept until the packets have been successfully received, or they timeout and are dropped.

Downlink frames are of the same length as uplink frames and contain the same number of packet slots as well as a control slot that is used by the BS to accept admission requests, and to provide transmission instructions to MTs that have requested permission to transmit packets. When responding to reservation requests the BS specifies the slot(s) and the corresponding number of packets that may be transmitted in those slots in the next frame. The BS transmits this message using the MTs primary PN code; hence no MT identification number is needed.

WISPER designates each slot to support a certain BER on a frame-by-frame basis. To use the bandwidth efficiently, packets that have equal or similar maximum BER specifications are transmitted in the same slot. In other words, the packet with the most stringent BER requirement determines the maximum number of packets that may be transmitted simultaneously in that timeslot. Figure 2.12 shows an example slot assignment containing multiple traffic classes. If, in any given frame, there are more outstanding packets than those that can be transmitted, the packet transmission order is determined by a novel scheduler. The scheduler prioritises packet batches according to their timeout values and the number of packets left in the batch.

		VOICE				
		VOICE				
		VOICE				
		VOICE				
		VOICE	VIDEO			
		VOICE	VIDEO		AUDIO	
	VOICE	VOICE	VIDEO	AUDIO	AUDIO	
	DATA	VOICE	VOICE	DATA	AUDIO	DATA
Slot Number:	1	2	3	4	5	6
Supported BER:	10^{-9}	10^{-3}	10^{-5}	10^{-9}	10^{-6}	10^{-9}

Figure 2.12 An example WISPER slot assignment.

WISPER is a flexible, reservation-based protocol that maximizes the throughput as well as minimises packet loss. Its BER scheduling scheme is simple to implement in that only one power level is required for all traffic classes as opposed to the several power levels required in a regular slotted-CDMA protocol using the conventional power control scheme with different power levels for each traffic type.

2.3.4 Wideband CDMA MAC Protocol for Real-Time and Non-Real-Time Services

In [36], Roobol et al., propose a WCDMA MAC protocol for real-time and non-real-time services to be used on top of the WCDMA physical layer adopted by the Japanese standardization body, ARIB, and their European equivalent, ETSI. The WCDMA physical layer is capable of multiplexing data streams from different services on a single MT. The different BER requirements of the various data streams are met at the physical layer by applying different types of FEC coding schemes. For example, services requiring very low BERs may use a convolutional code and a Reed Solomon code, whereas services with more relaxed BER requirements may only use a convolutional code. It is also possible to use no FEC at all. Furthermore, it is possible for MTs to transmit with variable bit rates by changing the spreading factor of the spreading code. This may be done if the MT has a dedicated channel (DCH), each having a dedicated code, at its disposal. Each DCH also has an associated physical control channel (PCCH) used for power control and rate information. It is possible to change the data rate every 10ms. When a DCH is set up its transmission rate, error code, interleaving depth and repetition/puncturing scheme (for rate matching) is specified as a transmission format (TF).

There are two basic methods to transmit packet data. Firstly, short packets may be transmitted in an ALOHA fashion using the random access channel (RACH), a channel common to all MTs in the cell. For longer packets a MT will use the RACH to request a DCH. After evaluating if the required resources are available the network will reply with a set of possible TFs. If the load is low the network may indicate the specific TF and the time the user can start transmitting. For heavier loads the MT has to send a capacity request before the BS replies with a scheduling message containing the specific TF and time at which the MT may initiate transmission. While a MT is transmitting the network also has the ability to increase/decrease the DCH's transmission rate according to the network load. Once the transmission is complete the MT will enter a holding state, maintaining the DCH for a certain time. If a new packet is generated during that time, the MT may transmit it immediately with the prescribed TF on the DCH. If the new packet is very large, however, the MT will first have to transmit a capacity request.

Real-time services have very similar allocation procedures. When a MT has data to transmit it issues a reservation request on the RACH. The BS responds with a set of TFs, and the MT may immediately begin transmission using any of the TFs in the set. This degree of freedom allows for variable rate transmissions, however, if the network load becomes too high the network may send a resource limit message reducing the allowed TF set to a subset of itself. The MT uses the PCCH associated with the DCH to indicate to the BS which TF it is using.

The MAC should also support multiple services transmitted from the same MT. One method to achieve this is to transmit each service with a different code. However, the physical layer is capable of multiplexing bit streams originating from different services, which leads to a less complicated implementation. It is up to the MAC layer to control this multiplexing process. For example, if a MT has been assigned TFs for a real-time stream as well as a packet data stream it may find that transmitting at the maximum bit rate on each service leads to an aggregate output bit rate that is too high for the DCH. Therefore, the MAC could implement a threshold on the output bit rate with the TF of the one service dependant on the other service's TF such that the aggregate output bit rate never exceeds the threshold.

This WCDMA MAC protocol deals with a very complex and flexible physical layer. It is essentially a demand assignment protocol relying on complex resource management. Extensive use of ALOHA type access in the RACH for short packet transmissions and request messages could severely degrade the performance of the protocol, not only because of excessive packet collisions, but also because of the increase in the interference level caused by numerous non-fast-power-controlled packet transmissions.

2.4 A Summary of the Fundamental Design Choices

Akyildiz et al., in their overview of MAC protocols supporting multimedia traffic [18], summarised the important questions to be answered when exploring the ability of MAC protocols as follows:

- What channel access method is being used?
- What are the slot/code designations within each frame? Are frame lengths variable or fixed?
- How are resource assignment decisions made?
- How are different traffic types effectively integrated and QoS constraints managed?
- For terminals contending for the same resources, how are unsuccessful transmissions resolved?

These questions, along with some other fundamental design choices, are discussed in the following.

2.4.1 The Multiple Access Scheme

Comparatively little research has been done on WATM over CDMA protocols as opposed to WATM over TDMA. One of the attractive features provided by CDMA is a graceful degradation of the wireless channel with increasing code usage. As the number of active mobiles increases more codes are required, leading to a graceful decrease in the quality of the wireless link. Thus, there is a trade off between the number of active codes and the quality of the radio channel. This may be exploited to provide varying qualities of service. Throughput may be increased for delay sensitive traffic by utilising more codes, while for loss sensitive traffic fewer codes may be used to improve the channel quality.

Many existing WATM projects have avoided CDMA because it was felt that the spreading bandwidth is impracticably large. These projects were designed around indoor environments with small BS cell sizes (<100m). With larger cell sizes, as in this project, more mobiles need to be serviced by a single base station. FDMA would require more frequency bands to achieve this, and hence a large overall bandwidth. TDMA would require an increased bit rate – which is difficult to provide over the longer distances. One also needs to remember that the same frequency band may be re-used by adjacent cells if CDMA is used, whereas a TDMA system requires the colouring of adjacent cells. The wide frequency band used by CDMA seems warranted when one considers the increased number of mobiles that may be supported by a single BS.

TDMA based protocols for WATM have been well researched. The Magic WAND project and WATMnet, among others, are excellent example of the extensive work performed in this area. This thesis is concerned with the implementation of a WATM protocol that may be described as a hybrid TDMA/CDMA protocol, also referred to as slotted-CDMA.

2.4.2 Separating the Uplink and Downlink

There are two basic techniques to separate the uplink and downlink traffic: frequency division duplexing (FDD) and time division duplexing (TDD). One of the major design goals in a WATM system is to minimise the transmission overhead per cell. In a TDD system the radio turn around time, between transmission and reception, is one of the limiting performance factors, hence it must be minimised. The turn around time is dependent on the design of the physical layer but is to some extent independent of the bit-rate. ORL's 1996 RATM achieved a turn around time in the order of 40 bit times, or approximately 1/10th of an ATM cell [25]. The throughput lost by the turn around time is a disadvantage of TDD as FDD has the advantage of not requiring a turn around.

However, for WATM implementations there is a strong argument in favour of TDD. The reason being that by utilising a dynamic TDD protocol the bandwidth allocated to the uplink traffic and that allocated to the downlink traffic may be dynamically adjusted as required. If there is no traffic waiting for the downlink then the excess bandwidth that was allocated to the downlink may be reallocated to some traffic waiting for uplink transmission. TDMA in particular is suited to TDD. The Magic WAND project demonstrates this with a portion of its frame dynamically allocated, on a frame-by-frame basis, to downlink traffic and another portion allocated to the uplink. This leads to very efficient bandwidth usage when the traffic profile is asymmetric.

However, when using dynamic TDD with CDMA a problem is encountered whereby the boundary between the uplink slots and the downlink slots cannot be moved as freely. If the boundary for one CDMA code channel is not the same as the boundary of another CDMA code channel, then a situation is encountered whereby transmission is required using one code at the same time as reception is required using another code. In other words, the turn around from reception to transmission, and vice versa, needs to be synchronised on all CDMA code channels. So, while some flexibility may be achievable, it is more limited than for the TDMA case.

For this reason one may argue that there isn't a significant advantage to using a TDD scheme over a FDD scheme for a WATM over CDMA implementation. Although the flexibility of trading bandwidth between uplink and downlink is lost, any spare bandwidth in a channel may be allocated to temporarily increasing error correction abilities, and/or temporarily increasing the number of contention slots – as appropriate.

2.4.3 The Channel Access Method

As previously mentioned, the three basic channel access methods are random access, fixed assignment, and demand-based assignment. The channel access method can seriously impact the delays experienced by user packets. For example, many of the protocols described above use a demand-based slot assignment scheme along with a random access scheme for reservation requests. Collisions that occur during the contention-based reservations will need to be resolved in a quick and efficient manner in order to achieve efficient utilisation of the available bandwidth. The use of mini-slots and piggybacking techniques can be useful to minimise the resources consumed by the reservation process.

Another option is to combine demand assignment with a fixed assignment scheme for reservation requests. Figueira and Pasquale propose such a technique in [37]. In their remote queuing multiple access (RQMA) protocol, real-time connections are assigned a backlog slot in which to transmit reservation requests. Thus, in every frame a real-time connection has a fixed mini-slot in which to update the BS. Best-effort connections have to contend for reservation slots or piggyback the request in the uplink transmission of one of its best-effort packets. But, there are some drawbacks with using a fixed assignment scheme for reservations. Since the number of real-time backlog slots is fixed, the maximum number of real-time sessions supported by a BS is also fixed. Also, when not all the backlog slots are being used the bandwidth allocated to them is wasted.

Most protocols take advantage of the BS's position as a gateway between the MTs and the network by utilising a centralised access technique. However, a BS's slot assignment process must implement a QoS based policy that incorporates information relating to BER, delay and throughput requirements. Such algorithms must, at the same time, be balanced against the corresponding complexity to reduce costs and implementation time.

2.4.4 Variable Frame Length or Fixed Frame Length?

The advantage of having a variable frame length, as used in RATM and WAND, falls away to some extent when using FDD. The advantage is in many ways linked to the advantage of using TDD. In adhering to a call's traffic contract (that specifies a cell transfer delay, peak cell rate, etc) the variable boundary between uplink and downlink frame sections, as well as the variable frame size can play a big role in providing the flexibility required to meet the ATM traffic contract's requirements.

When a scheduling algorithm constructs a frame in a TDMA/TDD scheme it may encounter a situation where it has to transmit a downlink cell near the end of the uplink frame section in order to meet the traffic contract. The cells at the end of the uplink section will still adhere to their traffic contract if they're transmitted in the next frame, so the scheduler may end the frame there and begin the next frame, and downlink section, in time to transmit the cell in question. This is a simplistic example of how a variable frame length is useful in TDD systems – the frame length is determined after the slot map is ready. It does, however, add to the complexity of the scheduler, as well as making it more difficult to detect the beginning of each frame when a MT is trying to synchronise itself with a BS.

Using FDD reduces the problem of adhering to a connection's traffic contract. In a FDD scheme there won't be any situations where an uplink cell can't be transmitted at the required time because that time is during the frame's downlink section. This is because with the FDD scheme cells may be transmitted on the uplink frequency at the same time as being received on the downlink frequency. But, there may still be an advantage to using a variable frame length with FDD. This was outlined by the RDRN project: a longer frame length results in a smaller overhead due to frame headers (as they're transmitted less often), but greater cell losses in the event of receive errors as error checking is applied on a per frame basis. So, if the channel conditions deteriorate the frame length may be dynamically adjusted to reduce cell losses at the expense of increased frame header overhead. If, however, the error checking is applied on a per cell basis instead of the per frame basis used in RDRN, then the increased cell loss with increased frame length falls away at the expense of slightly more overhead per cell.

A shorter frame length leads to a more rapid recovery from lost cells with ARQ schemes coming into effect more quickly as well as shorter intervals between the scheduling of cells resulting in reduced scheduling delays. A short fixed frame length may also allow a MT to synchronise to a BS more quickly. Obviously a happy medium must be found between the frame header overhead and scheduling/error recovery delays.

2.4.5 The Base Station Cell – How big?

There are numerous factors one has to consider when selecting the BS cell size to support. Only allowing smaller cell radii may mean that the round trip delay between a BS and MT becomes negligible simplifying matters as far as synchronisation is concerned. Smaller cells also reduce problems of multi-path and attenuation leading to lower transmit powers and, hence, longer battery life for mobiles. Another advantage can be increased throughput with each BS serving fewer MTs therefore having more bandwidth for each.

The most obvious advantage to accommodating large cell radii is that the number of BSs required to cover a given area is decreased. This is obviously an important factor when large areas have to be covered, although the landscape characteristics will often limit the size of a cell even if the protocol doesn't. Another benefit of large cells is that hand-offs occur less frequently, which often decreases the likelihood of dropped calls.

2.5 Combining CDMA and WATM for Better QoS

What benefits are there in combining a CDMA physical layer with a WATM link layer when the ultimate goal is to provide QoS guarantees for multimedia traffic? Well, as previously mentioned, the 3G WCDMA physical layer incorporates the ability to vary the spreading factor and hence the bit rate of a CDMA code channel i.e. multi-rate CDMA. This may be carried out dynamically according to the quality of the wireless link and traffic requirements of the connection e.g. depending on the multi-path effects one may decrease the spreading factor, increasing the data rate, until some maximum BER level is achieved. Two further techniques have been suggested in the literature.

The first technique takes advantage of the power control techniques inherently needed in CDMA systems. Instead of requiring all MTs to minimise their transmission powers such that all the spread signals arrive at the BS with the same minimal but satisfactory power, one may set different power levels with higher priority traffic being power controlled to higher power levels than lower priority traffic. Essentially, different traffic classes are power controlled to be received with different signal to interference ratios according to their BER requirements. Details pertaining to such techniques may be found in [34], [38], and [39]. The drawback with such techniques is the increased complexity required in the physical layer implementation.

Akyildiz, Levine and Joe's WISPER protocol [35] suggested an alternate method to take advantage of CDMA in providing different service levels. The basic concept arises from the so called soft capacity offered by CDMA whereby each CDMA code channel sees the other spread signals as noise, so the more CDMA code channels there are in use, the greater this multi-user/multi-code interference and the worse the bit error rate. Thus, by adjusting the number of codes in use one can, to some extent, control the BER of the remaining code channels. Akyildiz, Levine and Joe proposed a slotted CDMA protocol that takes advantage of this property to schedule packets according to the BER requirements. The number of codes used on each timeslot is limited by the BER requirements of the packets being transmitted in the timeslot.

Of course, multi-user interference is not the only source of bit errors; multi-path effects will provide some lower limit for the BER although spread spectrum techniques are more resilient to fading channels than narrowband techniques. In WISPER each VC's mapping between BER and multi-user interference is fixed for the duration of its existence. However, the possibility remains to dynamically adjust this interference to compensate for a fading channel as required. This would, however, further complicate the entire CAC and QoS guarantee issue. A fast power control algorithm, as is typical of DS-CDMA systems, does to some extent negate the need to even vary the relationship between BER requirement and multi-user / multi-code interference for each MT.

Chapter 3 The Proposed WATM MAC Protocol

This chapter concentrates on the proposed WATM MAC protocol, dealing with design choices and describing how the protocol functions. The focus is on the manner in which the MAC layer controls access to the available bandwidth and, in particular, how the scheduling algorithm dishes out bandwidth to ensure QoS guarantees are met.

3.1 *An Overview of the MAC Protocol*

The implemented MAC protocol is designed to function within a typical wireless cellular network environment as portrayed in Figure 3.1. It uses a slotted, direct sequence (DS) CDMA system with the uplink and downlink separated by frequency division duplexing (FDD) allowing transmissions to occur simultaneously in both directions.

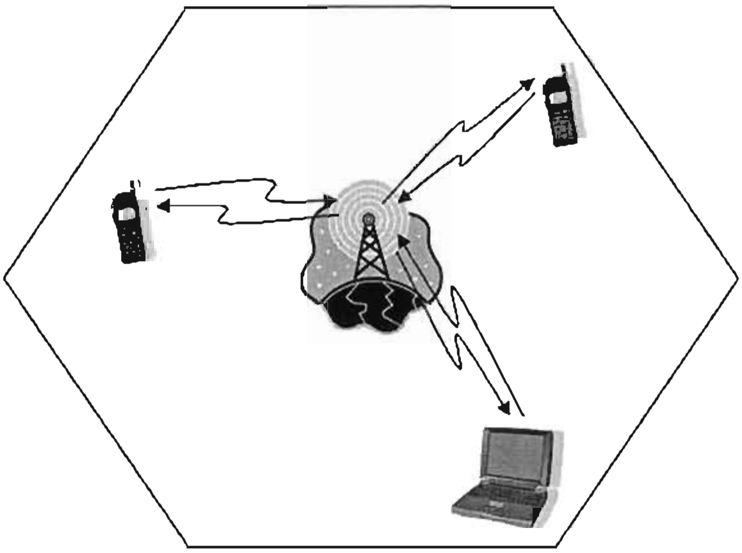


Figure 3.1 MT – BS cellular architecture.

3.1.1 CDMA Code Assignments

There are two possibilities for the way in which CDMA codes are assigned and used. The first approach is to have a fixed set of codes available for use, with each MT and BS required to support transmission/reception on all codes simultaneously. A more flexible approach would be to assign a primary pseudo noise (PN) code to each MT. A MT supporting simultaneous transmission on *n* codes can derive these codes by multiplying its primary PN code by *n* different codes from a fixed set of orthogonal codes (e.g. Walsh codes). This is the method used in the WISPER protocol [35] and allows different MTs to support different transmission rates by being able to transmit up to *n* packets simultaneously on different codes.

To simplify matters, not only in simulations of the protocol but also in the implementation carried out on a software radio architecture developed by Alcatel Altech Telecoms, the protocol presented here assumes the first technique of requiring each MT to support a fixed set of codes. Assigning unique PN codes to each MT requires longer spreading codes to ensure enough codes are available for allocation. It also increases the complexity of the physical layer with higher spreading factors or spreading codes that span more than one data bit required. BSs and MTs will also need to be able to dynamically change the PN codes in use. The baseband signal processing implementation is carried out in another project. This implementation is based in the software radio's FPGAs (field programmable gate arrays). So, to simplify the FPGA baseband processing and to ensure that the software radio has the resources required for the BS implementation, the physical layer supports simultaneous transmission and reception on a fixed set of codes rather than dynamic codes assignments.

3.1.2 Frame Structure

Each CDMA code channel is slotted and structured into fixed length frames. This hybrid TDMA/CDMA approach, also termed slotted-CDMA or TD-CDMA, results in each frame effectively containing a matrix of slots that may be allocated to ATM VCs as required. The MAC layer makes use of a demand assignment, resource sharing technique to allocate the bandwidth. The BS runs a scheduling algorithm to perform these slot allocations but, obviously, the BS needs to be able to inform each MT in the cell of its slot allocations, and the MTs need to inform the BS of the ATM cells it has waiting for transmission in the uplink. These issues are explained next.

Figure 3.2 shows an example of an uplink and a downlink frame, each containing six timeslots with a set of six CDMA codes available. A frame header (FH) always occupies the first slot in the downlink frame, while each of the remaining slots may contain a WATM cell, or some specialised cell containing sequence number acknowledgements or MT association acceptance, or nothing. (The details of the various cell types are presented in the following sub-sections.) For each CDMA code, the frame header describes the structure of the remainder of the downlink frame as well as that of the corresponding uplink frame. So, MTs are informed about the schedulers slot allocations via the frame headers.

The uplink frame is offset from the downlink frame by the equivalent of two timeslots. This is to give each MT time to receive the frame header and interpret it before having to transmit its uplink frame as specified by the frame header. The last slot in each uplink frame is always a contention slot (CS) that is divided into two mini-slots in which MTs can transmit VC update mini-cells containing slot reservation requests and cell sequence number acknowledgements, or MT association request cells to allow the MT to register with the BS. Collisions may occur in the contention slot if two or more MTs select the same CDMA code and timeslot in which to transmit. It should be pointed out, though, that if the system is enhanced to allocate unique code sets to each MT then direct collisions would not occur in the contention slot. Instead, the multi-user/multi-code interference would just increase/decrease according to the number of MTs transmitting in the contention slot. This random access process for reservation requests and cell acknowledgements is only used when they can't be piggybacked onto other uplink cells already being transmitted.

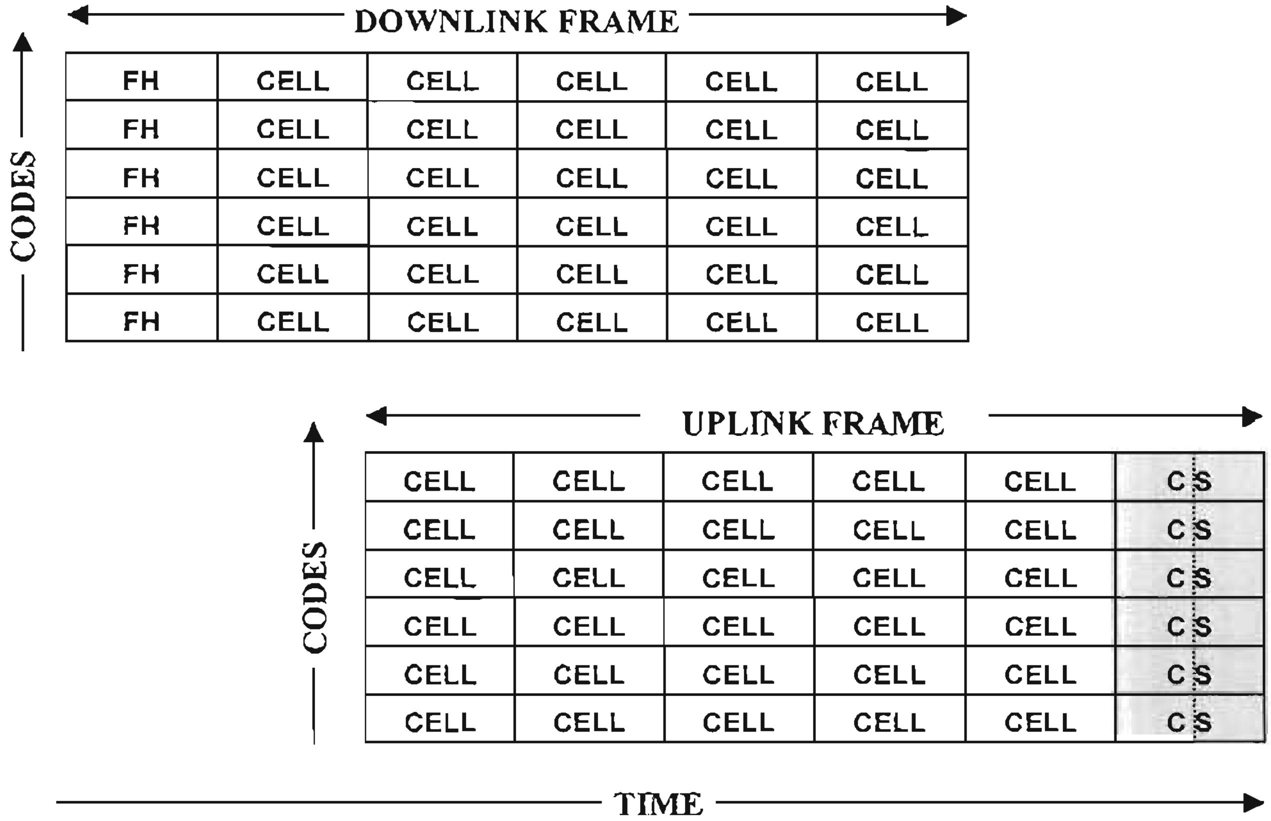


Figure 3.2 Uplink and downlink frame structures.

It is advantageous to have as many slots in a frame as possible in order to reduce the overhead caused by the frame header. However, the number of slots must be chosen such that the maximum cell delay requirements of the VCs can be met i.e. longer frames result in potentially longer scheduling delays experienced by cells. One also needs to take into consideration that some cells may need to be scheduled more than once if retransmissions are required. The length of a frame, in terms of the time it takes to transmit, will obviously depend on the bit rate over each CDMA code channel. Another factor that may limit the number of slots in a frame is the size of the frame header i.e. a frame header can only contain a limited number of slot allocations to describe its frame structure. Including some FEC information in the frame header is also desirable, as incorrectly received frame headers will result in MTs missing slots that are allocated to them. Often the frame length is chosen to coincide with the cell inter-arrival time of the most common traffic type (which is normally voice traffic). This is not necessarily the traffic type that consumes the most bandwidth, but rather that with the most connections, as the idea is to reduce the reservation request traffic in the contention slot. If most connections require one slot per frame then the reservation request for the next frame may easily be piggybacked onto the cell being transmitted in the current frame, freeing up the contention slot for connections without uplink cells on which to piggyback.

3.1.3 The WATM Cell

Every 53-byte ATM cell is converted into a 56-byte WATM cell before being transmitted over the wireless interface (Figure 3.3). This involves compressing the 28-bit GFC/VPI/VCI fields of the ATM UNI cell into a 16-bit WVCI (wireless VCI) in order to minimise the WATM header size – a procedure employed by many of the WATM protocols previously discussed. This 16-bit WVCI is only valid within the BS cell in which it is assigned, and means that each BS can theoretically support around 65000 connections. The 8-bit HEC field of the ATM cell is removed and instead a 16-bit CRC is appended to the end of the WATM cell. The CRC field enables error detection on the entire cell instead of just the cell header, as is the case with an ATM cell's HEC field.

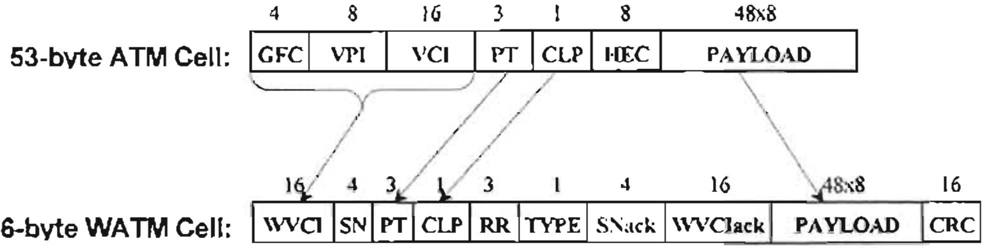


Figure 3.3 ATM and WATM cell structures.

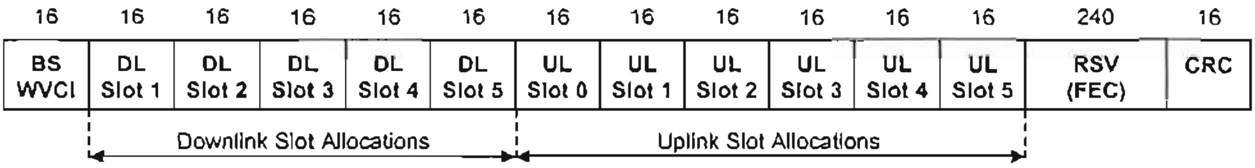
The cell payload remains unchanged, but some extra fields for cell sequencing (SN), slot reservation requests (RR) and acknowledgements (SNack) are included in the cell header. The manner in which these fields are utilised is as follows. The SN or sequence number field is used to maintain the sequence of the cells in each virtual channel. This is important as the ATM standards require that the underlying layers do not re-order the ATM cells. The cell sequence numbers are also used by the DLC to implement a retransmission technique (see Chapter 4).

The 1-bit TYPE field is used to indicate whether the SNack field is indeed a sequence number acknowledgement or if it is in fact another reservation request for the connection indicated in WVCIack. By using WVCIack to indicate the VC to which the acknowledgement or reservation applies, a MT with multiple connections may use cells transmitted from one connection to update the BS on the status of another connection without having to transmit in the contention slot.

In the uplink direction the WATM cell contains a 3-bit field, RR, in which MT uplink slot reservation requests are inserted. The 3-bits simply contain a binary number between 0 and 7 indicating the number of cells the VC has queued at the MT. Obviously the BS doesn't need to send slot reservation requests to MTs, so in the downlink these 3-bits may be used for something else. For example, they may be used as power control bits in which the BS indicates to the MT to either increase or decrease its transmit power (see Appendix A.2.4).

3.1.4 The Downlink Frame Header

The downlink frame header (FH) cell describes the contents of the following downlink frame as well as that of the corresponding uplink. The BS uses FH cells to inform the MTs in its cell of the slot allocations as allocated by the scheduler on a frame-by-frame basis. Figure 3.4 shows the format of a FH for a frame containing six timeslots. The first field is a WVCI field, as with ordinary WATM cells. In fact, every cell format begins with a WVCI field with certain identifiers reserved for certain cell formats. This enables a MT or BS to determine the format of a cell from the value of its WVCI field. Appendix A.2.1 contains the reserved WVCI values. For FH cells the value of the WVCI field must lie between 0010h and 001Fh, providing a simple check for such cells. Adjacent BSs will transmit different values in this WVCI field so that it may also be used by MTs to distinguish between FHs from different BSs. Thus, the FH's WVCI effectively doubles as a BS identifier.



16	16	16	16	16	16	16	16	16	16	16	16	240	16
BS WVCI	DL Slot 1	DL Slot 2	DL Slot 3	DL Slot 4	DL Slot 5	UL Slot 0	UL Slot 1	UL Slot 2	UL Slot 3	UL Slot 4	UL Slot 5	RSV (FEC)	CRC

Figure 3.4 Structure of the downlink frame header.

Although there are six slots per frame for the example FH, only five downlink slot allocations may be specified in the FH. This is because the first downlink slot in a frame is always a FH by default. The five downlink slot allocations are assigned to ATM connections by inserting the VC's WVCI into the corresponding FH allocation slot. This will enable an MT to only receive cells meant for it, and the receiver may potentially slip into a micro-powerdown mode while nothing is being sent to it, allowing an increase in MT battery life. As already mentioned, certain WVCI values are reserved for specialised cells or purposes. For example, a DL slot assignment containing 0000h is used to indicate an empty slot. Refer to Appendix A.2.1 for the other WVCI reservations.

For the uplink slot assignments all six slots in the frame are allocated in the same manner as the downlink slots. Even though the last slot is always reserved for a contention slot, it is still specified as such in the FH. This leaves the scheduler with the flexibility to perhaps schedule this slot for another purpose. The scheduler may also specify other slots to be contention slots instead of empty slots if appropriate.

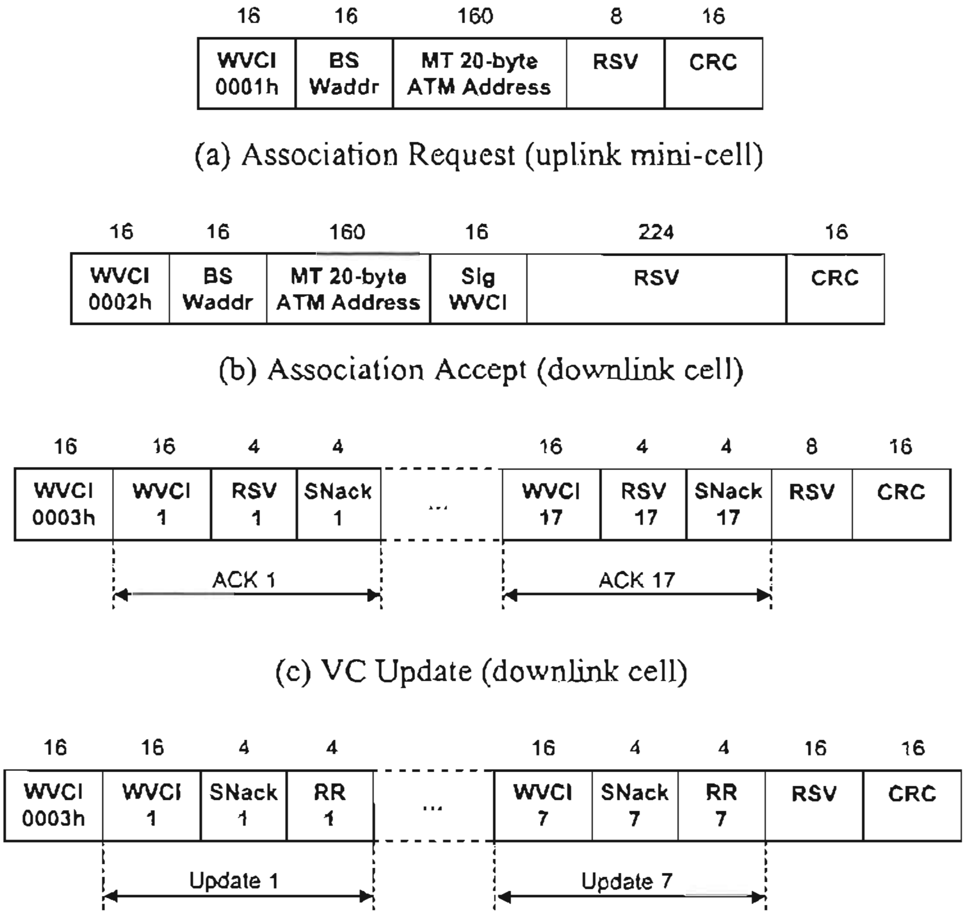
With the excess reserved (RSV) bits in the FH there is the potential to implement a FEC technique to decrease the likelihood of corrupt FHs. Although this option is there it has not been implemented at this stage. The 16-bit CRC is currently used to check for incorrect reception of FHs. When an MT fails to receive a FH correctly it may still receive the downlink cells being sent by the BS, but any uplink slots allocated to VC's belonging to the MT will be wasted. Note that if the frame length is increased to include more timeslots then some of the RSV bits shall be used to allocate these slots and, hence, there will be fewer bits for FEC.

3.1.5 Some Specialised Cells

There are a few specialised cells that are required in order to implement the DLC protocol, slot reservation requests and MT associations. In the downlink direction these cells are transmitted as broadcast cells so that all MTs receive them. In the uplink direction they are transmitted as mini-cells in the contention slots.

When a MT is first switched on it has to go through a number of states before it is ready to initiate new ATM connections. Firstly, in the physical layer, the MT synchronises to the CDMA code sequence and, hence to the MAC timeslots. This is achieved by monitoring a pilot signal that is transmitted by the BS. At this stage the MAC layer will be able to receive whatever the BS is transmitting in each slot. The MAC layer synchronises to the MAC frame as soon as a FH is received (identified by its WVCI). Next the MT has to associate itself with the BS. To do this, the MT randomly chooses a CDMA code from a fixed set of codes and monitors the FH on that CDMA code channel to determine when the contention slots occur. An association request mini-cell, depicted in Figure 3.5(a), is then transmitted in one of the uplink contention mini-slots.

As with all other cells the association request mini-cells starts with a WVCI field identifying it as such. Following this is a BS Waddr field, which identifies the BS with which the MT is registering. This is the BS identifier transmitted by the BS in its FHs. The MT's full 20-byte ATM address is also included. The BS requires this for the functioning of higher layers in procedures such as call set up and for mobility management. Finally a 16-bit CRC is appended for error detection. Like the WVCI field, all cells also have a 16-bit CRC appended.



(a) Association Request (uplink mini-cell)

(b) Association Accept (downlink cell)

(c) VC Update (downlink cell)

(d) VC Update (uplink mini-cell)

Figure 3.5 Special cell structures.

If the BS receives the association request correctly it will reply with an association accept cell [Figure 3.5(b)] in the downlink. All MTs in the BS's cell will receive the cell as it is broadcast. The association accept cell is identified by its reserved WVCI value of 0002h. Only MTs that are waiting for association will receive the cell. They can use the BS Waddr field to check that the correct BS sent it, and the 20-byte ATM address determines the MT for which the cell is intended. The association accept cell also contains some important information for the MT in the SigWVCI field. This is the MT's new signalling WVCI. Each MT is assigned a unique signalling WVCI to be used for the signalling channel between the BS and MT. WVCI values between 0020h and 03FFh are reserved for signalling WVCIs. This wireless signalling channel is used to set up connections, release connections, etc. A specialised wireless signalling protocol is used instead of implementing the ATM Forum's full UNI 4.0 specification. This allows a reduction in the number of signalling bytes transferred. For example, each MT's signalling WVCI also doubles as its WATM address, so the 20-byte ATM addresses are replaced with equivalent 2-byte addresses over the wireless interface. This customised signalling protocol is discussed in the next chapter. Once the association process is complete, the MT is ready to set up and receive calls.

To allow BSs to transmit cell acknowledgements (ACKs) that couldn't be piggybacked onto other WATM cells, there is a special VC update cell depicted in Figure 3.5(c). This cell is broadcast in the downlink by the BS and received by all MTs, which use its reserved WVCI value of 0003h to identify it. It may contain up to 17 ACKs with each ACK consisting of a WVCI and SNack pair. The WVCI indicates the connection for which the sequence number acknowledgement (SNack) is intended. So, with a single cell the BS can transmit 17 ACKs intended for different connections and different MTs. Notice that each ACK also includes a reserved field (RSV) that could perhaps be used to update MTs with some other information.

A similar cell is also transmitted by MTs in the uplink. This cell, shown in Figure 3.5(d), is also called a VC update cell, but it is a mini-cell because it is transmitted in an uplink contention slot. The uplink VC update mini-cell contains slot reservation requests as well as ACKs. It may contain up to 7 such updates. Each update consists of a WVCI, RR and SNack field. The WVCI and SNack fields serve the same purpose as for the downlink version of the cell, and the RR field, or reservation request, indicates the number of cells the MT has queued for the connection indicated in the WVCI. Unlike the downlink VC update cell, the uplink cell obviously requires all connections being updated to belong to the same MT. This justifies the smaller cell size for the uplink as opposed to the downlink.

3.1.6 Cell Lifetime (CLT)

VCs carrying real-time traffic generally have strict maximum cell delay requirements. For interactive applications these delay constraints are even stricter. Cells that are delayed by more than the specified requirement will be rendered useless at the receiver. In order to prevent this situation from occurring each VC has associated with it, a cell lifetime (CLT) that is the maximum time for which a cell may be queued before it is dropped. Although the cell is lost, there is no point in transmitting it if it won't be used at the receiving end. A VC's CLT also limits the size of the buffer required to queue cells waiting to be transmitted. Obviously there is a trade off between the likelihood of a cell being dropped and the average cell transfer delay experienced by the cells of a VC.

3.2 The Scheduler

3.2.1 Overview

The scheduler runs at the BS and has the important task of assigning slots to VCs such that their QoS requirements are met. The basic operation of the scheduler is exactly the same for the uplink and the downlink although the performance in the downlink is generally better. The main reason for this is that the BS knows exactly how many cells each VC has queued for transmission in the downlink, but for the uplink it has to rely on slot reservation requests received in contention slots or piggybacked onto other uplink cells. Naturally this means there is an extra delay in the scheduling of uplink cells while these reservation requests are transmitted to the BS. A number of factors, including the frame length and congestion in the contention slot, affect the length of this delay. The possibility of frame headers being received incorrectly by MTs also contributes to the difficulties in the uplink.

Remember that the frame structure of the slotted-CDMA wireless channel is such that each uplink and downlink frame has a matrix if slots to be allocated by the scheduler. This situation is depicted in Figure 3.6, where some scheduling scheme takes the cells queued for each VC and packs them into the slot matrix.

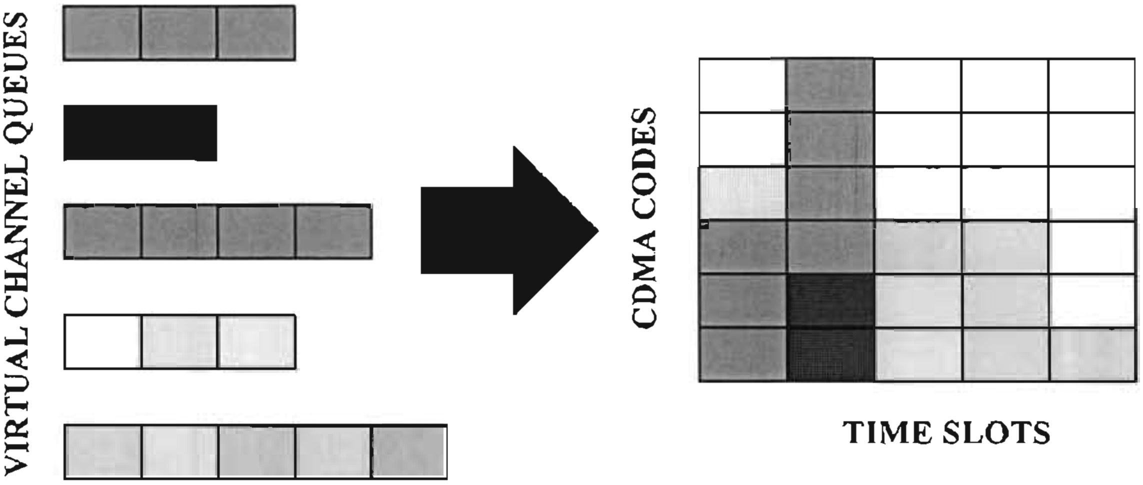


Figure 3.6 Scheduling Scheme.

When building a frame the scheduler goes through a number of iterations. Each iteration of the scheduler allocates a slot to a VC and can be divided into two steps. The first step involves choosing the next VC to be assigned a slot. VCs need to be chosen in an order that will result in their QoS requirements being met. The second step is to allocate a suitable slot in the frame to the VC. The slot must be chosen such that the BER requirements of the VC are adhered to.

3.2.2 Which VC gets the Next Slot?

This portion of the scheduler is based on the scheduling algorithm used in the ACTS Magic WAND project [30]. Every VC is assigned a token pool, the size of which is equivalent to the maximum burst size (MBS) specified for the VC. In other words, it is the maximum number of cells that may be transmitted, in one burst, at a rate above the mean cell rate (MCR) specified for the VC. This token pool resides at the BS. From the moment the VC is instantiated, tokens are added to its token pool at its specified MCR. One token is removed from the token pool for every slot allocated to the VC. Thus, at any instant, the number of tokens in the token pool gives an indication of the bandwidth consumed by the VC relative to the bandwidth guaranteed to it. Besides the token pool, each VC is also assigned a priority according to its traffic class where CBR traffic gets the highest priority, followed by real-time VBR, non real-time VBR, ABR, and finally UBR traffic (see Appendix A.2.2). When deciding which VC deserves the next slot, the scheduler picks the one that has, firstly, the highest priority and then the fullest token pool. Obviously the VC also has to have a cell waiting to be transmitted.

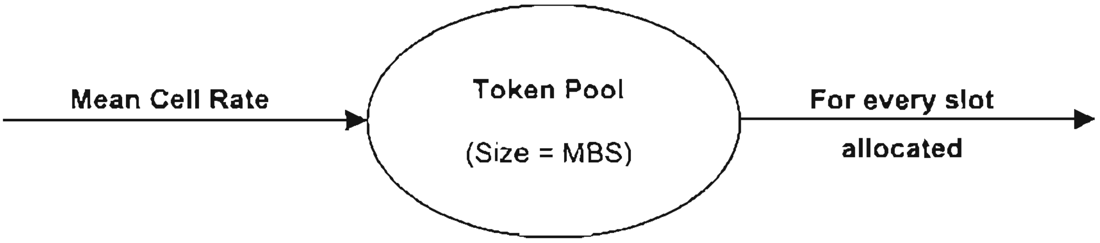


Figure 3.7 A VC's token pool.

Initially the scheduler was configured to choose the VC with the most tokens first, but it was discovered through simulation that scheduling according to the fullest token pool (number of tokens divided by size of token pool) tended to treat VCs of the same priority more fairly. This makes sense if one considers two VCs each transmitting at their specified MCR and each with their token pools half full.

3.2.3 Finding a Slot for the VC

Once the scheduler has chosen the next VC to be assigned a slot, it has to allocate the VC a suitable slot in the frame. This is done according to a maximum BER requirement specified for each VC and is based on the WISPER protocol proposed by Akyildiz, Levine and Joe in their paper entitled "A Slotted CDMA Protocol with BER Scheduling for Wireless Multimedia Networks" [35]. The basic idea behind the BER scheduling was explained in Chapter 2 and is repeated here for clarity. The idea is that the more cells there are transmitted (using different CDMA codes) in a timeslot, the greater the multi-user interference experienced by each cell and the higher their BERs. So, by adjusting the number of cells transmitted in a timeslot, the BER experienced by those cells may be varied. The scheduler takes advantage of this by mapping each VC's maximum BER requirement onto the maximum number of cells that may be transmitted in the same timeslot as a cell from the VC. The scheduler then searches for a suitable slot according to the following criteria (and in the following order):

1) An empty timeslot or a timeslot containing cells with the same BER requirements.

2) A timeslot containing cells with more stringent BER requirements.

3) A timeslot containing cells with less stringent BER requirements.

Note that for the third criteria, the timeslot containing cells with less stringent BER requirements must be able to be converted to the more stringent BER requirements of the VC being allocated a slot. In other words, the number of cells in the timeslot must be at least one less than the maximum number of cells allowed in a timeslot by the VC seeking a slot allocation. If none of the timeslots in the frame can accommodate the VC then it will have to wait for the next frame and the scheduler moves on to the next iteration. This scheduling procedure comes to an end once every timeslot contains its full compliment of cells, or the scheduler has iterated through all VCs with cells to transmit.

3.2.4 Filling the Frame

Once the scheduler has completed the procedure described above there may still be timeslots in the frame with the potential to hold more cells without breaking the BER requirements of any of the cells they already contain. For this reason, the scheduler may now assign slots to VCs that have cells awaiting transmission even if the VCs don't have any tokens. This prevents any slots from going to waste and helps to prevent QoS guarantees being broken, especially when a VC has to retransmit cells. When selecting VCs for this frame fill-up, the scheduler searches for suitable candidates according to the VC's traffic class priorities (Appendix A.2.2).

3.2.5 A Scheduling Example

To illustrate the workings of the scheduling algorithm the following example demonstrates how the scheduler allocates slots in the code and time domains to build up a frame. There are actually two instances of the scheduler running – one for the uplink and one for the downlink. This example only demonstrates one direction, but operation is the same for both directions. For the sake of this example it is assumed that only four CDMA codes are available and that a frame contains four timeslots. Figure 3.8 shows, step by step, how the available slots in the code and time domain are allocated to VCs by the scheduler. CBR1, VBR1, VBR2 and ABR1 are four VCs with cells to transmit. The current state of these VCs, after each step, is shown on the right hand side of the figure. Note that the *BER Limit* specified for each VC is the maximum number of cells per timeslot required to achieve the VC's BER requirements. On the left hand side of the figure the slot allocations are shown.

Initially all the slots are empty, as shown in Figure 3.8(a). The scheduler starts with the CBR VCs as they have the highest priority. In this case there is only CBR1, which has two tokens but only one cell. So the first available slot is assigned to CBR1 in Figure 3.8(b). Next the VBR virtual channels are allocated slots. VBR2 has the most tokens so it is assigned the first slot. The scheduler first looks for a timeslot containing cells with the same BER requirements or an empty timeslot, so the timeslot after the one assigned to CBR1 is chosen because CBR1 has a different BER requirement. Now both VBR1 and VBR2 have one token so they are both assigned a slot. Both have the same BER requirement of up to 4 cells in a timeslot, so they are both assigned slots in the same timeslot as seen in Figure 3.8(c). ABR1 is the next VC to be assigned slots. It has five cells waiting, but only four tokens. Its BER requirement only allows for up to two cells in a timeslot. Neither of the first two timeslots have this BER requirement, so the last two, which are currently empty, are used to assign 4 slots to ABR1. Figure 3.8(d) shows this allocation.

(a)

Code4				
Code3				
Code2				
Code1				
	Slot1	Slot2	Slot3	Slot4
BER Limit	4	4	4	4

	CBR1	VBR1	VBR2	ABR1
Priority	1	2	2	4
Tokens	2	1	2	4
Cells	1	3	2	5
BER Limit	3	4	4	2

(b)

Code4				
Code3				
Code2				
Code1	CBR1			
	Slot1	Slot2	Slot3	Slot4
BER Limit	3	4	4	4

	CBR1	VBR1	VBR2	ABR1
Priority	1	2	2	4
Tokens	1	1	2	4
Cells	0	3	2	5
BER Limit	3	4	4	2

(c)

Code4				
Code3		VBR2		
Code2		VBR1		
Code1	CBR1	VBR2		
	Slot1	Slot2	Slot3	Slot4
BER Limit	3	4	4	4

	CBR1	VBR1	VBR2	ABR1
Priority	1	2	2	4
Tokens	1	0	0	4
Cells	0	2	0	5
BER Limit	3	4	4	2

(d)

Code4				
Code3		VBR2		
Code2		VBR1	ABR1	ABR1
Code1	CBR1	VBR2	ABR1	ABR1
	Slot1	Slot2	Slot3	Slot4
BER Limit	3	4	2	2

	CBR1	VBR1	VBR2	ABR1
Priority	1	2	2	4
Tokens	1	0	0	0
Cells	0	2	0	1
BER Limit	3	4	4	2

(e)

Code4		VBR1		
Code3		VBR2		
Code2	VBR1	VBR1	ABR1	ABR1
Code1	CBR1	VBR2	ABR1	ABR1
	Slot1	Slot2	Slot3	Slot4
BER Limit	3	4	2	2

	CBR1	VBR1	VBR2	ABR1
Priority	1	2	2	4
Tokens	1	0	0	0
Cells	0	0	0	1
BER Limit	3	4	4	2

Figure 3.8 A scheduling example.

At this stage only CBR1 has any tokens available, but this VC doesn't have any cells to transmit. There are, however, still some timeslots (*Slot1* & *Slot2*) that can handle more cells without violating the BER requirements of any of the cells already in the slot. Also, VBR1 and ABR1 have more cells awaiting transmission. So, the scheduler performs a 'frame fill-up' procedure by allocating slots to these VCs even though they don't have tokens. The scheduler first looks for a slot for VBR1 as it has the highest priority. A slot in *Slot2* is allocated to VBR1 first as it already contains cells with the same BER requirements as VBR1. But, VBR1 still has another cell, so *Slot1* also takes a VBR1 cell without violating the BER requirements of the CBR1 cell already in the timeslot. Figure 3.8(e) shows these allocations. Finally, the scheduler looks for a slot for the last ABR1 cell. Unfortunately there are no timeslots left that will satisfy the 2 cells/slot BER requirement of ABR1, so this cell will have to wait for the next frame to be scheduled.

Chapter 4 The Proposed DLC Protocol

Now that the MAC layer has been defined the focus switches to the layer above it in the WATM protocol stack (Figure 2.1): the DLC layer. It is the responsibility of the DLC protocol to deal with transmission errors in the underlying layers and to ensure the sequence of the ATM cells is maintained before passing them on to the ATM layer. ATM was designed for the very low error rate environments achievable in modern wireline links, and hence does not provide strong error control with only the header of the ATM cell protected by a header error correction (HEC) field. Unlike the wireline links where errors are rare, usually single bit and randomly distributed, wireless links have high and variable error rates that often include burst errors where several consecutive bits are lost.

4.1 Error Control Schemes

In [40] Varshney discusses error control schemes that can be used in WATM networks. Figure 4.1 is based on a figure in [40] and summarises the various error control techniques available. Although not all the techniques would be applied at the DLC layer, the fundamental concept of each is briefly discussed.

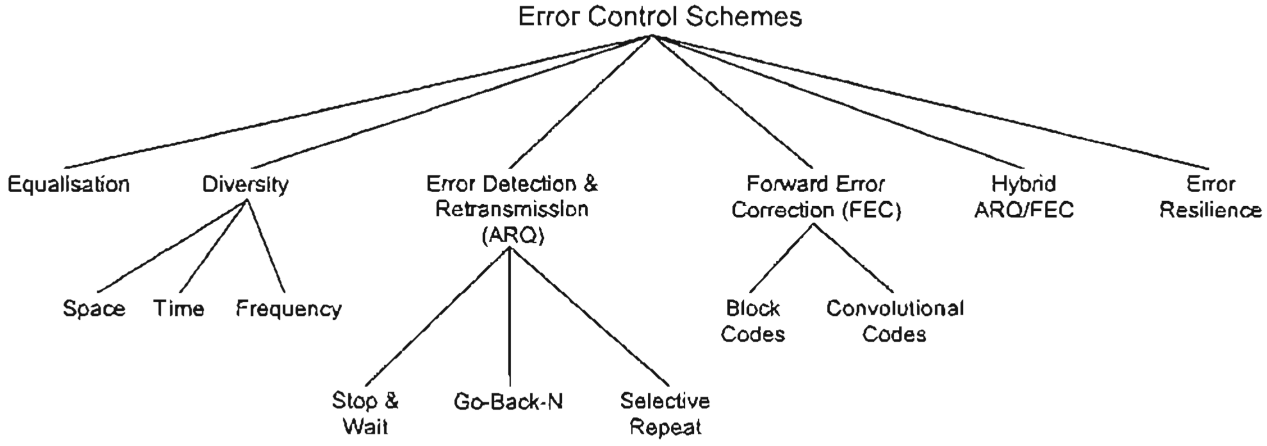


Figure 4.1 A classification of error control techniques.

Equalisation is used at the receiver to combat inter-symbol interference created by multi-path propagation of a signal. For narrowband systems, where there are a small number of strong multi-path signals arriving within a symbol period, channel equalisation may be used to correct inter-symbol interference. However, the wideband nature of CDMA systems means that the multi-paths may be more than one CDMA chip apart, so traditional equalisation is no longer suitable. Instead, a method that receives all the paths and combines them is required. A rake receiver synchronises to as many paths as possible by cross-correlating a reference pattern with the received signal. The receiver combines the signals from all these paths to produce one clear signal that is stronger than the individual components [41].

Diversity is a powerful communication technique to compensate for channel impairments. One can take advantage of time, spatial or frequency diversity. In fact, the rake receiver described above uses a time diversity technique to combat multi-path effects as it combines portions of a signal arriving at different times to reconstruct the original signal. Adaptive antenna arrays that implement beamforming techniques make use of spatial diversity to increase system capacity, while MC-CDMA and OFDM are two transmission schemes that apply frequency diversity techniques to combat frequency selective fading.

An error control scheme that is extensively employed in the DLC is referred to as automatic repeat request (ARQ). This involves an error detection scheme (e.g. CRC) at the receiver, which signals back to the transmitter when an error occurs so that the data containing errors can be transmitted again. So, in an ARQ system a feedback channel must be provided. There are three basic ARQ systems: stop-and-wait, go-back-n, and selective-repeat [42].

The stop-and-wait ARQ is the simplest to implement. When the receiver detects no errors in a received packet, it sends a positive acknowledgement (ACK) back to the transmitter. Upon receipt of the ACK, the transmitter sends the next packet. If, however, the receiver does detect an error in the packet, it sends a negative acknowledgement (NAK) back to the transmitter. In this case the transmitter sends the same packet again and then waits for an ACK or NAK. Obviously the limitation with this system is that it stands idle while waiting for an ACK or NAK. This will be extremely inefficient when packets experience significant propagation and processing delays.

In the go-back-n ARQ scheme the transmitter sends packets one after the other without waiting for an ACK or NAK. If the receiver detects an error in a packet it returns a NAK signal to the transmitter, which responds by returning to the packet indicated in the NAK and resuming transmission from that packet. The number of packets that are sent again depends on the propagation and processing delays. The go-back-n system offers significant improvement over the stop-and-wait scheme as it removes the idle period after the transmission of each packet. It does, however, have increase implementation costs as the transmitted packets have to be buffered at the transmitter in case a NAK is returned requiring that they be retransmitted.

In the selective-repeat ARQ the transmitter again sends packets in succession without waiting for an ACK or NAK. Here, if the receiver detects an error in a packet, it notifies the transmitter, which retransmits just that packet and then returns immediately to its sequential transmission. As may be expected, the selective-repeat ARQ has the highest efficiency of the three schemes. It is the most costly to implement though. For a WATM implementation that requires the cell sequence to be maintained, a selective-repeat ARQ would require cell buffering at both the transmitter and the receiver. A.2.3 discusses the probability of error in ARQ schemes, and also compares the throughput efficiencies of the three ARQ schemes as applied to the proposed protocol.

Another extremely important technique available for controlling transmission errors is referred to as forward error correction (FEC). An FEC scheme depends on coding to allow error correction, with the limitations that a relatively large number of redundant bits are required to achieve low error rates and that if there are too many errors, the code will not be effective. Hence, the efficiency of the code must be carefully considered. Also, good codes generally require complex processing and, therefore, expensive hardware. However, many applications have strict delay and delay variation tolerances for which ARQ schemes are not suited, and modern DSP and FPGA technologies have realised the processing power required to implement complex coding schemes (such as turbo codes). The two fundamental coding schemes are block codes and convolutional codes. For further information on these two schemes refer to [42]. The debate between FEC and ARQ schemes shall be discussed further in the next section.

Hybrid ARQ/FEC schemes have also been suggested. Such schemes would use error-correcting codes to correct as many errors in a packet as possible, and then use an error detection code to determine whether or not any errors remain in a packet, re-transmitting the packet if required. An important segment of such hybrid schemes are those that use a variable rate FEC coding scheme whereby the redundancy in each retransmission is increased until the message gets through. This results in low overhead when the channel quality is high, and increased overhead to deal with poor quality channels.

The last error control technique in Figure 4.1 is error resilience. This refers to the ability of an application to deal with errors that cannot be corrected for whatever reason. A good example of where error resilience techniques may be used is in video compression codecs. In [43] Zhang, Frater, Arnold and Percival investigate methods for error resilience in MPEG-2 video services for WATM networks. The methods looked at include concealment, temporal localisation and spatial localisation. An example of concealment in low motion areas of a video is to simply replace a lost block in a video frame with the corresponding information from the previous frame. Motion compensated concealment can improve the picture by using the motion vectors of surrounding blocks to estimate the motion vector of the lost block. Temporal localisation involves techniques to minimise the propagation of errors from one video frame to the next, and spatial localisation encompasses methods to minimise error propagation within a video frame by providing early resynchronisation of the elements in the bit stream that are coded differentially between blocks.

4.2 Maintaining QoS While Handling Transmission Errors

The unpredictable nature of wireless links makes it difficult to provide any guarantees for the quality of service provided to a virtual channel over such links. Typically, in wired networks, real-time scheduling assumes that all transmissions are successful and no provision is made for retransmissions. If retransmissions were to occur then the QoS of other virtual channels in the link may be jeopardised. That is, in allocating bandwidth to one connection for a retransmission, the scheduler could be denying service promised to another connection. Obviously this approach is impractical in wireless networks where transmission errors occur frequently and unpredictably.

Figueira and Pasquale's paper entitled "Providing Quality of Service for Wireless Links: Wireless/Wired Networks" [37] aims to support retransmissions while still using existing real-time scheduling disciplines by establishing a separate session called the (real-time) retransmission session. This session reserves a portion of the bandwidth to support the retransmission of real-time packets. Thus, if the transmission of a real-time packet is unsuccessful, the packet is assumed to have just arrived as part of the retransmission session. If the packet can be transmitted as part of the retransmission session without exceeding its original deadline, it is kept in the real-time queue with its original deadline and can be retransmitted without violating the QoS guarantee of the original session.

The simulation results presented by Figueira and Pasquale in [37] show that for fairly high error rates bandwidth reservation for retransmission provides a significant reduction in packet loss rate for fast fading channels. However, for slow fading channels, bandwidth reservation for retransmission did not provide a reduction in the packet loss rate even at high levels of reservation.

This is because when the channel fades for long periods of time, real-time packets are dropped (because their deadlines expire) before they can be transmitted successfully.

The alternative to retransmissions is to use some form of forward error correction (FEC). This is the approach used by today's cellular mobile phones that were designed exclusively for voice traffic. However, exclusive use of FEC techniques leads to excessive overhead if one has to support traffic for error sensitive applications such as email or database applications. So, a balance has to found between the number of bits that may be corrected and the number of FEC bits causing overhead. Ultimately what may be desired is a dynamic scheme that adapts the number of redundant FEC bits used to the quality of the wireless link in order to minimise the combined retransmission and FEC overheads. Such a scheme would really need to treat each BS-MT link separately, adjusting to each wireless channel according to its quality. This is because each MT may be in a different position and travelling at a different speed resulting in different link qualities for each. In practise such a scheme is difficult to achieve at the link layer in a wireless system with fixed length slots because the fixed length of each slot limits the dynamics needed to adjust the number of FEC bits in each packet. Another option is to implement such a scheme at some wireless ATM adaptation layer (AAL). Large PDUs from higher layer protocols are split into ATM cell payloads at the AALs for transportation through the ATM network. A wireless AAL receiving dynamic feedback on the quality of each VC's wireless channel from the wireless MAC layer could implement a dynamic FEC scheme on the PDUs as they're organised into cells. Such a scheme would also require dynamic renegotiation of a VC's traffic contract every time its coding rate changed.

Many interactive real-time applications will have delay constraints that are too stringent for the retransmission approach to be effective. Retransmitted cells will be delayed for too long rendering them useless at the receiving end. The RDRN project (Chapter 2.2.4) recognises this by only applying a go-back-n ARQ to non-real-time traffic while delay sensitive cells received with errors are simply dropped. The Magic WAND project (Chapter 2.2.5), on the other hand, blindly applies a go-back-n ARQ to all traffic, although it does drop cells that have been queued for longer than a specified time limit. As shown by Zorzi and Rao in [44], the typical ATM approach of simply dropping cells with bit errors has a big impact on the error rate seen by the higher layers. It may be substantially larger than the error rate on the raw channel. This is fairly logical if one considers that a few bits in error can result in an entire 48-byte payload being lost. Again, the RDRN project recognises this effect by dynamically adjusting the size of its frame and hence the amount of information that is lost when the frame contains an error. But, the smallest frame still occupies an entire ATM cell. In order to prevent interactive applications, for which ARQ techniques are too slow, from suffering from this effect, some technique whereby real-time cells containing errors may be passed on to the higher layers needs to be implemented. This will reduce the error rate seen by the higher layers, allowing FEC schemes to function effectively at this level.

Of course, the scheduling algorithm already incorporates a BER scheduling technique whereby each VC has a maximum BER requirement associated with it which is translated into a multi-user/multi-code interference limit i.e. the maximum number of CDMA codes that may be used in the same timeslot as a cell from the VC. An interactive application using FEC, and no ARQ, may take advantage of this BER scheduling to ensure adequate channel quality for its coding scheme to function successfully. In the current design each VC's interference limit is fixed for the duration of its existence. However, the possibility remains to dynamically adjust the interference limit as needed, although the VC would have to gain permission from the call admission control (CAC) to ensure that the change could be handled without affecting the QoS guarantees of other VCs. A fast power control algorithm, as is typical of DS-CDMA systems, would, to some extent, negate the need to vary the relationship between BER requirement and multi-user interference for each MT.

4.3 An Overview of the DLC Protocol

With the above issues in mind, the proposed DLC protocol recognises the differing requirements of the various traffic types with a mechanism to allow VCs to use a go-back-n retransmission technique or FEC techniques as appropriate. The DLC layer is tightly integrated with the MAC layer to provide a flexible service that supports the requirements of many different traffic types.

4.3.1 Supporting FEC

By default a VC uses the retransmission technique described in the next section, but the call set-up message may specify that the DLC is not to use retransmissions and must in fact accept the VC's cells even if they contain bit errors. This procedure is aimed at delay sensitive real-time traffic for which the ARQ scheme reacts too slowly. The idea is that redundant FEC bits may be included in the cell payloads at higher layers. As already mentioned, allowing real-time cells with errors to be passed on can reduce the error rate seen by the higher layers, allowing FEC schemes to function far more effectively at this level. Note that it is possible to determine to which VC a received cell containing errors belongs, even if the cell's header is corrupt, because the slot in which it was received was allocated to the VC so no other VC will transmit in that slot. Also, the 16-bit CRC appended to each WATM cell does not need to be applied to the entire cell if the cell is going to be accepted with bit errors anyway. So, it may be used to provide error detection on the cell header, or even to implement some FEC on the header. When used in conjunction with the BER scheduling scheme, a VC implementing FEC it provided with flexible control over the quality of its link.

An example of when a VC would be set up to accept bit errors is when a voice call is being made. The real-time nature of voice traffic imposes cell delay requirements that are too stringent for the ARQ protocol to deal with, so the same approach that is used in existing mobile networks may be used here as well. By allowing the FEC to be implemented at higher layers that may be application specific, rather than at the DLC layer that has to support all traffic types, the coding may be matched to the application and used to increase error resilience. For example, the GSM full rate (FR) speech codec compresses voice to a 13kbps bit stream that is coded up to 22.8kbps to allow for FEC. However, certain bits in the bit stream are more important for perceived speech quality than others, so each block of 260 bits is divided into 3 classes with the different classes coded differently according to their effect on the perceived quality [45].

It is all very well implementing such an application specific scheme in the DLC if the only traffic, or vast majority of traffic, is voice traffic. But, if one cannot make such an assumption because a variety of multimedia traffic types are to be supported, then it makes more sense for each application to implement error resilient FEC schemes appropriate to itself. That is the intention when introducing this optional scheme to allow cells containing bit errors to be accepted.

4.3.2 Supporting Retransmissions

As may be seen from the WATM protocols reviewed in Chapter 2, the go-back-n ARQ is the most popular ARQ scheme because it is relatively simple, yet effective. The selective-repeat scheme performs only marginally better but is far more complex and costly to implement (see A.2.3). For this reason the DLC protocol supports a go-back-n ARQ scheme, but it is optional so applications specify whether or not they want to use it during the call set up process.

In fact, the ARQ scheme used is termed a windowed go-back-n ARQ and is the same scheme used in the Magic WAND project and described by Meierhofer in [31]. For this purpose each WATM cell includes a sequence number (SN), sequence number acknowledgement (SNack) and 16-bit cyclic redundancy check (CRC) fields. At the receiver the cell's CRC field is used to detect transmission errors and its SN field is checked to ensure that the cell order is maintained. If the cell contains transmission errors then it is dropped, otherwise, if both checks are passed then the cell is acknowledged by sending a feedback message to the transmitter containing the cell's SN to the transmitter. This feedback message may be piggybacked in the SNack field of a WATM cell being transmitted in the reverse direction or, if there is no traffic in the reverse direction, the receiver generates a VC update cell that may contain a number of sequence number acknowledgements as well as any reservation requests a MT may have.

When a cell is received out of sequence the SN of its VC's last successfully received cell is acknowledged. Thus, if the sending end receives two successive acknowledgements that are exactly the same then it knows that a transmission error has occurred. In this case the transmitter resumes transmission from the cell after the double acknowledged one. In order to prevent cell transmissions from getting too far ahead of cell acknowledgements, a sliding window is used. The window length determines how far ahead of the acknowledgements the cell transmissions can get.

To emphasise the limitation with such a scheme that leads to the need to simultaneously support alternative techniques consider the following problem that is illustrated, in Chapter 6, by the simulation results of voice traffic using the ARQ scheme. A VC with a low mean cell rate may have a larger cell inter-arrival time than cell lifetime. If a retransmission is triggered by a cell arriving out of sequence, this would result in every unsuccessfully transmitted cell being dropped at the transmitting end before its retransmission is triggered, and can lead to an even worse performance for the VC than if the retransmission technique were not applied at all. The situation may be improved if, instead of waiting for a cell out of sequence to trigger a retransmission, as soon as a cell fails its CRC based error detection the retransmission request is triggered. Of course, the information in the cell header could not be used to determine the VC to which the cell belonged as it may be corrupt, so the slot in which it arrived would have to be used for this purpose; i.e. the slot would have been allocated to a VC, so it can be used to determine the cell's VC.

As the protocol stands, when a VC goes back for a retransmission it would have already used some of its scheduling tokens to transmit cells that weren't received correctly. This means that it may not have enough tokens to retransmit those cells again, as well as transmit the new cells arriving in the mean time, without relying on the schedulers frame fill up procedure. To solve this problem we draw on Figueira and Pasquale's technique of reserving a portion of the bandwidth for retransmissions [37]. For this purpose the BS instantiates a dummy retransmission VC with its own token pool, MCR and MBS. These parameters will determine what portion of the bandwidth is being allocated to retransmissions. To recover tokens for retransmissions a VC may 'steal' whatever tokens the dummy retransmission VC may have. This is easy enough for the downlink because the BS knows when a downlink VC goes back to retransmit and it knows how many cells need to be retransmitted. However, the BS does not have this information available for the uplink, so what happens instead is a VC 'steals' an uplink token whenever an uplink cell is received out of sequence. In addition, the VC is not able to use these retransmission tokens unless the last cell received was in sequence. This prevents the tokens from being used to transmit uplink cells that are out of sequence because the MT end of the VC has not yet gone back for retransmission. These retransmission tokens are also considered by the scheduler to have the highest priority so retransmissions are scheduled first.

In reserving a portion of the bandwidth for retransmissions, the scheme also limits the bandwidth that may be utilised for retransmissions. For example, if VC's are allowed to transmit retransmissions whenever they need to then one VC may 'swamp' the bandwidth with its retransmissions on its poor quality link while other VCs with good links are being denied slot allocations.

4.4 Call Admission Control

Obviously it is imperative that the call admission control (CAC) does not clog the system by accepting calls that can't be handled resulting in failure to meet users' QoS requirements. On the other hand, the CAC can't become too conservative, as it still needs to obtain high bandwidth utilisation. There are two fundamental methods by which the CAC can accept/reject calls. The first method is to estimate the aggregate traffic from the traffic descriptors declared for each VC upon connection and compare this to the node capacity [46]. If there is still some surplus capacity available then a new call can be accepted. In practise the difficulty is that the traffic descriptors are subject to statistical estimation and therefore they cannot be used as exact parameters. Hence, the estimation of the aggregate traffic has uncertainties.

The second method is to use a traffic measurement based CAC. In this method traffic flows are dynamically characterised by measurement with minimal a priori traffic description required [47]. This is the technique that shall be discussed further for possible implementation with the proposed link layer. To implement such a method the CAC needs to obtain feedback from the MAC layer with actual measurements of the available bandwidth. In order to achieve this the MAC layer keeps statistics on the number of free slots per packet. The problem with the proposed MAC protocol is that the number of free slots available to a new VC will depend on the VC's BER requirement because of the BER scheduling that adjusts multi-user/multi-code interference for each VC as required. For this reason a separate statistic must be kept for each level of BER requirement. The number of levels is equivalent to the number of CDMA codes in use. Every time the scheduler builds a frame, the available bandwidth statistic for each BER requirement is updated according to an exponential-weighted moving average technique as used in [47]. So, if s_i is the current measurement for the number of free slots per frame then the current estimate of the available bandwidth $\hat{s}_i$ is given by

$$\begin{aligned}\hat{s}_i &= ws_i + (1-w)\hat{s}_{i-1} \\ &= \hat{s}_{i-1} + w(s_i - \hat{s}_{i-1})\end{aligned} \qquad \textbf{Eq. 4-1}$$

where w denotes a weighting factor of the current measurement s_i. The weighting factor w must be chosen to provide a reasonably smooth estimate of the available bandwidth while reflecting the peak rate available to new connections. Depending on the burstiness on the traffic flows, this technique is able, to some extent, to filter out stochastic oscillations of the s_i measurements around the mean value.

One issue that needs to be considered is whether the number of free slots in a frame s_i should be counted before or after the scheduler implements its frame fill-up procedure; i.e. before or after any slots are allocated to VCs without tokens. The initial decision is to determine s_i before running the frame fill-up procedure, because such slot allocations are bonus allocations that potentially permit VC's to exceed their specified MCR and MBS. However, the results shown in Chapter 6 do indicate that a significant portion of the traffic tends to be transmitted by frame fill-up, slot allocations due to the bursty nature of some traffic as well as retransmissions.

Table 4-I shows the time taken for the free slots moving average to reach 90% of the actual value if there is a change in the number free slots. Note that it doesn't make any difference how large the change is, the time taken for the moving average to come within 90% percentage of the new value remains the same. The parameters used in the simulation are used here as well; i.e. a frame contains six timeslots with eight CDMA code channels available and a raw bit rate over each CDMA code channel of 384kbps. It follows that the maximum possible change in free slots is forty, so the moving average has to reach 36 to be within 90% of the desired value.

Table 4-I Time taken for moving average to reach 90% of final value under various weighting factors (assuming a raw bit rate of 384kbps).

Moving average weighting factor (w)	Time taken to reach 90% of final value (seconds)
0.01	1.6
0.005	3.3
0.001	16.4

It is not critical that the moving average responds immediately to changes in the available bandwidth. In fact, the bursty nature of some traffic types makes it essential that the average is taken over a sufficient time period so as not to reflect temporary fluctuations in available bandwidth. In other words the moving average needs to be relatively smooth with small fluctuations. On the other hand, the moving average needs to adjust fairly rapidly to reflect the correct average when VCs are set-up and released. A delay of 16.4 seconds, achieved for $w = 0.001$, may seem fairly large but, the results shown in Chapter 6 suggest that the moving average is not quite smooth enough even with this weighting factor.

Table 4-II shows the same time delays assuming the raw bit rate is only 128kbps, as used in the implementation results.

Table 4-II Time taken for moving average to reach 90% of final value under various weighting factors (assuming a raw bit rate of 128kbps).

Moving average weighting factor (w)	Time taken to reach 90% of final value (seconds)
0.01	4.9
0.005	9.8
0.001	49.2

Chapter 5 The Test-Bed Implementation

A significant portion of this project involves the development of a multimedia communications test-bed employing the WATM link layer protocol. The test-bed implementation is based around the software radio platform developed by Alcatel Altech Telecoms (AAT). This chapter discusses the software radio concept, and AAT's hardware architecture used for the implementation. This is followed by an overview of how the link layer is implemented in software using the software radio platform. The objective is not to produce a compact, optimised solution but rather to work towards a flexible system that may be used to investigate different techniques in order to optimise the protocol.

5.1 The Software Radio Concept

With international mobile roaming still not possible, largely due to multiple standards operating in various regions throughout the world as well as within the same geographical area, there is a strong need for a common global mobile communications standard. Unfortunately, with WCDMA proving to be the 3G standard of choice in Europe while CDMA2000 establishes itself in North America, it seems such a global standard is still some way off, despite the similarities of these two standards. So, for the time being there is a strong call for radios capable of supporting multi-standard operation and easy reconfiguration.

Conventionally, the approach to the implementation of multi-standard radios has been to use stacked radios containing multiple transceivers, one for each standard. However, as the complexity of the radio interfaces increases and the number of standards escalate, so this approach becomes more and more infeasible with hardware replacements required for every change in the wireless interface. Alternatively, a radio with the ability for dynamic reconfiguration in every architectural layer could provide a more effective solution to the multi-standard problem. Hence the widespread interest in the so-called software radio.

As explained by Hentschel et al. in [48], one of the fundamental ideas behind the software radio is the extension on digital signal processing toward the antenna. In other words, by replacing the regions of the transceiver where analogue signal processing has been dominant thus far with digital signal processing, a digital front-end is achievable allowing the system to be "softly" reconfigured. With digital implementations of front-end functionalities, such as down conversion and channel filtering, that are traditionally realised by means of analogue signal processing, different air interfaces may function on a given hardware platform.

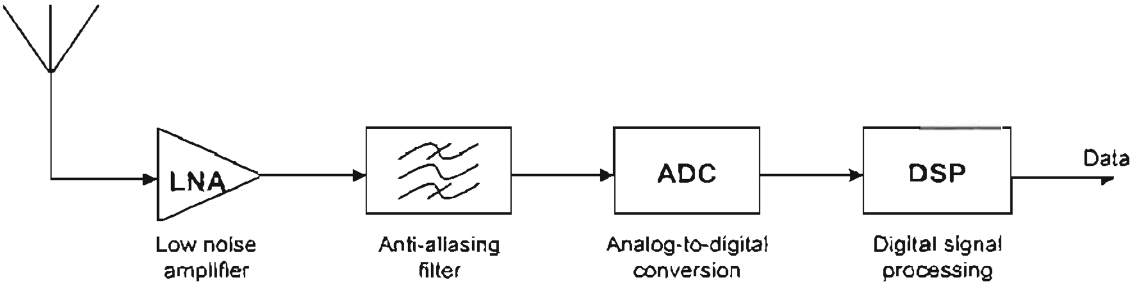


Figure 5.1 The ideal software radio receiver.

Figure 5.1 portrays the architecture of an ideal software radio receiver [48] featuring a minimum of analogue components. Both [48] and [49] explain the steep requirements of the ADC. Because the software radio needs to cope with numerous carriers coming from different sources, the ADC needs to cope with signals of very large bandwidth. Characteristics of RF signal propagation such as fading and shadowing combined with strong interference signals lead to dramatic, time varying differences in the RF power of received carriers. This means the ADC has to handle high dynamic ranges as well. The ADC performance requirements are a real challenge in the development of software radios and the digitisation of the bandwidth for all supported services is not currently feasible. Techniques to tackle this ADC challenge are suggested in [48] and [49].

Assuming the ADC is capable of digitising the received wideband signal with sufficient precision, the digital data produced must be processed in real-time in the DSP domain. In the simplest case the DSP domain could be just a single DSP device. The increasing performance of DSP processors is allowing for more of the radio interface processing to be performed in software. However, a single DSP processor cannot cope with the performance requirements, particularly for BS systems. A straightforward approach to get around this DSP bottleneck is to use multi-DSP structures where the overall functionality is divided between a number of devices including ASICs (application specific integrated circuits), FPGAs and DSP processors. In general ASICs and FPGAs are used to implement tasks that require high processing speed, but rarely require dynamic functional adjustment. DSP processors, on the other hand, are better suited to more complex procedures requiring dynamic adjustment.

In [50] Mitola and Maguire describe cognitive radio – a possible future extension to the software radio providing enhanced personalised services through a radio knowledge representation language (RKRL). Cognitive radios are required to have an advanced degree of "understanding" of the network, the user's goals and its own internal structure, thus enabling the radio to recognise common objects, events and local RF context to personalise the service offered to its user. For example, a cognitive radio would be able to hold an intelligent "conversation" with the network using RKRL to download an improved vocoder (voice coder), prompt the user to move closer to the doorway where the SNR is better, or activate real-time translation between two languages. The significant memory, computational resources and communications bandwidth required for cognitive radio means that such technology may not be deployable for some time, but the present research provides a glimpse of what can be expected in the future. However, parts of the cognitive radio concept are already being incorporated into MTs with the requirement that their location must be known to within 125m for emergency location reporting. This has opened the door for location aware services, such as flexible directory services.

5.2 Alcatel Altech Telecoms' Software Radio Infrastructure

Figure 5.2 provides an overview of the hardware infrastructure used in the development of the WATM test-bed. This is the architecture used by Alcatel Altech Telecoms (AAT) in the development of their software radio platform. AAT's main software radio board only contains re-configurable hardware, in the form of two Altera 10K100 FPGAs. This means that the software radio itself is missing the flexibility provided by a DSP processor. Although such a processor can't provide the high speeds of FPGAs, it is more suitable for complex, dynamic processing functions at slower speeds. In order to provide the flexibility of a DSP processor, AAT connected their software radio board to a Blue Wave Systems (BWS) PCI/C6200 applications board containing a Texas Instruments TMS320C6201 DSP processor. The BWS DSP board contains interfaces to an SCbus and an MVIP bus, so these two telecommunications buses provide the link between the software radio board and the DSP board. Four bi-directional 2048kbps serial links are connected for each bus. Although this configuration does include the functionality of a DSP processor in the system, it is not ideal. To make the system more flexible and compact it would be more suitable to incorporate a DSP processor into the software radio board. For example, the TMS320C6201 DSP processor on the BWS DSP board could be directly connected to the software radio board's FPGAs through its host-port interface (HPI). This would allow for faster and more flexible interaction between the FPGAs and DSP processor, with the FPGAs being able to directly access the DSP processor's external and internal memory as well as the two being able to interrupt each other.

The BWS DSP board also contains a 32-bit slave PCI interface that is used for rapid data transfer between the DSP board and its host PC. This means that the host PC can act as a data source/sink. AAT donated two of its software radio boards to the University of Natal's centre of excellence (CoE) in radio access technologies, allowing for two of the software radio configurations described above to be set up – one to act as a BS and the other as a MT. On the one side the host PC simulates an ATM network connected to a BS, while on the other side the host PC, acting as a MT, terminates the ATM connections.

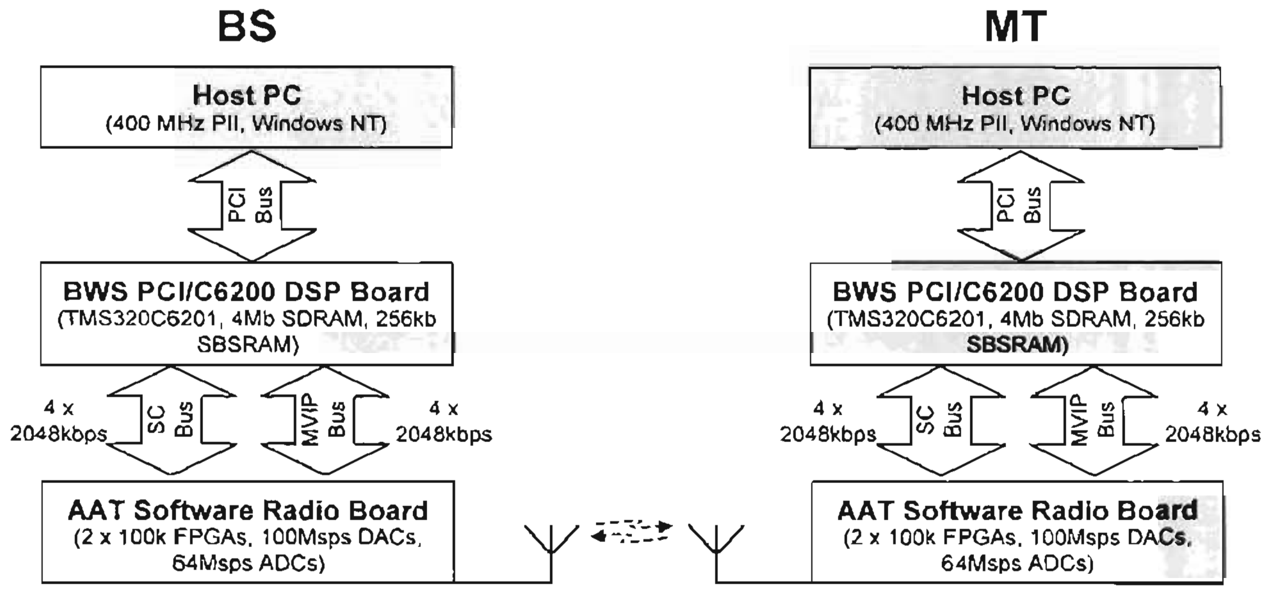

BS
MT
Host PC
(400 MHz PII, Windows NT)
PCI Bus
BWS PCI/C6200 DSP Board
(TMS320C6201, 4Mb SDRAM, 256kb SBSRAM)
4 x 2048kbps
SC Bus
MVIP Bus
4 x 2048kbps
AAT Software Radio Board
(2 x 100k FPGAs, 100Msps DACs, 64Msps ADCs)
Host PC
(400 MHz PII, Windows NT)
PCI Bus
BWS PCI/C6200 DSP Board
(TMS320C6201, 4Mb SDRAM, 256kb SBSRAM)
4 x 2048kbps
SC Bus
MVIP Bus
4 x 2048kbps
AAT Software Radio Board
(2 x 100k FPGAs, 100Msps DACs, 64Msps ADCs)

Figure 5.2 Hardware infrastructure.

Although this thesis is only concerned with the link layer implementation, the CDMA physical layer implementation is also being carried out in another project. The higher speed, bit and chip level processing required at the physical layer is more suited to an FPGA implementation than a DSP processor implementation, whereas the dynamic, memory hungry nature of the link layer lends itself to a processor implementation. For this reason the CDMA physical layer is implemented on the FPGA based software radio board, while the link layer runs on the DSP board. Because this thesis is not concerned with the physical layer implementation on AAT's software radio board, the architecture of this board shall not be discussed any further. However, the architecture of the BWS DSP board, along with its interfaces to the host PC and software radio board, is examined in more detail to complement the description of the link layer implementation.

5.2.1 The Blue Wave Systems DSP Board

As previously mentioned, the Blue Wave Systems (BWS) PCI/C6200 DSP board has, at its centre, a Texas Instruments TMS320C6201 DSP processor. This processor operates at 200MHz (5ns cycle time) and is capable of executing up to eight 32-bit instructions every clock cycle meaning it supports up to 1600 MIPS. The C6201 achieves this high processing rate through the use of Texas Instruments' VelociTI very long instruction word (VLIW) architecture that uses 256-bit wide packet fetches to internal program/cache memory. A block diagram of the DSP board, taken from the *PCI/C6200 Technical Reference Manual* [51], is shown in Figure 5.3.

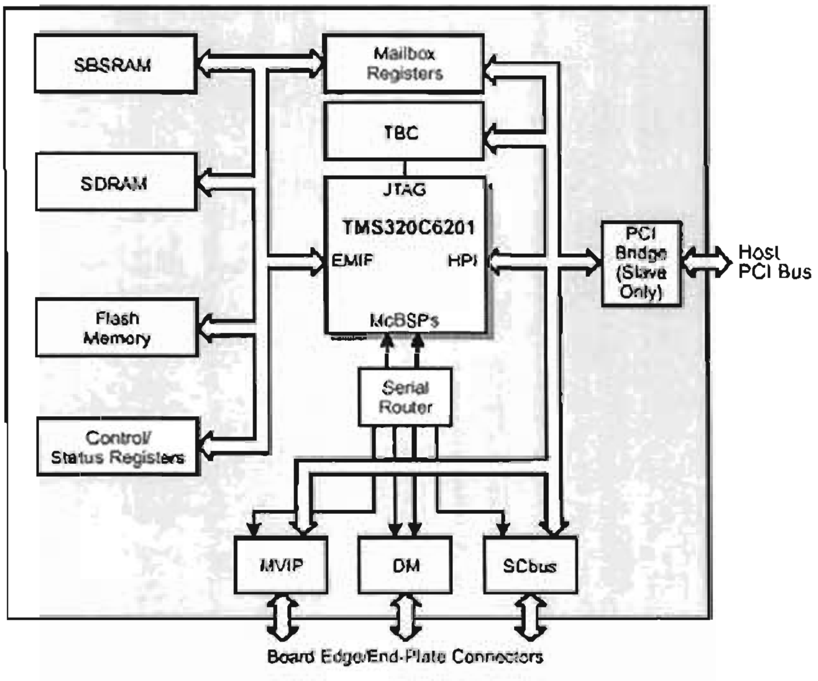


Figure 5.3 Blue Wave Systems PCI/C6200 block diagram.

Besides the 64kb of internal program memory and 64kb of internal data memory, the C6201's external memory interface (EMIF) provides the processor with access to 256kb of synchronous-burst static RAM (SBSRAM) and 16Mb of synchronous dynamic RAM (SDRAM). This provides more than adequate memory for cell buffering. A host port interface (HPI) enables a host PC to directly access the memory on the card. The HPI is itself accessible via a slave PCI bridge device. The PCI bridge incorporates a two channel DMA controller for data transfers between the PCI bus and DSP processor's host port. Two mailbox registers provide general-purpose message passing between the PCI bus master (host PC) and the C6201. These mailbox registers may be used to generate PCI or C6201 interrupts.

Two multi-channel buffered serial ports (McBSPs) provide the C6201 with interfaces to a Multi-Vendor Interface Protocol (MVIP) bus and an SCbus. The MVIP bus and SCbus are designed to carry telecommunications data between PC based boards via ribbon cabling. Appendix C contains further information on the functionality of these buses. Only the SCbus is currently used in the test-bed. It provides the interface between the link layer on the DSP board and the physical layer on the software radio board. Only four of the sixteen available bi-directional serial data lines of the SC data bus are used. Each data line offers 2.048Mbps and carries data in frames consisting of 8-bit packets or timeslots. There are 32 timeslots in a frame so each timeslot offers 64kbps. To increase the test-beds freedom of movement, the 5V logic levels of the SCbus are converted to RS485 differential lines between the DSP board and the software radio board. This enables the two to be separated by up to 1.2km (according to the RS485 specifications) instead of just a short ribbon cable. The signal delay through the Ics that performed the conversion to/from RS485 did, however, mean that the default SCbus configuration of sampling the bits at 50% of the bit width according to the clock signal resulted in erroneous data being received on the clock master side of the link. Nevertheless, the SC4000 IC that controls the SCbus does allow the sampling position to be configured to 75% of the bit width, solving this problem [52].

The C6201 DSP processor also has an internal DMA controller that provides four DMA channels that may be used to transfer data between internal and external memory, or between memory and a McBSP, etc. This DMA controller must not be confused with the PCI bridge device's DMA controller, which is used to transfer data between the C6201's HPI and the PCI bus.

A complete package of tools for use with the PCI/C6200 DSP board was used in the development of the test-bed's link layer software. These tools include Blue Wave Systems' IDE6000 software package comprised of a Windows NT/95 device driver, a tool for configuration file generation, a download monitor tool, and a host communications interface library (HCIL). The HCIL contains a library of functions that is used to provide applications running on the host PC with the ability to download and run DSP code on C6201 processor, transfer data between the host PC and DSP board, send/receiver interrupts, as well as control the configuration of peripherals such as the MVIP bus and SCbus. The DSP software running on the C6201 processor was written using Texas Instruments' Code Composer Studio, as well as their optimised C compiler, assembler and linker for the C6201.

5.3 Link Layer Software Overview

Now that the test-bed's hardware architecture has been described the focus shifts to the software modules. This section looks at the link layer software running of the DSP processor. Of importance are the dynamically created structures used to represent MTs and VCs, the scheduling module, the DLC module, and the software implementations of the interfaces with the physical layer (on the software radio board) and the ATM layer (on the host PC). The MT and BS link layers are fundamentally different in that the BS allocates slots while the MT follows these slot allocations. Nevertheless, there are many similarities between the two, allowing many identical, or near identical structures and modules to be used in both. For this reason, instead of developing two separate processes, one for the BS and one for the MT, a single software programme is coded for both with conditional compilation directives used to differentiate between them.

Figure 5.4 shows a block diagram of the link layer software architecture for the BS. The block diagram for MTs is almost identical to that for the BS. MTs have the same block structure, but there are differences in the way each block functions. For example, the *Mobile Container* would only hold one MT structure containing the parameters for the MT itself, and instead of performing slot allocations like the *BS Scheduler*, the *MT Scheduler* would send cells according to the schedule received from the BS.

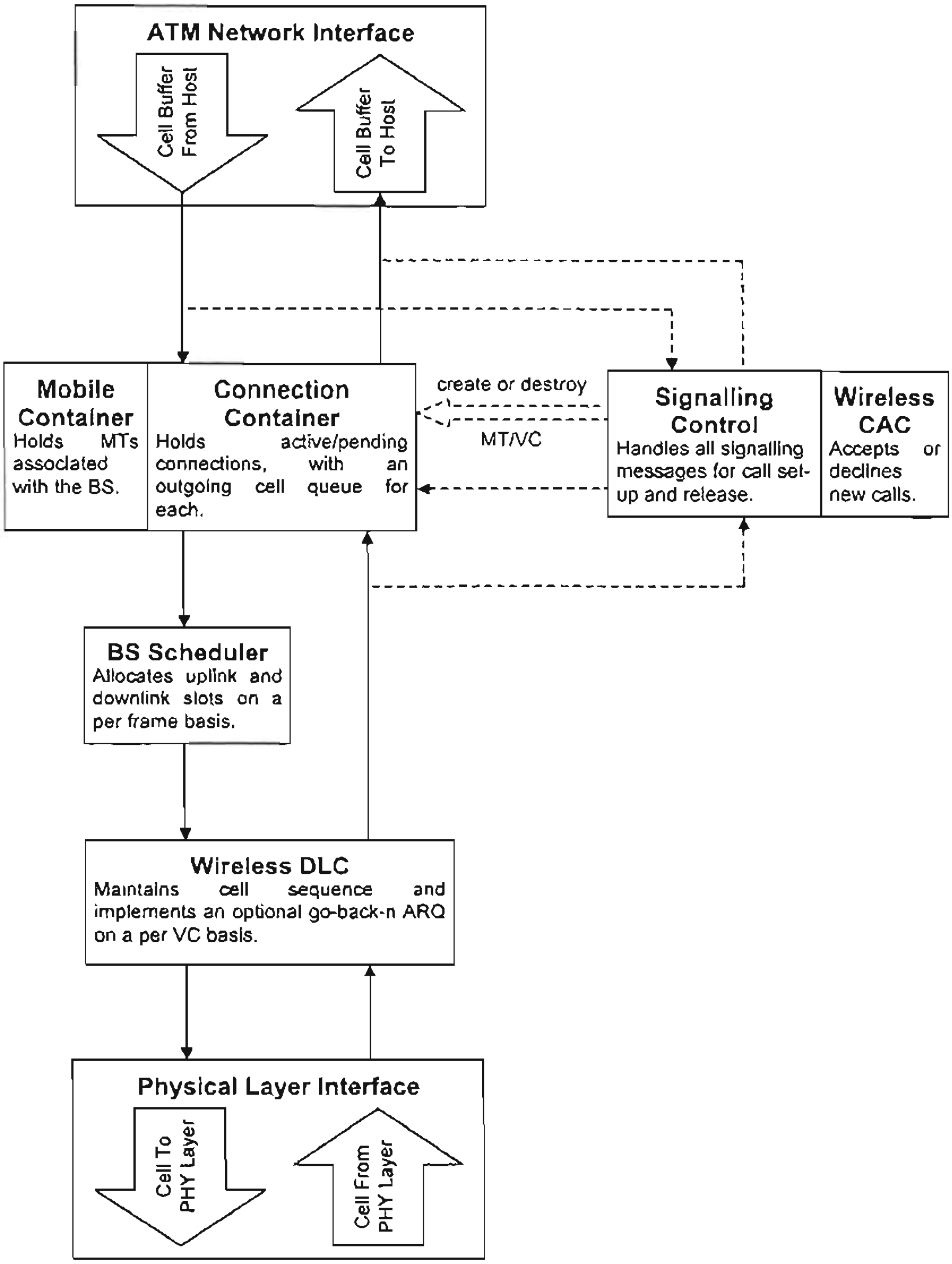

ATM Network Interface
Cell Buffer From Host
Cell Buffer To Host
Mobile Container
Holds MTs associated with the BS.
Connection Container
Holds active/pending connections, with an outgoing cell queue for each.
create or destroy
MT/VC
Signalling Control
Handles all signalling messages for call set-up and release.
Wireless CAC
Accepts or declines new calls.
BS Scheduler
Allocates uplink and downlink slots on a per frame basis.
Wireless DLC
Maintains cell sequence and implements an optional go-back-n ARQ on a per VC basis.
Physical Layer Interface
Cell To PHY Layer
Cell From PHY Layer

Figure 5.4 BS link layer software block diagram.

In Figure 5.4 the *ATM Network Interface* is in fact an interface to the host PC, which simulates an ATM network. This interface contains two cell buffers – one buffering cells from the host, and the other buffering cells to the host. This block is also responsible for routing cells into the correct connection buffer in the *Connection Container*, or passing them on to the *Signalling Control*. The *Connection Container* holds two linked-lists of connection structures – one containing active connections and the other containing pending connections (in the process of being set-up). Each connection structure describes the parameters and state of its VC including its token pool, reservation requests, traffic parameters, retransmission status, etc. The connection structure also contains a cell buffer for any of the VC's cells queued for transmission or awaiting acknowledgement of a successful transmission from the DLC. ATM cells received from the ATM network are converted to WATM cells before being queued in their VC's cell buffer and, conversely, WATM cells received from MTs are converted to ATM cells before being passed on to the ATM network. The *Mobile Container* is closely linked to the *Connection Container* and contains a linked list of MT structures currently associated with the BS. Each MT structure describes the parameters and state of its MT including information such as its full 20-byte ATM address and a link to the MTs signalling VC in the *Connection Container*.

The *Signalling Control* handles all ATM signalling messages received from the ATM network as well as WATM signalling messages received from MTs. ATM signalling messages are converted to WATM signalling messages and then placed in the appropriate MTs signalling VC, and vice-versa. To save time, instead of implementing the full ATM UNI specifications, a simplified signalling protocol that only includes the features needed for VC set-up and release to test the link layer, has been implemented. The *Wireless CAC* is closely linked to the *Signalling Control* and has the responsibility of determining whether or not the network can handle a new call. At this stage the *Wireless CAC* has not been properly implemented, accepting all new calls, although available bandwidth statistics that the CAC may need are collected.

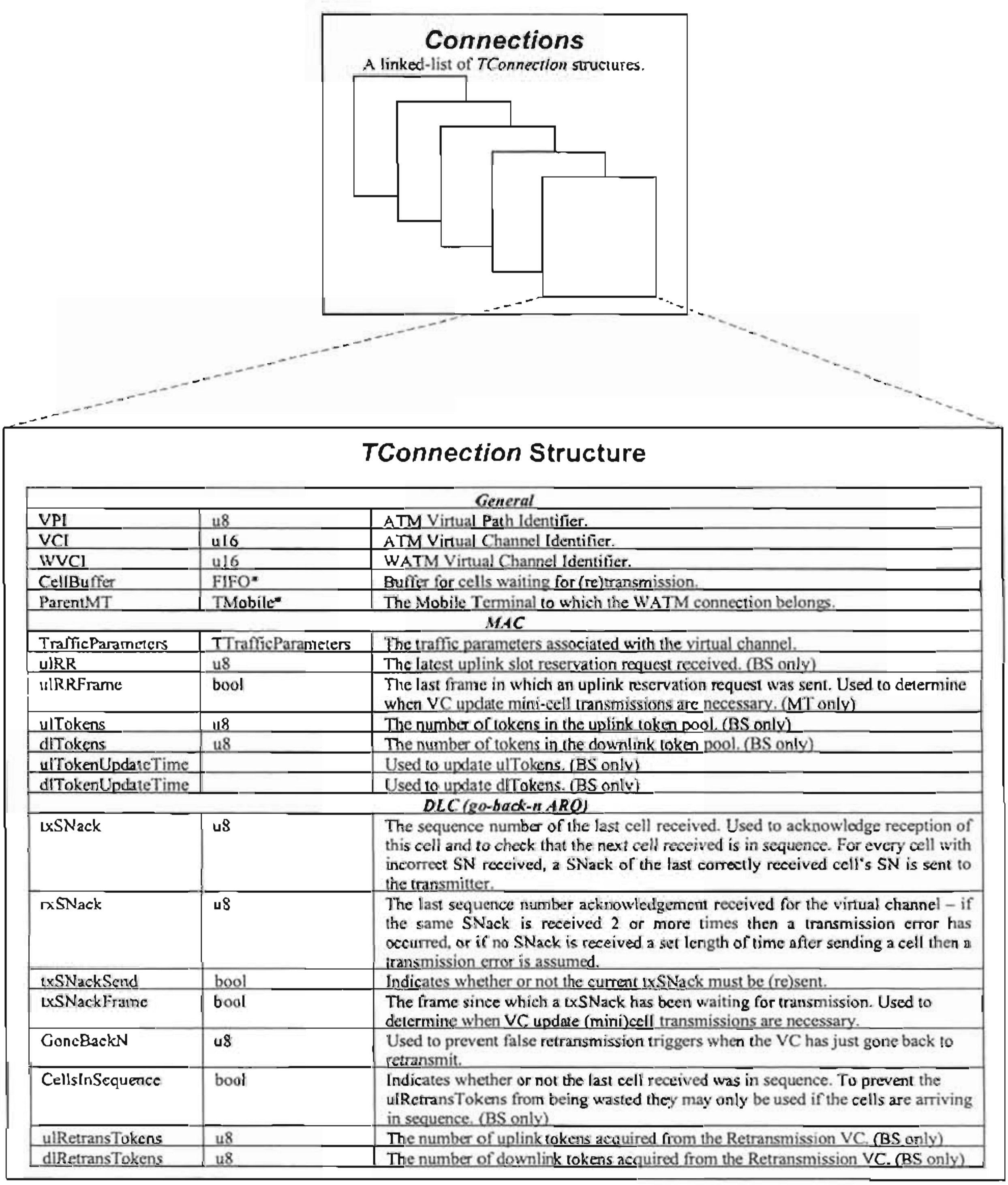
The *BS Scheduler* implements the previously described scheduling algorithm, performing uplink and downlink slot allocations, and building FHs describing the allocations in each MAC frame. This block also generates VC update cells when there are cell acknowledgements that can't be piggybacked onto any other cells scheduled for the current MAC frame. The FHs and scheduled downlink cells are passed on to the *Wireless DLC* where the correct SN, SNack and CRC fields are assigned. Following this the cells are passed on to the physical layer through the *Physical Layer Interface*. The *Wireless* DLC also receives cells from the physical layer. These cells are checked for errors before any cell acknowledgements or slot reservation requests are removed from them. If a cell in error belongs to a VC that has the ARQ scheme enabled, then a retransmission is triggered for that VC. The following sections describe the operations of each block in more detail.

5.3.1 Mobile and Connection Containers

The Mobile and Connection Containers store the parameters and current states of all the MTs and VCs respectively. MTs are constantly entering and leaving the BS's cell, and VCs are constantly being set-up and released, thus these containers need to be flexible and efficient in keeping track of new structure instances being created and old instances being destroyed. Many of the other blocks in the system rely on the information stored in these containers, so this information must be easily accessible to these blocks. For example, the *ATM Network Interface* needs to be able to quickly locate a specific connection according to its VPI/VCI so that it may route received cells belonging to that connection into the connection's cell buffer. Similarly, the *BS Scheduler* needs to be able to quickly locate VCs according to their WVCI in order to access cells that have been scheduled for transmission. The following sections describe the structures used to represent VCs and MTs.

Keeping Track of Virtual Channels

Each VC is represented by a structure of type *TConnection*. The *TConnection* structure contains all the parameters necessary to identify the virtual channel, implement the DLC's windowed go-back-n scheme, and implement the MAC scheduling algorithm. It also contains a *TTrafficParameters* structure that defines the traffic characteristics of the VC as specified during the call set-up process. The *TConnection* structure is used to represent VCs at the BS as well as the MT although there are a few parameters specific to either side. The entire collection of active VCs is stored in a linked-list of *TConnection* structures called *Connections*. A linked-list allows elements to be efficiently added to or removed from any position in the list. So, with each *TConnection* structure forming an element in the linked list, connections may be easily added and removed from the list.

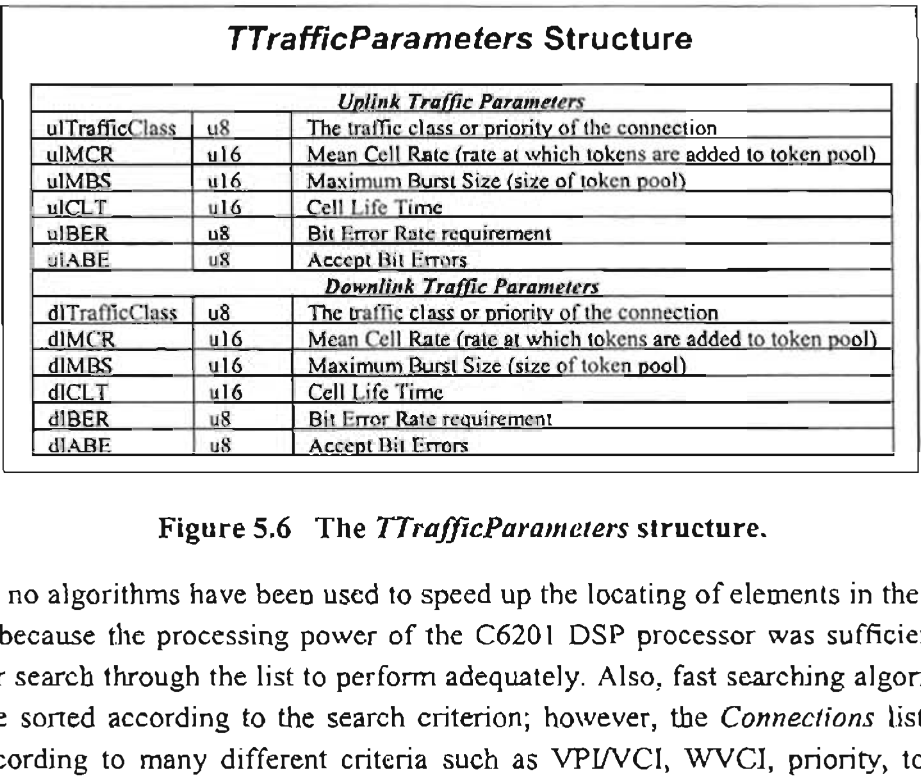


General		
VPI	u8	ATM Virtual Path Identifier.
VCI	u16	ATM Virtual Channel Identifier.
WVCI	u16	WATM Virtual Channel Identifier.
CellBuffer	FIFO*	Buffer for cells waiting for (re)transmission.
ParentMT	TMobile*	The Mobile Terminal to which the WATM connection belongs.
MAC		
TrafficParameters	TTrafficParameters	The traffic parameters associated with the virtual channel.
ulRR	u8	The latest uplink slot reservation request received. (BS only)
ulRRFrame	bool	The last frame in which an uplink reservation request was sent. Used to determine when VC update mini-cell transmissions are necessary. (MT only)
ulTokens	u8	The number of tokens in the uplink token pool. (BS only)
dlTokens	u8	The number of tokens in the downlink token pool. (BS only)
ulTokenUpdateTime		Used to update ulTokens. (BS only)
dlTokenUpdateTime		Used to update dlTokens. (BS only)
DLC (go-back-n ARQ)		
txSNack	u8	The sequence number of the last cell received. Used to acknowledge reception of this cell and to check that the next cell received is in sequence. For every cell with incorrect SN received, a SNack of the last correctly received cell's SN is sent to the transmitter.
rxSNack	u8	The last sequence number acknowledgement received for the virtual channel – if the same SNack is received 2 or more times then a transmission error has occurred, or if no SNack is received a set length of time after sending a cell then a transmission error is assumed.
txSNackSend	bool	Indicates whether or not the current txSNack must be (re)sent.
txSNackFrame	bool	The frame since which a txSNack has been waiting for transmission. Used to determine when VC update (mini)cell transmissions are necessary.
GoneBackN	u8	Used to prevent false retransmission triggers when the VC has just gone back to retransmit.
CellsInSequence	bool	Indicates whether or not the last cell received was in sequence. To prevent the ulRetransTokens from being wasted they may only be used if the cells are arriving in sequence. (BS only)
ulRetransTokens	u8	The number of uplink tokens acquired from the Retransmission VC. (BS only)
dlRetransTokens	u8	The number of downlink tokens acquired from the Retransmission VC. (BS only)

Figure 5.5 *Connections* – a linked-list of *TConnection* structures.

The *Connections* linked list and the *TConnection* structure are depicted in Figure 5.5. The members of the *TConnection* structure are grouped into general, MAC-related and DLC-related members.

The significance of each member is described in the figure, and the application of some of them will be explained in more detail in the following sections. Figure 5.6 shows the members of the *TTrafficParameters* structure that is included in the *TConnection* structure. Only the traffic parameters required to test the protocol are included in the structure (and, hence, in the call set-up signalling messages). Note that the uplink and downlink parameters are specified separately so that each connection may have entirely different uplink and downlink characteristics. The traffic class parameter simply specifies one of the five ATM traffic classes and is used as a priority by the scheduler (see Appendix A.2.2). The MCR is specified as a fixed-point number in cells per second. Of the 16-bits used to specify the MCR, 10-bits represent the integer portion and 6-bits represent the fraction; i.e. the MCR is effectively scaled up by a factor of 64. This means that MCRs may be specified in increments of 0.015625 cells/s with a maximum specifiable MCR of approximately 1024 cells/s. The MBS determines the maximum number of tokens a connection may have at any one time, and the CLT specifies the maximum number of frames for which a cell may be queued. A connection's BER requirement is specified as the maximum number of cells that may be transmitted, using different CDMA codes, in the same timeslot as a cell from that connection. Finally, the ABE parameter determines whether or not the DLC should accept cells containing bit errors. Obviously, if bit errors are accepted then the DLC doesn't apply the go-back-n ARQ to the VC.



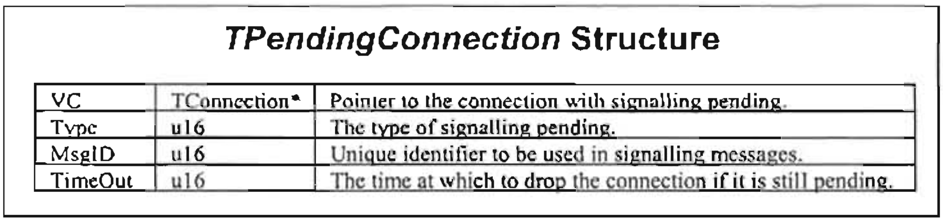
TTrafficParameters Structure

		Uplink Traffic Parameters
ulTrafficClass	u8	The traffic class or priority of the connection
ulMCR	u16	Mean Cell Rate (rate at which tokens are added to token pool)
ulMBS	u16	Maximum Burst Size (size of token pool)
ulCLT	u16	Cell Life Time
ulBER	u8	Bit Error Rate requirement
ulABE	u8	Accept Bit Errors
		Downlink Traffic Parameters
dlTrafficClass	u8	The traffic class or priority of the connection
dlMCR	u16	Mean Cell Rate (rate at which tokens are added to token pool)
dlMBS	u16	Maximum Burst Size (size of token pool)
dlCLT	u16	Cell Life Time
dlBER	u8	Bit Error Rate requirement
dlABE	u8	Accept Bit Errors

Figure 5.6 The *TTrafficParameters* structure.

At this stage no algorithms have been used to speed up the locating of elements in the *Connections* list. This is because the processing power of the C6201 DSP processor was sufficient to allow a simple linear search through the list to perform adequately. Also, fast searching algorithms require the list to be sorted according to the search criterion; however, the *Connections* list needs to be searched according to many different criteria such as VPI/VCI, WVCI, priority, tokens, etc. A simple solution to reduce the average search time is to position the most frequently accessed connections at the top of the list. For example, connections with high cell rates will be accessed more frequently than connections with low cell rates. Thus, the *Connections* list may be sorted in descending order according to the MCR specified for each connection, which decreases the time taken to locate the most frequently accessed connections. Note that for the BS the list needs to be sorted according to the downlink MCR, whereas the MTs require the list to be sorted according to the uplink MCR.

The *Connection Container* also includes a linked-list of pending connections. These are connections that are in the process of being set-up. Pending connections are represented by the *TPendingConnection* structure shown in Figure 5.7, and are stored in a linked list structure called *PendingConnections*. The *TPendingConnection* structure includes an MsgID member that specifies a unique identifier to be used in all signalling messages associated with that pending connection. The Type member indicates the type of signalling pending; e.g. call set-up or call release. If a connection has been pending for longer than a specified period then it is destroyed. For this reason *TPendingConnection* also includes a TimeOut member. The VC member points to a *TConnection* structure representing the pending connection. For an example of how the *PendingConnections* list is used consider the call set-up process. When the BS receives a call set-up message from the ATM network it first checks with the *Wireless CAC* to make sure it can handle the call, and then it creates a *TConnection* structure with its parameters initialised to the requirements of the call. This is placed in a *TPendingConnection* structure and added to the *PendingConnections* linked-list. A call set-up message is passed on to the MT. If no call-connect message is received from the MT before the timeout expires then the *TConnection* and *TPendingConnection* structures are destroyed. Similarly, if the MT returns a call-release message then both structures are destroyed. However, if a call-connect message is received from the MT then the *TConnection* structure is placed in the *Connections* linked-list, and the *TPendingConnection* structure is destroyed.



***TPendingConnection* Structure**

VC	TConnection*	Pointer to the connection with signalling pending.
Type	u16	The type of signalling pending.
MsgID	u16	Unique identifier to be used in signalling messages.
TimeOut	u16	The time at which to drop the connection if it is still pending.

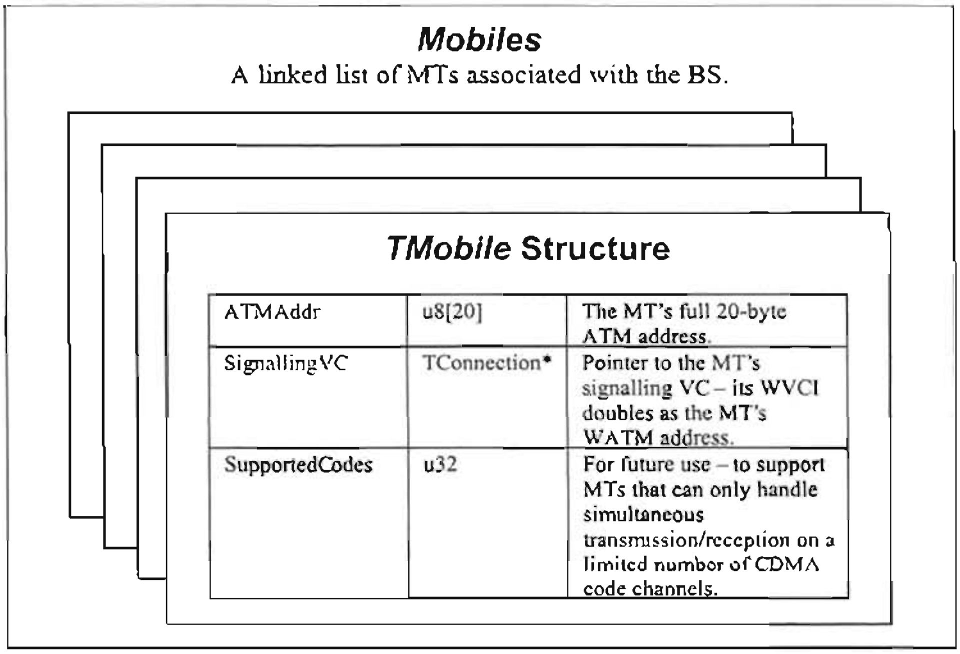
Figure 5.7 The *TPendingConnection* structure.

The *Connection Container* includes a library of functions that provide an interface for the other blocks in the system to create and destroy connections, as well as add, locate and remove connections from the *Connections* list. Similar functions also exist for pending connections.

Keeping Track of Mobile Terminals

A BS needs to maintain a record of all the MTs associated with it, as well as all the VCs associated with each MT. This is so that when a MT disassociates from a BS, all VCs connected to the MT may be redirected to the MT's new BS. The BS keeps track of MTs in a linked-list of *TMobile* structures called *Mobiles* (Figure 5.8). Every MT that associates itself with a BS is added to the BS's *Mobiles* linked-list. Thereafter, each VC that is connected to the MT is represented by a *TConnection* structure, which contains a pointer to a *TMobile* structure, linking the VC to its parent MT.

Each MT also has a special signalling VC associated with it. Signalling VCs are used to carry all signalling traffic between a BS and a MT. The WVCI assigned to the signalling VC also doubles as the MT's WATM address and is used in place of its full 20-byte ATM address in the wireless signalling. In fact, because each MT has its own signalling VC with a unique WVCI, a signalling cell's WVCI value may be used to indicate the MT's address without requiring an additional field for this in the signalling message. Signalling VCs are represented by *TConnection* structures in the *Connections* linked-list just like any other VC, but each MT's *TMobile* structure also contains a pointer to its signalling VC, allowing it to be immediately located for transmission of signalling messages. The *Mobile Container* block depicted in Figure 5.4 contains the *Mobiles* linked-list as well as a number of functions for other blocks to interface with the *Mobile Container*. These functions allow for the dynamic creation and destruction of *TMobile* structures, as well as functions to add, locate and remove MTs from the linked-list.



ATMAddr	u8[20]	The MT's full 20-byte ATM address.
SignallingVC	TConnection*	Pointer to the MT's signalling VC – its WVCI doubles as the MT's WATM address.
SupportedCodes	u32	For future use – to support MTs that can only handle simultaneous transmission/reception on a limited number of CDMA code channels.

Figure 5.8 *Mobiles* **– a linked-list of** *TMobile* **structures.**

Buffers for Queued Cells

Each *TConnection* structure (Figure 5.5) contains a pointer to a cell buffer that is used to queue cells awaiting transmission or retransmission. Arriving cells are added to the tail of the buffer, and cells are only removed from the head of the buffer once their correct reception has been acknowledged, or their cell lifetime (CLT) expires. In this sense the buffer is a FIFO (first in first out) buffer, and hence the structure used to represent the buffer has been called FIFO. However, the cell buffer module is more than just a simple FIFO buffer. It also keeps track of which cell is next in line to be transmitted over the wireless link. This is done via a cursor that points to the correct cell. Each cell's CLT expiry is tracked and enforced by the cell buffer. The buffer also handles the cell sequence numbering, and ensures that a retransmission starts from the correct cell when the go-back-n ARQ comes into play. Figure 5.9 illustrates the format of the VC cell buffer.

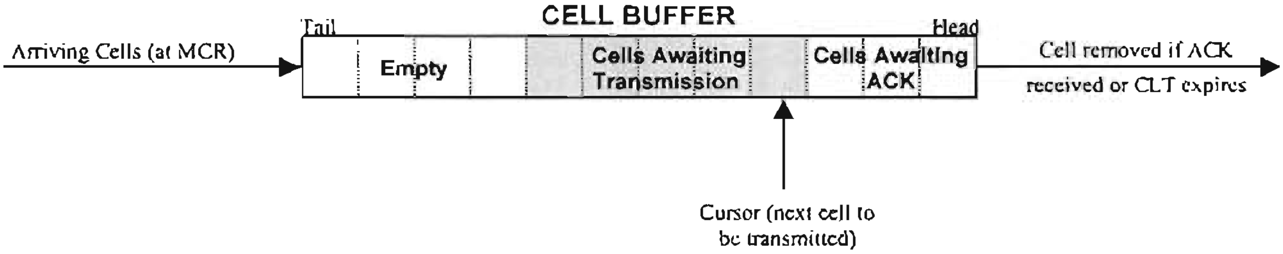

CELL BUFFER
Tail
Head
Arriving Cells (at MCR)
Empty
Cells Awaiting Transmission
Cells Awaiting ACK
Cell removed if ACK received or CLT expires
Cursor (next cell to be transmitted)

Figure 5.9 The virtual channel cell buffer.

The size of a cell buffer is fixed for the duration of a VC's existence. This is for performance reasons as it avoids the constant allocation and de-allocation of memory. Instead of shifting along all the cells in the buffer memory every time a cell is removed from the head, a circular buffer format is used with pointers keeping track of the head and tail within the buffers allocated memory.

Since each buffer's size is fixed once it has been created, it is important that an appropriate size is specified when creating the buffer. Obviously the buffer must be large enough to prevent buffer overflows, but it mustn't be excessively large so as to waste memory. The buffer size required is going to depend on the connection's mean cell rate (MCR) and CLT. For CBR traffic, cells are added to the buffer's tail at a rate equal to the MCR of the connection, and there could potentially be a period equal to the connection's CLT during which no cells are removed from the buffer. So, the buffer would have to be able to hold at least MCR × CLT cells if a buffer overflow is to be avoided. This situation could also be used to approximate the VBR case with some safety factor, perhaps dependant on the maximum burst size (MBS), used to increase the buffer size by a certain factor in order to cater for bursts above the MCR present in VBR traffic. Eq. 5-1 is used to determine the buffer sizes allocated to each CBR and VBR connection:

$$BUFFER\ SIZE = \lceil MBS + MCR \times CLT \rceil \qquad \textbf{Eq. 5-1}$$

where the units of MBS is cells, MCR is cells/s, and CLT is seconds. In fact this equation could be used for all traffic types. For ABR and UBR traffic it is more difficult to predict what the necessary buffer size is. These traffic classes will be used for bursty data so the MBS will need to be included in the equation to calculate the buffering needed. Because the ABR and UBR traffic classes are aimed more at best effort services than real-time services, they will tend to have less stringent delay requirements and, hence, longer CLTs. If the formula suggested above is again used to calculate the cell buffer size, these CLTs may lead to excessively large cell buffers being required in order to guarantee prevention of buffer overflows. Some upper limit on a VC's cell buffer size may be required, however the 16Mb of SDRAM available in the DSP board does provide plenty of memory for cell buffering. The simulation and implementation results presented in the next chapter further investigate the required cell buffer sizes. Some examples of cell buffer sizes calculated using Eq. 5-1 are shown below in Table 5-I.

Table 5-I Example cell buffer sizes for various traffic parameters.

Mean Bit Rate (kb/s)	MCR (cells/s)	CLT (ms)	MBS (cells)	Buffer Size (cells)
9.6	21.4	50	2	4
9.6	21.4	100	10	13
9.6	21.4	200	25	30
33.6	75.0	50	2	6
33.6	75.0	100	10	18
33.6	75.0	200	25	40
64	142.9	50	2	10
64	142.9	100	10	25
64	142.9	200	25	54
128	285.7	50	2	17
128	285.7	100	10	39
128	285.7	200	25	83

5.3.2 The BS Scheduler

In order to efficiently implement the scheduling algorithm, the characteristics of each WATM VC must be readily available to the scheduler. For example, the scheduler needs to know how many uplink and downlink tokens a VC has, as well as the number of uplink and downlink cells waiting to be sent. The VCs need to be grouped according to the manner in which the scheduler needs to access them. More specifically, the algorithm requires the VCs to be accessed according to their priority (or traffic class), as well as the number of tokens in the VCs token pool. This needs to be done separately for the uplink and the downlink. The *Connections Container* does not provide efficient access to VCs according to these criteria. The scheduler will also need to keep track of the number of waiting cells. As it allocates slots to VCs the scheduler will need to decrease the waiting cell count, but it can't do this in the actual *TConnection* structure representing the VC, as this would be equivalent to indicating that the cell has already been transmitted. The number of waiting cells specified in the *TConnection* structure must only be updated when the cell as actually transmitted, not when it is scheduled. On the other hand, the uplink and downlink token counts located in the *TConnection* structure can be updated by the scheduler as it allocates slots to VCs.

VC Representation for the Scheduler

In order to deal with the issues mentioned above, temporary scheduling structures, grouped according to VC priority, are filled with only those VCs that have cells to transmit. Separate structures are filled for the uplink and the downlink. This means that when the scheduler is choosing the next VC to assign a cell to it only needs to search through those VCs with the specific priority being scheduled at the time, and only VCs that actually have cells to transmit. Only one pass through all the VCs in the *Connections Container* is needed to fill the temporary structures and thereafter only passes through relevant VCs are made in order to carry out the scheduling process. This is as opposed to a pass through all the VCs in the *Connections* linked-list every time a slot is allocated. The temporary scheduling structures are located in the C6201's internal data memory rather than the slower access SDRAM where the *Connections* linked-list is stored. A VC is represented in the temporary scheduling structure by a pointer to its *TConnection* structure in the *Connections* linked-list, as well as an integer containing the number of cells the VC has queued. This allows the scheduler to decrement the VCs token count in the *Connections Container*, but keep track of the number of waiting cells without affecting the counts in the *Connections Container*.

Scheduling Algorithm Pseudo Code

A pseudo code version of the scheduler is presented below. This implements the scheduling algorithm described in Chapter 3. The pseudo code implements 'fullest token pool' scheduling, as opposed to the 'most tokens' scheduling also mentioned in Chapter 3. The slot allocations are performed according to the BER scheduling technique described in Chapter 3. The timeslot before a new MAC frame begins a function called `fill_scheduler_connections` is run to place references to appropriate VCs in the temporary scheduling structures as explained above. This function has not been included in the pseudo code below. The temporary scheduling structures are used by the scheduler to efficiently locate VCs of specific priorities with cells waiting. The scheduler then runs the `build_downlink_frame` and `build_uplink_frame` functions, which iterate through the temporary scheduling structures – choosing the next VC to be assigned a slot in each iteration. The allocate_downlink_slot_to_connection and allocate_uplink_slot_to_connection functions are called to assign appropriate downlink and uplink slots to the chosen VC respectively. Only the build_downlink_frame and allocate_downlink_slot_to_connection pseudo code functions are presented below because the uplink versions of the functions are almost identical. The main difference being that for the uplink, reservation requests are used to determine the number of waiting cells each VC has instead of the actual cell counts that can be used for the downlink.

```
Note: As defined in Appendix A.2.2, a VC's priority is related to its traffic class with
the CBR class awarded the highest priority of 1, followed by rt-VBR, nrt-VBR, ABR and
finally UBR with the lowest priority of 5.
-------------------------------------------------------------------------------------------
Function build_downlink_frame()

/* FULLEST TOKEN POOL SCHEDULING */
For Priority = 1 to 5
{
   For TokenPoolLevel = 10 to 1
   {
      While (slots are being allocated) Do
      {
         For each VC with dlTokens/dlTokenPoolSize = TokenPoolLevel
         {
            If the VC has a cell to transmit
               Then allocate_downlink_slot_to_connection()
         }
      }
   }
}

/* FRAME FILL-UP */
For each slot in downlink frame
{
   If slot is not full
      Then find a virtual channel with a cell to transmit that will fit into the slot.
}
-------------------------------------------------------------------------------------------
Function allocate_downlink_slot_to_connection()

For each timeslot in the downlink frame
{
   If (slot is empty) or (slot contains cells of same BER requirement & slot not full)
      Then
      {
         allocate free code slot to the VC.
         Decrease VC's downlink token count.
         Exit function
      }
}
For each timeslot in the downlink frame
{
   If (slot contains cells with more stringent BER requirement & slot not full)
      Then
      {
         allocate free code slot to the VC.
         Decrease VC's downlink token count.
         Exit function
      }
}
For each timeslot in the downlink frame
{
   If (slot contains cells with less stringent BER requirement & slot can be converted to
       BER requirement of the VC without becoming too full)
      Then
      {
         allocate free code slot to the VC.
         Decrease VC's downlink token count.
         Exit function
      }
}
-------------------------------------------------------------------------------------------
```

The scheduling procedure, as implemented, does have a potential problem in that it may consistently choose one VC before another even when they both have the same token count. This is because the order in which the scheduler searches for the next VC to be assigned a slot is dependant on the order of the VCs in the *Connections* linked-list. This may lead to VCs closer to the head of the list tending to get preference over VCs closer to the tail. Of course, if a VC loses out in one MAC frame it will have more tokens in the next frame than the VC to which it lost out. So, one can argue that this is made up in the next frame. However, the next frame may be too late – the cell's CLT may have caused it to be dropped by then. A partial solution to this problem, would be to pass from head-to-tail through the *Connections* linked-list on every even frame, and from tail-to-head on every odd frame, when filling the temporary scheduling structures. An even better solution would be to start passing through the list from the VC that would have been assigned the next slot had the frame not been full. This problem only takes effect when there is congestion over the wireless link and multiple consecutive MAC frames are completely full. In reality the CAC should prevent this situation from occurring.

Updating VC Token Pools

One other important aspect related to the scheduler that needs to be dealt with is how the number of tokens in each VC's token pool is updated. Tokens need to be added to the token pool at a rate equal to the connection's MCR, which is specified as a 16-bit fixed-point number in cells per second. Of the 16-bits, 10-bits represent the integer portion of the number and 6-bits represent the fraction. In other words, the number is scaled by 64 resulting in a MCR with a resolution of 0.015625cells/s. One token is removed from a VC's token pool for every slot assigned to the VC. This is carried out easily enough in the `allocate_downlink_slot_to_connection` and `allocate_uplink_slot_to_connection` functions described above. However, some sort of timer is going to be necessary to add tokens to token pools at the correct rate.

The TMS320C6201 DSP processor has two 32-bit timers/counters available. They may be configured to increment a 32-bit counter at a quarter of the CPU clock rate. A 200MHz clock is used on the DSP board so the counter register increments at 50MHz, which means the timer period is 20ns. Being a 32-bit counter it takes 85.899 seconds ($2^{32} \times 20$ns) to overflow. Because the scheduling algorithm only runs once every frame, the token counts only need to be updated once every frame. If one of the C6201's timers/counters is configured to function as described above then the tokens in each VC's token pool may be updated using this nanosecond counter.

Another method to update the token counts would be to use the timing associated with the MAC frame. The scheduler runs once every frame and every frame is a fixed length. So, if token updates are performed before every scheduler run, then they will be performed at regular intervals equivalent to the frame length. This negates the need for another timer to be set up. A pseudo code example that takes into account the fact that the MCR is stored as a 16-bit fixed-point number is shown below.

```
Function update_connections_tokens()

TokenUpdateTime += SECONDS_PER_FRAME
TokenIncrease = Integer(TokenUpdateTime * MCR) >> 6
TokenUpdateTime -= (TokenIncrease << 6) / MCR;
Tokens += TokenIncrease
If Tokens > MBS Then Tokens = MBS
```

5.3.3 Wireless DLC

The main function of the Wireless DLC is to implement the windowed go-back-n ARQ on a per VC basis as described in Chapter 4. However, not every VC makes use of the ARQ scheme, and some of the specialised cells need to be dealt with in their own special way. Every cell received over the air interface is passed through the DLC, which handles them in the following manner:

- *Select Handler*: Each cell is passed to an appropriate handler according to the slot in which it was received i.e. depending on the WVCI allocated to the slot, the cell is passed to a Frame Header handler, Association Request handler, Association Receive handler, VC Update handler, or Standard Cell handler. Note that some of these handlers only apply to MTs or BSs.

- *Apply FEC*: The handler applies FEC if appropriate. Although no FEC has been implemented at this stage, the frame header and VC update cells do have the potential to include FEC.

- *Perform Error Detection*: The handler uses the 16-bit CRC field included in all cell types to perform error detection.

- *Handle Cell*: The handler then deals with the cell according to its type. For example, the Standard Cell handler would locate the *TConnection* structure associated with the VC in order to determine whether or not the VC is making use of the ARQ scheme. If errors were detected in the cell and it is using ARQ, then the cell is dropped and a flag is set in the *TConnection* structure indicating that a retransmission is required. If, however, the VC has been configured to accept bit errors, then the cell is accepted and routed to its destination after being converted from the WATM cell format to the ATM cell format. If, on the other hand, no errors were detected in the cell, then any slot reservation requests or sequence number acknowledgements piggybacked onto the cell are removed before routing the cell to its destination.

Probably the most involved part of the DLC is keeping track of cell sequence numbers in order to implement the go-back-n ARQ while maintaining the cell sequence. The path of sequence numbers and sequence number ACKs through the wireless system is illustrated in Figure 5.10. Every *TConnection* structure contains a txSNack and an rxSNack variable. TxSNack holds the next sequence number acknowledgement that needs to be transmitted. If a VC receives a cell whose sequence number doesn't succeed the sequence number in txSNack then the cell is out of sequence, and the sequence number in txSNack is re-acknowledged. RxSNack contains the last sequence number acknowledgement received. If the next ACK received by the VC is the same as rxSNack then a retransmission is triggered. In other words, the cell transmissions resume from the cell with the sequence number succeeding rxSNack.

This process may seem fairly straight forward, but complications appear when a VC's cell lifetime causes cells to be dropped before they've been acknowledged, or possibly even before they've been transmitted at all. This requires some careful cell renumbering, the details of which are not included.

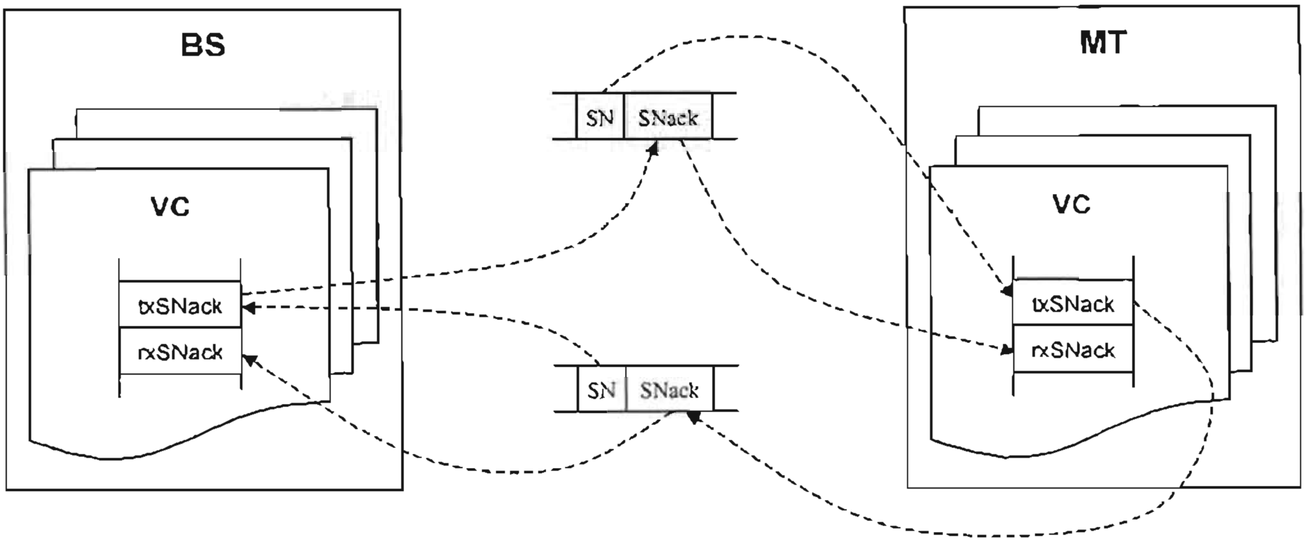


Figure 5.10 The sequence number path.

5.3.4 Physical Layer Interface (SWR – DSP)

As previously mentioned the McBSPs provide interfaces to an MVIP bus and an SCbus. Each bus provides four bi-directional 2Mbps links with the physical layer board. Each of these 2Mbps links is divided into frames of 32 timeslots with each timeslot holding a single byte (8-bits). Only the SCbus is used in the implementation, although the MVIP bus could be configured in an identical fashion. Appendix C.1 describes the SCbus in more detail. As illustrated in Figure 5.11, the DSP board contains an SC4000 switching device that is configured to receive data at 8Mbps from the McBSP and de-multiplex it into the four 2Mbps streams on the SCbus, as well as receive data from the four 2Mbps streams and multiplex them into an 8Mbps stream going to the McBSP. The four SCbus data channels are denoted SD0, SD1, SD2 and SD3. The (de)multiplexing is performed on a byte level, so every fourth byte that the McBSP transmits/receives corresponds to data from a single 2Mbps SCbus channel. Furthermore, of these fourth bytes, every 32nd byte corresponds to one of the 32 timeslots in the SCbus frame.

Each timeslot in an SCbus channel effectively offers 64kbps (2Mbps divided by 32 timeslots). Initially the physical layer will only support 64kbps data rates over each CDMA code channel. By only making use of SD0, each timeslot is used to stream the data for one CDMA code channel to and from the physical layer at 64kbps. Thus, a maximum of 32 CDMA code channels is supported in the MAC layer. The MAC layer does, however, support higher bit rates over each of the 32 CDMA code channels. This is achieved by using SD0 and SD1 to achieve a 128kbps bit rate. By combining SD0, SD1 and SD2 a bit rate of 192kbps is achievable, and so on. The scheme used to de-multiplex the data stream received at the McBSP is illustrated in Figure 5.12.

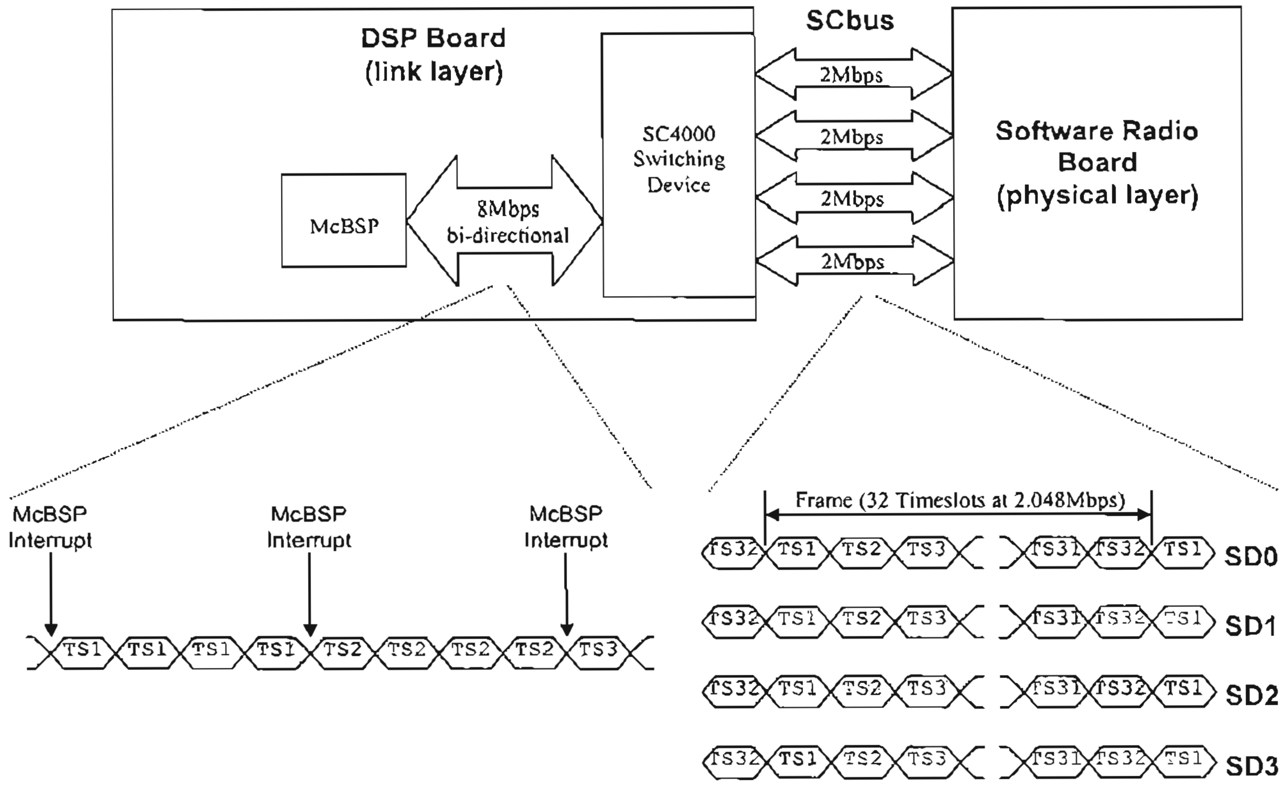


Figure 5.11 The MAC Layer – PHY Layer Interface

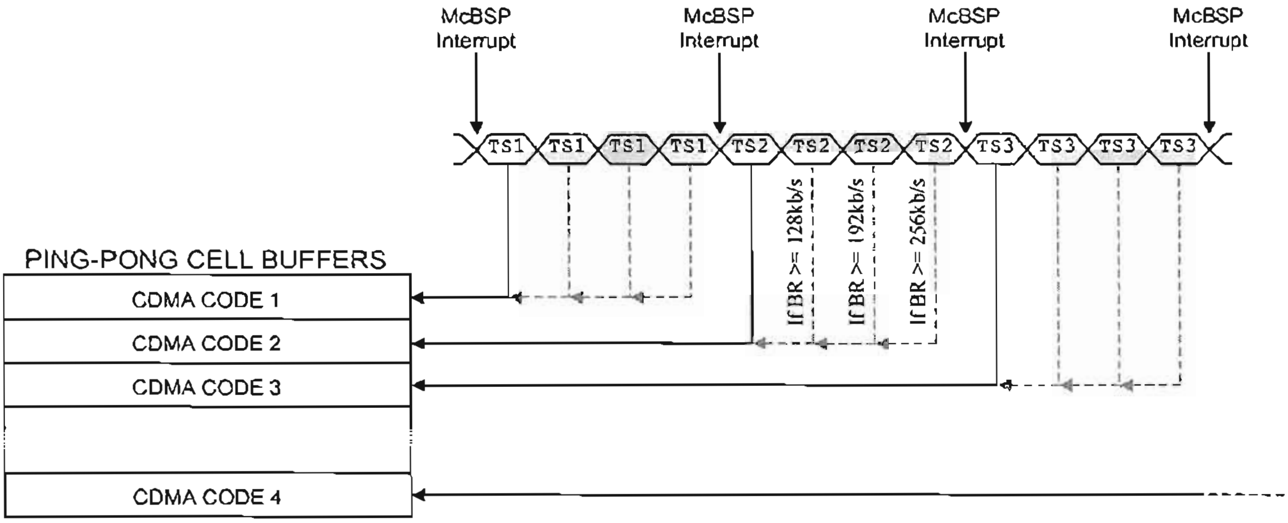


Figure 5.12 De-multiplexing the received data stream.

The above configuration means that the McBSP must be configured for a data rate of 8Mbps. It is also configured to generate transmit and receive interrupts after every 4 bytes (32-bits) are transmitted or received respectively. The receive interrupt service routine (ISR) keeps track of the current timeslot, placing the received byte(s) into the appropriate cell buffers. If the bit rate is set to 64kbps then only one of the four bytes received in each interrupt is valid. For a bit rate of 128kbps, two of the four bytes received in each interrupt are valid, and so on. A ping-pong cell buffering scheme is used to receive cells on each CDMA code channel. This means that while the next cell is being received from the McBSP into one section of the buffer, the cell that was previously received into the other section of the buffer is processed and routed to its destination. The ping-pong buffering concept is explained in more detail in the next section. A similar scheme is used by the transmit ISR to transmit cells on each code channel, except that the cells are routed directly from their respective VC cell buffers instead of from specialised transmit buffers.

5.3.5 ATM Network Interface (Host PC – DSP)

Each DSP board is located in a host PC. This host PC acts as source and sink for ATM cells. On the BS side the host PC imitates an ATM network, allowing VCs to be set-up to carry a variety of traffic models in order to test the performance of the wireless link layer protocol. On the MT side the host PC functions as the end user's terminating node. The DSP Board includes a 32-bit slave PCI interface that is used for rapid data transfers between the board and the host PC. A PLX Technology PCI9060 bridge device is used to provide this PCI interface [51]. The entire memory space of the C6201 DSP processor can be accessed from the PCI bus via the processor's host port interface. The PCI9060 also includes a two channel DMA controller that can be configured to transfer data between C6201 memory and the PCI bus.

Two 8-bit mailbox registers are provided for the purpose of passing application-specific messages between the host PC and the C6201. These registers are mapped into the C6201 memory and, hence, into the slave PCI interface as well. Each mailbox register can only be used to pass messages (represented by 8-bit numbers) in one direction. On the C6201 and the host PC the arrival of a new message can be used to trigger an interrupt. The host communications interface library provided with the DSP board includes functions to send and receive mailbox messages and carry out data transfers with or without the use of the PCI9060's DMA controller. If a large block of data is to be transferred then it is quicker to perform a DMA transfer, but if only a single word is to be transferred then an ordinary memory transfer is more efficient due to the time taken to set up a DMA transfer. Using these application-specific interrupts and data transfers, the flow of ATM cells between the DSP board and host PC is controlled.

A simple manner in which to transfer cells across the PCI bus would be to use a basic buffering method. At the transmitting end, cells would be added to the buffer until it is full. A mailbox message would then be sent to the receiving end indicating that the buffer is full. The receiving end would then remove all the cells from the buffer and send a mailbox message back to the transmitting end, indicating that the next block of cells may be placed in the buffer. Such a buffering scheme is inefficient in that only one of the ends is accessing the buffer at any one time. To improve the buffering scheme ping-pong buffers are used. A ping-pong buffer is divided into two buffer sections called the ping buffer and the pong buffer. The idea being that while the transmitting end is placing cells in the one buffer section, the receiving end is removing cells from the other buffer section. As soon as the transmitting or receiving end has completed its processing it sends a mailbox message to the other end indicating this. When both ends are ready the two ends switch buffer sections. By alternating between the ping and pong buffers in this way the amount of time each end spends idle is kept to a minimum.

Of course, with the ping-pong buffer being used for real-time cell transfers, waiting for a buffer section to fill up before switching sections would impose an unnecessary delay on cells, especially if the rate at which cells are being generated is low. To prevent such delays a buffer switch is triggered as soon as the receiving end is ready and the transmitting end has generated at least one cell.

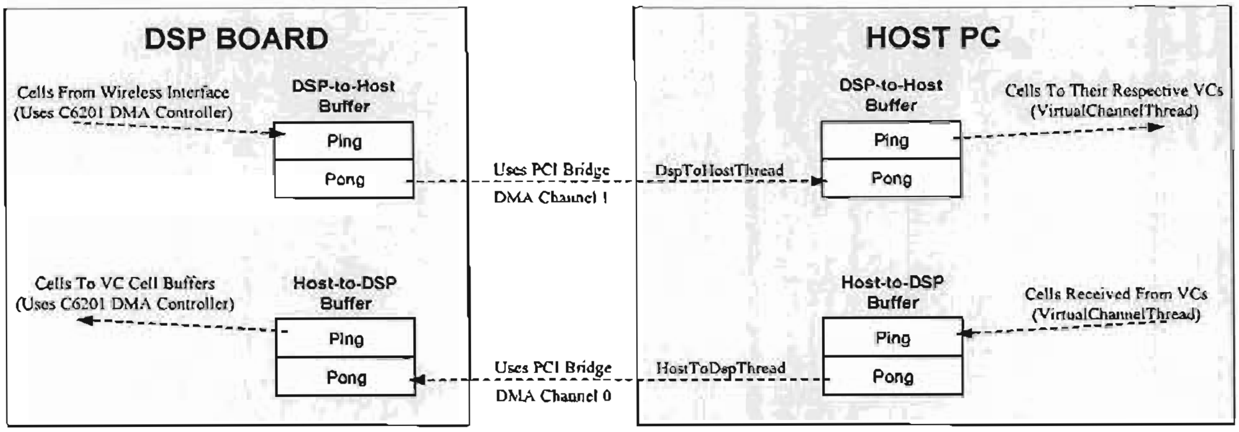


Figure 5.13 Host PC – DSP Data Flow

Two ping-pong buffers are configured for the purpose of transferring cells across the PCI bus – one for each direction. Figure 5.13 illustrates the operation of these buffers. One of the PCI9060 DMA channels is used for transfers from the host PC to the DSP board, and the other is used in the opposite direction. On the host PC separate threads are used to control the transfers in each direction.

5.3.6 Signalling Control – A Simple Signalling Protocol

Although the signalling control does not form part of the link layer, it is necessary to implement a signalling protocol in order to set-up and release VCs to test the link layer protocol. Instead of implementing the entire ATM UNI signalling protocol, which would take some time, a simple custom signalling protocol is implemented. The WATM signalling messages used over the wireless link are compressed to fit into single ATM cell payloads. This is made possible by the use of 16-bit WATM addresses instead of the full 20-byte ATM addresses. As previously mentioned, each MT is assigned a 16-bit signalling WVCI upon association with a BS, which doubles as a WATM address for the MT. The ATM signalling messages used between the DSP board and the host PC are different to the WATM messages used over the wireless interface, but only the WATM messages are discussed below. The following WATM signalling messages are defined…

1	1	20	2	1	2	2	2	1	1
MsgType 01h	MsgID	ATMAddress (of Src/Dst)	New WVCI	UL Traffic Class	UL MCR	UL MBS	UL CLT	UL BER	UL ABE

1	2	2	2	1	1	6
DL Traffic Class	DL MCR	DL MBS	DL CLT	DL BER	DL ABE	RSV

Figure 5.14 WATM_CALL_SETUP signalling message.

Figure 5.14 shows the structure of the WATM_CALL_SETUP message. All WATM signalling messages have *MsgType* as their first field. This is used to specify the type of signalling message e.g. call set-up, call connect, call release, etc. The *MsgID* field contains a unique identifier used to link all messages relating to the same VC. The *NewWVCI* field is set to 0000h when the MT initiates the call as it is up to the BS to assign WVCIs to each connection. The BS will indicate the WVCI to be used when it sends the WATM_CALL_CONNECT message to the MT. When another terminal has initiated the call and the BS is forwarding the call set-up message to the MT, the BS specifies the WVCI to be used for the VC in the *NewWVCI* field. Note that an ATM address of the source is not required in a MT to BS WATM_CALL_SETUP message, as the BS will know the ATM address of the source MT from the signalling WVCI used to transmit the cell – each MT has a unique signalling VC associated with it. So, when the WATM_CALL_SETUP message is transmitted from MT to BS the *ATMAddress* field contains the ATM address of the destination. However, in the BS to MT direction the ATM address of the destination isn't required because the MT that receives the message will know that it is the destination. So, for the BS to MT direction the *ATMAddress* field contains the source's ATM address. The remaining fields contain the traffic parameters required by the user. (See Appendix A.2.3 for details.) The call set-up message contains separate traffic parameters for the uplink and the downlink directions. The WATM connections are bi-directional by default, but the uplink and downlink may have completely different traffic parameters – even different traffic classes.

1	1	2	44
MsgType 02h	MsgID	New WVCI	RSV

Figure 5.15 WATM_CALL_CONNECT signalling message.

Figure 5.15 shows the structure of a WATM_CALL_CONNECT signalling message. The BS or MT that sends this message will do so in response to the reception of an ATM_CALL_CONNECT message. The BS/MT that transmits the CALL_CONNECT message will also activates the VC to which it applies on its end of the link. The other end of the link will be activated by the BS or MT that receives the message. The unique identifier contained in the *MsgID* field is used to identify the correct VC in the *PendingConnections* linked-list to activate. The newly connected VC has its WVCI set to the value held in the *NewWVCI* field of the WATM_CALL_CONNECT message. It is always the BS that determines the WVCI value, so if the WATM_CALL_CONNECT message is sent from BS to MT then the BS chooses an available WVCI to place in the NewWVCI field; otherwise, if a MT is sending the WATM_CALL_CONNECT message it uses the NewWVCI specified in the original WATM_CALL_SETUP message it received. This would have been sent by the BS and hence, the BS would have chosen the WVCI value.

In order to terminate a call, or turn down a call set-up request, a WATM_CALL_RELEASE message is sent. The structure of a WATM_CALL_RELEASE message is very similar to the WATM_CALL_CONNECT message. The main parameter included in the message is the WVCI of the connection being released. For the case where a call set-up request is being rejected and a WVCI hasn't been assigned to the connection yet, the *MsgID* field identifies the pending VC to which the cell release message applies. A MT or BS will respond to a call release message with a WATM_CALL_RELEASE_COMPLETE message, which is also very similar to the WATM_CALL_CONNECT message.

Figure 5.16 and Figure 5.17 illustrate the call set-up and call release process. Although the signalling functionality that has been implemented is fairly crude, it provides the ability to set-up and release VCs, which is all that is required to test the protocol.

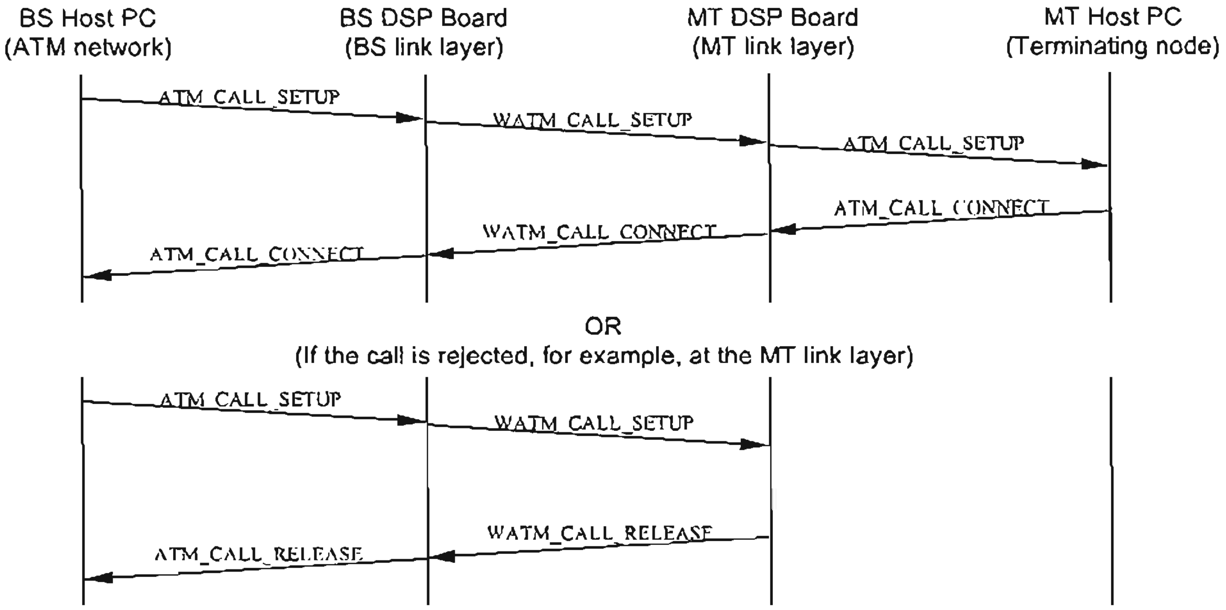


Figure 5.16 Setting up a new virtual channel.

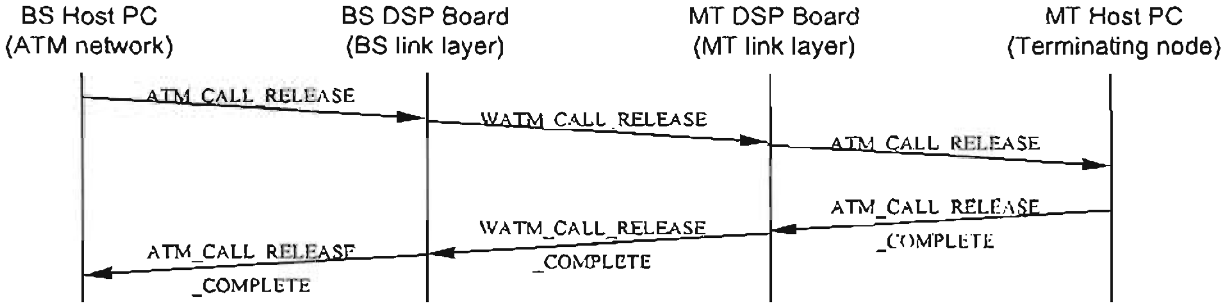


Figure 5.17 Releasing an existing virtual channel.

5.4 Host PC Software Overview

The main functions of the host PC, on both ends of the wireless link, include terminating the ATM connections, allowing connections to be set up and released, and generating traffic to send over each connection in order to test the performance of the WATM link layer protocol. The host PC is also required to gather performance statistics from the DSP board. In order to carry out these requirements in real-time a multi-threaded Windows NT 4.0 application was developed using Borland C++ Builder.

Figure 5.18 contains a block diagram of the structure of the host PC software. There are three main blocks of code in the software. The *DSP Interface* handles all communications with the DSP board. To do this, the software uses Blue Wave System's host communications interface library (HCIL) provided with the DSP board. The DMA based ping-pong buffering scheme used to transfer ATM to/from the DSP board was described in Chapter 5.3.5, but the *DSP Interface* does more than just the transfer of ATM cells. It also retrieves statistics on the performance of the link layer e.g. the average and maximum times taken for the scheduler to build a MAC frame. Two threads are used to control the DSP interface: the *DSP-To-Host* thread controls ATM cell transfers to the DSP board, and the *Host-To-DSP* thread controls ATM cell transfers from the DSP board. The HCIL functions only allow one thread within a single application can access a particular DSP system at any one time. Therefore, these two threads use critical sections to prevent simultaneous access of the DSP board.

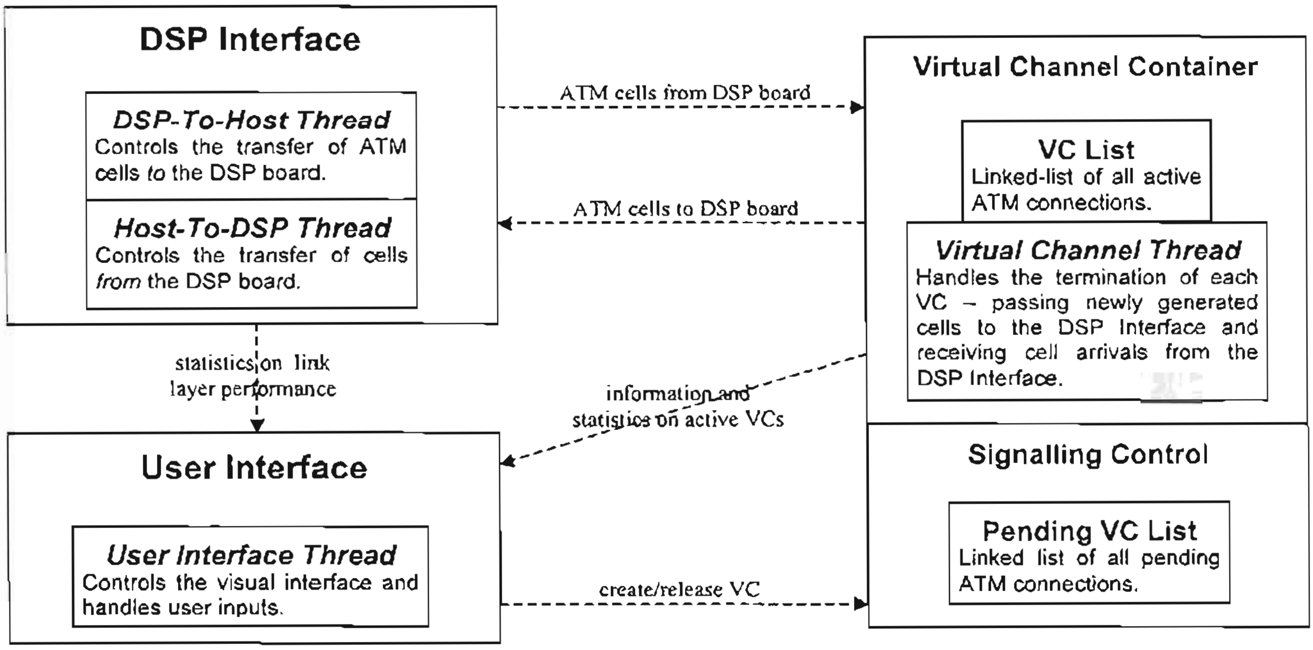

DSP Interface
DSP-To-Host Thread
Controls the transfer of ATM cells *to* the DSP board.
Host-To-DSP Thread
Controls the transfer of cells *from* the DSP board.
ATM cells from DSP board
ATM cells to DSP board
Virtual Channel Container
VC List
Linked-list of all active ATM connections.
Virtual Channel Thread
Handles the termination of each VC – passing newly generated cells to the DSP Interface and receiving cell arrivals from the DSP Interface.
statistics on link layer performance
information and statistics on active VCs
User Interface
User Interface Thread
Controls the visual interface and handles user inputs.
create/release VC
Signalling Control
Pending VC List
Linked list of all pending ATM connections.

Figure 5.18 Block diagram of the host PC software.

All ATM cells received from the DSP board are passed on to the *Virtual Channel Container*, where they are routed to their appropriate VCs. All active VCs are represented by a TVirtualChannel class and stored in a linked-list called *VC List*. The parameters with which a TVirtualChannel class is instantiated determines the VCs traffic parameters and hence the rate at which it generates new cells. A number of classes are derived from TVirtualChannel to implement traffic with different characteristics. For example, classes exist to implement voice, video, audio and computer data traffic. The characteristics of these traffic types are described in the next chapter. A Virtual Channel thread controls the generation of new cells by each VC, and passes these cells on to the *DSP Interface*.

The *Virtual Channel Container* is closely associated with the *Signalling Control*. The *Signalling Control* receives requests to create and release VCs from the *User Interface*. These requests are handled by generating the appropriate signalling messages and adding new VCs to the *Pending VC List*, or moving existing VCs from the *VC List* to the *Pending VC List* respectively. The *Signalling Control* also handles signalling messages received from the DSP board.

Finally, the *User Interface* provides a visual display of active VCs with their average transmission rates and received throughput rates. These are retrieved from the *Virtual Channel Container*. Other statistics such as the number of available slots and the processing time required by the scheduler are retrieved from the *DSP Interface*.

Chapter 6 How does the Protocol Perform?

In order to investigate the performance of the proposed link layer protocol, results from both simulations and the actual implementation are used. The simulation model was developed using Mil3's Opnet Modeler 6. A number of different traffic models, each generating traffic with distinct characteristics and requirements, are used to test the protocol. This chapter presents the traffic models, followed by the simulation and implementation parameters used and a discussion of the results.

6.1 The Traffic Models

The following traffic models were used to generate traffic in order to test how the protocol handles various traffic characteristics. The traffic models are based on those used in [35]. The simulation assumes a higher raw bit rate over each CDMA code channel than is used in the actual implementation, so some of the traffic models generate higher cell rates for simulation runs than for implementation runs. These differences are explained in the traffic model descriptions below.

6.1.1 VBR Voice Traffic

The purpose of the voice traffic model is to generate speech traffic with the characteristics of an ordinary telephone conversation. A typical conversation contains periods of talking and listening, so this model generates patterns of talk-spurts and gaps. Inside each talk-spurt there are mini-spurts and mini-gaps that model the short spurts and pauses that punctuate continuous speech. It is assumed that the length of the conversations, as well as all spurts and gaps, have exponentially distributed durations and the durations are statistically independent of each other. A 9.6kbps data rate is generated during the mini-spurt periods of a conversation. If, however, the WATM connection carrying the voice call is configured to accept bit errors rather than use the go-back-n ARQ scheme, then it is assumed that redundant FEC information must be included in the voice traffic. A half-rate FEC scheme is assumed resulting in a 19.2kbps data rate being produced during each mini-spurt. The average durations of the spurts and gaps are shown in Table 6-I.

Table 6-I Average durations for the voice traffic model.

Parameter	*Average Duration (s)*
Principal Talk-spurt	1.000
Principal Gap	1.350
Mini-spurt	0.275
Mini-gap	0.050

6.1.2 VBR Video Traffic

The video traffic model attempts to imitate the traffic characteristics of a variable rate video signal that may be used in a videophone or videoconferencing application. Each state of this multiple state model generates a continuous bit stream for a certain holding time. The state's holding durations are determined from an exponentially distributed random variable, and assumed it be statistically independent. The bit rate generated in each state is obtained from a truncated exponential distribution with minimum and maximum bit rates defined. Table 6-II shows the values used for the various model parameters. Note that different values are used for the simulation and implementation.

Table 6-II Parameter values for the video traffic model.

Parameter	*Simulation Value*	*Implementation Value*
Mean State Holding Time	160 ms	160 ms
Maximum Bit Rate	128 kbps	64 kbps
Mean Bit Rate	96 kbps	48 kbps
Minimum Bit Rate	64 kbps	32 kbps

6.1.3 CBR Audio Traffic

This is a fairly simple traffic model that generates a continuous bit stream to simulate a constant bit rate audio source such as digital FM stereo audio. For the simulation a constant 128kbps stream is generated, whereas the implementation is tested with a 52kbps stream.

6.1.4 ABR Computer Data Traffic

The computer data traffic model is a straightforward model characterising traffic from an application such as web browsing. Data message lengths are assumed to be exponentially distributed with a mean length of 10kbytes. The message inter-arrival time is also generated from an exponential distribution with the mean time between data messages equal to 60 seconds.

6.1.5 ABR Email Traffic

E-mail traffic is modelled using the size distribution shown in Figure 6.1. This distribution is an approximation of the empirical size distribution used in [35]. Each e-mail message's size is generated according to the distribution with an average size of approximately 3000 bytes.

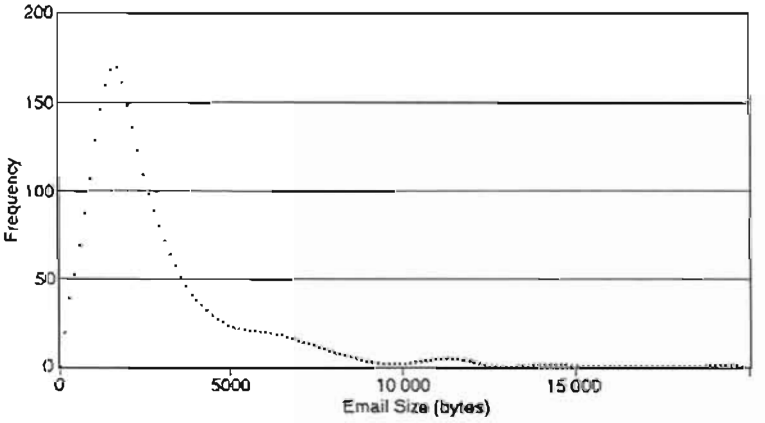


Figure 6.1 E-mail message size distribution.

6.2 Simulation Results

6.2.1 Simulation Assumptions and Parameters

The simulation assumes 8 CDMA code channels are available for use, with a raw bit rate of 384kbps over each channel. The 384kbps corresponds to the IMT-2000 requirement for a pedestrian environment. A MAC frame was chosen to contain only six timeslots resulting in a frame length of 7.25ms. Although this results in a fairly high overhead caused by the frame header, it allows faster feedback for retransmissions as well as good FEC built into the frame header to reduce the likelihood of incorrect reception. Of the total 448 bits in a frame header, 192 contain information and 256 are redundant FEC bits. The simulation model assumes that up to 27 bits in error are correctable. It is assumed that all cells transmitted in the uplink contention slot are received correctly, but this does not mean that all sequence number acknowledgements and reservation requests are successfully transferred because those that are piggybacked onto other cells are still lost if the cell is not received correctly. The dummy retransmission VC is set up to have an MCR of 25 cells/s and an MBS of 10 cells.

The simulation model uses Opnet Modeler's default radio pipeline to simulate the radio channel. This includes stages that calculate the interference noise between packets, and the bit errors in each packet [53]. The pipeline was configured such that it produced the BERs shown in Table 6-III.

Table 6-III Average Simulation BERs for timeslots containing an increasing number of cells.

Cells per Timeslot	*Average BER*
4	~2.0 x 10^{-7}
5	~1.5 x 10^{-6}
6	~8.0 x 10^{-6}
8	~1.2 x 10^{-4}

By changing the number of active VCs in each simulation run, the traffic load in each run is varied. The ratio of audio, video and voice VCs is maintained at 1: 1: 5 respectively. In this manner the performance of the protocol is evaluated under increasing traffic loads. In the following section the traffic load is specified in terms of the number of voice VCs. So if the load is specified as 20, this implies that there are 20 voice VCs, 4 audio VCs and 4 video VCs. The email traffic is fixed at 50 messages per hour for all simulation runs.

6.2.2 VC Configurations for the Various Traffic Models

Every time a VC is set up, the traffic parameters and QoS specifications of the VC have to be negotiated according to the requirements of the traffic that is to be transmitted over the new VC. It is important that these parameters are carefully selected if the VC is to be provided with adequate service. For example, a maximum burst size (MBS) must be at least large enough to send the average number of cells that a traffic source generates within a single MAC frame; e.g. A CBR audio VC transmitting at a payload bit rate of 56kbps has a cell rate of 145.8 cells/s. If the MAC frame contains six timeslots over 128kbps CDMA channels, then the frame length is 21.375ms, and the CBR audio VC will need to transmit 3.12cells per frame. This means that the VC will not be allocated sufficient slots unless its MBS is at least four, regardless of what mean cell rate (MCR) is specified. The VC configurations that the simulation uses for each traffic model are presented below.

Voice Traffic

The voice traffic model generates a variable bit rate stream alternating between 9600bps (or 19200bps with FEC) and zero. Obviously voice traffic is delay sensitive, and hence virtual channels that are set up to carry this traffic are given real-time variable bit rate (rt-VBR) status. The maximum acceptable cell delay, or the cell lifetime (CLT), is assumed to be 30ms. It is also assumed that voice traffic can handle relatively high BERs, so up to eight other cells may be transmitted in the same timeslot as a cell containing voice traffic. The MBS of the VC is set to five cells. In other words, up to five cells may be transmitted in one burst at a rate above the mean cell rate specified for the VC. With the CLT set to 30ms and traffic only being generated at a maximum of 9600bps (40ms between cells), there should never be more than a single cell queued at any one time, so the MBS is larger than necessary. The traffic parameters for voice VCs are summarised in Table 6-IV.

Table 6-IV Virtual channel configuration for voice traffic.

Parameter	*Value*
Traffic Class	rt-VBR
Mean Cell Rate (MCR)	25 cells/s
Maximum Burst Size (MBS)	5 cells
Cell Lifetime (CLT)	30 ms
Maximum Cells Per Slot (BER)	8 cells

Video Traffic

The video traffic model generates a variable bit rate stream with the rate varying between 64kbps and 128kbps. A MCR of 250 cells/s corresponds to the mean bit rate of 96kbps produced by the exponential distribution used to determine the video bit rate. In order to accommodate bursts of up to 128 kbps (or 333.3 cells/s) with a mean burst time of 160ms, the MBS is set to 54 cells. This is obtained by multiplying the mean burst length by the maximum bit rate. The video traffic is assumed to be fairly sensitive to bit errors, hence a maximum of five cells may be transmitted in the same timeslot as a video cell. If the video stream is to be used in interactive applications such as video conferencing, it will be sensitive to transmission delays. The CLT is set to 60ms, which is not as stringent as the voice CLT. This is because the far greater throughput requirements of video traffic needs a more lenient CLT in order to avoid large cell losses. Table 6-V summarises the configuration of video VCs.

Table 6-V Virtual channel configuration for video traffic.

Parameter	*Value*
Traffic Class	rt-VBR
Mean Cell Rate (MCR)	250 cells/s
Maximum Burst Size (MBS)	54 cells
Cell Lifetime (CLT)	60 ms
Maximum Cells Per Slot (BER)	5 cells

Digital Audio Traffic

With the digital audio model generating traffic at a constant bit rate of 128kbps, VCs carrying this type of traffic use the constant bit rate (CBR) service class. A MCR of 334 cells/s corresponds to a payload bit rate of 128kbps. While digital audio traffic, such as a digital radio station broadcast, is classified as real time traffic, it is not interactive traffic like the voice and video traffic described above. Therefore, the receiving end can buffer the incoming data to "absorb" delay and jitter effects. For this reason, the CLT of a digital audio VC is more relaxed than that for voice and video traffic. The CLT is set to 75ms. This CLT can result in up to 25 cells awaiting transmission in the VC's cell buffer, although the MBS is set to 10 for the simulation. The digital audio traffic is assumed to be more susceptible to bit errors than voice traffic, with the maximum number of cells per slot set to six. The traffic parameters are summarised in Table 6-VI.

Table 6-VI Virtual channel configuration for digital audio traffic.

Parameter	*Value*
Traffic Class	CBR
Mean Cell Rate (MCR)	334 cells/s
Maximum Burst Size (MBS)	10 cells
Cell Lifetime (CLT)	75 ms
Maximum Cells Per Slot (BER)	6 cells

E-mail Traffic

E-mail traffic is not sensitive to delays so available bit rate (ABR) VCss are used for this type of traffic. The MCR is set to 75 cells/s, which corresponds to a payload bit rate of 28.8kbps. A MBS of 50 cells allows email messages to be transmitted in large bursts, taking advantage of any available bandwidth. Because all the cells in an entire email message may arrive in one quick burst, the CLT associated with each cell is dependent on the number of cells in the email message. The CLT for the VC is set to 50ms times the number of cells in the email message which means that, on average, each cell may be queued for up to 50ms before being dropped. E-mail messages are sensitive to bit errors so a maximum of four cells may be transmitted together in a timeslot containing e-mail traffic.

Table 6-VII Virtual channel configuration for e-mail traffic.

Parameter	*Value*
Traffic Class	ABR
Mean Cell Rate (MCR)	75 cells/s
Maximum Burst Size (MBS)	60 cells
Cell Lifetime (CLT)	50 ms * email size (cells)
Maximum Cells Per Slot (BER)	4 cells

6.2.3 Performance Evaluation

A number of simulation runs have been carried out with various sections of the protocol being omitted in some. These include a run assuming a perfect radio channel with no bit errors (Perfect Channel), a run with no error correction included in the protocol (No GBN), with a go-back-n ARQ implemented but no dummy retransmission VC (GBN), with a go-back-n ARQ and a dummy retransmission VC (GBN, RVC), and finally with a go-back-n ARQ, dummy retransmission VC and the voice traffic accepting bit errors as it includes FEC information (GBN, RVC, ABE). The following figures show plots of the simulation results, but to prevent the graphs from becoming too cluttered only those results with the most significant differences are shown on each graph, and only the uplink results are included. One hour of traffic is simulated to obtain each point.

Figure 6.2 (a), (b) and (c) show the throughput of voice, video and audio traffic respectively. Remember that the traffic load is specified in terms of the number of active voice VCs, where the ratio of voice, video and audio VCs is kept at 5: 1: 1 respectively. The voice and video traffic show congestion from a traffic load of 25 onwards whereas the audio traffic, with a higher priority because of its CBR service class, only shows congestion in an imperfect channel for loads above 30. Note that the voice throughput doubles when its VC is set up to accept bit errors (ABE). This models the inclusion of FEC information in the voice traffic. Consequently, the video traffic, which has the same priority, shows increased congestion, but still only from traffic loads of 25 onwards. Due to its higher priority, the audio traffic is not affected by this increase in voice throughput, and hence the audio plot for this case is not shown. In Figure 6.2(a) it is clear that the go-back-n ARQ does not adequately handle transmission errors for voice traffic – the throughput with an imperfect channel is lower than for a perfect channel even for relatively low traffic loads. The ARQ scheme handles errors more effectively for video traffic even though it has the same priority as the voice traffic. This is because voice VCs are configured with a far more stringent CLT than video VCs.

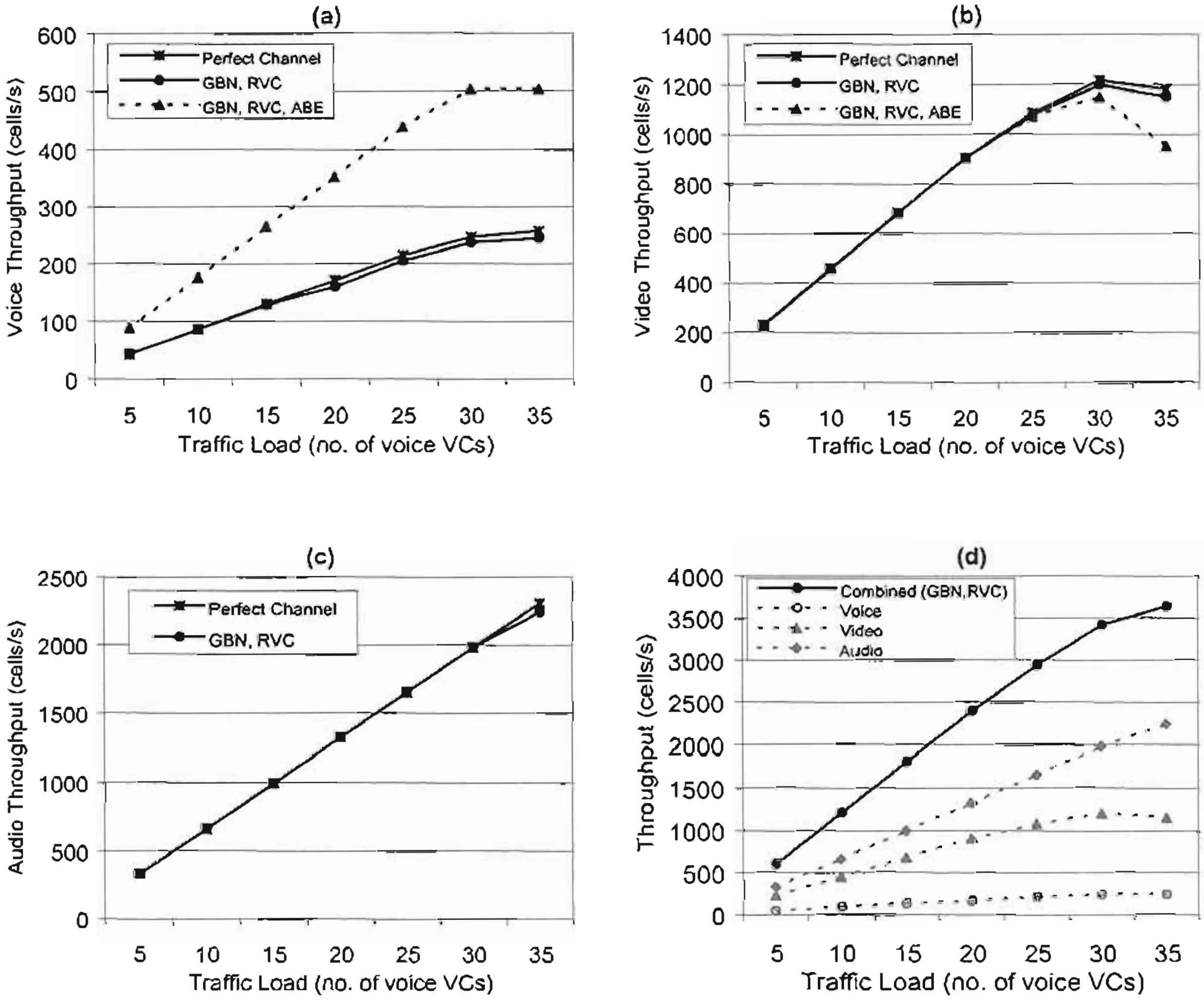


Figure 6.2 Throughput performance of (a) voice, (b) video, (c) audio and (d) combined traffic.

Figure 6.2(d) shows the combined throughput performance for all traffic types under increasing traffic loads. This is for an imperfect channel with the go-back-n ARQ and retransmission VC in use. Although the majority of VCs are voice VCs, their combined throughput is smaller than that of audio and video traffic because of their low transmission rate. Email traffic was also included in the simulation but its throughput is too small to be shown on the graph.

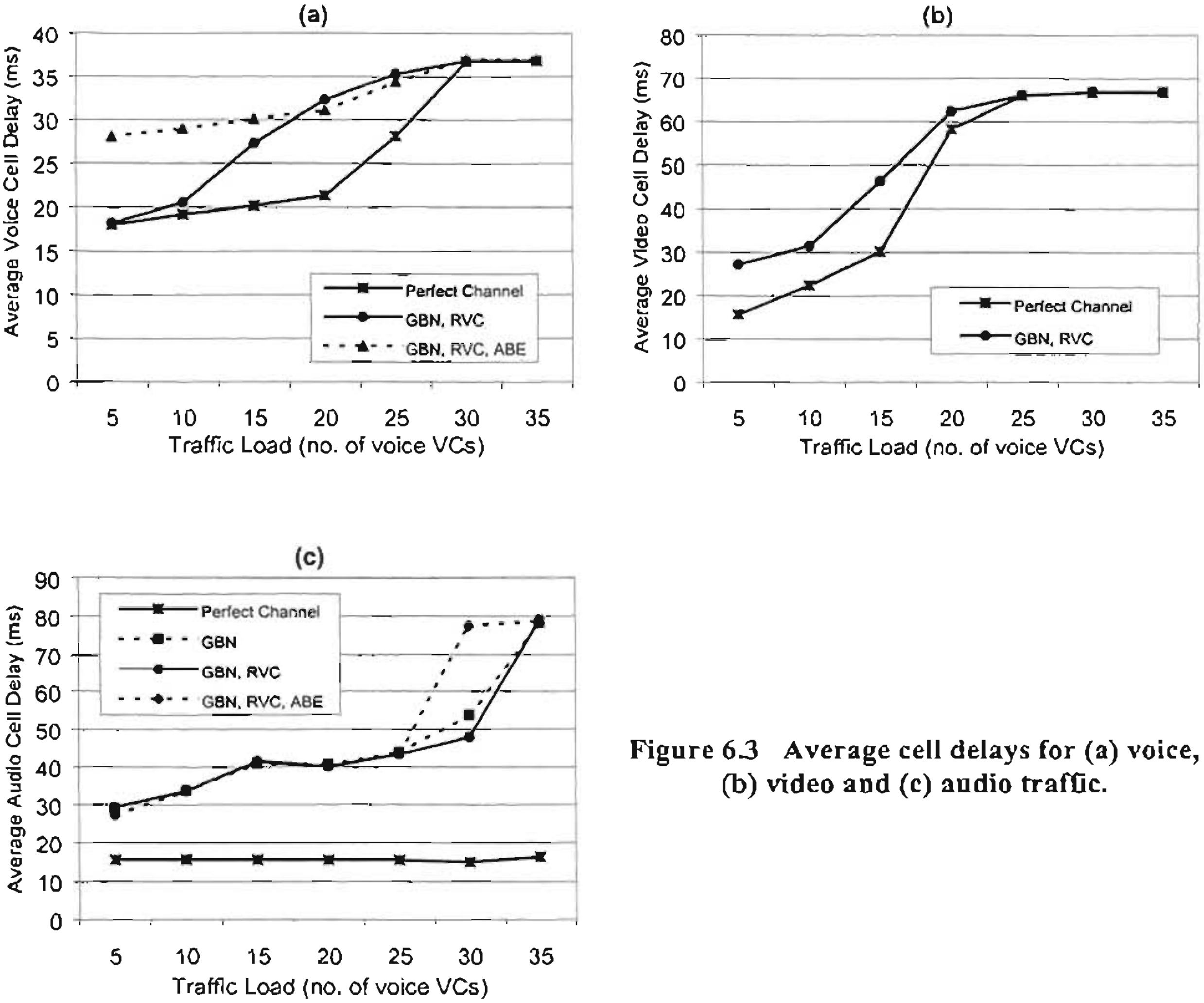


Figure 6.3 Average cell delays for (a) voice, (b) video and (c) audio traffic.

The average cell delays for voice, video and audio traffic are shown in Figure 6.3 (a), (b) and (c) respectively. As would be expected, these delays are considerably smaller for a perfect channel than for an imperfect channel with the go-back-n ARQ in use. Although, for voice and video the average cell delay does approach the maximum possible delay determined by the CLT, even for the perfect channel. Note that the maximum possible cell delay is one frame length greater than the CLT specified for the VC as the CLTs are only enforced once every frame. When voice traffic uses ABE, the average cell delays experienced by voice cells at low traffic loads are greater than those experienced when voice traffic uses the go-back-n ARQ. This is because of the higher throughput required to include FEC information when ABE is used. However, for medium to high traffic loads, voice using FEC has smaller cell delays than voice using ARQ. This is because of the delays introduced by ARQ while retransmitting cells.

The average cell delay experienced by video traffic reaches its CLT limit at a lower traffic load than voice traffic. This is despite the fact that they both have the same priority and voice has a stricter CLT. The higher video cell delays can be explained by the fact that the video traffic sources have periods during which they generate traffic at higher rates than scheduling tokens are being generated. On the other hand, voice traffic sources never generate cells at a rate greater than the VCs specified MCR, which is the rate at which scheduling tokens are generated.

The average cell delay for audio traffic transmitted over a perfect channel, shown in Figure 6.3 (c), remains relatively unchanged for all traffic loads plotted. This is because audio VCs have the highest priority which means the other traffic suffers delayed slot allocations before audio traffic. When bit errors are generated over the wireless channel, the ARQ retransmissions result in larger cell delays, but only at the highest traffic load of 35 does the cell delay approach the maximum delay determined by the CLT. Note that when the retransmission VC isn't used, the cell delays are slightly longer at higher traffic loads. This is because the retransmission VC reduces the ARQ's dependence on the frame fill-up procedure.

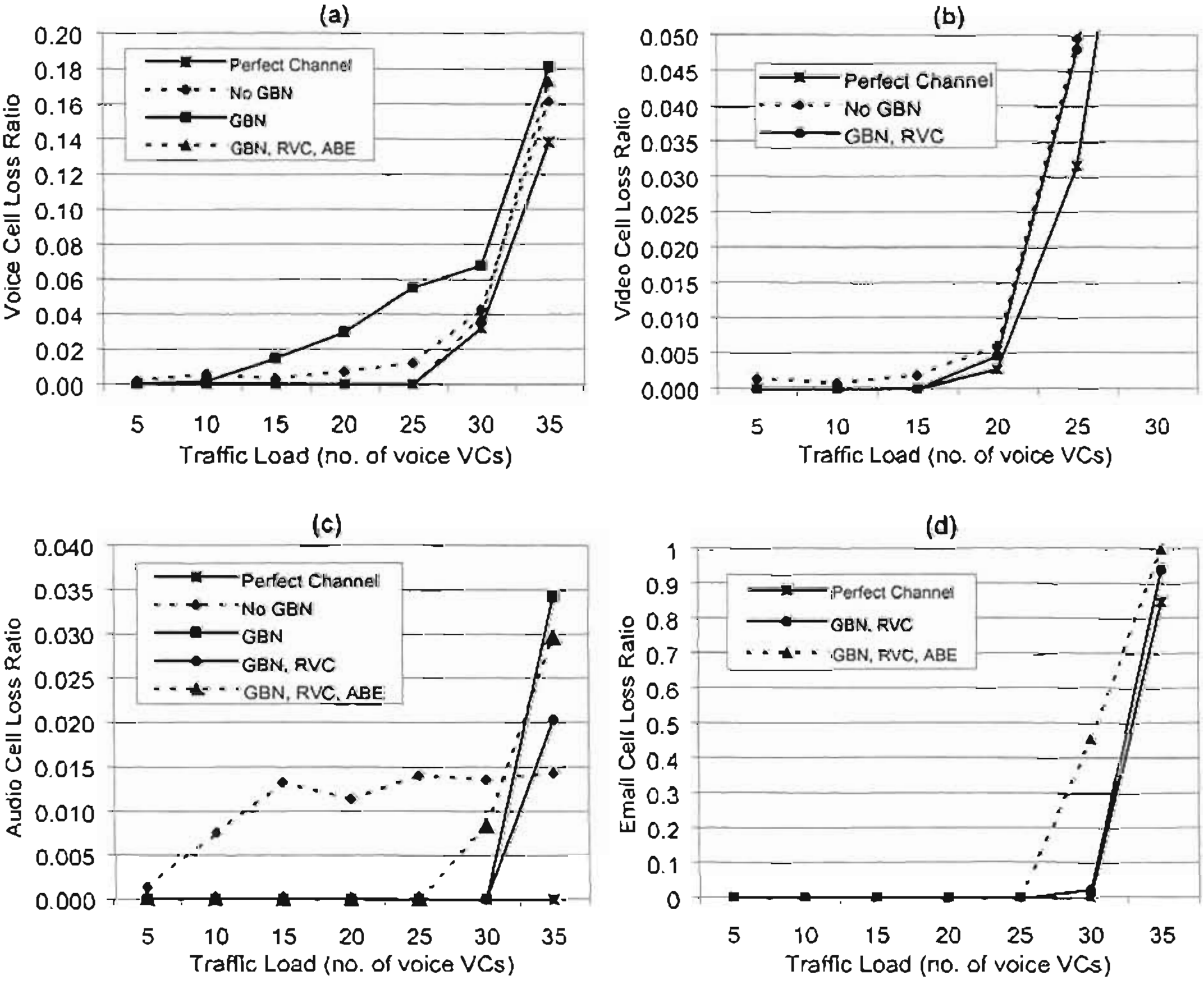


Figure 6.4 Cell loss ratios for (a) voice, (b) video, (c) audio and (d) email traffic.

The voice, video, audio and email cell loss ratios are presented in Figure 6.4 (a), (b), (c) and (d) respectively. For voice, the cell losses are even worse when using a GBN ARQ than without. As previously mentioned, this is because the voice cells are dropped due to their CLT before a retransmission can be carried out. However, when the voice traffic uses FEC, cells are only dropped once the channel becomes congested, which is for traffic loads above 25. These cell payloads will contain bit errors but the half-rate FEC scheme assumed will be able to correct the majority of them.

The video cell losses are considerably less with the go-back-n ARQ than without. The ARQ corrects all errors for traffic loads up to 15. Thereafter, congestion in the link causes the video cell losses to increase fairly rapidly. It is noticeable that the video traffic shows congestion effects at lower traffic loads than the other traffic types. The scheduling algorithm does not handle the bursts in video traffic very well, often relying on the frame fill-up procedure to allocate slots to the extra cells in a burst. It is possible, however, to increase the MCR and MBS specified for video VCs in order to improve their performance.

Audio traffic benefits even more from the go-back-n ARQ. Its cell losses without retransmissions are considerably higher than those of video because of the less stringent BER requirements with which its VCs are configured. Using retransmissions results in near perfect recovery from transmission errors until congestion sets in at a traffic load of 30 (or 25 when voice uses FEC and hence requires more bandwidth). Although there is an improvement for video and audio traffic when using the dummy retransmission VC it is fairly small and can't be properly shown in the plots. This is because the scheduler's frame fill-up procedure caters fairly successfully for the retransmissions until the channel becomes congested. Hence, one can notice the improvement in the audio cell loss ratio when using the retransmission VC at a traffic load of 35.

Email traffic doesn't experience any cell losses for traffic loads below 25. However, for higher traffic loads the cell losses increase rapidly. The go-back-n ARQ keeps cell losses close to those experienced purely due to congestion when a perfect channel is used. As would be expected, it is again apparent that congestion begins at lower traffic loads when voice uses FEC rather than ARQ.

In the above plots it is clear that using FEC instead of ARQ for voice traffic has an adverse effect on the other traffic classes. This is because the increased throughput required by the FEC information results in congestion at lower traffic loads. However, the improvement in the voice traffic performance outweighs the decreased performance of the other traffic types. For example, audio traffic congestion starts from traffic loads above 25, instead of 30, when voice uses FEC, but the performance of voice traffic is adequate for traffic loads up to 25, instead of only 10 as is the case when ARQ is used.

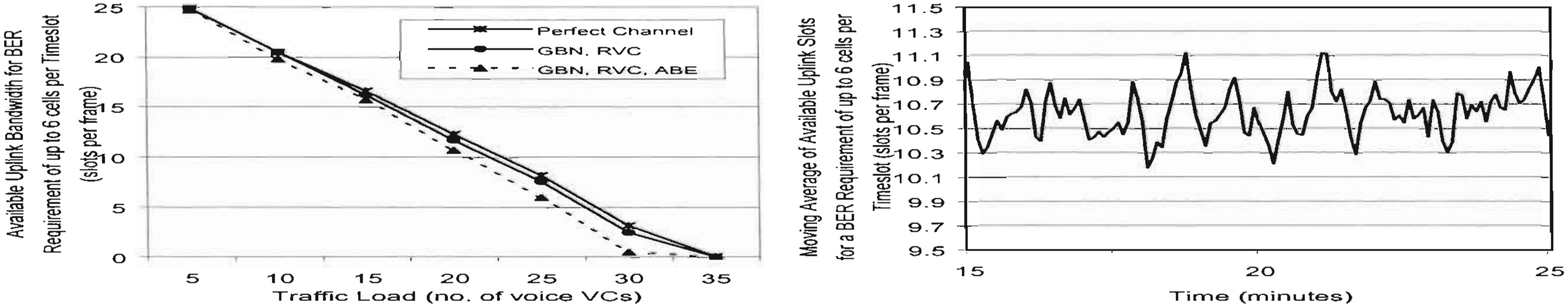


Figure 6.5 Available uplink bandwidth for a BER requirement of up to six cells per timeslot.

The plots shown in Figure 6.5 indicate the average number of free slots available per MAC frame for a BER requiring no more than six cells in a timeslot, and measured using the moving average shown in Eq. 4-1. The weighting factor, w, was set to 0.001 and the measurement was performed before the scheduler carried out its frame fill-up procedure. Considering the congestion effects shown in other results, it would appear that a significant amount of traffic is being transmitted in the frame fill-up even with the dummy retransmission VC being used. However, as would be expected, there is less bandwidth available when retransmissions are used on an imperfect channel than when a perfect channel is used. Also as expected, when the voice traffic throughput increases to include FEC information the available bandwidth decreases. Figure 6.5(b) shows how the free slots moving average varies during a ten minute period with a traffic load of 20 and voice using ABE. The moving average varies fairly rapidly between 10.2 and 11.2 slots per frame during this ten minute period. This fluctuation of approximately one slot per frame indicates that a smaller weighting factor may be needed to smooth the moving average.

The average number of tokens available in the retransmission VC provides an indication of the bandwidth consumed by retransmissions. Figure 6.6 shows that from traffic loads of 25 onwards the retransmission VC battles to provide enough tokens for retransmissions. Also, when voice traffic uses FEC rather than retransmissions there are slightly more tokens available, as would be expected.

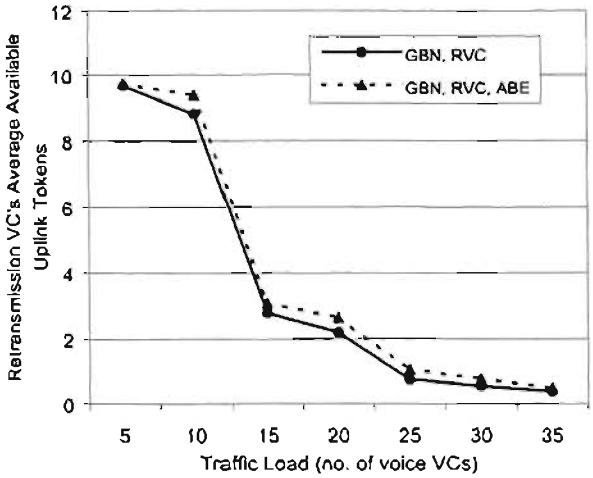


Figure 6.6 Average available uplink retransmission tokens.

6.3 Implementation Results

6.3.1 Implementation Assumptions and Parameters

For the purpose of collecting results from the implementation, the BS DSP board was connected directly to the MT DSP board thereby omitting the physical layer and air interface. This allows for a controlled environment in which the BER experienced over the 'air interface' may be manipulated. It is assumed that 15 CDMA code channels are available for use, with a raw bit rate of 128kbps over each channel. The 128kbps is fairly close to the IMT-2000 requirement for 144kbps in a vehicular environment. As in the simulation, a MAC frame was chosen to contain only six timeslots, which results in a frame length of 21.375ms. Because only one MT is available in the implementation, there is no contention in the uplink contention slot and hence all cells transmitted in the contention slot are received correctly. This does not mean that all sequence number acknowledgements and reservation requests are successfully transferred because those that are piggybacked onto other cells are still lost if the cell is not received correctly. The dummy retransmission VC is set up to have an MCR of 70 cells/s and an MBS of 10 cells.

In order to generate bit errors, a technique similar to that used in Opnet Modeler's default radio pipeline is used [53]. However, instead of calculating the exact number of bit errors in each packet as Opnet's radio pipeline does, all that is required is to determine if the packet contains any bit errors at all. In a segment of length N bits ($N = 448$ for a WATM cell), the probability that no errors occur $P_0 = (1 - p)^N$ where p is the BER. Hence, for each cell received, a uniform random number R between 0 and 1 is generated. If $R < P_0$ then the cell doesn't contain any bit errors. Table 6-VIII shows the BERs used for increasing multi-user interference.

Table 6-VIII Average BERs for increasing cells per timeslot (as used in implementation).

Cells per Timeslot	*Average BER*
1	5×10^{-6}
2	6×10^{-6}
3	7×10^{-6}
4	8×10^{-6}
5	9×10^{-6}
6	1×10^{-5}
7	3×10^{-5}
8	5×10^{-5}
9	7×10^{-5}
10	9×10^{-5}
11	2×10^{-4}
12	4×10^{-4}
13	6×10^{-4}
14	8×10^{-4}
15	1×10^{-3}

As with the simulation, by changing the number of active VCs in each implementation run, the traffic load in each run is varied. The ratio of audio, video, computer data and voice VCs is maintained at 1: 1: 1: 5 respectively. In this manner the performance of the protocol is evaluated under increasing traffic loads. In the following section the traffic load is specified in terms of the number of voice VCs. So if the load is specified as 20, this implies that there are 20 voice VCs, 4 audio VCs, 4 video VCs and 4 data VCs.

6.3.2 VC Configurations for the Various Traffic Models

The VC configurations used for each traffic model are summarised in Table 6-IX. The parameter settings are chosen along the same lines as those used in the simulation, but have been adjusted to comply with the slower bit rate used in the implementation.

Table 6-IX Virtual channel configurations used for implementation results.

Parameter	*Units*	*Voice*	*Video*	*Audio*	*Data*
Traffic Class		rt-VBR	rt-VBR	CBR	ABR
Mean Cell Rate (MCR)	cells/s	50	125	135.4	75
Maximum Burst Size (MBS)	cells	3	26	7	10
Cell Lifetime (CLT)	frames	5	12	15	213
Maximum Cells Per Slot (BER)	cells	15	7	8	5

6.3.3 Performance Evaluation

In order to evaluate the performance of the implementation, the throughputs obtained for each traffic type under increasing traffic loads are plotted below. Figure 6.7(a) shows the throughputs for perfect channel conditions i.e. when no bit errors are introduced over the wireless channel. For this situation the throughput matches the generated traffic more-or-less exactly. However, when bit errors are introduced the situation changes somewhat. As shown in Figure 6.7(b) the throughputs remain relatively close to the generated traffic until traffic loads of 20. Thereafter, audio and video traffic throughput declines rapidly, although voice and data throughputs continue to follow the generated traffic quite closely. The reason being that voice traffic is using ABE and data traffic has very large CLTs.

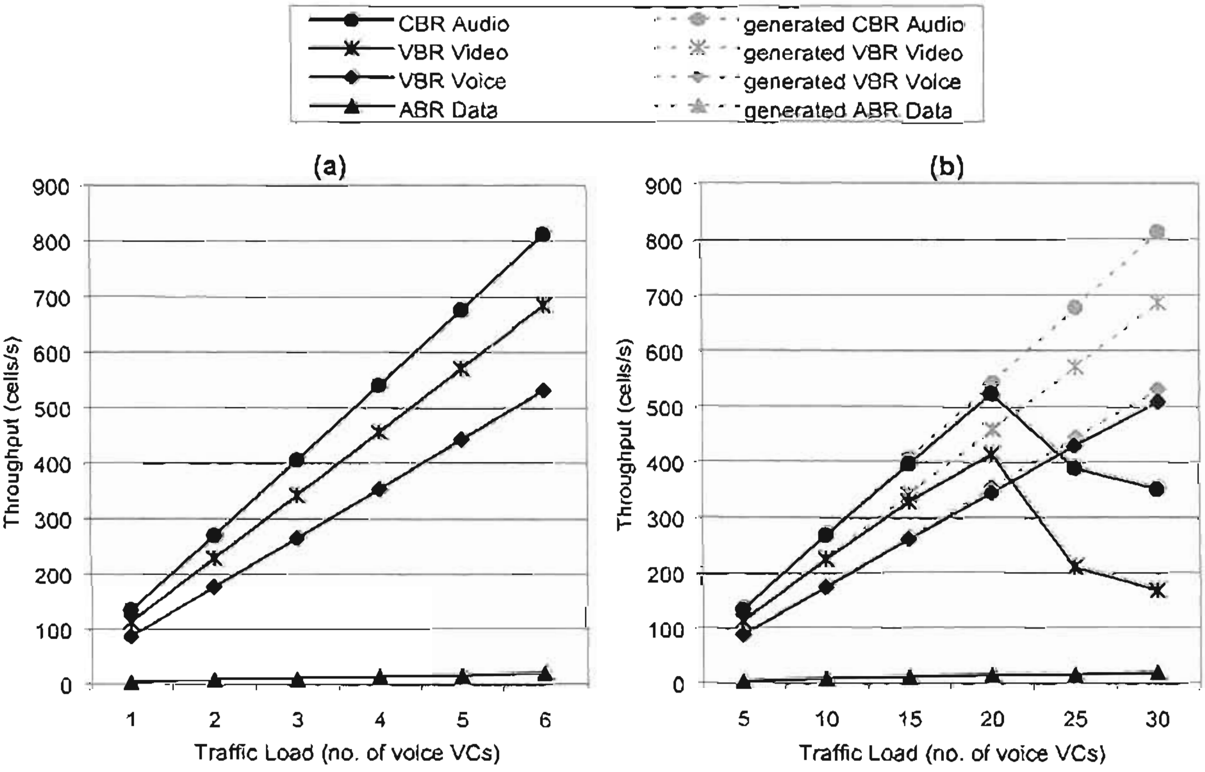


Figure 6.7 Uplink throughputs achieved in implementation for (a) a perfect channel and (b) an imperfect channel.

Audio and video generates a large amount of retransmission traffic which results in congestion at much lower traffic loads than when a perfect channel is used. This is confirmed by Figure 6.8 which shows the average number of free slots per frame under increasing traffic loads with and without bit errors on the channel. It is clear that when bit errors are introduced a large portion of the bandwidth is used for retransmissions. The bit rates required by audio and video traffic are relatively high when compared to the 128kbps raw bit rate available over each CDMA code channel. This decreases the throughput efficiency of the go-back-n ARQ scheme as the number of cells that are dropped before a NAK reaches the transmitting end is relatively large (see Appendix B.2). A selective-repeat ARQ may significantly improve the performance of the protocol in this situation.

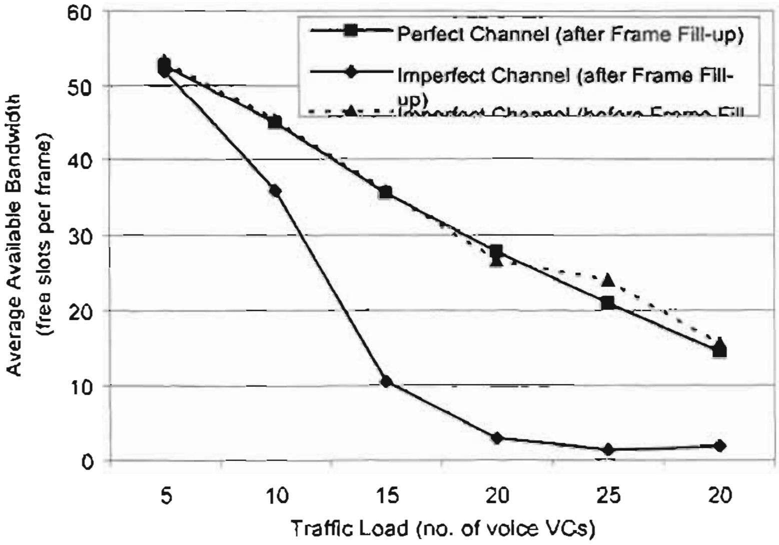


Figure 6.8 Available uplink bandwidth for a BER requirement of up to 15 cells per timeslot.

Figure 6.8 also compares the average number of free slots for an imperfect channel before and after the frame fill-up procedure. The number of free slots is considerably less after frame fill-up indicating that a significant portion of the traffic is transmitted in slots assigned by the frame fill-up procedure.

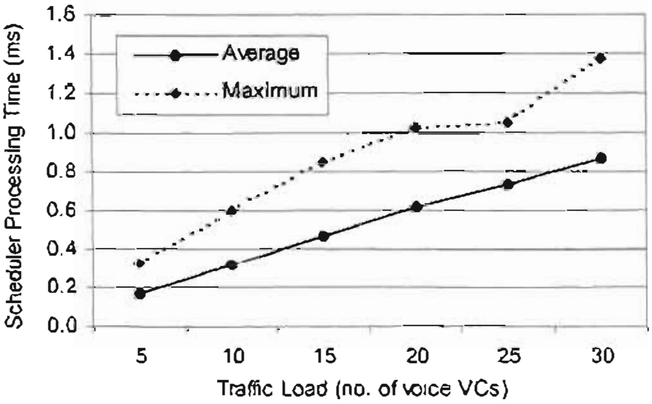


Figure 6.9 Scheduler processing time.

One important factor in the implementation of the protocol is the processing time required by the scheduler to allocate slots for each uplink and downlink frame. The scheduler performs the slot allocations during the last timeslot of each downlink frame so that the frame headers are ready to be transmitted in the first timeslot of the following downlink frame. This means that the scheduler must take less than one timeslot period to complete the slot allocations. For the 128kbps system used to gather the implementation results, one timeslot has a length of 3.56ms. Figure 6.9 shows the average and maximum processing time required by the scheduler for increasing traffic loads. Even for high traffic loads when the system is congested the maximum processing time does not exceed 1.4ms, indicating that the TMS320C6201 DSP processor has sufficient processing power to handle the protocols scheduling algorithm. However, if the bit rate were to double, halving the timeslot period, the scheduler would take too long to complete the slot allocations for higher traffic loads.

Chapter 7 Conclusion

This dissertation began by introducing the current state of mobile communications and how the world has changed as a result of the mobile revolution. The future of mobile and the technologies driving this future were discussed. The first chapter also outlined the ATM philosophy and the reasons for extending it to the wireless domain. This was followed by a description of the functionality promised by the so-called 3G wireless networks, and an explanation of the underlying concept of CDMA and the fundamental benefits it provides.

Chapter two explained the requirements of the various layers in a WATM system, especially the MAC and DLC layers. The chapter then went on to describe the evolution of WATM protocols, pointing out the improvements made during the 1990's. This was followed by a review of existing CDMA based MAC protocols and a discussion on how a CDMA based WATM protocol could improve the QoS offered by the wireless network. The effects of some fundamental issues relating to the design of a wireless protocol, such as the base station cell size and the multiple access scheme, were also examined.

The next chapter described the MAC layer of the proposed slotted-CDMA (hybrid TDMA/CDMA) based WATM protocol. The protocol combines some of the techniques used in the previously discussed protocols. The demand-based assignment MAC layer takes advantage of the soft capacity of the CDMA physical layer to perform flexible BER scheduling that supports a variety of traffic classes. Each VC is assigned a priority according to its service class. Slots are allocated according to the VC's traffic contract. Chapter four presented the DLC protocol, which allows each VC to take advantage of a windowed go-back-n ARQ, or implement some application specific FEC technique at a higher layer according to its error requirements and delay constraints. A special retransmission VC allows a portion of the bandwidth to be reserved for retransmissions. Some issues relating to a CAC algorithm for the protocol were also discussed as the CAC is complicated by the BER scheduling technique used by the MAC layer.

Chapter five discussed the actual implementation of the link layer protocol. This was carried out on a Blue Wave Systems DSP board that forms part of AAT's software radio architecture. The chapter began with a discussion of the software radio concept and how it promises to bring a new degree of flexibility to radio interfaces, allowing radios to adapt to their local environment. AAT's software radio platform is a good first step towards what may be described as the ultimate software radio, but there is still plenty of room for improvement. For example, the FPGA based radio board would benefit from the inclusion of a DSP processor, as the complexity of many algorithms is extremely difficult to implement on an FPGA. Also, the anti-aliasing filters used for the receiver are fixed, limiting the range of received signals that may be successfully sampled. The implementation details of each block in the BS and MT link layer software were discussed. In particular, the interface to the physical layer and between the various blocks, the structures used to represent MTs and VCs, and the scheduler implementation were explained. Besides the link layer, the software running on the host PCs was also described. This software was used to generate traffic in order to test the performance of the link layer implementation.

The performance of the protocol, and slight variations thereof, were shown in chapter six. Results were collected from both a simulation of the protocol developed in Mil3's Opnet Modeler, as well as from the actual implementation. The results have shown the scheduling algorithms support for multimedia traffic and the improvements obtained through the use of a retransmission technique. However, the requirement for an alternative error correction method was demonstrated by the inability of the go-back-n ARQ to deal with errors in the voice traffic. Hence the inclusion of the mechanism to accept cells even if they contain bit errors, allowing FEC techniques to be applied at higher layers.

A number of possibilities for future work on the protocol have also been identified. As far as the MAC protocol is concerned, there are a few possible improvements that may be investigated. The first is to allow CBR traffic to be automatically assigned uplink slots without having to send uplink reservation requests. This would potentially reduce congestion in the contention slot. It is also possible to investigate the extension of this concept to rt-VBR traffic. For example, if a rt-VBR connection's average cell rate drops below its specified mean or minimum the connection may be assigned an uplink slot even if no reservation request has been received. This may counteract congestion in the contention slot, ensuring that the real-time traffic contract is upheld. Another idea worth investigating is the use of a polling scheme to gather uplink reservation requests from MTs, instead of relying on the contention slot.

One of the difficulties with the current DLC set up is that an application specific FEC scheme can't be applied in the DLC with the go-back-n ARQ scheme applied above it. At the moment, if a FEC scheme is applied at higher layers then the DLC may be specified to accept bit errors. If a retransmission scheme is also required then this has to be implemented at higher layers as well. It would be possible to implement a fixed FEC scheme to all cells, but this approach lacks flexibility – in many cases a FEC scheme may be far more effective if designed to match the application with more important bits being more heavily coded then less important bits. A good example of this is the coding scheme used for voice in GSM. The software radio concept does, however, posses the potential for an elegant solution to this problem. It would be possible for an application to make available its FEC module via the network. A software radio could then download this FEC module via the network, and implement this application specific FEC scheme on the appropriate VC.

An important component of a wireless cellular system that has not been investigated in this thesis is a hand-off procedure. Incorporating hand-offs into the ATM network is a fairly involved process because of the strict maintenance of cell ordering required. In order to maintain a VCs QoS during a hand-off, some VCs will require low delays to be the most important criteria, while other VCs will stress the requirement for no or low cell losses. Because the physical layer is CDMA based, the system has the advantage of being able to easily incorporate a soft hand-off technique.

Finally, the implementation of an appropriate CAC algorithm that prevents congestion over the wireless channel while maximising bandwidth utilisation is of extreme importance. Although this thesis did mention the idea of using an exponential-weighted moving average to measure the available bandwidth, this was not the focus of the thesis and hence, no algorithm that uses this free bandwidth measurement to determine whether or not a call can be handled was investigated.

Appendix A Some Protocol Specification Details

A.1 The ATM Cell Format

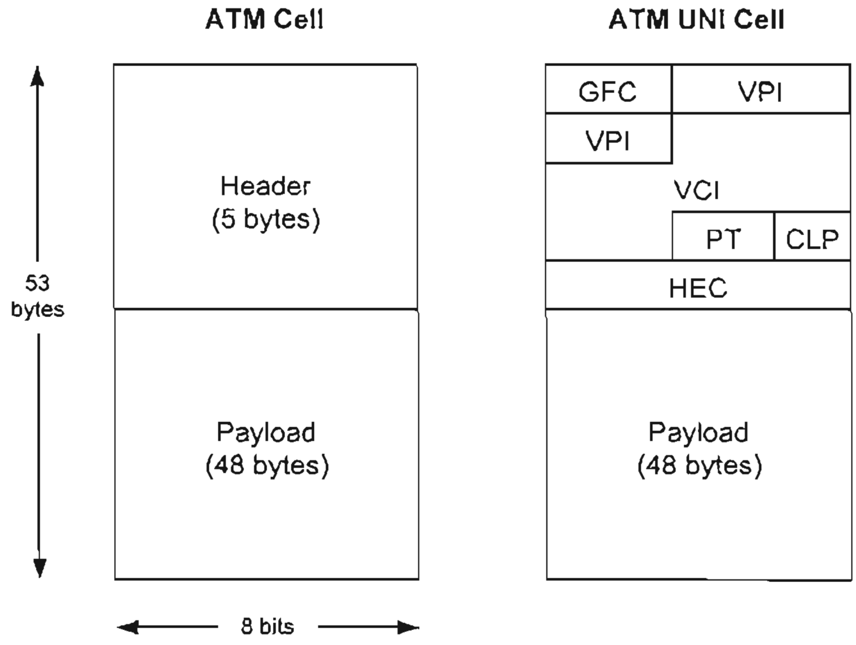


Figure A.1 ATM Cell Format. [11]

An ATM cell consists of a 5-byte header and a 48-byte payload. The cell header can be one of two formats: User-Network Interface (UNI) or Network-Network Interface (NNI). UNI cells are used between ATM switches and ATM end points whereas NNI cells are used between ATM switches in the network. Figure A.1 shows the fields in an ATM UNI cell. The purpose of each field is summarised below:

Generic Flow Control (GFC) Provides local functions such as supporting more than one station on a single ATM interface. (Generally set to a default value and not used.)

Virtual Path Identifier (VPI) Used in conjunction with VCI to specify the next destination of a cell as it progresses, from switch to switch, through an ATM network.

Virtual Channel Identifier (VCI) Used in conjunction with VPI to specify the next destination of a cell as it progresses, from switch to switch, through an ATM network.

Payload Type (PT) The first bit indicates whether the payload contains user data or control data, the second bit indicates congestion and the third bit indicates whether the cell is the last cell in an AAL5 sequence.

Cell Loss Priority (CLP) Indicates whether the cell should be discarded if congestion is encountered.

Header Error Correction (HEC) An 8-bit cyclic redundancy check on the cell header.

The NNI cell header is exactly the same as the UNI cell header except that the GFC field is removed and the VPI extended in its place allowing for larger ATM trunks in the ATM network.

A.2 Specifications for the Proposed WATM Protocol

A.2.1 WVCI Allocations

When a MT is associated with a BS it is allocated a wireless virtual channel identifier (WVCI) to be used as its signalling VCI with the BS. This 16-bit WVCI is only valid within the cell and doubles as a WATM address for the MT within the cell. This signalling WVCI is used in the WATM signalling protocol instead of the MT's full 20-byte ATM address, and hence reduces the signalling traffic. There is a range of WVCIs reserved for use as signalling WVCIs. Similarly, other WVCI values are reserved for special functions. Table A-I shows all reserved WVCI values, and ranges of WVCI values.

Table A-I WVCI Allocations.

WVCI	Reserved Usage
0000h	*Empty Slot:* Only used in the frame header to allocate an empty slot in which nothing is transmitted.
0001h	*Association Request Cell:* Uplink mini-cell transmitted in an uplink contention slot.
0002h	*Association Accept Cell:* Only transmitted in the downlink.
0003h	*VC Update Cell:* A downlink SNack cell broadcast by the BS, or an uplink SNack & RR mini-cell transmitted in the uplink contention slot.
0004h...000Eh	*Reserved* for future specialised cells.
000Fh	*Uplink Contention Slot* (only used in the frame header to allocate an uplink contention slot)
0010h...001Fh	*BS Wireless Addresses:* 16 available addresses used to distinguish between the BSs of adjacent cells. A BS advertises its Waddr in its downlink frame header by using its Waddr in the frame header's WVCI field. In this way a MT knows that a slot contains a frame header if its WVCI is in this range, and it can distinguish between frame headers from different BSs by the value of the WVCI field.
0020h...03FFh	*Signalling WVCIs:* 991 WVCIs reserved for use with signalling VCs. Each MT is assigned a signalling VC, so up to 991 MTs may be associated with a BS at any one time.
0400h...FFFFh	*Standard WVCIs:* Allocated to ordinary WATM connections.

A.2.2 Scheduling Priority Assignments

Every VC is assigned a scheduling priority for the wireless link according to its ATM service class. There are five service classes defined by the ATM Forum. These are constant bit rate (CBR), real-time variable bit rate (rt-VBR), non real-time variable bit rate (nrt-VBR), available bit rate (ABR), and unspecified bit rate (UBR). CBR and rt-VBR are considered to be real-time service classes, and together with nrt-VBR offer guaranteed services. These guaranteed services are supported via admission control, bandwidth reservation, policing, scheduling, etc. ABR and UBR are considered to be best effort services. The VC priority assignments used by the scheduler follow logically from the requirements of the various traffic classes. Table A-II shows the priority assignments with 1 being the highest priority, and 5 the lowest.

Table A-II Scheduling priority assignments.

Service Class	Priority
CBR	1
rt-VBR	2
nrt-VBR	3
ABR	4
UBR	5

A.2.3 WATM Traffic and QoS Parameters

The following parameters define the traffic characteristics and QoS requirements of a WATM virtual channel in the proposed protocol. They must, therefore, be specified when a VC is set up in order for the scheduling algorithm to allocate adequate bandwidth to the VC. The traffic parameters for the uplink and downlink are specified separately because, although a VC is bi-directional by default, it may have completely different traffic requirements for the each direction. Table A-III summarises the traffic parameters.

Table A-III WATM Traffic Parameters.

Parameter	Description
Traffic Class	Effectively a priority assigned to the VC and used by the scheduler. (See Appendix A.2.2 for the traffic class to priority mappings.)
Mean Cell Rate (MCR)	The rate at which tokens are added to the VC's token pool. (Specified in cells per second.)
Maximum Burst Size (MBS)	The maximum number of cells that may be sequentially transmitted at a rate above the specified MCR. Determines the size of the VC's token pool. (Specified in cells.)
Cell Life Time (CLT)	The maximum length of time for which a cell may be queued before it is dropped. (Specified in MAC frames.)
Bit Error Rate (BER)	The maximum acceptable bit error rate that the VC can handle. (Specified in codes per timeslot i.e. multi-code interference.)
Accept Bit Errors (ABE)	Specifies whether or not cells containing bit errors are accepted and passed on, or simply dropped and retransmitted if their CLT isn't exceeded.

A.2.4 Possible Power Control Bit Assignments

The WATM cell's RR field is only applicable to uplink cells. A possible use of this field in the downlink is for power control – an important component of any CDMA system. Table A-IV illustrates how this 3-bit field may be used by BSs to control the transmission powers of MTs.

Table A-IV Example use of Power Control Bits

BITS	Power Change (db)
000	No change
001	+1
010	+2
011	+3
100	Reserved
101	-1
110	-2
111	-3

Appendix B Performance Comparison of ARQ Schemes

The following comparison is based on the performance of ARQ systems, as discussed by Taub and Schilling in [42] (pages 580 – 583). The performance of ARQ systems may be measured in two ways: by the probability of error, and by the transmission efficiency. These two methods of performance measurement are discussed below.

B.1 Probability of Error

In an ARQ system, block codes are used for error detection. The only time that errors pass through the ARQ system is when a received packet has a sufficient number of errors and looks like a valid codeword, so the errors go undetected. For an arbitrary (n, k) block code, where n is the length of the codeword (i.e. the number of bits in the packet including redundant bits) and k is the number of information bits in the message (i.e. the length of the packet excluding redundant bits), there are 2^n possible received packets and 2^k possible messages. Thus, an error will occur if one of the other $2^k - 1$ messages is received. To upper bound the probability of error P_e it is assumed that all 2^n possible received messages are equally likely. Then P_e is

$$P_c \leq \frac{2^k - 1}{2^n} \approx 2^{-(n-k)} \qquad \textbf{Eq. B-1}$$

So, for the (448, 432) CRC code used in the WATM cell structure (Figure 3.3), the probability of error is bounded by $P_e \leq 2^{-16} \approx 1.5 \times 10^{-5}$.

B.2 Throughput Efficiency

The throughput efficiency of an ARQ system is defined as the ratio of the average number of information bits accepted at the receiver per unit time to the number of information bits that would be accepted per unit time if ARQ were not used and all information bits were assumed to be error free. Although all the ARQ schemes will yield the same error rate (as described above), their throughput efficiencies are different. Equations for the throughput efficiencies of the three ARQ systems are derived in [42], but the derivations are based on a transmission channel dedicated to a single connection. The following throughput efficiency derivations apply to the demand assignment protocol proposed in this thesis where the channel's bandwidth is shared between the active connections.

The probability that the receiver accepts any particular packets is P_A and it is assumed that, on average, a packet is scheduled for transmission in a timeslot in the middle of a MAC frame. Further, it is assumed that the CAC prevents congestion in the wireless channel, but that the traffic throughput is very close to the maximum so that the scheduler's frame fill-up procedure has an insignificant effect. It is also assumed that the DLC's retransmission VC has not been implemented so that there is no bandwidth reserved for retransmissions. If no ARQ is applied and all packets are accepted then packets will be transmitted, on average, at the connections MCR. In other words, the average time between packet transmissions $\bar{\tau}_p$ is given by

$$\bar{\tau}_p = \frac{1}{MCR}$$

B.2.1 Stop-and-Wait ARQ

The probability that a single transmission is all that is required for acceptance of the packet is P_A. The probability that only two transmissions will be required is $(1 - P_A)\ P_A$. That is the product of the probability that the first transmission is rejected, $(1 - P_A)$, and that the second transmission is accepted, P_A. The average number of transmissions required for acceptance of a single packet is the sum of the products of the number of transmissions j and the probability of requiring j transmissions, $P_A(1 - P_A)^{j-1}$. Hence, the average number of transmissions required is

$$\begin{aligned} N_{S\&W} &= 1\cdot P_A + 2\cdot P_A(1-P_A) + 3P_A(1-P_A)^2 + \ldots \\ &= \frac{1}{P_A} \end{aligned}$$

The following variables are defined and illustrated in Figure B.2: The *slot time* T_S is the length of a timeslot in seconds. The *feedback time* T_F is the total time it takes for a feedback message (ACK or NAK) to be returned to the transmitting end after a cell transmission is completed. As shown in Figure B.2, it is assumed that, on average, two contention slots (CSs) are required for a feedback ACK or NAK to be successfully transmitted in the uplink. The *allocation time* T_A is the total time until the next slot allocated to the VC is reached after the transmitting end receives the feedback message.

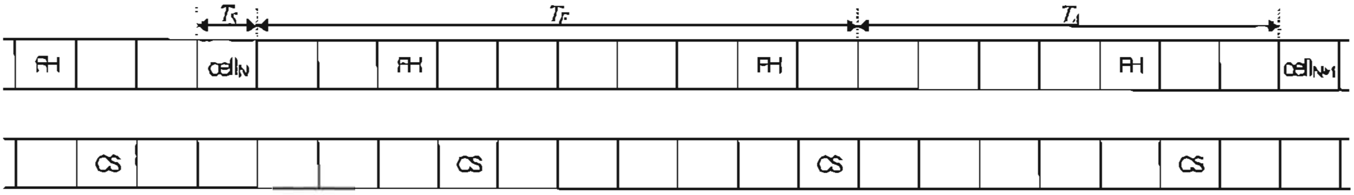


Figure B.2 Stop-and-wait ARQ delays.

Hence, the total time devoted to a single transmission attempt is the greater of $T_S + T_F + T_A$ and $\bar{\tau}_p$, denoted $max(T_S + T_F + T_A\ ,\ \bar{\tau}_p)$; i.e. only if the connections packet inter-arrival time is less than the feedback delay caused by the stop-and-wait ARQ, will the ARQ deteriorate the connection's throughput. The average time taken to transmit a packet is given by

$$\bar{\tau}_{S\&W} = \frac{max(T_S + T_F + T_A, \bar{\tau}_p)}{P_A}$$

from which it follows that the throughput efficiency of the stop-and-wait ARQ is

$$\eta_{S\&W} = \frac{\bar{\tau}_p}{\bar{\tau}_{S\&W}} = \frac{P_A}{MCR \times max(T_S + T_F + T_A, \bar{\tau}_p)} \qquad \textbf{Eq. B-2}$$

B.2.2 Go-Back-N ARQ

For the go-back-n ARQ, the number of packets that are lost when an error is detected is $\lfloor 1+T_F \times MCR \rfloor$. Therefore, $\lfloor 1+T_F \times MCR \rfloor$ packets have to be retransmitted to recover from the error. Following the same procedure used for the stop-and-wait ARQ, the average number of transmissions required for acceptance of a single packet is

$$\begin{aligned} N_{GBN} &= 1 \cdot P_A + (1 + \lfloor 1 + T_F \times MCR \rfloor) \cdot P_A(1-P_A) + (1 + 2\lfloor 1 + T_F \times MCR \rfloor) \cdot P_A(1-P_A)^2 + \ldots \\ &= 1 + \frac{\lfloor 1 + T_F \times MCR \rfloor (1-P_A)}{P_A} \end{aligned}$$

The time devoted to a single transmission attempt is $\bar{\tau}_p$, so the average time to transmit a packet is

$$\bar{\tau}_{GBN} = \bar{\tau}_p N_{GBN} = \bar{\tau}_p \left(1 + \frac{\lfloor 1 + T_F \times MCR \rfloor (1-P_A)}{P_A} \right)$$

from which it follows that the throughput efficiency of the go-back-n ARQ is

$$\eta_{GBN} = \frac{\bar{\tau}_p}{\bar{\tau}_{GBN}} = \frac{P_A}{P_A + \lfloor 1 + T_F \times MCR \rfloor (1-P_A)} \qquad \textbf{Eq. B-3}$$

B.2.3 Selective-Repeat ARQ

As for the stop-and-wait ARQ, the average number of transmissions required for acceptance of a single packet in the selective-repeat ARQ is

$$N_{SR} = \frac{1}{P_A}$$

However, the average time devoted to a single transmission attempt is $\bar{\tau}_p$. Therefore, the average time taken to transmit each packet is

$$\bar{\tau}_{GBN} = \bar{\tau}_p N_{SR} = \frac{\bar{\tau}_p}{P_A}$$

And so, the selective-repeat ARQ's throughput efficiency is given by

$$\eta_{SR} = \frac{\overline{\tau}_p}{\overline{\tau}_{SR}} = P_A \qquad \textbf{Eq. B-4}$$

B.2.4 Comparison of Throughput Efficiencies

The following graphs compare the throughput performance of the three ARQ schemes. As in the simulation, the raw bit rate over each CDMA code channel is selected to be 384kbps with a MAC frame containing six timeslots. This implies a slot length T_S of 1.2ms. Following the diagram in Figure B.2, the feedback time T_F is 11.9ms, and the allocation time T_A is 8.3ms. Figure B.3 and Figure B.4 compare the throughput efficiencies of the ARQ schemes for increasing MCRs with P_A = 0.99 and P_A = 0.99 respectively. It is clear that for low cell rates and low error probabilities the performance of the go-back-n ARQ matches that of the selective-repeat ARQ very closely. However, with higher probabilities of error the go-back-n ARQ efficiency deteriorates relatively rapidly as the cell rate increases. The performance of the stop-and-wait ARQ under increasing cell rates deteriorates rapidly when compared to both the go-back-n and the selective-repeat ARQ schemes. Note that the best throughput efficiency achieved by any of the ARQ schemes is equal to the probability of packet acceptance P_A. However, if the retransmission VC is used it is possible for the go-back-n and selective-repeat ARQ schemes to recover completely from transmission errors with throughput efficiencies of one.

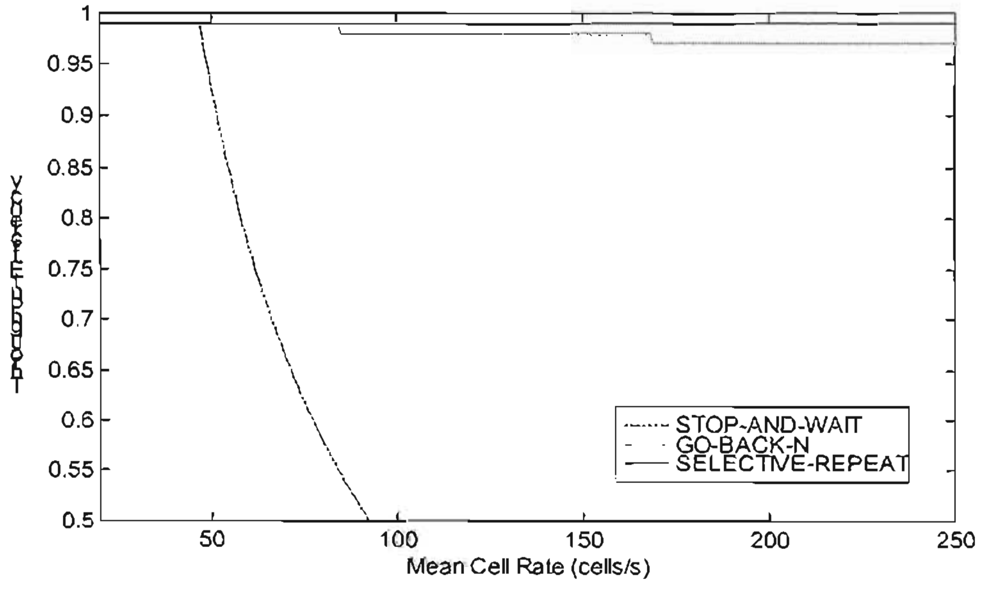


Figure B.3 ARQ throughput efficiencies for $P_A = 0.99$

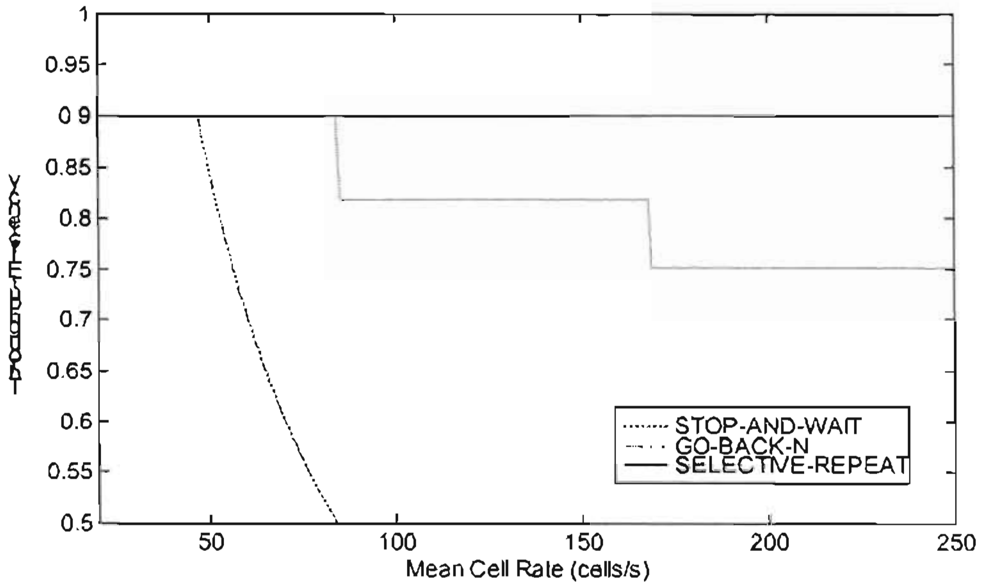


Figure B.4 ARQ throughput efficiencies for $P_A = 0.9$

Appendix C Details of the DSP Board Hardware

C.1 The SCbus

The standard SCbus includes both a message bus and a data bus, although the message bus is not implemented in the PCI/6200 DSP board. The data bus contains sixteen bi-directional serial data lines, SD0 to SD15. Each line carries data in frames of 8-bit packets referred to as timeslots. The bandwidth equivalent of each timeslot is fixed at 64kbps, so the number of timeslots in a frame depends on the clock frequency of the SCbus: 2.048MHz, 4.092MHz or 8.192MHz. The DSP board uses a clock frequency of 2.048MHz, which means a frame consists of 32 timeslots.

Figure C.5 shows the SCbus data and control signals. Besides the SD[0..15] data lines, the SCbus also uses two clock signals, SCLK and SCLKx2N, to determine the bit boundaries of the data stream. The frequency of SCLK determines, and is the same as, the data rate of the 16 data lines. The frame synchronisation pulse, FSYNCN, is specified to have a frequency of 8kHz

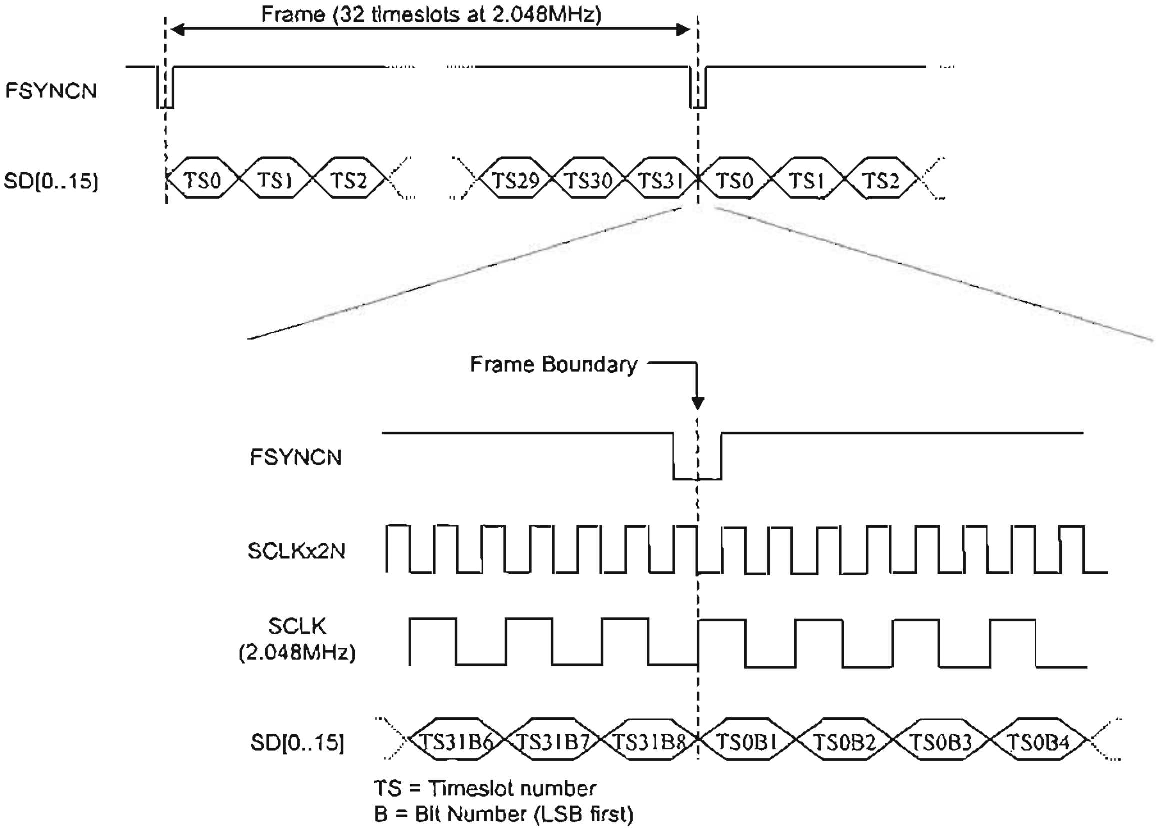


Figure C.5 SCbus data and control signals [52].

C.2 The MVIP Bus

The clock and timing specifications of the ST-BUS, another common bus specification, are a subset of those for the SCbus. For a clock frequency of 2.048MHz the SCbus satisfies the ST-BUS specification. The MVIP (Multi-Vendor Interface Protocol) architecture uses the ST-BUS format.

C.3 The McBSP

The Texas Instruments TMS320C6201 DSP processor has two McBSPs. The Blue Wave Systems PCI/C6200 DSP board interfaces the two McBSPs to MVIP switching compatible and ST-bus compliant devices. Figure C.6 shows the structure of a McBSP. Each port has separate pins for transmission (DX) and reception (DR), with control information (clocking and frame synchronisation) communicated via four other pins: transmit clock (CLKX), receive clock (CLKR), transmit frame synchronisation (FSX), and receive frame synchronisation (FSR).

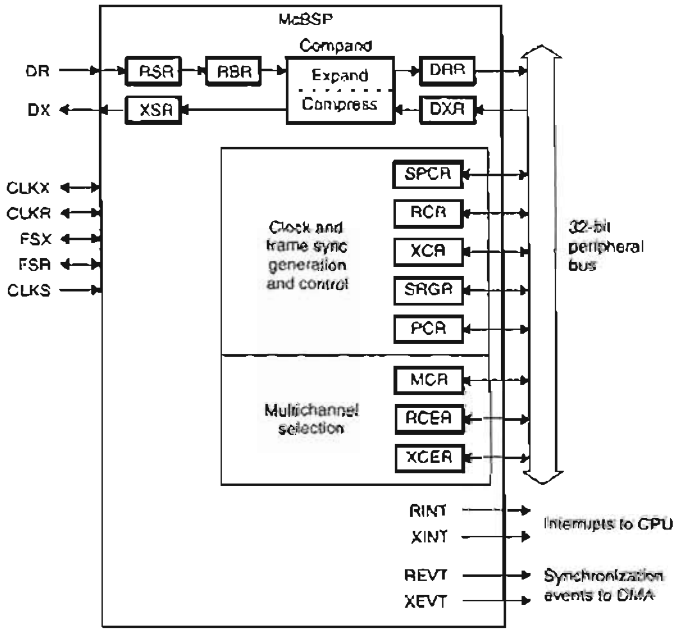


Figure C.6 McBSP Block Diagram [1].

The TMS320C6201 communicates with the McBSPs via 32-bit wide control registers accessible through the internal peripheral bus. Either the CPU or the DMA controller reads received data from the data receive register (DRR), and writes data for transmission to the data transmit register (DXR). Data written to the DXR is shifted out to DX via the transmit shift register (XSR). Similarly, data received on the DR pin is shifted in to the receive shift register (RSR), copied into the receive buffer register (RBR) and then into the DRR.

The McBSP is configured via the serial port control register (SPCR) and pin control register (PCR). The receive and transmit control registers (RCR and XCR) configure parameters of the receive and transmit operations respectively. For more detail on these registers refer to the "TMS320C6000 Peripherals Reference Guide" [54].

C.4 The DMA Controller

The TMS320C6200 on chip DMA controller provides four DMA channels that may be independently configured to transfer data from one memory mapped location to another without the intervention of the CPU. Each DMA channel has the following set of registers that must be configured before starting a DMA transfer:

- *Primary Control Register* – Used to configure the transfer.

- *Secondary Control Register* – Used to enable interrupts to the CPU and to monitor the channels activity.
- *Transfer Counter Register* – Used to keep track of the transferred elements.
- *Source Address Register* – Memory location from which the element is transferred.
- *Destination Address Register* – Memory location to which the element is transferred.

In addition to the registers above there are several global DMA registers that may be used by any channel to perform more complex transfers:

- *Global Address Registers* – Used as either a split address or an address reload value.
- *Global Index Registers* – Used to control address updates during a transfer.
- *Global Count Reload Registers* – Used to reload the transfer counter register of a DMA channel.

One additional register, the *auxiliary control register*, is used to set the priority of the auxiliary channel with respect to the four DMA channels and the CPU. The auxiliary channel allows the host port to access the C6201's memory space. For controlling the DMA versus CPU priority, each DMA channel can be independently configured in high priority mode. However, the DMA controller implements a fixed priority between DMA channels with channel 0 having the highest priority and channel 3 the lowest priority.

A Texas Instruments document entitled "TMS320C6000 DMA Example Applications" [55] describes a number of DMA channel configurations starting with a simple contiguous block move example. Other examples include a data sorting transfer, servicing a peripheral with a synchronised data transfer, and a ping-pong transfer. The initial implementation of the McBSP driver that interfaced with the SCbus used a combination of all three of these techniques. One DMA channel was configured to multiplex four WATM cells onto the McBSP, automatically re-initialising itself to transfer the next four cells using a ping-pong technique. Another DMA channel was configured to de-multiplex the received cells into ping-pong buffers in data memory. However, in order to allow for the raw bit rate over each CDMA code channel to be easily adjustable in the code, the final implementation did not use the DMA to service the McBSP. This is because the DMA controller could not achieve the complex multiplexing required fro certain bit rates. Nevertheless, the DMA controller allows for very flexible transfer schemes as illustrated by the following three examples.

C.4.1 Data Sorting Transfer

Many applications involve the use of multiple data arrays and often the arrays are available with the first elements arranged so that they're adjacent, the second elements adjacent, and so on. This is the manner in which data is received from the four 2Mbps SCbus channels via the McBSP running at 8Mbps; i.e. the first byte received is the first byte of channel 0, the second byte received is the first byte of channel 1, the third byte received is the first byte of channel 2, etc. But, the application may require all the elements of each array to be contiguous. Of course, the reverse situation may also be true. The DMA controller may be configured to rearrange the data into the desired format. Figure C.7 demonstrates this process.

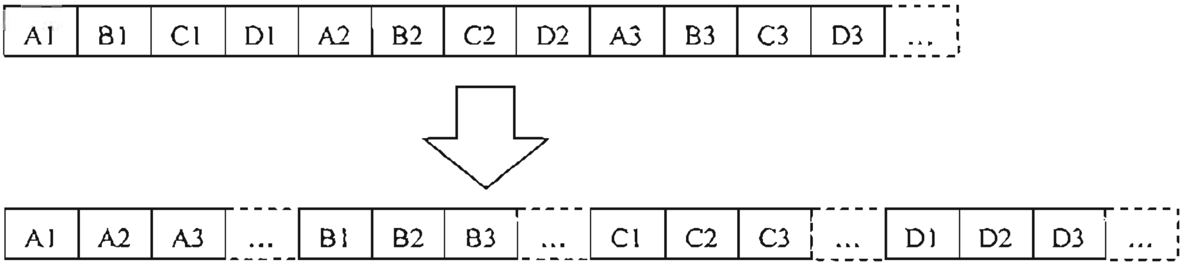


Figure C.7 Data Sorting Example

C.4.2 Synchronised Data Transfer

Synchronisation allows DMA transfers to be triggered by events such as interrupts from internal peripherals (such as McBSPs and Timers) or external pins. Three types of synchronisation can be enabled for each channel: read synchronisation, write synchronisation, and frame synchronisation. With read synchronisation each read transfer waits for the selected event to occur before proceeding. With write synchronisation each write transfer waits for the selected event to occur before proceeding. With frame synchronisation each frame transfer waits for the selected event to occur before proceeding.

To transfer data to and from a McBSP it is necessary to use read synchronisation to read from the data receive register (DRR) and write synchronisation to write to the data transmit register (DXR). Refer to Appendix C.3 for details on the McBSP. The McBSP issues a transmit event (XEVT) when a value has been copied from the DXR to the transmit shift register (XSR), signifying that the most recent data has been transferred out. Similarly, a receive event (REVT) is issued when a value has been copied from the receive buffer register (RBR) to the DRR, indicating that a new data value has been received. Hence, a DMA channel must be configured to wait for these events when synchronizing data transfer to a McBSP.

C.4.3 Ping-Pong Transfer

When data flow requires that a peripheral be repetitively serviced, as is the case for the McBSP in this application, it is appropriate to configure the DMA to automatically reprogram itself for subsequent transfers. This is achieved by running the DMA channel in auto-initialisation mode, and providing reload values for the source and destination addresses as well as the transfer counter. Transfers to and from two types of buffers may be implemented in this way: circular buffer transfers and ping-pong buffer transfers. In the ping-pong implementation a dual buffering scheme is used. Simply put, there are two input buffers and two output buffers. While the DMA transfers data to/from the one buffer, the CPU processes the data in the other buffer. The benefit of this is greatly improved when the ping and pong buffers are in different memory blocks, as there is no contention between the CPU and DMA when they are accessing different blocks.

References

[1] *ITU Telecommunication Indicators Update*, April-May-June 2000, http://www.itu.int/ti/update/archive.htm

[2] *World Telecommunication Development Report 1999: Mobile Cellular*, Executive Summary, ITU, October 1999, http://www.itu.int/ti/publications/wtdr_99/wtdr99.htm

[3] *"German mobile auction sets record"*, CNN.com, August 15, 2000, http://europe.cnn.com/2000/WORLD/europe/08/15/germany.mobile/index.html

[4] Dr Brian G. Marchent, Fujitsu Europe Telecom R&D Centre, *"Development of ATM and IP Networks for Future Mobile Multimedia Systems"*, pp. 7/1 – 7/32, Sixth IFIP Workshop on Performance Modelling and Evaluation of ATM Networks, Ilkley, West Yorkshire, U.K., July 1998.

[5] *ATM Wireless Access Communication System (AWACS)*, ACTS Project AC228, http://infowin.analysys.com/ACTS/RUS/PROJECTS/ac228.htm

[6] *Wireless Broadband CPN/LAN for Professional and Residential Multimedia Applications (MEDIAN)*, ACTS Project AC006, http://www.infowin.org/ACTS/RUS/PROJECTS/ac006.htm

[7] *System for Advanced Mobile Broadband Applications (SAMBA)*, ACTS Project AC204, http://www.infowin.org/ACTS/RUS/PROJECTS/ac204.htm

[8] *Wireless ATM Network Demonstrator (WAND)*, ACTS Project AC085, http://www.infowin.org/ACTS/RUS/PROJECTS/ac085.htm

[9] Riaz Ahmad, *"Next Generation of Broadband Network Architectures – Traffic Flow Control and End-to-End Performance"*, ATM '98: Sixth IFIP Workshop on Performance Modelling and Evaluation of ATM Networks, Participants Proceedings: Part 2 – Research Papers.

[10] M. Markaki, E. Nikolouzou and I. S. Venieris, *"Performance Evaluation of Scheduling Algorithms for the Internet"*, ATM & IP 2000: Eighth IFIP Workshop on Performance Modelling and Evaluation of ATM & IP Networks, Participants Proceedings: Tutorial Papers, T52.

[11] *"Internetworking Technology Overview"*, "Chapter 21. Asynchronous Transfer Mode (ATM) Switching", Cisco Systems Inc. http://www.cisco.com/univercd/cc/td/doc/cisintwk/ito_doc/index.htm

[12] Jouni Mikkonen, *"Wireless ATM Overview"*, Mobile Communications International, Issue 28, pp. 59 – 62, Feb 1996, available on the Magic WAND web site: http://www.tik.ee.ethz.ch/~wand/

[13] Ramjee Prasad and Tero Ojanperä, *"An Overview of CDMA Evolution toward Wideband CDMA"*, IEEE Communication Surveys, Fourth Quarter 1998, Vol. 1 No. 1, http://www.comsoc.org/pubs/surveys

[14] *"Multiuser Detection"*, Sergio Verdu, Cambridge University Press, ISBN 0-521-59373-5.

[15] Motorola Cellular Infrastructure Group, *"CDMA Technology & Benefits – An Introduction to the Benefits of CDMA for Wireless Telephony"*, March 1996. http://www.motorola.com/NSS/Technology/cdma.html

[16] Sunil Jagannath, *"An Adaptive Data Link Layer Protocol for Wireless ATM Networks"*, M.S. Thesis, University of Kansas, June 1997 http://www.tisl.ukans.edu/RDRN/

[17] *"Wireless Trends"*, Bell Labs Technology: Trends and Developments, April 1997.

[18] Ian F. Akyildiz, Janise McNair, Loren Carrasco Martorell, Ramon Puigjaner and Yelena Yesha, *"Medium Access Control Protocols for Multimedia Traffic in Wireless Networks"*, pp. 39 – 47, IEEE Network, July/August 1999.

[19] Xinri Cong (cong@cis.ohio-state.edu), *"Wireless ATM – An Overview"*, http://www.cis.ohio-state.edu/~jain/cis788-97/wireless_atm/

[20] James Aldis, *"On the Choice of Physical Layer Techniques for Wireless ATM"*, ACTS Summit 98, available on the Magic WAND web site: http://www.tik.ee.ethz.ch/~wand/

[21] Damian Gilmurray, Alan Jones, Oliver Mason, John Naylon and John Porter, *"Wireless*

ATM Radio Access Layer Requirements", ATM Forum Contribution 96-1057, Olivetti Research Limited.
http://www.cam-orl.co.uk/radio/atm96-1057.html

[22] D. Raychaudhuri and N. Wilson, *"Multimedia Personal Communication Networks: System Design Issues"*, pp. 259 – 288, Proc. 3rd Workshop for Third Generation Wireless Information Networks, Rutgers University, New Brunswick, NJ, April 1992.

[23] The ATM Forum: Technical Specifications Spec Watch
http://www.atmforum.com/atmforum/specs/specwatch.html

[24] John Porter and Andy Hopper, *"An ATM based protocol for Wireless LANs"*, Olivetti Research Limited, April 1994.
http://www.cam-orl.co.uk/abstracts.html#25

[25] J. Porter, A. Hopper, D. Gilmurray, O. Mason, J. Naylon and A. Jones, *"The ORL Radio ATM System, Architecture and Implementation"*, Olivetti Research Limited, January 1996.
http://www.cam-orl.co.uk/abstracts.html#68

[26] D. Raychaudhuri et al., *"WATMnet: A Prototype Wireless ATM System for Multimedia Personal Communication"*, IEEE Journal on Selected Areas in Communications, Vol. 15, No. 1, January 1997, pp. 83 – 95.

[27] Ricardo J. Sánchez, Joseph B. Evans, Gary J. Minden, Victor S. Frost and K. Sam Shanmugan, *"A Rapidly Deployable Radio Network – Implementation and Experience"*, Proceedings of ICUPC '98, Florence, Italy, October 1998.
http://www.tisl.ukans.edu/RDRN/documents/papers/icupc-98.pdf

[28] Ricardo J. Sánchez, Fadi F. Wahhab, Joseph B. Evans, Victor S. Frost and Gary J. Minden, *"Design and Evaluation of an Adaptive Data Link Control Protocol for Wireless ATM Networks"*, Proceedings of GLOBECOM '98, Sydney, Australia, November 1998.
http://www.tisl.ukans.edu/RDRN/documents/papers/globecom-98.pdf

[29] The Magic WAND (ACTS Project AC085) web site, http://www.tik.ee.ethz.ch/~wand/

[30] Ioannis Dravopoulos, Nikos Pronios, Anastasia Andritsou, Ioannis Piveropoulos, Nikos Passas, Dimitris Skyrianoglou, Geert Awater, Jan Kruys,
Neda Nikaein, Alain Enout, Stephane Decrauzat, Thomas Kaltenschnee, Thorsten Schumann, Jürg Meierhofer, Stefan Thömel and Jouni Mikkonen, *"Deliverable 3D5: Wireless ATM MAC – Final Report"*, ACTS Project AC085: Magic WAND.
http://www.tik.ee.ethz.ch/~wand/DOCUMENTS/docs/AC085_3d5.pdf

[31] J. Meierhofer, *"Data Link Control for Indoor Wireless ATM Networks"*, Wireless 98 Proceedings, pp.517-525, Calgary, Canada, July 1998.
http://www.tik.ee.ethz.ch/~wand/DOCUMENTS/papers/3etn064b_ps.zip

[32] A. Brand and A. Hamid Aghvami, *"Multidimensional PRMA with Prioritized Bayesian Broadcast – A MAC Strategy for Multiservice Traffic over UMTS"*, pp. 1148 – 1161, IEEE Transactions On Vehicular Technology, Vol. 47, No.4, November 1998.

[33] G. A. Cunningham, *"Delay versus Throughput Comparisons for Stabilized slotted Aloha"*, pp. 1932 – 1934, IEEE Transactions on Communications, Vol. 38, No. 11, 1990

[34] R. Majoor and F. Takawira, *"A MAC Protocol for Wireless ATM over CDMA"*, pp. 155 – 160, Proc. Comsig '98, Cape Town, September 1998, IEEE Catalog No. 98EX214.

[35] Ian F. Akyildiz, David A. Levine and Inwhee Joe, *"A Slotted CDMA Protocol with BER Scheduling for Wireless Multimedia Networks"*, pp. 146 – 158, IEEE/ACM Transactions on Networking, Vol. 7, No. 2, April 1999.

[36] Christiaan Roobol, Per Beming, Johan Lundsjö and Mathias Johansson, *"A Proposal for an RLC/MAC Protocol for Wideband CDMA Capable of Handling Real Time and Non Real Time Services"*, pp. 107 – 111, Proc. IEEE VTC, Ottawa, Canada, May 1998.

[37] Norival R. Figueira and Joseph Pasquale, *"Providing Quality of Service for Wireless Links: Wireless/Wired Networks"*, pp. 42 – 51, IEEE Personal Communications, October 1999.

[38] Sudhir Ramakrishna and Jack Holtzman, *"A Scheme for Throughput Maximization in a Dual-Class CDMA System"*, pp. 830 – 844, IEEE Journal on Selected Areas in Communications, Vol. 16, No. 6, August 1998.

[39] J. T. Wu and E. Geraniotis, *"Power Control in Multimedia CDMA Networks"*, pp. 789 –

793, Proc. IEEE VTC, 1995.

[40] Upkar Varshney, *"Error Control Techniques for Wireless ATM Networks"*, pp. 104 – 110, Proc. IEEE International Performance, Computing, and Communications Conference, Phoenix/Scottsdale, Arizona, U.S.A., February, 1999

[41] *"Implementation of a WCDMA Rake Receiver on a TMS320C62x DSP Device"*, Texas Instruments Application Report SPRA680, July 2000.

[42] H. Taub and D. L. Schilling, *"Principles of Communication Systems, Second Edition"*, McGraw-Hill, 1986, ISBN 0-07-100313-4

[43] Jian Zhang, Michael R. Frater, John F. Arnold and Terence M. Percival, *"MPEG 2 Video Services for Wireless ATM Networks"*, pp. 119 – 128, IEEE Journal on Selected Areas in Communications, Vol. 15, No. 1, January 1997.

[44] Michele Zorzi and Ramesh R. Rao, *"On the Impact of Burst Errors on Wireless ATM"*, pp. 65 – 76, IEEE Personal Communications, August 1999.

[45] GSM 05.03 Version 5.2.0 (gsmts_0503v050200p), August 1996, *"Digital cellular telecommunications system (Phase 2+); Channel coding"*.

[46] Janos Levendovszky, Cs. Vegso and E.C. van der Meulen, *"Nonparametric decision algorithms for CAC in ATM networks"*, Sixth IFIP Workshop on Performance Modelling and Evaluation of ATM Networks, Ilkley, West Yorkshire, U.K., July 1998.

[47] Xinping Guo, Colin Pattinson, Tony Dacre and Rebecca Strachan, *"Connection Admission Control Based on Traffic Measurements"*, Sixth IFIP Workshop on Performance Modelling and Evaluation of ATM Networks, Ilkley, West Yorkshire, U.K., July 1998.

[48] Tim Hentschel, Matthias Henker and Gerhard Fettweis, *"The Digital Front-End of Software Radio Terminals"*, pp. 40 – 46, IEEE Personal Communications, August 1999.

[49] Apostolis K. Salkintzis, Hong Nie and P. Takis Mathiopoulos, *"ADC and DSP Challenges in the Development of Software Radio Base Stations"*, pp. 47 – 55, IEEE Personal Communications, August 1999.

[50] Joseph Mitola III and Gerald Q. Maguire Jr, *"Cognitive Radio: Making Software Radios More Personal"*, pp. 13 – 18, IEEE Personal Communications, August 1999.

[51] *"PCI/C6200 Technical Reference Manual"*, Version 0.90, Blue Wave Systems, June 1998.

[52] *"SC4000 Universal Timeslot Interchange Data Sheet"*, VLSI Technology Inc.

[53] "Chapter Ps/Ra: Pipeline Stages / Radio Links", *"OPNET General Models"*, part no. D00122, MIL3 Inc., pp. 69 – 110.

[54] *"TMS320C6000 Peripherals Reference Guide"*, Texas Instruments Literature Number: SPRU190C.

[55] David Bell, *"TMS320C6000 DMA Example Applications"*, Texas Instruments Application Report SPRA529.