Multi-Media Wireless Access Networks: A Traffic Engineering Perspective

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Dedicated to my family and friends,

who at every step showed me the best way to go.

And to my supervisor and friend, Moshe Zukerman,

who gave his time and knowledge so generously.

Abstract

This dissertation addresses the fundamental question of optimal strategies to minimise the amount of transmission bandwidth required in multi-media Wireless Access Networks. Bandwidth is minimised under a set of realistic and disaster traffic conditions, subject to meeting quality of service requirements. Interactions between several network layers, as well as capacity required for signalling necessary to access the medium, are considered. Through comprehensive modelling of the operation of wireless access systems, under a variety of traffic flows, guidelines are provided for wireless network provisioning.

Although a layering model is very important to simplify telecommunications processes, a more inclusive approach is essential to globally optimise transmission parameters and link dimensioning. This thesis leads in the consideration of performance related interactions between the different layers including: Physical, Medium Access Control (MAC), Data Link, and Transport layers. The approach complements significant research performed by others on physical layer coding, particularly Turbo Coding, as well as work on enhancing Automatic Repeat Request (ARQ) protocols for retransmission of lost or erroneous packets. In each of these two bodies of research, a local optimisation has improved the performance in each layer separately, without consideration of changes to system performance as a whole.

MAC wireless protocols are analysed and synthesised in the thesis and one is developed using deadlock models to optimise transmission resource allocation. MAC structures capable of dynamically varying the amount of capacity allocated to each user are studied, keeping in mind

effects of any dynamic alteration in net capacity required by the real-time adjustment of the physical layer modulation and Forward Error Correction as well as higher layer ARQ retransmission.

A final major outcome relates to recovery of systems after network 'failure'. The transient or nonstationary congestion period triggered by the backlog of packets due to retransmission after a failure or sustained period of low transmission capability is modelled. Neglecting solutions to the survivability questions involving the use of yet more bandwidth to ensure QoS, this thesis provides detailed statistics on the performance of proposed systems under abnormal 'stress' traffic noting the relative performance of a range of Contention Resolution Algorithms (CRAs), and hence demand assigned MAC protocols, under extreme inter-station correlation.

Using three novel signalling channel capacity allocation schemes to study tree based and *p* persistence CRAs for inclusion in the Remote to Base Control section of a wireless MAC, a series of deadlock models are developed. One outcome of these models is a demonstration of the enhanced performance *Q*-ary tree CRAs under all of the Basic, Modified Basic and Binomial Deadlock. Another is a new understanding of the effect of request piggybacking on recommendations arising from each of the developed models.

Declaration

This is to certify that

- The thesis comprises only my original work towards the PhD except where indicated in the Preface,
- Due acknowledgement has been made in the text to all other material used,
- The thesis is less than 100,000 words in length, exclusive of tables, maps, bibliographies and appendices.

FRASER KEITH CAMERON, March 2001

Table of Contents

Figures

Tables

Acronyms

AAL	ATM Adaptation Layer
ABR	Available Bit Rate
ACK	Acknowledgement
APRMA	Aggressive PRMA
ARQ	Automatic Repeat Request
ATC	ATM Transfer Capability
ATM	Asynchronous Transfer Mode
AWGN	Additive White Gaussian Noise
BCH	Bose-Chaudhuri-Hoacquenghem
BER	Bit Error Rate
BIN	Binomial Deadlock Model
BS	Base Station
CAC	Connection Admission Control
CBR	Constant Bit Rate
CCS_M	Cyclic CMS Sharing with Multiple CMSs per Slot
CCS_S	Cyclic CMS Sharing with Single CMS per Slot
CDMA	Code Division Multiple Access
C_MAC	Core MAC
CMS	Contention Minislot
CQF	Channel Quality Feedback
C-PRMA	Centralised PRMA
CRA	Contention Resolution Algorithm
CRC	Cyclic Redundancy Code
CRI	Contention Resolution Interval

CSMA	Carrier Sense Multiple Access
D^2MA	Demand Distributed Multiple Access
DL	Data Link
DLC	Data Link Control
DS	Data Slot
EBE	Error Based Efficiency
EBSN	Explicit Bad State Notification
FAP	Free Access *Q*-ary Tree Protocol
FCS	Full CMS Sharing with Multiple CMSs per Data Slot
F-CPR	Fair Centralised Priority Reservation
FDM	Frequency Division Multiplexing
FDMA	Frequency Division Multiple Access
FEC	Forward Error Control
FIFO	First In First Out
FRCV	Fast Recovery Algorithm
GMSK	Gaussian Minimum Shift Keying
GPRS	General Packet Radio Service
HC	Head End Controller
HFC	Hybrid Fibre Coaxial
IETF	Internet Engineering Task Force
IP	Internet Protocol
LAN	Local Area Network
LIFO	Last In First Out
MAC	Medium Access Control
MASCARA	Mobile Access Scheme based on Contention And Reservation for ATM

MDR-TDMA	Multi-services Dynamic Reservation TDMA
MHP	Mobile Host Protocol
MPEG	Motion Picture Experts Group
MLAP	MAC Level Access Protocol
MPRMA	Minipackets PRMA
msSTART	Multi-slot n-ary Stack Random Access Algorithm
MT	Mobile Terminal
NACK	Negative ACK
OSI	Open Systems Interconnection
PGF	Probability Generating Function
PDV	Packet Delay Variation
PPA	Practical Probability of Absorption
PRMA	Packet Reservation Multiple Access
PRMA/DA	PRMA / Dynamic Allocation
PRMA-DSV	PRMA – Data Steal over Voice
PSK	Phase Shift Keying
QAM	Quadrature Amplitude Modulation
QBE	Queueing Based Efficiency
QI	Queue Identifier
QoS	Quality of Service
QPSK	Quadrature Phase Shift Keying
R-ACK	Request - ACK
RAMA	Resource Auction Multiple Access
R-B	Remote to Base
RLC	Radio Link Control
RS	Reed-Solomon

RTT	Round Trip Time
Rx	Receive
SCOP	Specific Connection Oriented Protocol
S-MAC	Supervisory MAC
SNR	Signal-to-Noise Ratio
TCP	Transmission Control Protocol
TDD	Time Division Duplex
TDM	Time Division Multiplexing
TDMA	Time Division Multiple Access
Tx	Transmit
UBR	Unspecified Bit Rate
UPC	Usage Parameter Control
VBR	Variable Bit Rate
VC	Virtual Channel
WAND	Wireless ATM Network Demonstrator
WATM	Wireless ATM
WCDMA	Wideband CDMA
WDM	Wavelength Division Multiplexing
WTCP	Wireless TCP

1. INTRODUCTION

1.1 Looking Forward

Wireless communications are becoming an ever more important feature of global communications, with recent times recording explosive growth in their popularity and capability [HKL97]. Commencing with the emergence of first generation wireless cellular telephones 15 years ago, wireless communications are now in their second generation, comprising systems with digital voice and data transmission. Within the past decade or so, communication technologies have changed from analogue (first generation) to digital (second generation), from bulky high-powered equipment to more convenient lower powered handsets, and from a one-tier cell structure to a hierarchical cell structure.

The challenge at hand is the development of a new wave of wireless systems – flexible networks capable of carrying multimedia traffic at very high rates. This third generation of communications comprises a comprehensive set of 'seamless communications' for users, stationary or moving, delivering applications such as the electronic newspaper, multimedia teleconferencing, and portable computing reflecting projected communications requirements [Pra97], [Goo90], [HKL97]. According to current design approaches, we expect future telecommunication networks to be based on a relatively simple core wireline network together with many sophisticated, complex, and expensive (mainly wireless) digital access networks. As such, access networks incur much of the cost of network construction and maintenance. Their efficient operation and accurate dimensioning has the potential to allow significant savings.

Any mobile network should be capable of interfacing with wired multimedia networks supporting Asynchronous Transfer Mode (ATM) [Sch96], a technology based on a store and forward network for fixed size cells, and supporting Quality of Service (QoS) requirements [Sta94]. ATM may be a significant future carrier of wired multimedia traffic and so the use of a modified ATM over the wireless link furthers the move toward a single compatible standard for communications [Sch95]. Spectrum has already been reserved for the high-performance Local Area Network (LAN) HIPERLAN and SUPERnet at 5 GHz [CHR97]. Transport Control Protocol/Internet Protocol (TCP/IP) is likely to be the communication protocol suite for transport over the wireless access network and the Internet Engineering Task Force's (IETF's) Mobile IP Working Group is currently developing IP capable of allowing a mobile host to change networks entirely transparently, with the user to remaining connected to the best possible network as he or she moves [Joh96].

1.2 *Focus of this Thesis*

This thesis investigates teletraffic issues in the design and development of multi-media wireless networks systems. It aims, through comprehensive modelling of the operation of these systems under a variety of traffic flows, to provide guidelines for multi-media wireless networks network provisioning and survivability. It is worth noting that, although the aim of this project considers Wireless ATM

(WATM) to be significant amongst technologies sampled, most aspects of this work are general across wireless (and even wired) multi-service access networks. Such networks include wireline access networks including IEEE 802.14 Hybrid Fiber Coaxial (HFC) and, in more abstract form, many other wireless and wireline multi-service access systems including General Packet Radio Service (GPRS) and Enhanced GPRS (EGPRS).

The thesis establishes principles of multi-media wireless network dimensioning, leading to the development of specific guidelines on how to perform cost effective design and dimensioning of future access networks. The fundamental question of the best strategy to minimise the amount of bandwidth required under given realistic traffic conditions, subject to meeting specified Quality of Service (QoS) requirements for the different services, is addressed [Zuk98]. The project provides an efficient Medium Access Control (MAC) protocol as well as a detailed insight into optimal operation, performance and dimensioning of access networks. Results presented here give telecommunication providers the basic principles as well as specific guidelines on how to perform cost effective design and dimensioning of future wireless multi-service access-networks.

Unfortunately, it is not possible to obtain a brute force optimal dimensioning solution by relying on a comprehensive simulation that takes into consideration all aspects of the problem. Hence, to approach the question of how to optimise the use of the bandwidth required for wireless access, subject to meeting specified QoS requirements for the different services under realistic scenarios of traffic conditions, a comprehensive set of capacity requirements must be taken into account. These include:

- Traffic dependent signals related to capacity reservations and Automatic Repeat Request (ARQ);
- Capacity required for meeting QoS requirements such as queuing delay (to avoid protocol time-outs and packet retransmissions) and packet loss (due to buffer overflow);
- Retransmissions of erroneous packets; and
- Modulation and error correction redundancies.

Although Open Systems Interconnection (OSI) layering is very important to simplify telecommunications processes, a more comprehensive approach is essential to globally optimise transmission parameters and link dimensioning. In particular, we consider performance related interactions between the different layers: the Physical, MAC, Data Link (DL), and Transport layers. This approach compliments the significant research performed on physical layer coding, where progress has been made in error correction using Turbo coding [BGT93] [HHR97], and other work that has enhanced ARQ protocols for retransmission of lost or erroneous packets. In each of these two bodies of research, a local optimisation was performed to improve the performance of each of these layers separately, even though each layer has the potential to interfere with the performance of the other, requiring a global approach.

Another important part of this work centres on the general area of analysis and synthesis of wireless MAC protocols. A wireless MAC protocol is synthesised using the above mentioned global optimisation results to increase the efficiency of transmission resource allocation. In general terms, MAC structures capable of dynamically varying the amount of capacity allocated to each user are analysed, keeping in mind a dynamic alteration in net capacity required by the real-time adjustment of the physical layer Higher Level Modulation (HLM) and

Forward Error Correction (FEC) as well as higher layer ARQ retransmission. The MAC protocol proposed in this thesis has some similarities to modern multiservice MAC protocols described in [IZ98a], [IZA97], [IZA97a], and [NBJ97].

In evaluating the use of CRAs in a new MAC system, it is important to note that, in any wireless network, atmospheric and other interference effects lead to transitory periods in which the system cannot transmit data. Hence the recovery of the system after network ‘failure’ is a central issue. A related issue is the transient or nonstationary congestion period triggered by the backlog of packets due to retransmission after a failure or sustained period of low transmission capability. Hence it is necessary to address the survivability of MAC proposals contained in this thesis under transient traffic patterns, including periods of extreme load and traffic correlation as numbers of remote stations attempt to retransmit after failure of the wireless link.

Solutions to survivability questions involving the use of yet more bandwidth to ensure QoS are wasteful. A more promising mode of analysis is to investigate the performance of systems under abnormal ‘stress’ traffic and to suggest approaches to minimise extra network provisioning so as to guarantee limited transient response in the wireless network. In wireless access systems, it is the remote to base control section (see Chapter 2, below) that is the most sensitive to traffic characteristics and volume.

Collision based capacity request signalling channel relying on the Slotted ALOHA multi-access principle is studied for inclusion in a proposed MAC protocol for HFC and wireless access networks. Chapter 6 studies the performance of a *p* persistence contention resolution algorithm (CRA), subject to extreme inter-station correlation, by means of a discrete-time Markov chain analysis. That research contributes valuable understanding to the performance of the S-ALOHA CRA, however, it is vital to also consider other CRA contenders for the hybrid MAC.

Finally, this thesis completes and extends the research from the author's contribution in [IZC00] to consideration of three signalling channel capacity allocation schemes using tree-based algorithms for the resolution of contention in the R-B Control section of a proposed MAC. The work leads to a deadlock model for a multi-service medium access protocol employing a form of multi-slot *Q*-ary tree algorithm reported by the author in [CZI00]. Multi-slot Stack Random Access Algorithm (msSTART) is proposed for use in IEEE 802.14 HFC networks and Chapter 6 outlines the performance of *Q*-ary tree CRAs in the wireless environment using the Basic, Modified Basic and Binomial Deadlock models. Finally, the thesis considers the effect of request piggybacking on the above research.

1.3 Subdivision of this Thesis by Chapter

Chapter 2: An Overview of 3rd Generation Wireless Issues provides a brief overview of some of the issues that the dissertation discusses in later chapters. It provides a brief account of the path to 3rd generation communications as well as the research issues outstanding.

The chapter includes an outline of currently evolving wireless technologies, and an evolution path to the future. Discussing proposed and current schemes of FEC rate adaptation as well as MAC and DLC issues, the structure of the wireless access network to be studied in the remainder of the thesis is outlined.

Chapter 3: Inter-Layer Interaction covers some guidelines for development of FEC, ARQ, and Modulation Gain based enhancements of multi-media, multi-access wireless networks. In this Chapter, a basic limit for the use of FEC in such access networks will be presented. A layer-based optimisation for a more global solution is rejected, and the effect of traffic correlation on suggestions is considered.

Applying a model for analysis that separates queueing based efficiency from error-based efficiency, the thesis develops a set of rules for the use of FEC and ARQ in a multi-access, multimedia wireless access network. The outcome is a rule describing a limit for the use of FEC in any wireless network. By then investigating the effect of different channels and small block Data Link Control (DLC) schemes, Chapter 3 is able to give some insight into when FEC should still be avoided, despite network conditions falling within our limit for use of FEC. Numerical results using a unit of error based efficiency measured in (bits/s)/Hz are provided.

Chapter 4: TCP over Wireless Access Networks describes a subset of issues arising out of the use of the congestion control mechanism in TCP/IP over wireless access networks. The chapter commences with a brief overview of the two TCP schemes most implemented in today's Internet. A number of proposed solutions for the so called 'slow start problem' as it is relevant to wireless access networks are then reviewed. After commenting on each of the categories of solution, a DLC small block ARQ approach is adopted and described, before methods for optimising block size in the proposed DLC small block ARQ scheme are demonstrated.

Chapter 5: Contention Resolution Algorithms furthers the concept of a CRA and its use in both the random assignment environment and hybrid demand assigned MACs. Considerations in the design of a Demand Assigned MAC are discussed in Chapter 2 and a proposal for a demand assigned MAC with central control is outlined. Central to that proposal is the use of a CRA to regulate bandwidth reservation by the mobile terminals In this chapter we specify a number of CRAs that will be further studied in Chapter 6 and review efficiency statistics for each algorithm.

Chapter 6: Deadlock Models for a Multi-Service MAC Protocol: Comparing p Persistence and Tree Based CRAs commences by providing a number of models for efficiency evaluation of the *p* persistence CRA proposed for use in the Multi-services Dynamic Reservation Time Division Multiple Access (MDR-TDMA) MAC that is described in Chapter 2. The development and analysis of a suite of disaster scenarios leads to development of a set of conditions that will lead to deadlock. Using the channel signalling schemes introduced in the chapter, we describe the effect of each scheme on the stability of the CRA region of the proposed wireless MAC, and hence its effect on the system as a whole.

Using a Markov chain analysis, a mathematical model is developed to describe the performance of each CRA under disaster scenarios. This model is tested using a system simulated using C++ for the case of *p* persistence and the Pseudo-Bayesian Estimator (an adaptive *p* persistence algorithm).

The chapter goes on to extend this analysis to other CRAs, comparing the *p* persistence solution to that of tree based CRAs including Multi-Slot *Q*-ary Stack Random Access Algorithm (msSTART). The chapter quantifies the extensive gains in efficiency and stability that can be achieved via the employment of these more advanced protocols in the remote to base signalling section of the demand assigned TDMA frame. Further, gains from the use of piggybacking future requests on the back of previously assigned data frames are evaluated. The CRAs are investigated under a series of channel signalling schemes including: Full CMS Sharing; Multi-CMS Cyclic CMS; and Single CMS Cyclic Sharing (Priority Access) and the effect of errors on both *p* persistence and the tree based CRAs are evaluated.

1.4 Contributions of this Thesis

Below is a list of the major contributions of this thesis. Contributions are sorted in approximate order of appearance with section numbers indicating where the point is first discussed in the thesis. Relevant publications are shown in parentheses. The fact that the contributions span a number of diverse topics is a legacy of the inter-layer nature of this research and the need for a global and integrated solution to issues of dimensioning in wireless multi-access multi-media networks.

1. Development of an analytical model for wireless efficiency optimisation that considers all of: FEC redundancy, ARQ retransmission, and adaptive modulation at the transmitter and receiver for both real-time and non-transparent traffic (Chapter 3, [CZG98a] [CZG98]).
2. Extension of a separation of Error Based Efficiency and Queueing Based Efficiency, and the provision of a set of quantitative results for effects on overall system efficiency of a number of realistic

disturbances in the wireless channel (Chapter 3, [CZG98a] [CZG98]).

3. Development and evaluation of a rule that gives a hard limit for the use of FEC in any wireless multi-media access network under any conditions (Chapter 3, [CZG99a]).
4. Quantitative analysis for when FEC should be used in systems displaying different levels of error correlation (Chapter 3, [CZ01]).
5. Criticism of a number of solutions proposed in the literature to issues associated with TCP over wireless for the most common forms of TCP/IP currently incorporated in the Internet infrastructure (Chapter 4, [CZG99a]).
6. Verification and further development of a Link Layer solution to TCP 'slow start' utilising the analysis in Chapter 3 to extend a means of optimising ARQ packet size (Chapter 4, [CZG99a]).
7. Development and analysis of 3 deadlock models for the investigation of survivability for MDR-TDMA MAC R-B Control regions (Chapter 6, [IZC00] [CZI00] [CZ99a] [CZI99])
8. Verification by simulation of analytical results for the survivability of the *p* persistence under deadlock conditions (Chapter 6, [IZC00] [CZI00]).
9. Investigation and verification by simulation of the effect of 3 signalling schemes on the throughput and survivability of the *p* persistence CRA (Chapter 6, [IZC00] [CZI00]).
10. Development and verification of analysis for the survivability of tree based CRAs under three developed models simulating deadlock conditions (Chapter 6, [IZC00] [CZI00] [CZI99] [CZ99a]).
11. Development and verification by simulations an analysis of the effect of 3 signalling schemes on the throughput and survivability

of the tree based CRAs under deadlock conditions (Chapter 6, [IZC00] [CZI00] [CZI99] [CZ99a]).

12. Development of analysis and results for the effect of data request piggybacking of all CRAs considered (Chapter 6, [CZ99a]).
13. Development of analysis and results for the effects of errors on the performance of the CRAs considered (Chapter 6, [IZC00] [CZI00] [CZI99] [CZ99a]).
14. Provision of recommendations and guidelines for the use of msSTART as the CRA for the R-B Signalling portion of the proposed MDR-TDMA wireless MAC (Chapter 6, [IZC00] [CZI00] [CZI99] [CZ99a]).

1.5 Publications by the Author Related to this Thesis

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2. AN OVERVIEW OF 3RD GENERATION WIRELESS ISSUES

2.1 Introduction

According to current projections, future wireless personal communicators – commonly termed 3G Wireless – will, in the near future, have extensive capabilities beyond those available today. Using the currently marketed WAP as a mere indicator of the first possibilities of the medium, many innovative services are predicted for a handset that has the potential to become a versatile, go-anywhere device for both voice, and all other forms of communications services.

Support for multi-media implies higher service standards and reliability as well as significant additional capacity, clearing the way for a host of new services. This chapter presents an introduction to a number of engineering challenges remaining in the push towards higher bandwidth, more flexible mobile communications. The reader is introduced to the focal points of the later chapters of this thesis and provided with information on current and near term wireless access network design.

The chapter commences with a description of application traffic types that must be accommodated in a future wireless network as user needs evolve in sophistication. Using that base, the work then describes the interrelated nature of the physical, data link and network OSI layers, outlining reasons for the necessity to take a global approach to efficiency optimisation in wireless access network design.

Keeping in mind the interrelationship and flow on effects of optimisation in any single area, efficiency tradeoffs in FEC, ARQ, Adaptive radio and MAC use are described and the benefits and drawbacks of each technique are considered in preparation for Chapter 3. A number of considerations and design parameters in all of FEC, ARQ, adaptive modulation and MAC specification are outlined and several current proposals for wireless network architectures in each of these areas are reviewed. Finally the chapter ends with a brief overview of network congestion and fairness criteria.

2.2 Application Traffic Types

Any study of wireless access networks must first define the types of traffic for which the network must be designed. In this section a quick introduction to the diverse forms of probable traffic is given to introduce the reader to the challenges facing a traffic engineer in the provision of timely service capable of satisfying user demand. Flexibility to implement a wide range of communications applications over the communications infrastructure is required as the traffic density and characteristics are highly dependent on applications running over the wireless link.

Multimedia applications can be classified into two broad classes depending on the characteristics of the generated traffic. These categories are real-time applications, and non-real-time applications. Audio and video are examples of real-time traffic. They are highly delay sensitive, becoming useless if not transported to their destination within certain delay bounds. A file download, on the other hand, is an example of a non-real-time application that is not delay sensitive.

Voice traffic is a low bandwidth audio application and is characterised as constant bit rate traffic as audio codecs generate packetised voice samples at a constant rate within a talkspurt. In order not to inhibit normal conversation, transmission delay including multiple access delay as well as transmission and propagation delay, together with any delays associated with multiplexing in an integrated voice/data environment, must not exceed 300ms for full duplex voice. Moreover, delay cannot fluctuate significantly between talkspurts if the natural rhythm of the speech is to be preserved. Excessive delay leads to discards of segments of the speech using a strategy known as speech clipping. Typically clips larger than 50ms must not occur with a probability greater than 2% [GS85].

As well as having larger bandwidth requirements than their audio class counterparts, video class applications are able to tolerate higher error rates. Videophone type applications can be classified as constant bit rate connections, and interactive video or video on demand, with their more bursty nature, can be classified as variable bit rate services.

In the non-real-time class, packets do not demand stringent delay requirements, however, most of the applications in this class cannot tolerate errors to the same level as the real-time applications. A neat division of this class of applications is to distinguish interactive and non-interactive services. Typical examples of the former are FTP, WWW, Telnet, and database applications. Examples in the latter division would be electronic mail, paging and fax. In general the bandwidth requirements of the non-interactive class are less by comparison with the interactive class.

2.3 Network Congestion

This section moves away from introducing traffic issues from a user perspective and sets out some principles for network focussed traffic engineering. In particular, it sets out issues relating to network congestion that must be kept in mind when analysing issues in Chapter 3 and 4 of this thesis.

Network congestion occurs when there are insufficient network resources to transmit traffic sent to that network by the end nodes. Data are lost either through a network component ‘dropping’ packets via buffer overflow, or via the correctly delivered data violating its QoS limits, usually through excessive delay. Some information is transmitted transparently, that is, information that is lost in either of these ways is simply not available at the receiver leading to user noticeable degradation in higher layer application performance. Other transmission mechanisms are non-transparent, meaning that data integrity can be maintained by retransmission of data. However, throughput is severely restricted when retransmissions take place, especially where protocols retransmit in batches or blocks including both the lost packet and a number of the successfully transmitted packets.

[ZC94] defines *goodput* as the useful throughput of a network. That paper contains useful figures displaying the relation between goodput and wasted redundancy as the load on the network increases. As the offered load increases beyond a threshold in a network lacking congestion control, the useful throughput decreases rapidly as a proportion of the utilised capacity of the network until all capacity is consumed for retransmissions – a phenomenon known as *congestion collapse*.

The role of *congestion control* is to reduce the amount of redundant transmissions and to maintain the goodput as a reasonable percentage of the net capacity of the network. There are many strategies for this including:

- Utilising more efficient ARQ policies (including *selective repeat* policies although these are considered somewhat complex and TCP, OSI Transport Protocol Class 4, and others do not implement them);
- Requiring users to follow an *adaptive control scheme* whereby load is reduced when congestion is experienced (as followed by all of the TCP variants following RENO);
- Optimising the network adaptively (changing the level of FEC and ARQ redundancy utilised by the network in response to continual testing of the state of the external redundancy);

Note though, referring to the second point, that current version of such schemes can achieve congestion control only if all users act in a co-operative manner. Hence it is possible for a malicious user to continuously send packets, causing the other compliant users' adaptive control algorithms to detect high network utilisation and thus back off. For this reason, it is essential for the network operator to introduce mechanisms ensuring that under overload conditions, no user gets more than a *fair* share of the network resources. Under this scenario, if some users fail to adapt their rates under network congestion, then they suffer without affecting the goodput of the co-operative users [Iva97].

2.4 The Fairness Criterion

In addition to concerns about congestion collapse, there is a concern regarding ‘fairness’ for best effort traffic. For wireless access networks in which TCP/IP is used, equitable sharing of bandwidth depends on all flows are running compatible congestion control algorithms RFC2914. For TCP this means congestion control algorithms conformant with current specifications in RFC793, RFC1122, and RFC2581.

In [ZC94] fairness is viewed as the treatment of each user in a way that maximises the thoroughput of all users whilst minimising bandwidth requirements – the *maxmin* criterion [Ber92]. No user can achieve a higher bandwidth than a controlled user. In particular [Iva97] identifies the following features from [ZC94]. The scheme must:

- Penalise only those users that cause overload;
- Allow for prioritised apportionment of capacity;
- Deal with throughput rather than delay fairness; and
- Achieve globally efficient outcomes.

RFC2914 identifies that, even in the internet world (where TCP achieves only the last two criteria required of an architecture by [ZC94]), the issue of fairness has become increasingly important for the following reasons:

- Using window scaling, individual TCPs can use high bandwidth even over high propagation delay paths;
- With the growth of the world wide web, users increasingly want high bandwidth and low delay communications that do not share with less urgent communications during times of low resources;
- The proliferation of TCP implementations has seen the introduction of some which fail to implement the TCP congestion avoidance

mechanisms either because of poor engineering, or deliberately so as to claim more bandwidth;

- Alterations in granularity even in correctly implemented TCP can cause more aggressive performance.

In the wireless environment, however, it is probable that the second criterion from [ZC94] (reported on the previous page but not implemented in TCP) is the most important because some forms of projected applications in the wireless world are likely to require more bandwidth than others. A scheme that allocates bandwidth without regard for the type of transmission considered, limits the capabilities of the medium. Indeed, under congestion conditions, it may become commercially important to prioritise connections by determining user and connection value. Additionally, services could be characterised so as to hide congestion from a high value user by performing a progressive restriction of non-delay critical connections. Sections 6.4.1 and 6.5.7.1 add important insight to the design of wireless access architectures capable of effectively implementing the second criterion of maxmin fairness.

2.5 Challenges and Interactions Between Layers

Personal and terminal mobility will require better area coverage, improved signal quality and reliability, as well as the ability to interface with wired broadband multimedia networks based on TCP/IP and/or ATM [CS93] in order to satisfy user traffic demands. Before such a broadband radio network can provide multimedia services based on specified QoS requirements and be integrated with the existing fixed networks, certain well-known obstacles must be recognised. These include:

- The limitation of radio channel capacity versus that of fibre optics; and
- Higher processing delay due to special requirements of radio channels leading to powerful error correction, equalisation, voice encoding and interleaving – limitations significantly affecting the performance of radio based networks [Sch95].

Hence, unlike the case in wired optical fibre networks, where QoS requirements are mainly related to cell or packet loss probability due to congestion, in wireless broadband networks it is necessary to also consider access delay and channel loss. Indeed, the most significant difference between wired and wireless transmission is that wireless (especially mobile) transmission is very error prone.

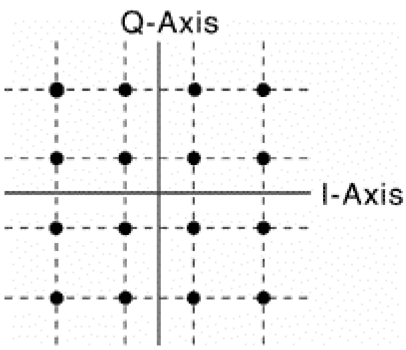

Figure 1. Graphical representation of a transmission constellation for 16 Quadrature Amplitude Modulation (QAM).

In a *transparent mode* of transmission (used by real-time video or audio connections), the two options to improve the Bit Error Rate (BER) and meet QoS requirements are:

- To use FEC; and/or

- To reduce the number of symbols in the transmitted constellation (Figure 1 shows the transmission constellation for 16 QAM and Figure 2 shows the transmission constellation for 4-PSK and 8-PSK) [Pro95] [BCE99].

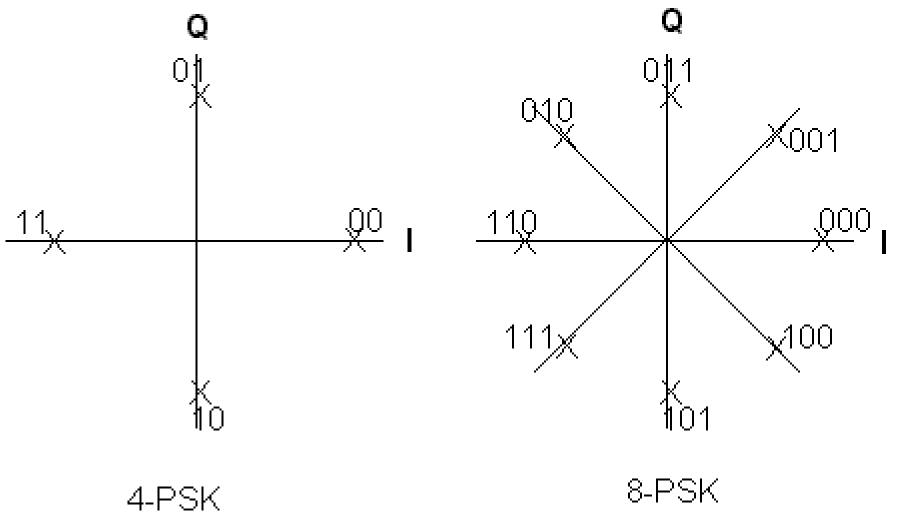

Figure 2. Graphical representations of the constellations for Quadrature Phase Shift Keying (4-PSK) and 8-PSK.

In *non-transparent mode*, all of the above, as well as an ARQ protocol (Figure 3), can be used to avoid or recover errors.

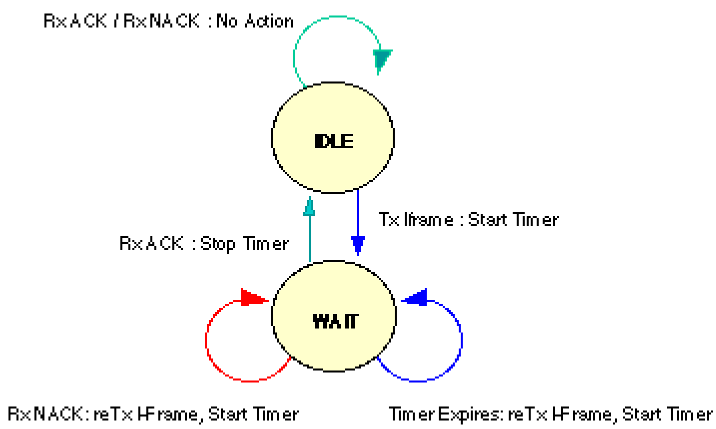

Figure 3. State Diagram for a simple stop and wait ARQ protocol.

In considering these mechanisms for error detection and recovery, it is important to consider the interrelated effects of their use. An increase in the size of the number of symbols in the transmission constellation gives additional capacity per unit bandwidth, however, the penalty is an increase in BER at a given Signal to Noise Ration (SNR) and, with it, an increase in the required number of retransmissions by the higher layer protocols (Link Layer ARQ or end-to-end ARQ).

On the other hand, retransmissions of erroneous blocks of data (ATM cells, packets or TCP windows, depending on the level at which ARQ is implemented) increase the amount of bandwidth required for the transport of a set user bit rate, adding redundancy. So too does the use of a more powerful FEC. Further, the size of the ARQ retransmission frame can have a significant effect on efficiency where propagation delay and header redundancy must be traded off against added waste from the retransmission of large blocks containing few errors. Thus there is a clear trade-off between modulation and (re)transmission redundancy.

The requirements for any multi-media, multi-access, wireless access network specification can be separated into two categories: the radio access protocol layers to handle the wireless channel specific functions; and the mobile protocol extensions for mobility management functions to handle personal terminal mobility. We focus on the radio access layers, which consist of [NBJ97]:

- A radio Physical Layer, capable of high-speed Physical Layer transmission and reception;
- A MAC layer for efficient sharing of the available bandwidth among multiple users, along with QoS management;
- A DLC layer to overcome radio channel impairments; and
- A wireless control layer for radio resource management.

A reference WATM stack is shown in Figure 4.

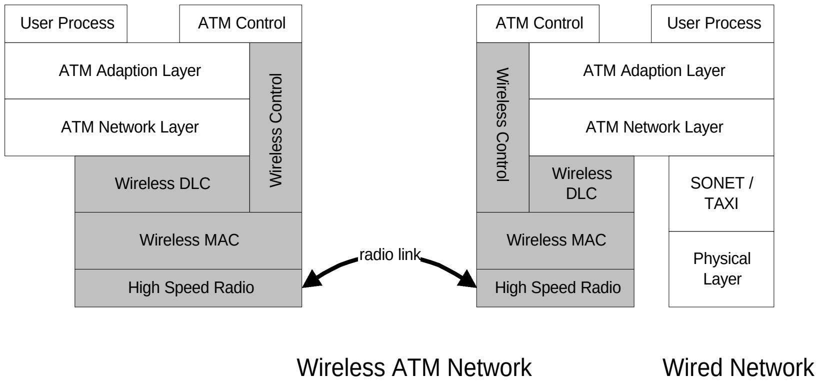

Figure 4. Interface between the fixed and wireless networks, relative position of the DLC and MAC layers.

This study shows that adaptive modulation is a powerful tool for performance optimisation – the size of the constellation is an essential part of data services in wireless networks and the choice of that constellation has a high impact on the BER. Further we demonstrate that it is possible to find an optimal constellation for use at a particular SNR per bit.

2.6 Radio Physical Layer

WATM and other generic broadband wireless access, possibly using TCP/IP, requires a high-speed radio modem capable of providing reasonably reliable transmission in microcells and picocells. The systems will operate on various frequency bands according to national regulation, however this is likely to be associated with the 5GHz U-NII band in the US and the HIPERLAN band in Europe [NBJ97]. The modem must be able to support burst operation with relatively short preambles and may be required to be capable of transmitting a number of constellations of symbols.

This section introduces the issues associated with FEC and adaptive modulation that will be further considered in Chapter 3 of the thesis and describes in depth FEC and adaptive modulation advances in current and near term wireless access technologies.

2.6.1 Physical Constraints of the wireless environment

Unlike transmission in the wireline environment where it is possible to increase the available transmission spectrum by using another wavelength or installing another cable, wireless transmission is limited by available spectrum. Communication is impaired by interference and multipath propagation causing fading and delay spread [MNK97]. In this environment, communication channels linking base stations and terminals are unpredictable and highly variable with time, their capacity is limited by comparison with fibre optics, and there is a higher processing power needed at terminals due to special requirements of radio channels (powerful error correction, compression, equalisation, voice encoding and interleaving) [CM97]. Overcoming such physical constraints requires the introduction of coding and other recovery features that introduce delay to the link, meaning that maximum delay criteria must be considered alongside satisfaction of CLR requirements in determining QoS. Hence the design of a hybrid wireless access networks will need to satisfy a greater set of QoS constraints that those of a purely wired network. Further, the error correction capability of the physical and data Link Layers will be of vital importance to technology [CM97], [ZHG98], [Ray97], [XNY95], [NBJ97].

2.6.2 Forward Error Correction

An important issue for the viability of multi-media wireless networks will be the development of efficient yet powerful methods for error correction in multi-media wireless networks. Excessive FEC redundancy (code rate R too small) will reduce useful data transmission bandwidth and insufficient FEC redundancy (R too large) will cause excessive and inefficient retransmission at the Link Layer ARQ.

Although the use of ARQ is generally better for the correction of burst errors (due to atmospherics, multipath fading and multi-user interference) and the FEC best for random noise error, use of an interleaver can distribute burst errors making the use of FEC efficient for both types of errors.

2.6.3 Adaptive Modulation

Adaptive modulation is defined as the ability to alter the number of bits transmitted over a wireless link at any particular instant via the use of real-time channel feedback mechanisms.

The motivation for an adaptive modulation scheme is to adjust the method of transmission to ensure that the maximum efficiency is achieved over the link at all time. In general, a modulator accepts a number of binary digits at one time and transmits a single symbol over the channel. This provides Modulation Gain – a term that we denote G throughout the thesis, representing the increase in capacity in the units (bits/sec)/Hz on a wireless link as a result of having altered the size of the symbol constellation. For the case of transmission of multiphase signals including Phase Shift Keying (PSK), channel bandwidth is simply the bandwidth of the equivalent lowpass signal pulse for PSK. We assume that bandwidth is approximately equal to the reciprocal of the duration of the pulse (for the case of a single sideband). Hence

$$G = \log_2 M \qquad (1.)$$

That is, G which is given by G = (data rate / bandwidth required) is defined similarly to spectral efficiency where M is the number of symbols in the modulation constellation.

Later chapters of this thesis will consider tradeoffs in these areas in depth. Further, proposals to counter some of the drawbacks of the use of FEC and adaptive modulation will be outlined and evaluated. This chapter continues by introducing some specific frameworks using adaptive FEC and modulation in current and already specified wireless access networks.

2.6.4 Specific Design – EGPRS and Other Advanced Systems Utilising FEC

2.6.4.1 Introduction

Standardisation is currently underway to Enhanced Data rates for GSM Evolution (EDGE). Enhanced Circuit-Switched Data (ECSD) and Enhanced General Packet Radio Service (EGPRS) are enhancements for circuit-switched and packet-switched data respectively for service over a TDMA frame. 8-ary PSK is being used in EDGE and better radio link control (RLC) procedures are being defined using link adaptation and incremental redundancy to achieve better delay / throughput performance over a wider range of operating conditions.

Figure 5 depicts a likely commercial migration strategy to 3G services, and Figure 6 presents a block diagram of an EDGE modulated system. Figure 5 can be used as a road map to the various technologies described in this section and demonstrates the amorphous nature of the term ‘3G Wireless’.

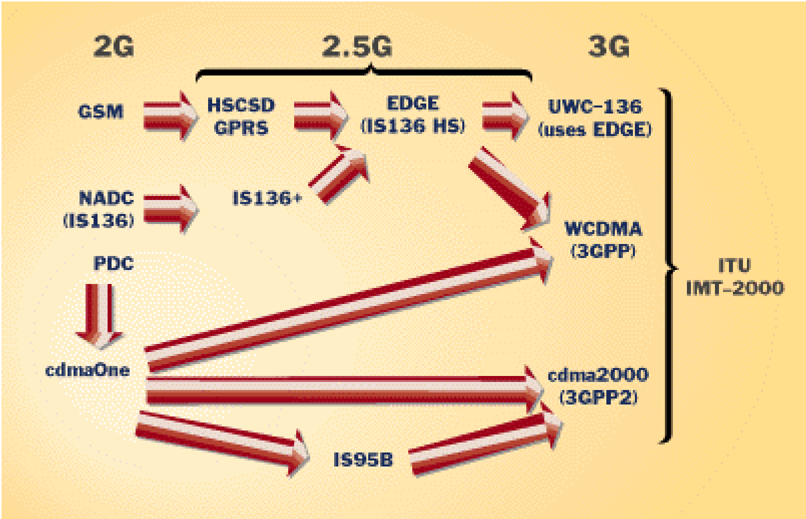

Figure 5. Migration paths to 3G for service providers [Sha00].

Figure 6 is one example of one section of the physical layer workings of a wireless access technology. The serial bit stream from a source is converted into 3-bit words and mapped to the 8-PSK constellation using Gray encoding. The symbols are then rotated by $3\pi/8$ radians to ensure that the envelope of the signal does not go to zero. Next, the symbols are unsampled and filtered using a linearised Gaussian filter (similar to, but different from the method used for GSM) [Sha00].

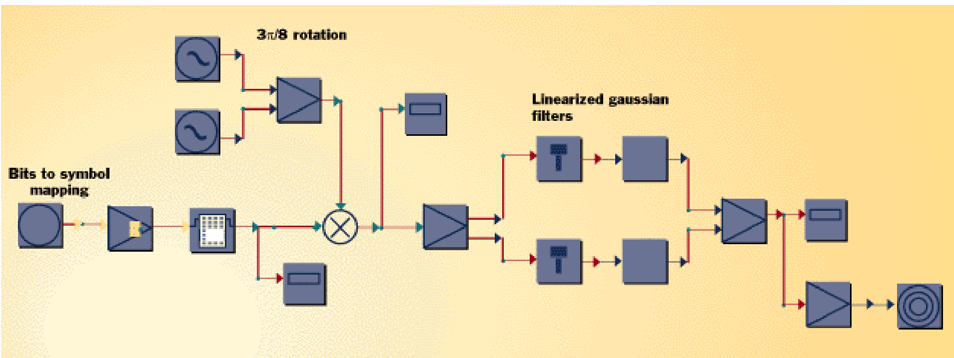

Figure 6. The EDGE modulator [Har00].

The following sub-sections describe a number of technologies in detail, commenting on the nature of adaptivity in the physical layer for each.

The differences between incremental and non-incremental redundancy and the benefits for a system employing incremental FEC redundancy together with adaptive modulation are highlighted. Further, the mechanical means for the implementation of adaptive systems is explored. In Section 2.6.4.6 below, is a summary of current and proposed rate adaptation structures.

2.6.4.2 Incremental Redundancy – GPRS-136

This section gives one working example of a feedback scheme for the alteration of the transmitted symbol constellation in real-time. The reader is introduced to incremental redundancy in both transmitted signal constellations and FEC for the wireless link, giving the example of GPRS-136 that has been defined for the IS-136 TDMA standard [BCE99].

Adaptive Modulation

Adaptive modulation is defined in Section 2.6.3, above. GPRS-136 uses adaptive modulation and incremental redundancy to achieve higher throughput than traditional deployments wireless systems. In this technology, adaptive modulation is provided via a switching between constellation sizes (referred to as M later in this thesis) of 4, 8, and 16 as a function of the channel quality between the base station and any Mobile Terminal (MT) in given cell. The different number of constellation points means that different amounts of information can be transmitted in any instant – in this case 2, 3, or 4 bits per symbol depending on the size of the constellation. The M = 4 level has been standardised as quaternary PSK (DQPSK) and M = 8 is coherent 8-PSK. The M = 16 level is yet to be standardised.

[BCE99] and [NBK00] describe the mechanics of transmission. As the different modulation types allow the transmission of 2, 3 and 4 bits per symbol, then that number of radio link protocol data or parity blocks are mapped to the TDMA slot. A data segment header is added for each slot, together with a parity/control sub-block pointer to indicate the composition of the slot in terms of the number and position of data or parity blocks. A CRC is added and the header is encoded using an $R = ½$ convolutional code.

The choice of modulation is made according to a periodic Channel Quality Feedback signal (CQF) provided by the receiver. This indicates the maximum constellation size allowable under the prevailing channel conditions. Importantly, blocks transmitted using a particular modulation scheme can be retransmitted in the case of error even where the modulation has changed as the burst headers are capable of providing sufficient information to the receiver for the recognition of differently packaged transmissions.

The transmitter uses the CQF together with knowledge of the offered load to adapt the chosen modulation scheme [BEN99].

Adaptive FEC

In GPRS, FEC is also incremental, allowing a close match between the channel SNR and interference. Each GPRS-136 radio link protocol data block is coded into D data sub-blocks and D parity sub-blocks. A $R = ½$ convolutional code is used, then the output of one gene rator is mapped into D sub-blocks termed data blocks, and the output of the other generator is mapped similarly into another D sub-blocks termed parity sub-blocks. Transmission of the data sub-blocks results in FEC free transmission, whereas including parity sub-blocks achieves a code rate of

$$R = \frac{D}{(D+1)}, \quad \ldots \quad , \frac{1}{2}. \qquad (2.)$$

Information relating to earlier sub-blocks must be used with information received later for the use of incremental redundancy, so the ability to carry out this form of channel correction will depend on the memory available at the receiver, as it must store information corresponding to information that has not yet been successfully decoded. The receiver must decode the header of each of the received blocks to determine how to soft combine the received block with previously stored information in order to decode the data segment successfully.

2.6.4.3 No Incremental Redundancy – GPRS

By contrast, GPRS uses adaptive coding with Gaussian Minimum Shift Keying (GMSK) modulation and the scheme has no incremental redundancy. This allows the header and the data to be encoded together. Different block sizes are defined, each at a different code rate, but the RLC must be held at the same code rate that is used for the initial transmission of the block if ARQ is required. This is significant as it implies that ARQ recovery may fail when the channel deteriorates. Including measurement reports in the ARQ status feedback messages provides feedback.

2.6.4.4 Incremental redundancy – EGPRS

Another defined technology, EGPRS, uses nine modulation and coding schemes which are defined with four using GMSK and five using 8-PSK. Retransmissions are allowed in any defined format from the same family of MCS as the original transmission, that is, using the same modulation scheme.

EGPRS allows a transition between link adaptation, which operates differently to, but in the same broad fashion as GPRS-136, and incremental redundancy. The transitions depend only on the amount of available receiver memory.

2.6.4.5 Rate of adaptation

The speed of adaptation required in order to maximise efficiency in any telecommunications technology depends on the nature of the channel. Currently, proposals for link adaptation are best described as slow rate adaptation, as the data rate is adapted to slower variations in fading and distance loss over the cell coverage area [NBK00]. Faster rates are not practicable because:

- The feedback interval is constrained to be of the order of tens to hundreds of milliseconds. The use of fast feedback bits for closed loop rate control even in Code Division Multiple Access (CDMA) systems (where there is a fast closed loop power control feedback) has not yet been specified; and
- Rate adaptation structures are currently being defined only at the physical layer frame or slot level. There are no structures to permit modulation or coding variation on a symbol by symbol basis within a slot or frame transmission.

2.6.4.6 Summary of Current and Proposed Rate Adaptation Structures

Table 2.1 summarises the preceeding sections with a concise summary of rate adaptation structures that are close to finalisation. Of particular interest are the limited peak data rates and the modes of CQF.

System or standard	Method of rate adaptation	Method of indicating format	Channel quality feedback	Peak data rate
CDMA IS-95 Rev B	*M* supplemental code channels each at 8 or 14kb/s	Supplemental channel assignment Message (SCAM)	Supplemental channel request message, pilot strength measurement	64kb/s
cdma 2000	Variable rate supplemental code channel – variable spreading and coding	SCAM, also considering blind rate detection	Supplemental channel request message, pilot strength measurement msg, power control bits	614.4 kb/s per Walsh code, obtains 2048 kb/s
UMTS Wideband CDMA (WCDMA)	Variable rate traffic channel – variable spreading and coding	Transport format combination indicator (TFCI) identifies format of each frame	Measurement report: (1) Pilot strengths; (2) SINR; (3) BER, BLER	2048 kb/s (6 Walsh codes)
GPRS	Time slot aggregation, adaptive coding	Separately coded field, “stealing bits”	Measurement report in ARQ status message: (1) Signal and interference; (2) BER; (3) Signal variance	160 kb/s
TDMA 136+ (GPRS – 136)	Time slot aggregation, adaptive modulation and incremental redundancy	Separately coded data field type (DFT)	Channel quality feedback: (1) In uplink ARQ status message; (2) In downlink packet channel feedback	44.4 kb/s
EGPRS	Time slot aggregation, adaptive coding, adaptive modulation, incremental redundancy	Separately coded field, “stealing bits”	Measurement reports in ARQ status message: (1) Signal and interference; (2) Bit error rate; (3) Fading rate	473.6 kb/s

Table 2.1. Summary of rate adaptation for packet data services in second and third generation cellular standards [NBK00].

2.6.4.7 Overall Effect

[Eri99] studies the overall efficiency of the EGPRS Channel Coding and shows that the combination of the adaptive FEC and adaptive modulation is capable of providing a smooth degradation of system capacity with increasing noise, ensuring that enhanced efficiency is obtained at most SNR levels. That study is completed over a TU3 channel with and without frequency hopping.

Chapter 3 further evaluates the interaction between different layers in achieving that effect.

2.7 Medium Access Control Architecture and Multi-access Techniques

An efficient MAC protocol is required to share available bandwidth among the multiple users. It must schedule CBR and VBR Virtual Channels (VCs) as well as ABR traffic classes while maintaining high utilisation of the radio link. [BG92] describes packet radio networks in which not all nodes can hear the transmissions of all other nodes. In this section, however, we are interested in developing principles for a MAC protocol in which stations are positioned in cells and hence must communicate within a set area with a central base station. Such a MAC protocol simply involves the interaction of a single controller with a changing set of mobile terminals as they enter, leave, and transmit in the cell.

Such a set of design parameters has much in common with current mobile telephone configurations, and with certain wired MAC protocols. Examples of the wired cousins of the MAC described here include Fair Centralised Priority Reservation (F-CPR) and MAC Level Access Protocol (MLAP).

The following sub-chapter categorises approaches to communications resource sharing, complementing this in following sections with details and critique of a number of specific examples. We then propose a MAC structure for a wireless access network MAC that we will use as a basis for analysis in later chapters.

2.7.1 Sharing Communications Resources

There are two diametrically opposite multi-access paradigms used for sharing a communication resource among many competing users [Pro95]:

- The *reservation based* multi-access approach which is similar to circuit switching; and
- The random multi-access approach, which is similar to the philosophy of packet switching.

Reservation based switching allows a single user to take a portion of the communication resource for themselves, and usually leads to a subdivision of the resource at the physical layer. These divisions often take the form of frequency-separated channels, or different time slots within a framing structure. Examples are Frequency Division Multiplexing (FDM), Time Division Multiplexing (TDM) and Wavelength Division Multiplexing (WDM). This method is able to provide absolute guarantees of bandwidth tending, however, it tends to be inefficient as such schemes are unable to capture the benefit of multiplexing gain. Further, the usability of these schemes depends on the physical layer characteristics of the medium in question.

On the other hand, random access networks rely on an unsegmented, shared physical transmission medium and the application of some added access control mechanisms (usually at a higher layer) to ensure transmission. Two subcategories of random access are:

- Algorithms that avoid collisions via polling or demand based scheduling; and
- Algorithms that handle collisions using a collision resolution process. Each of these Contention Resolution Algorithms (CRAs) set out rules that MTs must follow after a collision is detected on the randomly assigned link to ensure that MT will eventually achieve error free access to the link for transmission. Chapter 5 defines a number of CRAs that will be considered in the MDR-TDMA frame proposed in Chapter 6.

An example of the first form of random access network is Demand Assigned Multiple Access (DAMA) or the more simple polling solutions, an example of the latter is the *p* persistence CRA.

CRAs and their performance are very important for random access networks and hybrid networks that segment frames into fixed and random access portions. The CRA acts as a means for co-ordinating transmissions from the mobile stations to the base station so that collisions are minimised and throughput is maximised across the link. Each set of rules for transmission in the random access environment has a unique maximum throughput and each responds differently to changing traffic characteristics. One of the major contributions of this thesis is the study of CRAs under disaster conditions.

2.7.2 Fixed Assignment

Techniques such as Frequency Division Multiple Access (FDMA) and TDMA have been considered inappropriate for integrated wireless networks because they cannot achieve efficient radio spectrum utilisation in a bursty environment [KM97]. The dominant services in the broadband environment are Variable Bit Rate (VBR) (bursty) services.

On the other hand, fixed assignment protocols perform very well with Constant Bit Rate (CBR) connections in terms of both service quality and channel efficiency, however performance decreases dramatically when the requirement is to connect many infrequent users with variable bit rate connections. This effect is triggered as the protocols must provision their (unchangeable) connection size to meet the bandwidth required for the declared peak so as to satisfy QoS requirements. Hence a large portion of bandwidth is wasted during periods that the user is idle.

CDMA is a special case in this category, having both fixed and random assignment capabilities. Advantages include: close to zero channel delay; bandwidth efficiency; and excellent statistical multiplexing. Advances using CDMA have been limited by transmission rates, power control problems and base station complexity [Abr94], [MJL93], [Wil93]. These challenges have focussed near term attention on other forms of multiple access schemes, however, note the further comments on CDMA in Section 2.7.4 below.

2.7.3 Random Assignment

Many random assignment schemes exhibit poor throughput and collapse at peak times, their large access delay as a result of the contention resolution process making them inappropriate for deployment in a future wireless broadband network.

Although carrier sense multiple access with collision detection (CSMA/CD) does provide wireline networks with relatively good performance characteristics, difficulty is encountered in the wireless environment due to the difficulty in sensing remote carriers in the presence of a closer transmission. The signal from the local transmitter overloads the receiver disabling the ability to sense remote transmissions [KM97].

2.7.4 Fixed and Random Assignment (CDMA)

Spread spectrum has in the past not been considered a viable method of providing higher data rates and packet multiple access for the following reasons [KN99]:

- Spreading limits the permissible data rates in the limited wireless spectrum;
- Signal acquisition for packet access using spread spectrum requires large overhead delay; and
- Dynamic slotted time-division access can be more efficient compared with CDMA.

As Kumar and Nanda point out, the challenge is to design an access control mechanism that allows one or more users to get high data-rate bursts without impacting the QoS to the voice users. In [KN99] they design and analyse a scheme that has the following features:

- Acquisition delays and overhead are minimised through the use of a low overhead acquisition and tracking code channel even when idle. This has a double use as a dedicated signalling channel for burst control and signalling.
- A single user is able to access the entire bandwidth by limiting spreading. Data rates and QoS are controlled by variable spreading gain and coding with the dynamic assignment of additional code channels for higher rate transmissions.

There are many advantages to this approach if remaining difficulties can be overcome. One specific advantage, relevant to this section on MAC design, is that the efficient use of spectrum can be managed via a single parameter: aggregated interference over time. For the case of dynamic slotted time division access, however, resource management over the dimensions of time, frequency, and reuse must be implemented.

Indeed there is a growing understanding that spread spectrum signalling does not have to lead to spectral inefficiency in multi-user wireless systems. This increasing consensus is illustrated by the choice of Wideband Code Division Multiple Access (WCDMA) for some third generation cellular standards. Further, recent results referred to in [Ver00] indicate that the spectral efficiency of CDMA can be multiplied by four by increasing receiver complexity and increasing the number of users per chip from ¼ (typical of current systems) to 2. In the same article, Verdú points out that there is a current trend away from a philosophy of uncoordinated transmissions. Rather, recent systems contemplate packet scheduling based on individual demand and delay as well as the instantaneous fade levels experienced by each user.

2.7.5 Demand Assignment

Most broadband wireless MAC proposals have been based on a dynamic TDMA/Time Division Duplex (TDD) protocol with a centralised control based radio access protocol for the wireless link [XNY95], [NBJ97], [Ray97]. A Supervisory MAC (S-MAC) is used to schedule traffic dynamically, and a Core MAC (C-MAC) is used to multiplex and de-multiplex arriving and departing traffic.

Users are required to provide information about their future needs for bandwidth, which is communicated from each MT to the S-MAC via random assignment using a CRA (see Section 2.7.3). Bandwidth should be assigned to any mobile terminal on demand. This form of demand assignment avoids the drawbacks of fixed assignment as bandwidth not assigned during the idle period of a VBR source can be granted to an Available Bit Rate (ABR) connection. It also avoids waste associated with contending data packets on the wireless link as the central controller, based on demand requests from mobile nodes, assigns data slots.

Demand assigned MAC proposals are stable under a wider range of traffic loads and can guarantee a predictable QoS. The cost of these systems is increased complexity associated with the implementation of an efficient request reservation mechanism.

A number of specific demand assigned MAC proposals are now considered focussing on single channel (access point based) networks.

2.7.5.1 Demand Assignment with Distributed Control

Demand Distributed Multiple Access (D^2MA)

The D^2MA protocol calls for the reservation and scheduling functions to be performed by the mobile stations independently with only passive operations remaining the responsibility of the base station. These tasks include detecting collisions between reservation requests emitted by the mobile stations, and broadcasting the results on the down link [LMM98].

D^2MA implements a virtual ATM multiplexer based on a Frist In First Out (FIFO) distributed queue. The virtual multiplexer is composed of two subsystems:

- A booking subsystem utilised by MTs to require the uplink capacity needed for the emission of the cells; and
- A distributed queuing subsystem regulating the transmissions of the cells over the uplink.

When a MT has a cell for transmission, the D^2MA protocol causes a reservation attempt to be submitted to the booking subsystem. A reservation attempt is submitted by randomly setting a single reservation bit out of the available b bits to 1. The choice of any particular reservation bit is completely random and the reservation attempts are anonymous (meaning that the identity of the MT is not revealed to the base station). They are handled using an ALOHA scheme.

Any reservation attempt is repeated until it is successfully accepted, and the cell is inserted in the transmission queue. When that occurs, the MT tracks the cell's progress through the queue, waiting for its turn to transmit without contention. If a collision occurs, the reservation attempt is aborted and must be submitted again. A MT discovers a collision by analysing the Request Acknowledge (R-ACK) field emitted by the Base Station (BS) on the downlink.

There are many more variations of this protocol described in the literature. These protocols are substantially similar to the original but have been further developed to emphasise efficiency, fairness, or lesser complexity.

2.7.5.2 Demand Assignment with Central Control

Packet Reservation Multiple Access (PRMA)

Rutgers WINLAB proposed PRMA for packet-oriented voice transmission over wireless channels [KM97], [GVG89]. The authors sought to improve bandwidth efficiency of fixed assignment TDMA via the use of a slotted ALOHA signalling channel. Although having the advantage of being a simple protocol for demand assignment, PRMA cannot be used for WATM networks due to extremely variable channel access delay (which does not suit the stringent requirements of WATM QoS).

There have been many proposals for enhancing the performance of PRMA which concentrated on improving channel efficiency more and providing some kind of fairness for data applications, neglecting the variable channel access delay problem [KM97]. These include:

- Aggressive PRMA (APRMA);
- PRMA - Data Steal over Voice (PRMA-DSV);
- Minipackets PRMA (MPRMA);
- Centralised – PRMA (C-PRMA); and
- PRMA / Dynamic Allocation (PRMA / DA).

Resource Auction Multiple Access (RAMA)

In this scheme, MTs send requests by transmitting an identification sequence on a transmission/acknowledgement basis. After the transmission, the BS acknowledges one of those signals and the remainder of the MTs drop out of the ‘auction’. This form of allocation has significant Packet Delay Variation (PDV) since new connection requests and idle connections compete against each other in the Remote to Base (R-B) contention auction. Further, the process itself adds significant overhead to the system [Ami93].

Distributed Queuing Request Update Multiple Access (DQ-RAMA)

DQ-RAMA configures the uplink and the downlink on a slot-by-slot basis. The uplink slot contains a single contention free Data Slot (DS) and several random access Contention Mini-Slots (CMSs). The downlink contains a single contention free DS and one or more contention free Acknowledgement (ACK) minislots together with a contention free permission mini-slot which allows a MT to transmit its data in the next uplink DS. When a terminal transmits its request for bandwidth in one of the CMSs, it listens to the ACK slots to determine whether its request has been heard by the S-MAC. Once it receives an ACK, the MT stops contending for access in a CMS and waits to be identified in the permission slot. It then uses the next DS.

New connection requests and idle VBR, ABR, and Unspecified Bit Rate (UBR) connections contend equally in the CMSs for reservation of a DS creating a channel access delay problem for already admitted idle VBR, ABR and UBR services. Further, there is the potential for a synchronisation mismatch between ACK receipt and permission grant for DS transmission. For the case of time sensitive traffic at peak periods this synchronisation problem can be significant.

Mobile Access Scheme Based on Contention and Reservation for ATM (MASCARA)

MASCARA is the MAC protocol for the Wireless ATM Network Demonstrator (WAND) [PPM97]. It is similar to the scheme described in Section 2.7.5.3 below. Bandwidth requests are sent via a contention period, or can be piggybacked to the DSs (see Section 6.5.8 below for further details).

MASCARA uses the Prioritised Regulated Allocation Delay-Oriented Scheduling scheme, which implements a leaky bucket approach after sorting connections based on their traffic class. Tokens are generated at a fixed rate equal to the mean cell rate and the size of the pool is equal to the burst size of the connection (as declared at the time of connection setup). Once the connections are sorted, traffic is granted access in priority order from CBR down to UBR connections in turn.

This algorithm has the attractive benefit of being work conserving – given that a connection has been able to signal its readiness to transmit to the S-MAC, the channel will never be idle when there are ATM cells requiring transmission [PPM97].

2.7.5.3 Multiservices Dynamic Reservation – TDMA (MDR-TDMA)

MDR-TDMA is another form of centrally controlled demand assigned TDMA system which consists of contention free data transmission periods for the up and down links, separated by a contention period in which mobile terminals may compete to reserve bandwidth.

Proposed Frame Structure

A sample frame structure is shown in Figure 7: downlink information including control information and ATM (or other format) cells are transmitted at the start of the TDMA frame. This section contains: the preamble (frame sync & training sequence); the frame header (position and size of subframes within the frame); and the wireless network control signals and data (base to remote data cells). The base station controls the uplink bandwidth (one or more slots) dynamically via the frame header.

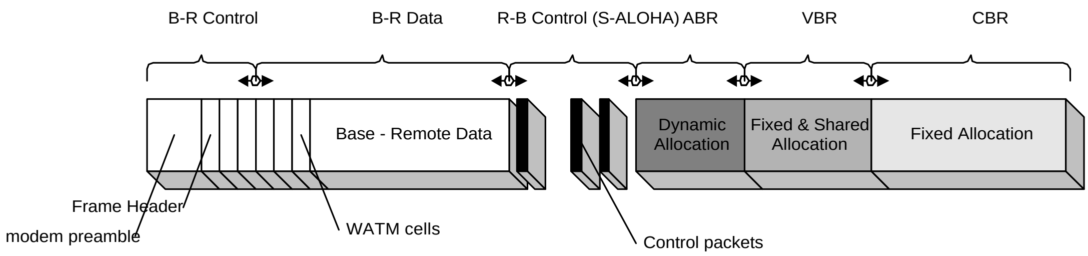

Figure 7. Proposed frame structure for a MDR-TDMA wireless MAC.

The uplink subframe consists of an uplink control region (R-B Control) for remote to base control information transmission (using a contention resolution algorithm), followed by allocated ABR, VBR and CBR data slots (scheduled by the S-MAC and hence contention free). The downlink contains ABR, VBR and CBR data slots in positions assigned by the S-MAC. A typical allocation of resources might be a total of 20% (12% R-B control, 8% B-R control) of the bandwidth allocated to the control regions, with 6% added overhead from frame headers and preamble, making the maximum available utilisation 74% [XNY95].

Dynamic scheduling and the preparation of the 'schedule table' which determines frame structure and content for each frame is completed by the S-MAC. Following the receipt of requests for bandwidth allocation from mobile terminals via R-B Control, and from the base station DLC for downlink transmission, the S-MAC schedules slots for both the uplink and the downlink for all remote stations in the cell. QoS requirements of each link must be taken into account in the allocation of data transmissions and Connection Admission Control (CAC) is a part of the base station S-MAC.

It is important that the S-MAC should implement max-min fairness [BG87] (see Section 2.4 above), as well as dynamic bandwidth control including 'graceful degradation' to cope with nodes entering and leaving the cell [Sch95]. An allied proposal is related to the exploitation of spare WATM radio bandwidth in low load periods using techniques such as "above-the-peak ABR". This new ATM Transfer Capability (ATC) allows transmission at rates higher than those specified in the traffic contract when there is spare WATM capacity. Together with the use of Transmission (Tx) and Reception (Rx) buffers subject to delay constraints, the ATC would help WATM to match the higher wired network data rates [LZG96].

The C-MAC is the interface between the DLC and the wireless physical layer, using the schedule table supplied by the S-MAC to multiplex/de-multiplex transmission and reception for all VCs. Its function at the base station and the terminal are identical. CBR circuits are assigned slots periodically according to their bit rate, whereas VBR connections are assigned slots via a Usage Parameter Control (UPC) based statistical multiplexing algorithm e.g. [BG87]. ABR connections are allowed demand based dynamic reservation of unused C/VBR slots in each frame on a burst-by-burst basis, possibly on a round robin system to stop one VC burst clogging all other connections.

Down- and Up-Link Considerations

Downlink transmissions are controlled by the S-MAC and can be operated with high efficiency (about 80% or greater) and low delay [KM97]. This portion of the link is not a critical driver of overall system efficiency.

By contrast, Figure 7 shows that the uplink is subdivided into the R-B Control period (contention based) and data transmission (contention-free) sub-periods. Assuming that the combined peak capacity of the mobile terminals within the cell controlled by the S-MAC is far greater than the net bandwidth available, then it is clear that the R-B Control section will be a key performance indicator for the system. Provisioning for this section of the frame will have a large impact on overall system delay characteristics.

Channel access delay is classified into connection and packet (burst) delay and is defined as the time required for the bandwidth request (uplink) and the data (uplink) packets to access the channel respectively [KM97]. The connection access delay is experienced in the R-B Control subchannel as well as the scheduling algorithm in the S-MAC and will have a significant impact on retransmission requests. Thus it is a critical contribution to the stability of the system and its performance at peak times. Packet access delay is also important. This statistic determines the size of queues at the mobile nodes and the centralised controller, as well as leading to PDV, which can be a serious problem for VBR applications. For this reason, any MAC algorithm must provide efficient algorithms to overcome these problems. Chapters 5 and 6 will investigate these issues in depth via R-B throughput and stability analysis under 'disaster' conditions.

The R-B Control Section

In all wireless systems, scheduling mechanisms for traffic reservation are seen as one of the most important areas to the proper functioning of the system. It is the primary role of the supervisory controller (S-MAC) operated at the base station or head end to receive reservations from the member stations in the shared area and allocate bandwidth on a share arrangement. Nodes contending for access to the controller must compete with each other to inform the Head end Controller (HC) of their need for a share of the bandwidth available to the population and distributed by the HC. Authors often refer to this contention period as containing CMSs and the remainder of a frame as containing DSs. While CMSs are subject to collision, the DSs are collision free, each station in the population informed of its transmission rights by the HC using a scheduling algorithm in the S-MAC. Hence it is critical to ensure the use of an acceptable performance CRA (see Section 2.7.3 above and Chapter 5 below) for the resolution of transmission requests in the Contention Resolution Interval (CRI). Indeed, one of the focuses of this thesis is on the contention resolution method of the MAC protocol, in particular, looking at Q-ary tree resolution methods (see Chapter 6 below).

2.8 Issues in Wireless DLC Design

Turning from MAC issues, the Data Link Layer provides for the reliable transfer of information, sending frames of data with the necessary synchronisation, error control, and flow control over the communications link [Sta94]. It may be divided into: the MAC for efficient sharing of the available bandwidth among multiple users along with QoS management (already considered); the DLC to overcome radio channel impairments; and Wireless Control for radio resource management [XNY95].

2.8.1 Efficiency and ARQ Block Size

Many wired packed networks are capable of very rapid switching largely because of the removal of error checking from the Link Layer on the assumption of a very high quality fibre link [CM97]. Wireless transmission is far from perfect and has greater error probability across the connection. This section outlines the NEC proposal for a WATM DLC, and shows the importance of error recovery at the Link Layer.

In order to use higher-level protocols such as TCP effectively, it is important that the wireless links appear to have characteristics similar to a fibre network to the higher layers to avoid the ‘slow start’ and other performance problems (see Section 4.4.8 for a fuller explanation). This requires a DLC capable of limiting effectively the flow of errors into the ATM network [CM97]. A further consideration is that efficiency has been shown to be a function of the size of an ARQ block, suggesting efficiency savings via the implementation of ARQ systems below the network and application levels. These are able to operate with smaller block size than the TCP retransmission system, thus increasing efficiency.

This issue with the use of TCP/IP over wireless links is a problem relating to TCP/IP's use of a large block Go-Back-*N* ARQ to test whether data have been correctly received at the destination node. TCP uses one of three retransmission strategies: first only, batch, or individual for error correction [Sta94]. Authors identify that the window protocol mechanisms built into TCP are designed to avoid network congestion and do not take account of characteristics of the wireless environment (which was not a consideration in the design of TCP). Hence, cell loss on any link of the network – which TCP assumes is wired thus having a low and stable BER – must be as the result of network congestion rather than an errored packet. In the situation of a cell loss in the network, the action of TCP is to reduce throughput in order to alleviate congestion including reducing window size – the 'slow start' phenomenon [BST95].

The concern of commentators is that this mechanism, designed to alleviate congestion, is of concern in the 'lossy' wireless environment, resulting in needlessly compromised utilisation through an entrenched incorrect assumption that the network was wired. An example is [BST95] where the concern is that 'slow start' may be wasteful in the case of Go-Back-*N* retransmission. Some authors have advocated change to TCP/IP to allow it to distinguish between congestion and wireless loss [BSA95], however recent research suggests that TCP, although acting for the wrong reasons may bring about a favourable network response in times of interference *contra* [XNY95], [Sch95].

Chapters 3 and 4 evaluate the trade-off between the use of FEC and ARQ over wireless links and establishes a bound for the use of FEC in wireless networks.

2.8.2 The NEC Proposal

The DLC outlined in this section provides reduced error propagation to the higher layer protocols and ensures sequential cell delivery to the higher layers with a guaranteed fixed total cell transmission delay used to recover transmission errors. Further, this DLC implements the interface between the MAC and higher layer protocols needed to support a demand driven medium-access strategy for multiservice wireless access [XNY95]. The result is a more efficient and flexible global structure (Figure 8).

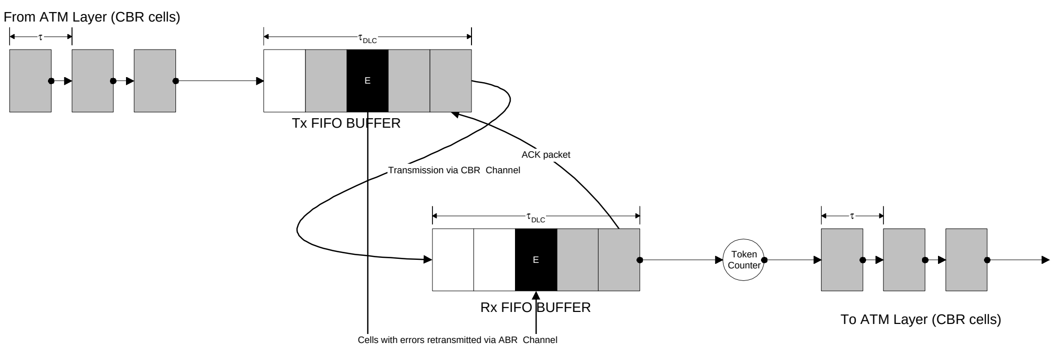

Figure 8. Block diagram of the DLC ARQ system (see Section 4.4.8 for further analysis).

Implementation of the DLC depends on the class of traffic being served, since QoS for WATM is a function of delay as well as CLR and blocking. ABR VCs wait without time limit to recover errors, whereas CBR and VBR traffic impose delay restrictions that are fixed, depending only on the size of the FIFO cell queue used by the Link Layer ARQ and the cell timer values.

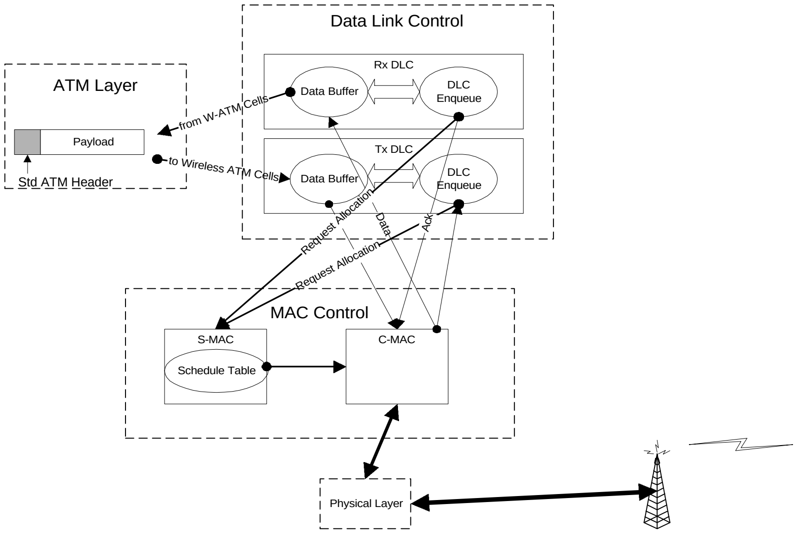

Figure 9. Architecture of the DLC and MAC protocol layers.

Figure 9 shows that the DLC (remote/base) has two components in the NEC design - Tx and Rx mode. These retain information for the Selective Repeat ARQ mechanism. Tx determines the cells to be transmitted/retransmitted (DLC enqueue) and deletes those for which timeout (this is the fixed DLC delay) has occurred, or for which an ACK has been received from the Rx buffer. The latter cells are passed to the ATM layer, and recovery of the former (if any) is left to higher layers. Cells not ACKed are retransmitted at the first subsequent opportunity by the C-MAC using ABR bandwidth that must compete for scheduling at the S-MAC with other ABR connections.

Figure 10 shows the format for a WATM data cell and ACK packet format.

A conflict exists, nevertheless, between the use of the ABR channel for retransmission, and the fact that lesser error correction in the DLC layer leads to lower data rates. In fact, utilisation of ABR channels for retransmission leads to large error rates for congested channels, since there may be very little spare bandwidth after allocation of VBR/CBR channels and retransmissions must compete with ABR traffic. In this scenario, errored cells are more likely to be timed out of the Tx buffer than retransmitted, resulting in lower supportable data throughput rates due to less efficient error correction at the TCP/IP level at a time when demand for data bandwidth is highest. This generates poorer service and breach of the QoS agreement in real-time applications such as voice, or in the worse case, congestion collapse.

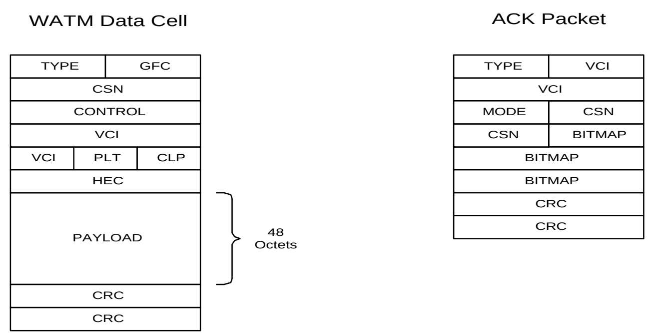

Figure 10. WATM data cell and ACK packet format.

One solution is to use the MAC to prioritise retransmitted packets over other ABR traffic [XNY95] so as to protect CBR and VBR VCs at the cost of ABR transmission. A more elegant solution is the allocation of a portion of bandwidth as a reserved retransmission VC, so as to guarantee QoS for non-blocked calls in a congested network. Both solutions require the consideration of quality performance tradeoffs and the addition of connection complexity to the system. Chapter 4 considers these issues in more detail.

2.9 Some Gaps in the Existing Literature

- There is extensive and ongoing work in the optimisation of specific OSI layers, however lesser attention has been paid to overall interactions between layers in optimising performance, with the exception of the well-known TCP ‘slow start’ problem.
- There is no consistent understanding of the optimal combinations of FEC, ARQ, adaptive modulation and other performance parameters for wireless multi-access networks under realistic traffic flows.
- No framework exists for the benchmarking of ‘disaster performance’ of access networks.
- Many authors take unrealistic traffic patterns as standard input in describing the performance of networks. Such an approach does not give ‘survivability’ statistics for the networks, nor does it give realistic indications of real performance.
- A complete design procedure for wireless multi-access networks has not been presented
- Accurate representations of alteration of system performance with input traffic change have not been presented for multi-access wireless networks.

3. INTER-LAYER INTERACTION

3.1 Introduction

This chapter optimises a comprehensive set of parameters aiming to minimise the bandwidth required for wireless transmission on a single link. It also provides important guidelines for wireless link dimensioning. A comprehensive set of coding, modulation and teletraffic issues are investigated to provide dimensioning guidelines for the use of FEC in wireless access networks, including:

- The effects of retransmissions of erroneous packets using ARQ protocols;
- Modulation efficiency;
- FEC redundancies; and
- The effect of traffic burstiness on loss and delay.

Simulations are used to derive Bit Error Rate values for a range of modulation efficiency and FEC Code Rate parameters and the parameters listed above are optimised to obtain maximal efficiency.

To evaluate the inaccuracy introduced by the simplifying assumption of the independence between the error process and the queuing process (see Section 3.2 below), a retransmission model is developed. This model is an enhancement of other reported retransmission models, as HLM and FEC are considered internally in its analysis.

Comprehensive results are obtained firstly for BERs over a wide range of SNR values on a given link by simulation. Simulations are then used to obtain efficiency results for a range of differing conditions and network set-ups which rely on not only a number of different coding schemes (such as Convolutional Coding, Bose-Chaudhuri-Hocquenghem (BCH) Coding, and Turbo Coding [BGT93] [HHR97]), but also use a range of different modulation gains and modulation scheme types. Note that some analytical results are available for BER in [Pro89] for different modulation levels when FEC is not used.

BER statistics generated by the simulation and a model (introduced in the next section) are then used to optimise both the code rate and HML level for the system under consideration so that efficiency (dimensioning) ratios for a wide range of situations and parameter values optimising FEC code rates and HML levels can be produced. Note that HML optimisation without considering retransmission has been performed in [Eri97].

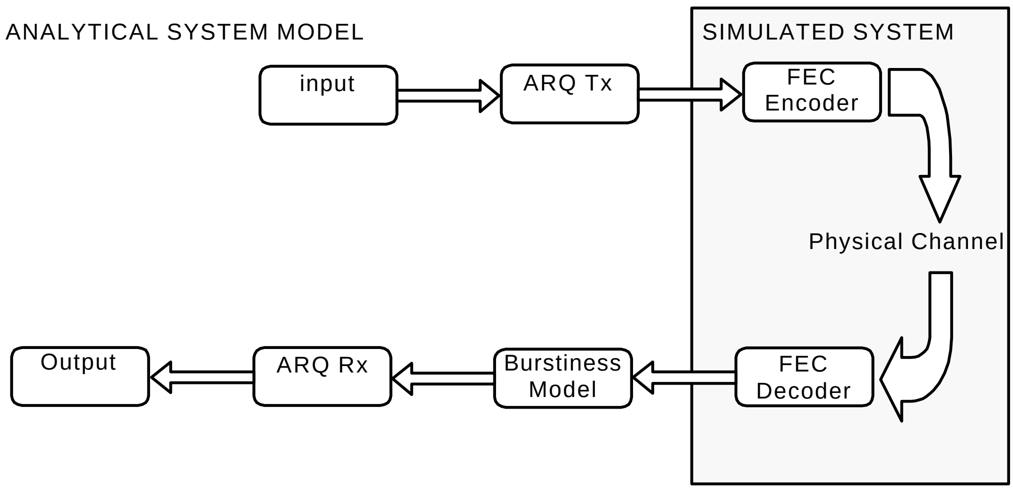

Figure 11. Overview of the model.

In using these observations to determine optimal dimensioning and transmission parameters it is important to consider:

- The mean rate of accepted traffic and its level of burstiness;

- The Modulation Gain;
- The size of the retransmitted block of data (called in this paper *block size*), i.e. ARQ block size (whether it be frame, packet or a single ATM cell) and hence the capacity allocated for retransmissions of erroneous blocks of data. It is important to clarify here that this chapter can be applied to WATM whereby the ARQ block size can be a single ATM cell within the data Link Layer ARQ retransmission as in [Ray97], otherwise the scope of the section is general to any wireless transmission;
- The capacity required by the coding scheme; and
- The capacity required for guaranteeing tolerable delay to avoid time-outs and congestion collapse.

The chapter is organised as follows: in the remainder of Section 3.2, the model and the methodology used for efficiency evaluation and optimisation is described. In Section 3.3, there is a general discussion of when FEC should be used, and a condition under which ARQ is sufficient and the use of FEC is not efficient is provided. In Section 3.4, the different modulation schemes considered in this chapter as well as the simulation process are descibed, and in Sections 3.5 and 3.6 numerical results for uncorrelated and bursty errors respectively are provided, allowing comparison between efficiency results for the various modulation schemes considered.

The results are used to motivate adoption of adaptive Modulation Gain and data block size in realistic mobile environments with continuously and rapidly changing SNR and traffic conditions.

3.2 The Models

This section commences with a reminder of the definitions of adaptive modulation developed in Section 2.6.3. The motivation for an adaptive modulation scheme is to adjust the method of transmission to ensure that maximum efficiency is achieved over a link at all times. In general, a modulator accepts a set of binary digits at one time, converting them to a single symbol for transmission over the channel.

This provides the Modulation Gain – a term denoted *G*, and used to represent the increase in capacity in the units (bits/sec)/Hz as a result of having altered the size of the symbol constellation. For the case of the transmission of multiphase signals including Phase Shift Keying (PSK), channel bandwidth is simply the bandwidth of the equivalent lowpass signal pulse for PSK. It is assumed that bandwidth is approximately equal to the reciprocal of the duration of the pulse (for the case of a single sideband). Hence

$$G = \log_2 M \qquad (3.)$$

That is, *G* which is data rate/bandwidth required from Eq.(3) is defined similarly to spectral efficiency, where *M* is the number of symbols in the modulation constellation.

3.2.1 Overview of the Retransmission Model

The retransmission model is now described. In [ZHG98] a model assuming the independence of Queuing Based Efficiency (QBE) from Error Based Efficiency (EBE) was set out. That paper explores the concept of QBE in depth. The analytical model presented in this section expands the concept of EBE, keeping the base idea of the analytical division between QBE and EBE.

3.2.1.1 QBE

The QBE is the ratio between the utilised capacity (including user data and overhead) required to provide a loss bounded data rate, and the overall channel capacity, related to the additional capacity required to handle burstiness in the traffic.

In Figure 12 and Figure 13, QBE values for different buffer sizes are presented for data and VBR video traffic respectively. In these figures, the QBE (= utilisation, or the proportion of time the server was busy) is calculated, using a single server queue model fed by the real trace and served by the minimal service requirement such that the QoS (loss probability of $1/10000$) is met.

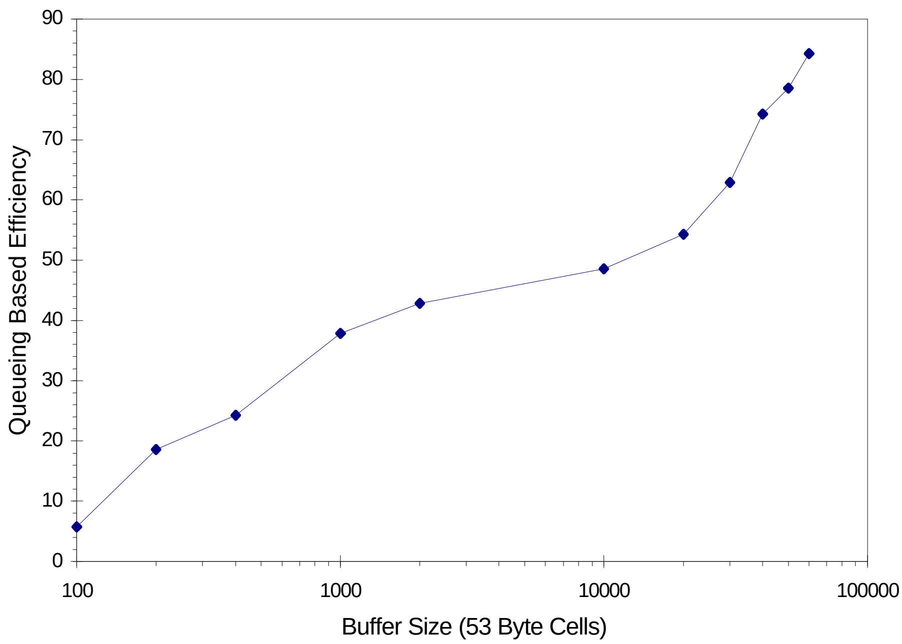

Figure 12. Link Utilisation vs. Buffer Size for measured Ethernet traffic with loss probability of 1/10000.

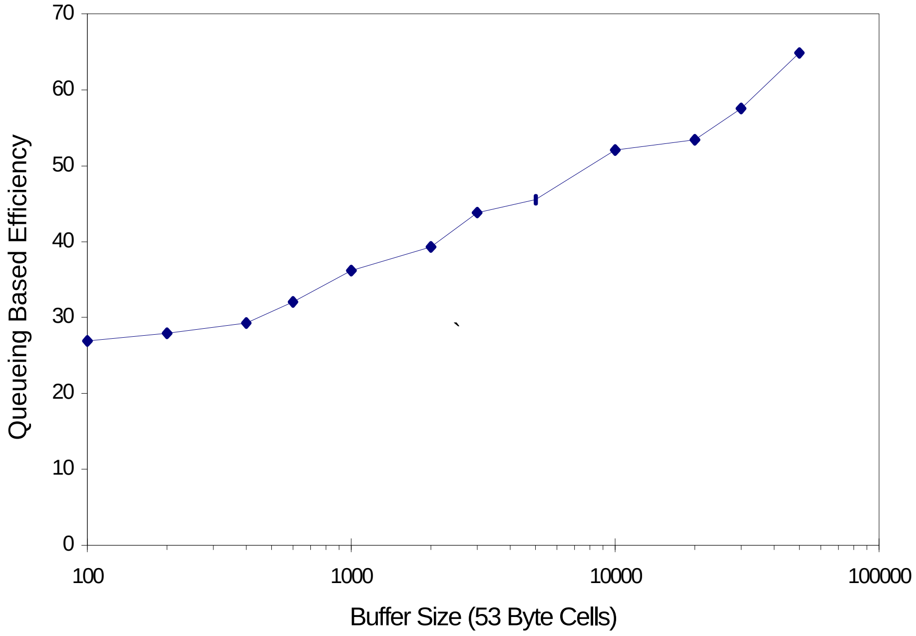

Figure 13. Link Utilisation vs Buffer Size for VBR video traffic (MPEG) with loss probability of 1/10000.

Denote the QBE using *r**. For ARQ (data) traffic (see Figure 12) a buffer of 1000 cells is a reasonable trade off between utilisation and delay giving *r** = 37.85%. For video (in Figure 13) a stricter delay requirement of a 100 cell buffer must be imposed, yielding *r** = 27.5%.

3.2.1.2 EBE

The other element of overall efficiency is the EBE. Because it takes the Modulation Gain (*G*) into account, this measure of efficiency is especially significant if adaptive modulation is used. The EBE is the ratio between the mean raw data rate and the raw data rate plus all overheads associated with error corrections and retransmission of erroneous blocks of data, namely, FEC and ARQ. It is adjusted for the Modulation Gain. Accordingly the EBE is a product of three factors. These are:

- The FEC code rate *R* defined by the ratio: *R* = (raw data) / (raw data + FEC redundancy);

- The ARQ efficiency denoted A and defined by the ratio: A = (raw data + FEC redundancy) / (raw data + FEC redundancy + error detection redundancy + additional capacity for ARQ retransmissions);
- The above defined Modulation Gain (G).

Overall Efficiency, or simply, Efficiency, which is the ratio between the mean raw bit rate generated by the customer and the required channel capacity, is calculated by multiplying the QBE by the EBE. Table 3.1 shows how we define throughput and required capacity for transparent and non-transparent transmission.

	Throughput	Required Capacity
ARQ (non-transparent)	Transmitted raw data since ARQ guarantees eventual correct delivery.	Raw data ARQ retransmission plus FEC overhead plus all other overheads at other levels including capacity adjustment for Modulation Gain (*G*).
Video (transparent)	Received error free raw data.[1]	Raw data plus FEC overhead plus all other overheads at other levels including capacity adjustment for Modulation Gain (*G*).

Table 3.1. Definition of throughput and required capacity for transparent and non-transparent transmission.

3.2.1.3 Optimal *EBE*

One of the main aims of this chapter is to derive the *Optimal* EBE denoted EBE^* and defined as the maximal value of EBE subject to meeting QoS requirements. In deriving EBE^*, the values of G (or

[1] Normally video throughput will also include erroneous bits, but we choose to exclude them to provide further insight into picture quality as a function of the SNR – see Figure 16.

alternatively M) and R are optimised for a given ARQ block size and a particular FEC scheme.

For the case of services requiring retransmission of erroneous messages by the higher layer protocols (Link Layer ARQ or end-to-end ARQ), retransmissions need to be considered when deriving the optimal value of G and R. Therefore additional capacity is required to allow for retransmissions of erroneous blocks of data (ATM cells, packets or TCP windows) and there is a clear trade-off between the level of G and the expected number of retransmissions. Given a set modulation constellations, it is shown that the required channel capacity for retransmissions is a function of the size of the ARQ block, FEC type and constraint length, as well as rate R chosen. If N is the ARQ block size in bits, then the probability that an ARQ block will be transmitted successfully is denoted by a and is given by:

$$a = \left(1 - e_{c,scheme}(G, R)\right)^{N}, \tag{4.}$$

where $e_{c,scheme}(G, R)$ is the BER of a connection with Modulation Gain G and code rate R for a given code and constraint length. An example of this notation is

$$e_{BCH(15,11),PSK}\left(2, \frac{11}{15}\right)$$

which would give the BER for a link which used BCH(15,11) coding and the modulation scheme was 4-PSK. Clearly, the result would also be dependent on the channel simulated.

Further, in Eq. (4) a limiting case is assumed via an assumption of the independence of errors at the output of the FEC decoder (Figure 11). This assumption will be further discussed in the next section.

Let W be the raw data, and b represent the ratio of the number of bits that would be transmitted in the absence of an ARQ Cyclic Redundancy Code (CRC) to the number of bits transmitted after the CRC is added. Then the total capacity required by ARQ retransmissions is:

$$\frac{W(1-a)+W(1-a)^2+W(1-a)^3+\ldots}{Rb} = \frac{W(1-a)}{Rba} \tag{5.}$$

so, assuming that Round Trip Time (RTT) is close to zero

$$A = \frac{\frac{W}{Rb}}{\frac{W}{Rb}+\frac{W(1-a)}{Rba}} = a \tag{6.}$$

Note that Eq. (6) follows from the assumption made that the delay due to retransmission is negligible. This is true in the case of wireless LANs and in cellular mobile network environments where the propagation delay is small when considered only over the final access link. However, it is not true in the case of satellite communications. Further, since the packed sizes used are relatively small (53 bytes) in simulations reported on in this chapter, it is questionable if RTT can be neglected for the case of a simple Stop-and-Go ARQ protocol. Rather, the approximation requires the use of a more sophisticated ARQ protocols such as Selective Repeat Protocol. This will keep capacity ideal on the packet level, but the complexity of the network is increased: it is either necessary to buffer packets, or reorder them.

Based on the above, the EBE of the system is given by:

$$EBE = GRa \quad . \tag{7.}$$

Recall that a is a function of G and R. The optimisation problem we consider here is:

$$EBE^{*} = \max_{G,R}[GRa] \; . \tag{8.}$$

To obtain optimal EBE, G and R must be optimised, with consideration given to the trade-off between the FEC redundancy and Modulation Gain versus the volume of retransmission of erroneous blocks of data. Since a is a function of G and R via the BER denoted $e_{c,scheme}(G,R)$ in Eq. (4), and since there is no analytical solution for the BER for a given set of G and R, we have to rely on simulations. We commence with the optimisation of Modulation Gain and then consider FEC for the optimised adaptive system.

3.3 Just ARQ or ARQ and FEC?

3.3.1 Independent Errors

In this subsection independent bit errors are assumed at the output of the FEC decoder, a scenario that, although unrealistic in the absence of an interleaver, allows the presentation of a limit for the use of FEC. Consider the case of not using FEC at all, and setting R to 1. In this case:

$$EBE = Ga \quad ,$$

where

$$a = \left(1 - e_{c,scheme}(G,1)\right)^N .$$

Then

$$\left(1 - e_{c,scheme}(G,1)\right)^N > R \ , \qquad (9.)$$

can be used as condition stating that FEC with rate R lower than $(1-e_{c,scheme}(G,1))^N$ should never be used. It is clear, however, that the inequality cannot provide a rule for when FEC rate R *should* be employed. This is because the effects of correlation between errors remaining after FEC may enhance the operation of the ARQ meaning that a higher FEC rate R than that indicated by the limit may be used.

3.3.2 Considering Channel Error Bursts

Now, if the assumption of independent errors at the FEC decoder output is relaxed, and data traffic requiring ARQ retransmissions is considered then, due to correlation between the error processes, the errors will concentrate in fewer ARQ blocks (where all other variables are constant) and the need for retransmission will be reduced. Hence the value of A as well as the EBE will increase. In this case, $A = a + c$, where c represents the improvement in the ARQ efficiency due to error correlation. c will vary with a large set of variables, most notably the error characterisation of the output of each different form of coding scheme.

Note also that in this case there is no need to use interleaving, so the delay is also improved. To conclude, under correlated errors, the EBE will be given by:

$$EBE = G\left[\left(1 - e_{c,scheme}(G,1)\right)^{N} + c\right] \quad . \qquad (10.)$$

Determining when code rate R should be used is a complex question that requires a clearer evaluation of the effect of burst errors on the model that is being developed. Burst errors appear in many media and the fading phenomenon in the wireless environment is a classic example. Hence the model is extended to capture the effect of error correlation.

Consider the effect of a correlation of errors appearing in the system before the operation of ARQ. The level of error and the strength of error correlation will depend on the type and rate of the code system used, as well as the nature of the wireless environment. The effect of the concentration of error bursts in single ARQ blocks on the efficiency of the system under study must be quantified, and it is important to determine their effect on the optimal balance of FEC code rate R and ARQ that should be used under realistic traffic conditions.

Since we wish to capture a wide range of burstiness characteristics, we will continue to obtain BER results for various coding and modulation gain schemes by simulation. We assume that the effect of burst errors in the channel can be captured through the use of a Gilbert loss model described in Figure 14, [Zor98] [ZR97].

The physical channel is allowed to be in one of two states: *good* or *bad*. A bit is lost if it is transmitted whilst the channel is in the *bad* state but a bit is correctly received if transmitted whilst the link is in the *good* state. u denotes the *Burstiness Factor* and it is used to model the correlation between channel errors in the model.

Between transmissions, the wireless link moves from the bad state to the good state with probability 1-*u* and from the good state to the bad state with probability 1-*q*. It remains in the good or bad state with probability *q* or *u* respectively. Transitions of the Markov chain happen at regular intervals and the time between two transitions is equal to the time to transmit one bit.

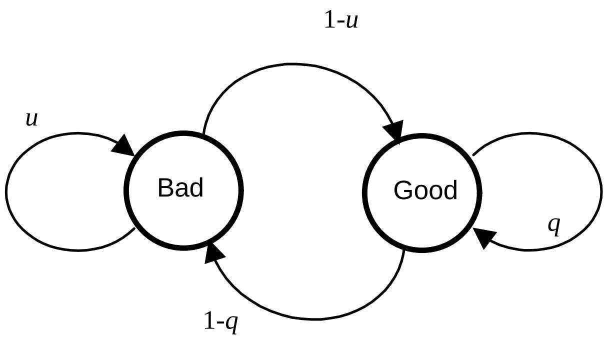

Figure 14. The Gilbert Loss Model.

Given that *u* is the probability of remaining in the bad state of the Gilbert Loss Model on any given transition, and *q* is the probability of remaining in the good state, then if S_B represents the mean time of sojourn in the bad state, and S_G represents the mean time in the good state,

$$S_B = \frac{1}{1-u} \tag{11.}$$

and

$$\frac{S_B}{S_G} = \frac{e_{c,scheme}(G,R)}{1-e_{c,scheme}(G,R)}$$

so that

$$S_G = \frac{1-e_{c,scheme}(G,R)}{e_{c,scheme}(G,R)(1-u)} \ , \tag{12.}$$

and

$$q = 1-\frac{e_{c,scheme}(G,R)(1-u)}{1-e_{c,scheme}(G,R)} \ .$$

Hence, given the BER for a given link (which is obtained by simulation) and an indication of the level of burstiness of the output after FEC (given by the Burstiness Factor u) it is possible to obtain the mean length of error bursts, and error free periods in the output traffic stream by using Eqs. (11) and (12).

The effect of the correlation of errors into a burst makes the use of ARQ over FEC more efficient compared to a situation of uncorrelated errors in a traffic stream. Let B_G be the event that an entire ARQ block of size N is without errors after FEC (using our previous notation, a = P(B_G)). Similarly, let B_B be the event that such an ARQ block is erroneous. Clearly, P(B_G)=1-P(B_B). Finally, let F_G and F_B be the event that the first bit in an N sized ARQ block is correct (good) or erroneous (bad) respectively.

Then, the probability that a block will not need to be retransmitted at the Link Layer is:

$$P(B_G) = P(B_G|_{F_G})P(F_G) + P(B_G|_{F_B})P(F_B) = a \ .$$

Since it is impossible to have a good block given the first bit is bad,

$$a = P\left(B_G\big|_{F_G}\right)\left(1 - e_{c,scheme}(G,R)\right).$$

The probability $P(B_G\big|_{F_G})$ is equal to the probability of having N-1 consecutive good bits, which is equal to $q^{N\text{-}1}$. Therefore,

$$a = q^{N-1}\left(1 - e_{c,scheme}(G,R)\right) .$$

Substituting for q, it is possible to express the probability that no retransmission will be required in terms of the BER and burstiness factor:

$$a = \left(1 - e_{c,scheme}(G,R)\right)\left(1 - \frac{e_{c,scheme}(G,R)(1-u)}{1 - e_{c,scheme}(G,R)}\right)^{N-1} \quad (13.)$$

Because Eq. (13) describes the efficiency of the ARQ system given its BER, it is used together with a simulation capable of describing the BER at the FEC decoder output for a range of values of G and R, and the selection of channels described in the next section.

In Section 3.5 we shall still assume independent errors in both the channel and after FEC, so that Eq. (10) provides a limit for R, being the best case for FEC. Use of FEC with code rate less than that limit, will lead to lower EBE than the transmission without FEC. In Section 3.6, we will investigate the effect of error correlation on our results in Section 3.5.

3.3.3 Considering user delay

3.3.3.1 ARQ

This subsection turns to a delay comparison between FEC and ARQ, commencing with the performance of ARQ alone (and assuming the

Gilbert Channel Model from the previous section). A pure ARQ system is considered which gives a conservative estimate of the performance achievable with retransmission techniques.

In this system, the information bit stream is segmented into blocks, and to each block a number of parity bits are appended such that b represents the ratio of the number of bits that would be transmitted in the absence of the parity bits to the number of bits transmitted after the parity bits are added. The parity bits are usually computed by using a CRC, and are used at the receiver for error detection. Typically b is very close to 1. The receiver computes the parity bits based on the bits it receives, and compares them against the received parities. If they match, the block is accepted as correct, otherwise retransmission is requested. In the latter case, the erroneous block is discarded.

To describe the ARQ retransmission scheme, analysis commences from the channel steady-state characteristics. A Markov chain captures the channel model,

$$\overline{X}(n) = (h(n), i(n)),$$

with state space

$$\{(h,l), 1 \le h \le 2, l \ge 0\}$$

where $h(n)$=1 and $h(n)$=2 respectively denote a bad and good channel state in slot n, and $l(n)$ is the number of packets in the queue at the beginning of slot n (at time nT, if T is the slot duration). If the arrival rate at the transmitter is l, then transition probabilities from the states are:

	(2, l)	**(1, l)**	**(2, l + 1)**	**(1, l + 1)**
(1, l)	$(1 - l)(1 - u)$	$(1 - l)u$	$l(1 - u)$	lu
(2, l + 1)	$(1 - l)q$	$(1 - l)(1 - q)$	lq	$l(1 - q)$

Table 3.2. Transition probabilities for the ARQ Model.

If

$$g = \left(\frac{l}{1-l}\right)\frac{(1-l)u + l(1-q)}{(1-l)(1-u)+lq} ,$$

and

$$p(1,0) = \frac{[(1-l)(1-u) - l(1-q)](1-q)}{(1-l)[(1-u)+(1-q)][(1-l)(1-u)+lq]} ,$$

are found from the normalization condition, then the steady-state distribution is:

$$\begin{aligned} p(1,l) &= g^{l} p(1,0) & l \geq 0 \\ p(2,l) &= \frac{l}{1-l} g^{l-1} p(1,0) & l \geq 1 \quad . \\ p(2,0) &= \frac{(1-l)(1-u)+lq}{(1-q)} \end{aligned} \qquad (14.)$$

If $f_{h,j}(m,n)$ is the probability that there are m successful slots in $\{0,1, \ldots ,n\text{-}1\}$, and that the channel state is j at time n, given that the channel state was h at time 0, then, the following recursion applies for n greater than or equal to 0:

$$f_{h,j}(m,n) = f_{h,1}(m,n-1)p_{1,j} + f_{h,2}(m-1,n) + d_{h,j}d(m)d(n) \ .$$

The probability that the packet is not transmitted within S slots, conditioned on the system being in state (h,l) in the slot of its arrival, is given by

$$P_S(h,l) = \sum_{m=0}^{l}\left(f_{h,1}(m,S) + f_{h,2}(m,S)\right) \quad ,$$

and the unconditional probability that a packet spends more than S slots in the queue is therefore

$$P_S = \sum_{l=0}^{\infty}\left(P_S(1,l)p(1,l) + P_S(2,l)p(2,l)\right) \ . \qquad (15.)$$

This result is used to place an important bound on the use of ARQ in a dimension other than efficiency. The bound on the system is a maximum probability that the system will take greater than S_{max} slots to deliver the ARQ blocks correctly. The value $1\text{-}P_{Smax}$ is the probability that a packet must be discarded despite correct reception at the receiver in order to satisfy the delay criterion of traffic QoS.

3.3.3.2 FEC

This subsection develops a delay analysis for FEC based solutions in the absence of ARQ, focussing on block codes, particularly BCH(N,K) codes. These codes are a generalization of Hamming codes and typically outperform all other block codes with the same N and K at moderate to high SNRs.

An incoming information stream is segmented into blocks of K bits, and N-K redundant bits are added to each block. The resulting K-bit blocks are transmitted on the channel, and the receiver makes a decision on the N information bits based on the N channel bits, independently for each block. The redundant bits are used to correct errors at the receiver and if more than t errors are present in a block, an erroneous decision occurs in the decoder.

Error correction codes perform better when errors are independent, while their performance is usually degraded in the presence of bursty errors. Interleaving is a technique to reduce bursty errors for error correction coding. The interleaver typically consists of a matrix with d rows and N columns. Code words are written as rows of the matrix, which are then read out by column to generate the bit sequence actually transmitted onto the channel. To make the interleaving work well, the interleaving depth d is properly chosen such that errors occur independently for each bit within a codeword (i.e., $(d\text{-}1)T_b \geq T_c$ where T_b and T_c respectively denote the bit time and the channel coherent time). Note however that the interleaving depth will be limited by the system delay constraint.

Turning to the relationship between the T_b and the total delay, D, assume that the output of the encoder starts filling the interleaving matrix at time 0. Then, the i-th bit of the encoded stream will be written in the matrix by time iT_b. Nothing will be transmitted until the matrix is full. At that point, the whole matrix is transmitted over the channel, so a delay of NdT_b is introduced in the transmitting end [Pro95].

At the input to the FEC decoder shown in Figure 11, the matrix is read out in the same order as it was written. This is significant as data bits can only be read out after the whole matrix has been received. For this reason, a second delay of NdT_b is introduced, this time at the receiver, bringing total delay to $D=2NdT_b$ as a result of interleaving.

If D_{max} represents the maximum allowable delay in the wireless access system remaining after delays due to decoding and all fixed physical delays other than interleaving are taken into account, then it is clear that, in order to satisfy QoS requirements, a maximum interleaving depth of

$$d = \frac{D_{\max}}{2NT_b} \qquad (16.)$$

may be used.

One advantage in the delay constrained system, is the constant nature of the FEC delay as against the dependence of ARQ delay on both the arrival rate and the channel quality.

3.4 The Simulation

Simulations used in this chapter were developed using the Matlab Communications Toolbox [The96]. The communications library form this software package was adapted to create a number of different wireless channels that were investigated. Both an Additive White Gaussian Noise Channel (AWGN) channel and a Rayleigh fading channel are simulated using the model just outlined that is able to capture the effects of changes in all of G, ARQ amount and block size, FEC, and burstiness of errors.

The channel reported on in Section 3.5 is an AWGN channel. This channel is used because of its non-bursty, memoryless nature. The ability of FEC to correct errors is reduced in channels that exhibit strong burst errors, however, ARQ becomes more effective, since many errors may be concentrated in a single retransmission. Hence by presenting a memoryless channel, it is possible to demonstrate the worst case for the use of ARQ. In Section 3.6 a Rayleigh Fading Channel is simulated and the burstiness analysis just presented is utilised. This allows the EBE results to vary with the mean length of any error burst.

Mechanics of the simulation are as follows: using MATLAB, coherent PSK signals are transmitted over the AGWN (or other) channel. We assume that Grey code is used in mapping bits to symbols so that two k-bit symbols corresponding to adjacent signal phases differ by only a single bit. This offers a significant benefit since the equivalent BER for the PSK signals is close to $1/k$ of the symbol error rate. Our choice of modulation scheme (coherent PSK) was selected for simplicity, however, the analysis presented is equally relevant for High Level Modulation including Enhanced Data rates for GSM Evolution (EDGE) [Eri97] which also uses PSK [BCE99]. A random bit stream is encoded by an FEC scheme before entering the channel and a FEC decoder is used after the channel to recover errors.

There are two choices at this stage of the simulation. It is possible to both obtain a BER for the system after the FEC decoder and then apply the burstiness analysis to study its effect on the system, or it is possible to simply apply the ARQ scheme to the raw bit stream as it appears. The latter is performed in Section 3.5 and the former is performed in Section 3.6.

Finally, as a part of the latter MATLAB simulation, ARQ is applied to bursty errors to validate the results of our analytical model. Errors are detected using the FEC (where included) or a CRC that varies according to the size of the ARQ frame under study and the simulation does not retransmit frames with undetectable errors. Real cases investigated by the author closely match the system when the burstiness analysis is applied later in this chapter.

A meaningful method of comparison of the various schemes investigated is based on the normalised data rate measured in bits per second per hertz of bandwidth versus the SNR per bit required to achieve a given error probability. The normalised capacity of the band limited AWGN channel due to Shannon [Pro95] is also shown in a number of this chapter's graphs. This represents the highest achievable bit rate-to-bandwidth ratio on the channel serving as an upper limit on the bandwidth efficiency of any type of modulation.

Generally the aim is to present a series of results that:

- Display the limit for the use of FEC in a realistic system, putting into practice the previous analysis; and
- Investigate the level to which burstiness is important in analysing the efficiency of our system, and in making choices as to the optimal *R*.

3.5 Results – Independent Errors

In this section numerical results displaying the strong connection between efficiency and Modulation Gain are presented. The imperative for the use of adaptive modulation in systems with highly variable noise levels is demonstrated. The section also uses these simulation results to illustrate the analytical limit for the use of FEC. Adaptive modulation can be implemented in real mobile networks using periodical reports from mobiles to base stations (as in GSM) however this chapter does not deal with practical implementation issues. Many of these have been considered in EDGE.

Figure 15 shows results of BER as a function of Signal-to-Noise Ratio (SNR) for a variety of constellation sizes in our PSK modulation scheme. The environment is an AWGN channel with no FEC applied.

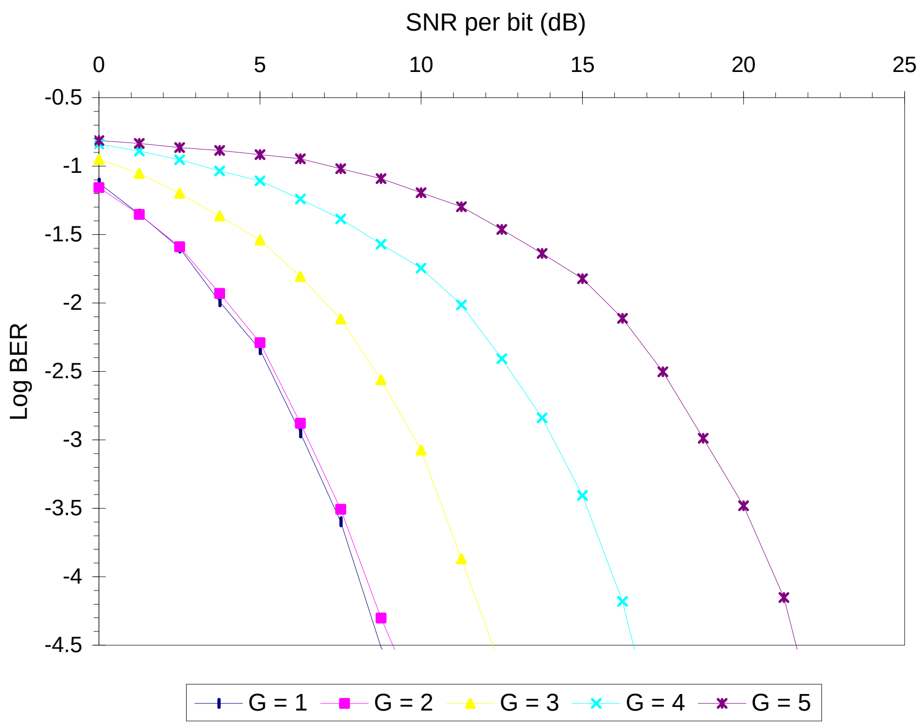

Figure 15. BER for PSK in an AGWN channel with no FEC.

3.5.1 Video with no FEC

Video data are an example case of a real-time service where there is no retransmission of erroneous messages. In this case, there is no waste of bandwidth due to ARQ, however, transmission at worse than set threshold loss rates cannot be tolerated.

Figure 16 can be used to obtain the efficiency of good data transmission for varying *G* as *EBE* has been defined in this case to be merely equal to the received error free data divided by transmitted data. This is just the data rate multiplied by the probability of correct transmission of any bit.

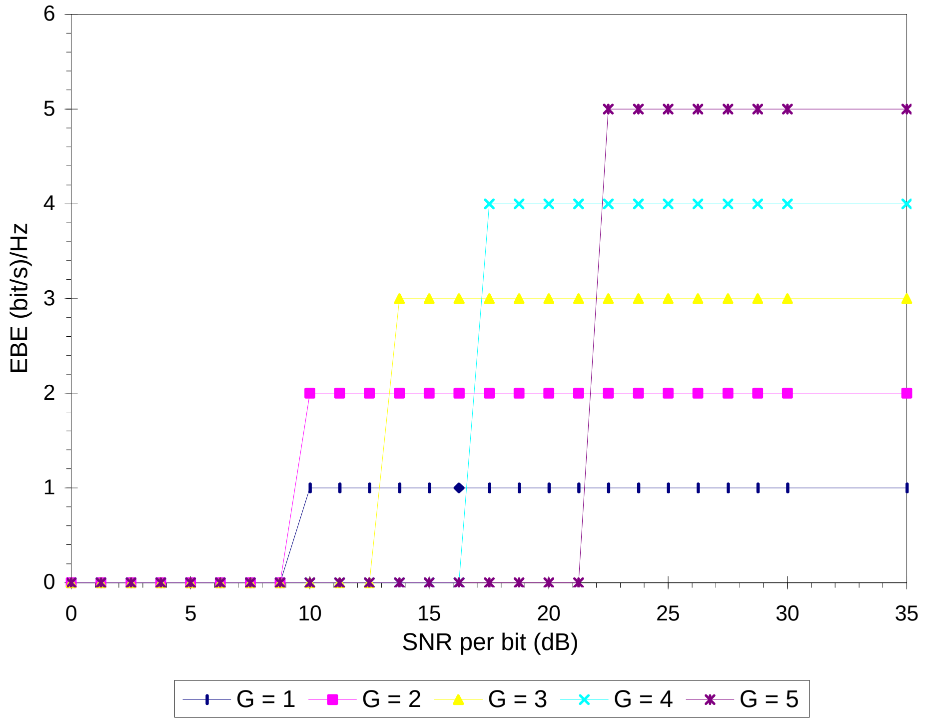

Figure 16. Comparison amongst modulation gains of efficiency of Video transmission (no ARQ and no FEC) as a function of SNR conforming to a QoS requirement over an AGWN channel.

Since large error rates can cause visible errors or even frame loss in video streams, threshold acceptable loss rates are set in real systems, usually with a BER of $10^{-4} – 10^{-6}$.

In such a situation, it is necessary to use the largest Modulation Gain such that $e_{c,scheme}(G,1)$ is within specified QoS requirements. Error greater than the threshold at which the QoS contract will be violated cannot be tolerated so, in Figure 16, the link utilisation is set as zero for each given G, wherever the error rate is higher than that allowed by the QoS parameters. Using a QoS threshold of 10^{-5} (a representative quality standard for medium compression MPEG transmission) utilisation characteristics are presented for the video stream for different G values. When the BER is above the threshold that is set as the minimum to satisfy the QoS requirements then, for each given value of G, as the

SNR increases, the quality of the real-time transmission will also benefit from decreased BER. Each dB rise gives approximately an order of magnitude decrease in the BER. For a given SNR, the jump to an increased *G* brings a quality decrease even where both values are above the threshold quality required by the system.

Figure 16 shows that each of the EBE outcomes for different *G* values go to zero at different SNRs, with lower gains more resistant to noise. It is also clear that the dynamic modulation gain optimisation case (this is just taking the maximum of all curves for a given SNR) has the best performance over all noise ranges since it is able to capture both the noise resilience of low *G* and the higher utilisation of high *G*. Finally, it is an interesting feature of this model that the above utilisation curve gives optimum SNR reference points for transfer between *G* levels.

3.5.2 Data with no FEC

The following results take ARQ into account. Figure 17 shows the probability of successful (no bit error) transmission of a single ATM cell (53 bytes) across the channel using PSK with no FEC. We observe that the probability of successful transmission of the packet experiences rapid degradation at a threshold SNR level that varies with the number of symbols in the constellation.

This rapid throughput decline is reflected in the overall efficiency of each constellation type in Figure 18, again using uncoded PSK in an AGWN channel. In that Figure EBE curves are presented for different values of *G* with an ARQ block size of one ATM cell.

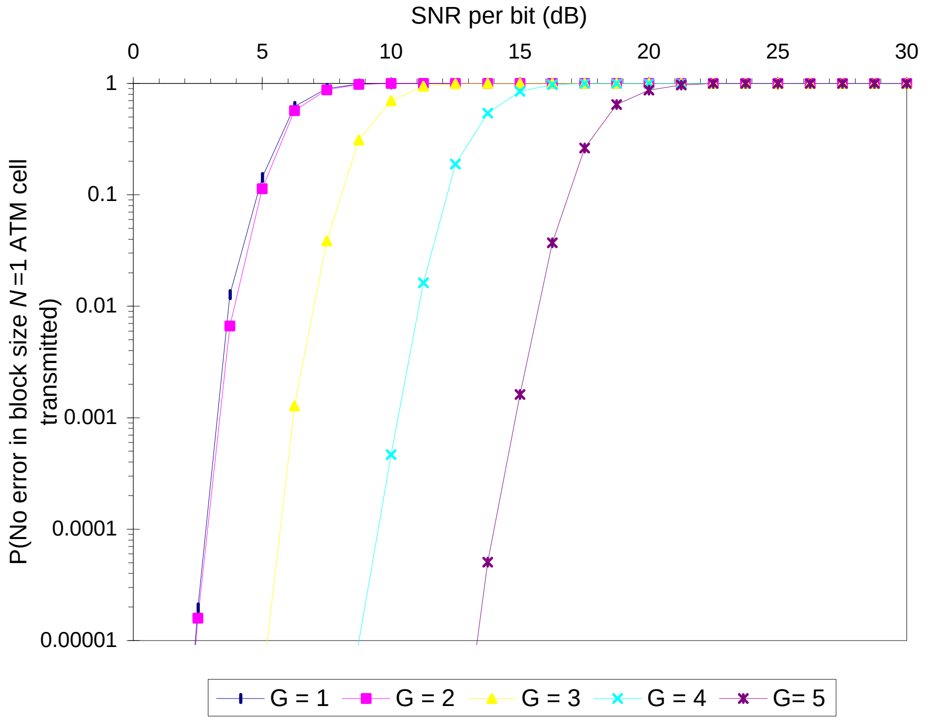

Figure 17. Comparison of the probability of error free transmission of a 53 byte cell across an uncoded AGWN channel using PSK modulation.

In Figure 19 efficiency results are presented for both fixed and adaptive *G* for ARQ block sizes of 1 and 25 ATM cells. The adaptive modulation scheme is compared with fixed gain modulation observing the importance of adaptive modulation and the effect of ARQ block size increases. The adaptive modulation scheme chooses the maximum EBE at any SNR of any Modulation Gain in Figure 18, which is given in Figure 19. Comparing the adaptive *G* case with any fixed *G* (in Figure 19, *G* = 4 is chosen) it is observed that if, for any SNR, the optimal modulation gain is chosen, then it is possible to improve significantly the EBE demonstrating the efficiency gain from using adaptive modulation in this simulation scheme.

From Figure 18, at SNR=23dB, the adaptive case (maximum EBE of *G*=2, 3, 4 or 5) provides four times the spectral efficiency of the fixed case (using *G*=2), although the adaptive case is still 10 of the normalised EBE units less efficient that the theoretical maximum channel capacity even after this significant improvement of 4 units (bits/s/Hz). It is further observed that, in the absence of an adaptive scheme, channel efficiency must be traded for ability to operate in noise if required coverage and network capacity are to be achieved.

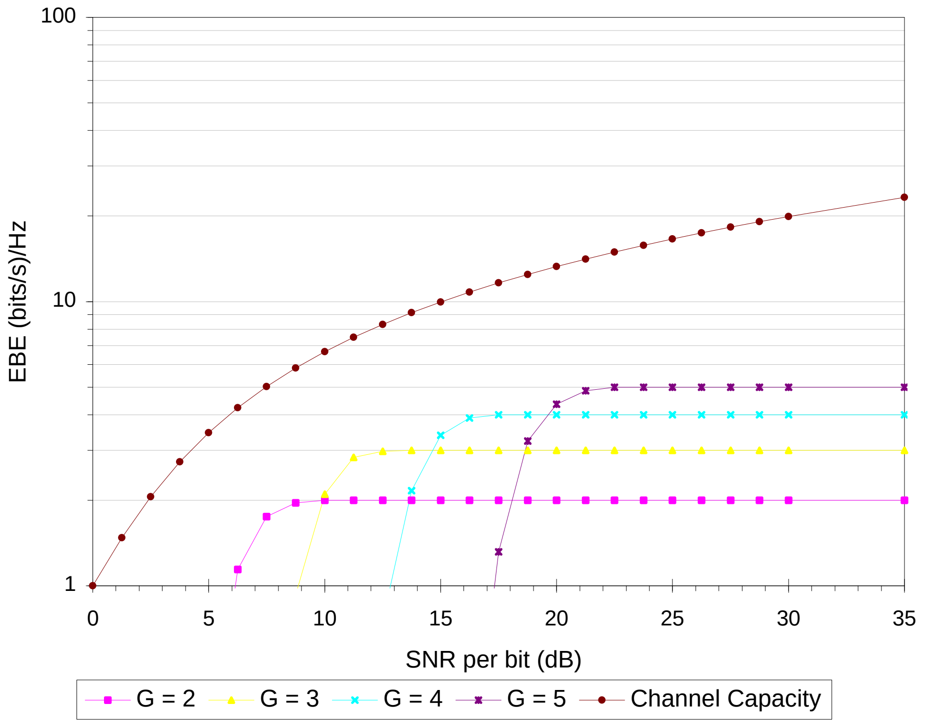

Figure 18. Comparison of EBE amongst modulation gains for uncoded PSK system with ARQ by constellation size for ARQ block size of 53 bytes over an AGWN channel.

For the case of larger ARQ block size (Figure 19), the fixed case has a near zero EBE at SNR = 15dB, however, the adaptive case is able to operate to 7dB in the absence of FEC. Figure 19 also shows the efficiency loss at lower values of SNR per bit for the larger frame size under the fixed and adaptive case (N = 25 ATM cells) as compared to using ARQ blocks of one ATM cell.

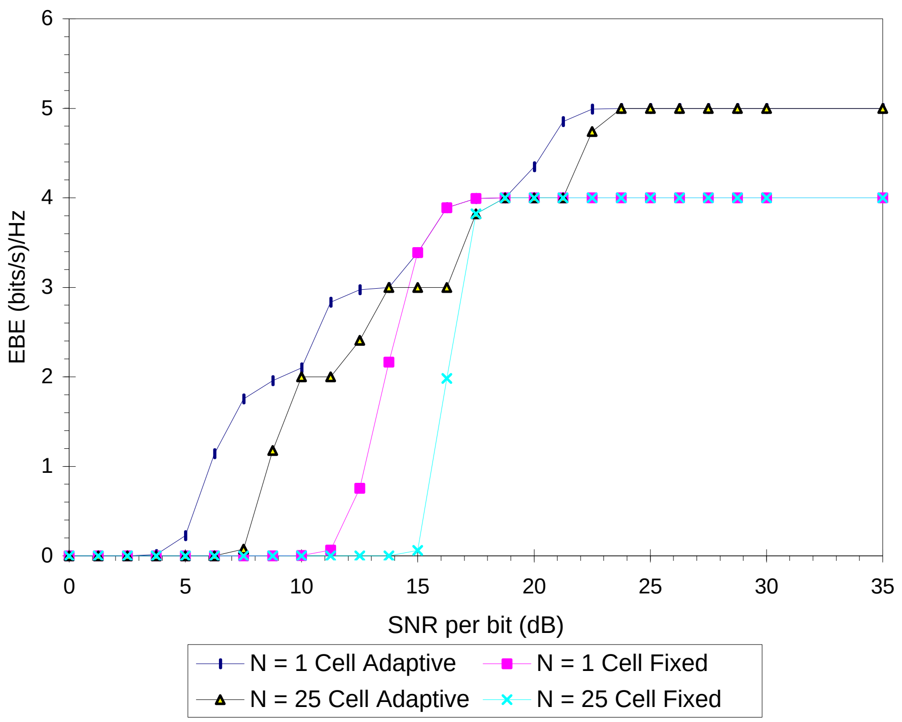

Figure 19. Comparison of EBE using adaptive and fixed modulation for our uncoded PSK system with ARQ block size equal to 1 and 25 ATM cells. The fixed case is $G = 4$.

The gradient of efficiency degradation per unit SNR for larger ARQ block size is greatly increased in both fixed and dynamic cases. Since the mobile environment is characterised by sudden and extreme variations in noise, this result shows that there must be a smaller latency in channel state feedback mechanisms for systems using larger ARQ block sizes, as they are more sensitive to changes in the channel state (see Section 2.6.4.5 above for further details on limits to the rate of adaptation in hybrid TDMA frames).

3.5.3 FEC Optimisation – Data

FEC is now considered in our adaptive modulation system from Figure 19. Independent bit errors in the system are assumed after modulation to provide the worst case for ARQ. The PSK system is simulated using an AGWN as before, but now includes various error-correction codes.

A selection of code rates and types were chosen for high gain and easy implementation, including simple convolutional codes with constraint length $L=3$, BCH, and Reed Solomon (RS) codes.

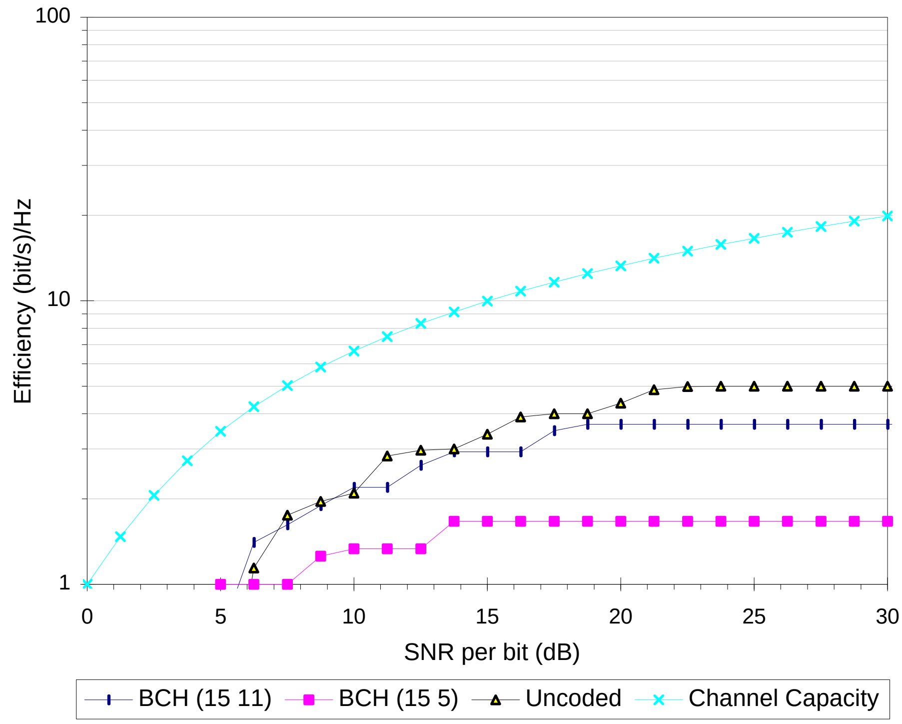

Figure 20. Efficiency comparison amongst coding schemes for a PSK system over an AGWN channel with ARQ for ARQ block size of 53 bytes (*N*=1 cell).

In this chapter, only block codes are presented, being a representative example. The aim is to optimise the FEC for the adaptive system described above, assuming first optimised adaptive Modulation Gain. For this case EBE is given by:

$$EBE^* = max_{\,G,R}\,[GR(1\text{-}e(G,R))^N\,]\,. \qquad (17.)$$

Figure 20 and Figure 21 use the adaptive modulation systems presented in Figure 19. They consider the efficiency gain to be made from the addition of FEC to a system with rapidly adaptive modulation and show that FEC with high R yet large gain is required to increase efficiency in such an adaptive scheme over the uncoded case.

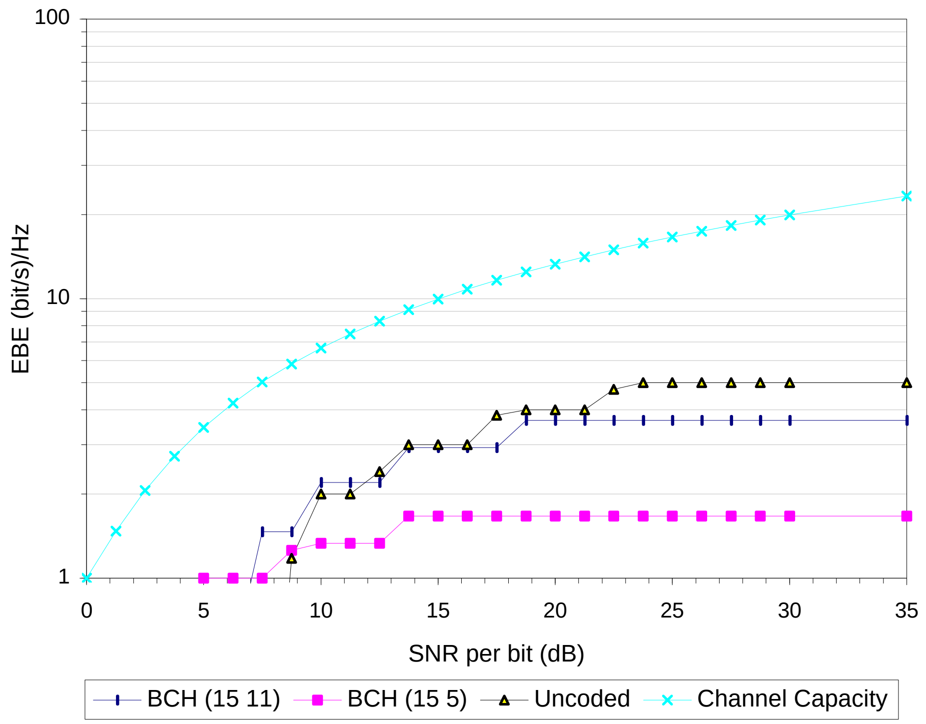

Figure 21. Efficiency comparison amongst coding schemes for a PSK system over a AWGN channel with ARQ for ARQ block size of 1325 bytes (*N*=25 ATM cells).

The Figures show that none of the FEC types were able to improve transmission efficiency (when adaptive modulation and ARQ are employed) except in high noise situations. This finding suggests that simple FEC schemes, of the type likely to be found in wireless terminals and handsets with constrained delay and complexity allowances, may not be of benefit to an adaptive system, whether with Go Back N or with Selective Repeat ARQ. Instead, the simulation suggests that a system relying exclusively on rapid modulation adaptation and ARQ (with small ARQ block size) without FEC out performs the same adaptive system with simple FEC except at extreme noise situations.

3.5.4 FEC Optimisation – Video

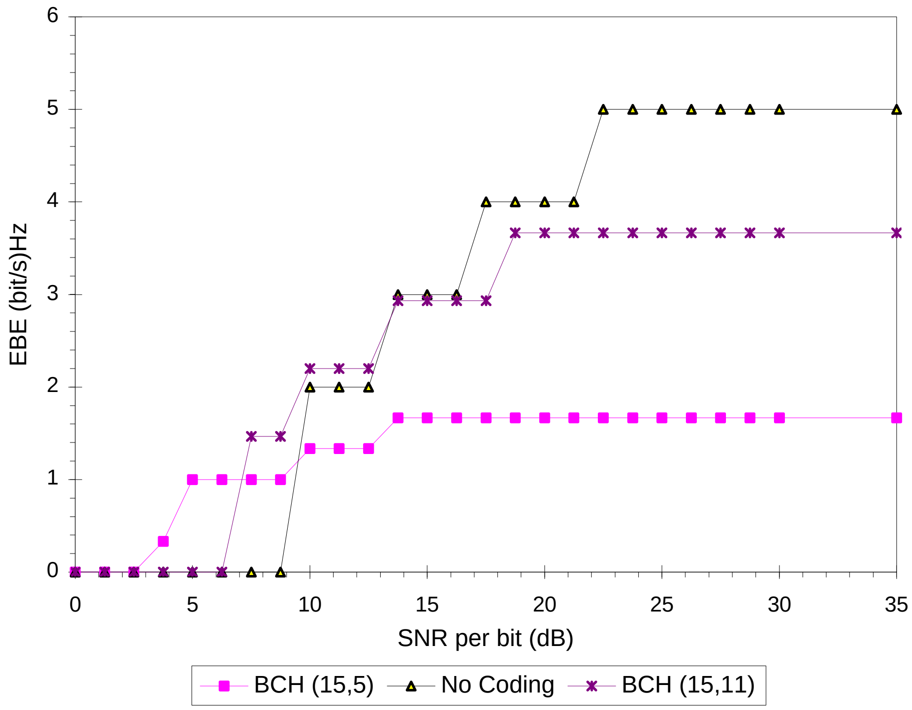

Figure 22. Efficiency comparison amongst fixed coding but adaptive modulation gains, optimised for simulation with QoS threshold discussed above. The system selects a value of *G* ranging between one and five dynamically to maximise system throughput for given SNR.

This study finds the above comments regarding non-transparent systems to also apply in the case of no ARQ. Figure 22 shows system efficiency when the threshold described in Section 3.5.1 is enforced. It is observed that, when FEC is optimised in conjunction to an optimisation over coding gain, little benefit is gained at higher SNRs since, after Modulation Gain optimisation, the added redundancy either outweighs the efficiency gain of the FEC. At lower levels of SNR, the cases are close, however there are gains, particularly for lighter codes except at high noise.

In [Tur97] a model of the General Packet Radio Service (GPRS) was simulated. It was observed there that in this selective repeat ARQ GSM system [Bra97], optimal performance was offered by the less powerful R = ¾ punctured convolutional code than other more powerful codes with increased redundancy. For the wireless case, the trade-off between FEC redundancy and ARQ Selective Repeat retransmission gave an optimum FEC rate point at R = ¾.

This simulation considers a more comprehensive set of parameters and includes an important optimisation across SNR. The results here support a tendency for the less powerful FEC codes to give enhanced efficiency over most SNRs, consistent with [Tur97]. However, it is noted that, in high loss situations, EBE is minimised by the use of increasingly powerful FEC codes as noise worsens.

3.6 *Results – Error Bursts*

In this section the effect of burstiness on efficiency is studied with the aim of validating observations in Section 3.5 that it is better to use uncoded, or high R FEC in a more realistic mobile environment – a Rayleigh Fading channel.

When errors appear in bursts, it is clear that more redundant information is required to clean the link using FEC where FEC is used alone. This result comes about since the BER in a small localised part of the traffic stream can be very high (and then zero for a long period of time) so that significant redundancy is required to clean the link during an error burst. The high level of redundancy is then wasted during the correlated periods of error free transmission.

In the simulations reported on in this chapter where there is an absence of interleaving, the increase in FEC redundancy required to clean the link is an increasing function of the burstiness, verifying that the independent error channel gives the best case for the operation of FEC.

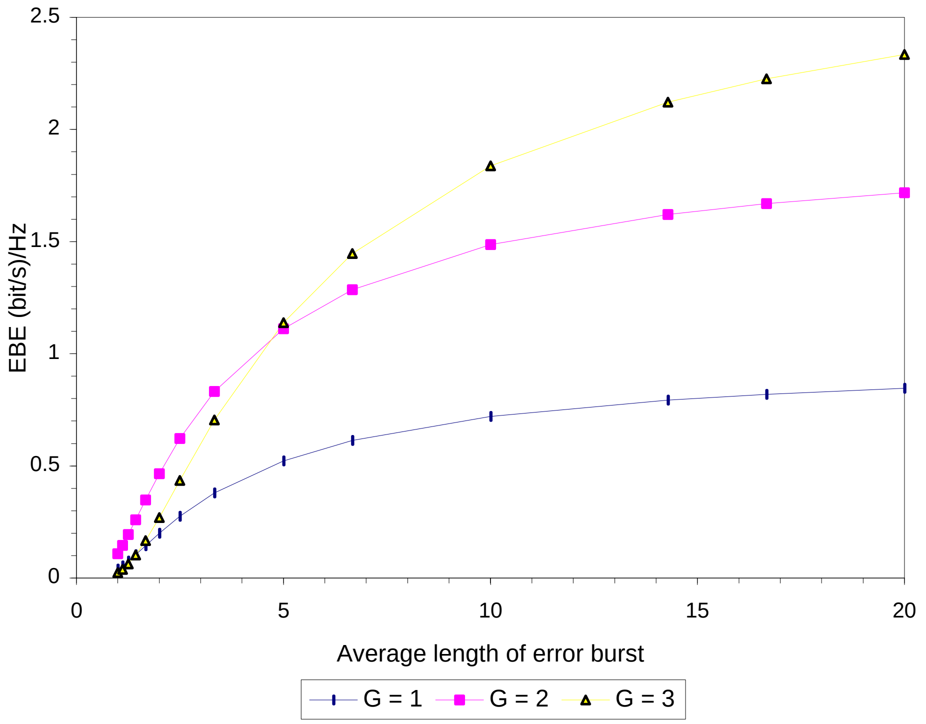

Figure 23. Effect of correlated errors on the simulation system. The channel is a Rayleigh Fading Channel with no FEC using an ARQ block size of 53 bytes. The efficiencies are given for a range of modulation gains at a fixed SNR per bit of 15dB.

For a fixed BER, however, as the burstiness factor (u) increases, so does q, the sojourn of the system in an error free state (see Section 3.3.2, above). This means that there is a lesser probability that the channel will move from the error free *good* state where there is a correlated session of error free transmission to the *bad* state in which the channel transmits bits that contain errors. Hence the probability that any given ARQ block is affected by errors is reduced as the errors congregate in a single block.

Figure 23 shows this phenomenon as the average length of an error burst increases. When we apply our burstiness analysis to the charts given in Section 3.5, we see significant improvement in efficiency for the uncoded case shown in the figure.

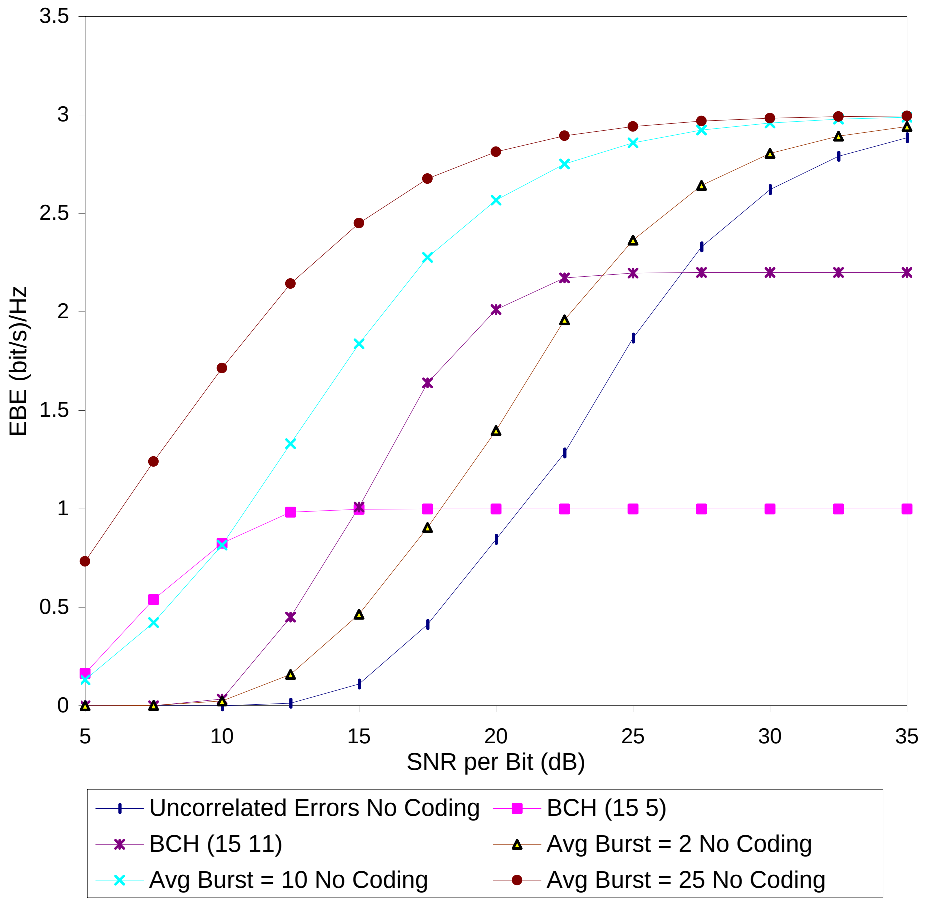

Figure 24. Efficiency comparison amongst coding schemes for a PSK system over a Rayleigh Fading Channel with ARQ block size of 25 ATM cells.

For the coded case, the effect of the FEC is to correct errors where they appear in bursts of less than a set length (this length depends on the FEC scheme under study). Where the errors appear in larger bursts, the FEC cannot operate to correct the errors and, failing to recover the errors, often adds to them.

For this reason, when offering the comparison between the coded and uncoded cases in Figure 24, best case for the FEC coded cases – that of independent errors are presented. On the other hand, Figure 24 shows the effect of burstiness for cases where there is no FEC coding. The FEC traces are for a Rayleigh Fading channel that uses interleaving to give a close to random placement of bit errors at the FEC interleaver output. The uncoded traces transmit bits across the same channel complete with the same interleaver, but if our burstiness analysis is then applied to the output of the interleaver, this results in dramatic efficiency gains in the system. Systems exhibiting average burst lengths of as little as 15 bits consecutively exhibit superior performance at almost all SNR values to any of our FEC examples. Further, the uncoded system has a superior maximum EBE to any of systems utilising coding.

Figure 25 presents a comparison between a wireless system incorporating interleaving and FEC against a system utilising neither. The diagram turns from the darker shade to the lighter shade when the interleaved system with FEC would have given a better EBE than the system without FEC or interleaving. Note that the results of such a comparison would be different were the comparison between a system utilising FEC *without* interleaving and a system not using FEC. Hence Figure 25 presents the best possible case for the use of FEC in line with our limit analysis in Section 3.3 above.

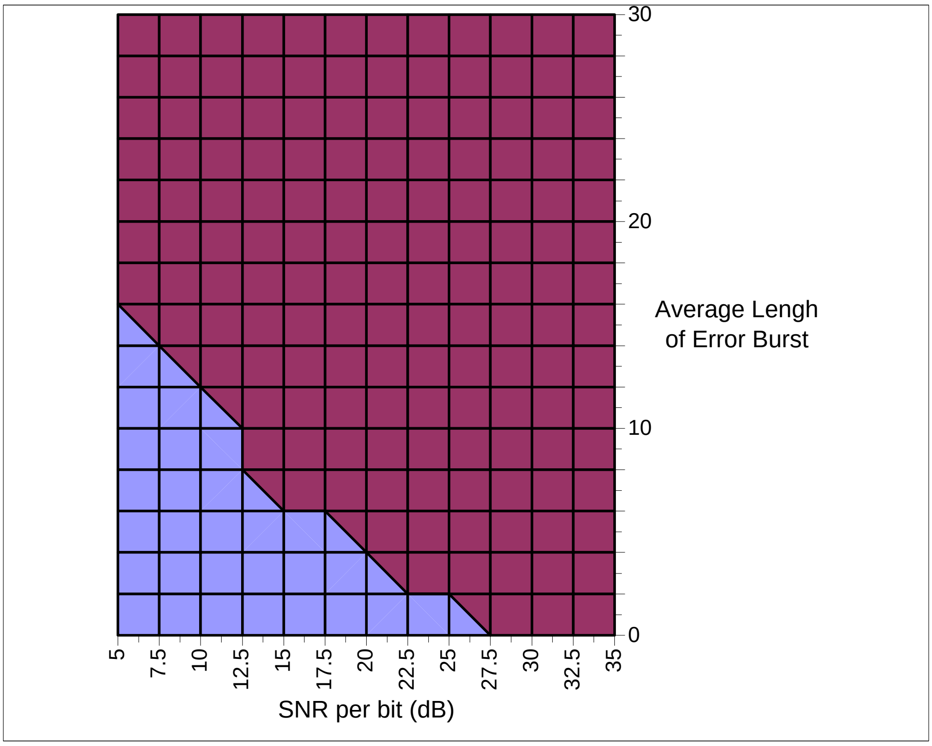

Figure 25. Comparison of efficiencies for an interleaved adaptive modulation PSK system over a Rayleigh Fading Channel with FEC and interleaving versus a non-interleaved system without FEC, both cases including ARQ block size *N*=25 ATM cells. The darker shade indicates superior performance without FEC and interleaving, and the lighter shade indicates superior performance with FEC and interleaving.

Referring back to previous observations and observing Figure 25, it is clear that a Rayleigh fading channel gives a strong preference to the use of uncoded transmission in conjunction with adaptive modulation over the channel. Because the channel is characterised by burst errors, a small block ARQ scheme better corrects the accumulating action of the channel than the use of simple FEC codes.

3.7 Conclusion

This chapter provides guidelines on optimising generic wireless access (or WATM) link dimensioning and coding parameters subject to meeting specified QoS requirements. Simulations used to derive BER values for a range of modulation efficiency and FEC Code Rate parameters, then optimised them to obtain maximal efficiency while taking into consideration a comprehensive set of coding and inter-layer issues.

The results compare the relative utility of a set of FEC, ARQ and Modulation Gain protocols in an error prone environment so that a limit on the use of FEC is established. Simulations over an AGWN channel show that adaptive modulation offers significant efficiency and coverage gains over the fixed system when ARQ is considered. The simulation over a Rayleigh Fading channel shows that observations in Section 3.5 about the relative value of ARQ over FEC in obtaining optimal efficiency retain their validity in realistic environments.

It is demonstrated that, in many instances, an adaptive modulation system utilising small block ARQ over the wireless link exhibits optimal EBE without FEC over wide SNR ranges. It is a very important observation that maximising efficiency is dependent on SNR as well as the trade-off between the FEC code, the ARQ and modulation gain. Further, results show that, even for certain realistic SNR conditions, the wastage due to errors can be very high because of retransmission. Finally it is demonstrated that a reduction in ARQ block size has a beneficial effect in non-transparent systems.

The chapter’s findings suggest that preference should be given to systems which adaptively change their parameters according to the instantaneous level of interference and noise and that optimum performance can be delivered by a system using packet by packet adaptive Modulation Optimisation rather than focussing on FEC based solutions.

4. TCP OVER WIRELESS ACCESS NETWORKS

4.1 Introduction

This Chapter aims to use the results from Chapter 3 to provide one solution to inter-layer interaction between the physical error process of a wireless access network and the error recovery systems of the higher layer TCP/IP. The chapter commences with a brief introduction to TCP RENO and TAHOE, the two most widely implemented TCP schemes in today's Internet. It goes on to consider a number of solutions to the 'slow start' issue that these versions of TCP present the wireless designer. Commencing with split connection schemes, and progressing through network layer schemes, random loss detection schemes and application level protocols, Link Layer ARQ schemes are settled upon as a preferred solution to the 'slow start' problem.

The addition of a Link Layer ARQ subsystem has the dual advantages of an efficiency increase to the overall system, regardless of the nature of the higher OSI layers, as well as providing a solution to issues with the use of TCP/IP over wireless access networks. It does so, at the cost of an additional layer of complexity. Before considering the inter-layer interactions between TCP and the performance techniques optimised in Chapter 3, operation of TCP TAHOE and RENO are first introduced.

4.2 Congestion Control in TCP TAHOE and RENO

TCP is a reliable window-based ACK clocked flow control protocol. It uses an additive increase – multiplicative decrease strategy for changing its window size under differing network conditions. Starting from one (or more) packets, the number of packets underway in the network is increased by one packet for each non-duplicate ACK until the source estimate of the network capacity is reached. Once this is reached, the source switches to the contention avoidance phase and continues to increase throughput at the rate of 1 packet for each window transmitted. The window size gives the maximum allowable number of unacknowledged packets (not counting retransmissions) that may be underway in the network at any given time. For an infinite data source, the connection will use its window to the fullest possible extent.

Losses are detected when consecutive ACKs sent to the source have the same “next expected” number. In this situation TCP retransmits the packet after a threshold (three) number of duplicate ACKs have been received (this is the fast retransmit option). If the packet loss is not detected (or Fast Recovery is not a part of the algorithm being used) then there will be an expiry of the timer.

It has been suggested [BM97] that one of the major problems with TCP and ATM over wireless links is that the window protocol mechanisms built into TCP are designed to avoid network congestion rather than assuming that loss is caused by transmission error. In other words, when packets are lost, TCP reduces its throughput in order to alleviate congestion. TAHOE [Jac88], the first implementation to include congestion control, sets the window after such an error to one packet and uses slow start to increase the window again which deteriorates the performance as a result of the low bandwidth utilisation during slow start.

Other versions (Reno, New Reno, SACK) implement a Fast Recovery (FRCV) algorithm to retransmit the losses so that the ACK clock is preserved and no Timeout occurs [BAD99]. TCP RENO is similar to TCP TAHOE except that it tries to remain in congestion avoidance unless there is a timer expiry. Packet loss (detected via duplicate ACKs) result in the window size being cut in half. If a timer expiry does occur, then the window size is reduced to one and slow start is used to grow the window back to half its value at the time of the timer expiry. See [FV91] and [ZSC91] for simulation studies.

The TCP TAHOE protocol [Jac88] uses the following algorithm in determining what should be the transmit window:

```
<Initialise>
Set cwnd = segsize;
Set ssthresh = 65535 bytes;

<After every non-repeated ack>
if cwnd <= ssthresh, set cwnd = cwnd+segsize; <Slowstart>
else set cwnd = cwnd+[segssize*segsize/cwnd]; <Congestion
Avoidance>

<When 3 duplicate ACKS arrive, or upon round-trip timer
expiry>
ssthresh = min{cwnd,rwnd}/2; <Rate Backoff>
cwnd = segsize;

<In all cases, the transmit window>
twnd = min{cwnd, rwnd};
```

Where: *cwnd* is the congestion window; *ssthresh* is the slow start threshold; *rwnd* is the receiver window; *twnd* is the transmit window and *segsize* is the maximum segment size indicated by the mobile. Figure 26 demonstrates the operation of the protocol in the presence of losses over the link.

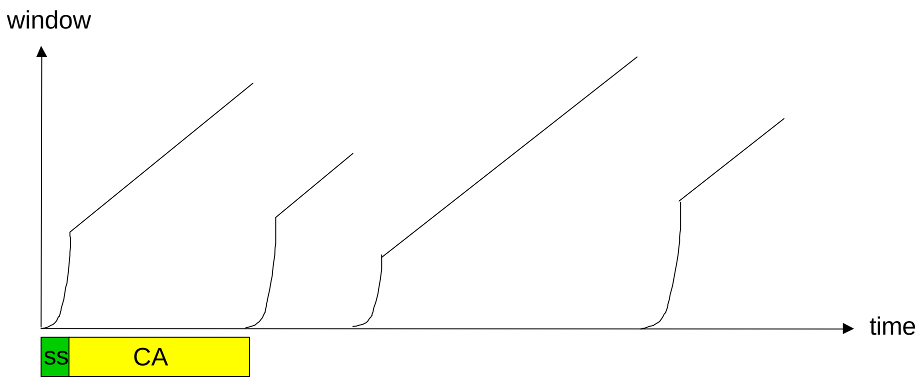

Figure 26. Demonstrating the response of TCP TAHOE to loss on the wireless link. A loss is recorded at each return of the window size trace to one [Low00]. SS represents the Slow Start period and CA represents the Contention Avoidance period.

The TCP RENO protocol [Jac90] uses the following algorithm in determining what should be the transmit window:

```
<Initialise>
Set cwnd = segsize;
Set ssthresh = 65535 bytes;

<After every non-repeated ack>
if cwnd <= ssthresh, set cwnd = cwnd+segsize; <Slowstart>
else set cwnd = cwnd+[segssize*segsize/cwnd]; <Congestion
Avoidance>

<When 3 duplicate ACKS arrive, Fast Retransmit>
ssthresh = max{twnd/2,2*segsize};
cwnd = ssthresh+3*segsize; <retransmit missed segment>
cwnd = cwnd + segsize; <for each duplicate ack – window
inflation>
cwnd = ssthresh; <when the ack for the retransmitted segment
arrives>
```

```
<Upon timer expiry – same as TAHOE>
ssthresh = min{cwnd,rwnd}/2; <Rate Backoff>
cwnd = segsize;

<In all cases, the transmit window>
twnd = min{cwnd, rwnd};
```

Figure 27 demonstrates the operation of the protocol in the presence of losses over the link.

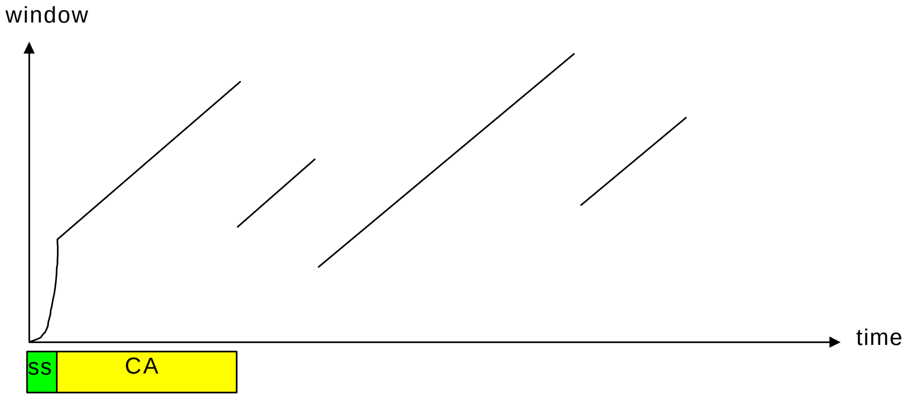

Figure 27. Demonstrating the response of TCP RENO to loss on the wireless link. A loss is recorded at each discontinuous jump in the window size trace [Low00]. Abbreviations as in Figure 26.

4.3 *TCP Slow Start – Analysing the Window Size and Throughput*

This section develops a model to analyse the performance of the wireless access link when TCP is used.

Consider a single TCP connection on an error-free bottleneck link of speed C packets/s [CLM99]. In the presence of a round trip delay of time t (constant), packet loss occurs on average when the window size reaches $Ct+B$ where B is the size of the bottleneck buffer. At this time, the window size drops down to one and increases exponentially during slow start to size $(Ct+B)/2$. It then increases almost linearly during congestion avoidance to $Ct+B$ at which point there is another loss. The slow start phase is relatively short or, in the case of FRCV, may not be entered at all. Hence it is the congestion avoidance period that largely governs the throughput. To fully use the bottleneck resource, the window size must be larger or equal to the product Ct at all times. Where B is greater or equal to Ct, the condition is satisfied for the entire congestion avoidance phase, so that if node buffering scales with the bandwidth delay product, utilisation is very good.

Moving now to a wireless link, if it is assumed that any packet can be lost with probability d, where each packet loss is independent, the window size will be reduced more often due to these random losses and the largest window size may never be attained. If the largest window size attained on congestion on average is $2W$ then the next congestion phase will commence on average at W. If there is no congestion, the number of packets transmitted as the window evolves from W to $2W$ is $1/d$, linking the window size directly to the error rate. If $2W$ is much smaller than $Ct+B$ (the maximum window size attained by TCP without random loss) then the window size, and hence throughput, will be severely reduced.

Hence if the round trip time is large and the loss rate is significant, the TCP connection may never operate at an offered rate that fully utilises the available bandwidth. Indeed, if the product $(Ct)^2 >> 1$ then random loss causes serious deterioration in TCP throughput. This means that the residual end-to-end loss probability seen by TCP after correction by lower layers must scale inversely with the square of the bandwidth delay product of the connection [CLM99].

As Chaskar *et al* point out [CLM99], the analysis above suggests the following features for an efficient deployment of TCP over wireless:

- The end-to-end packet loss seen by the TCP should be less than $1/(Ct)^2$;
- If an end-to-end DLC small block ARQ is utilised as the method for error recovery (see Section 4.4.7 below and Section 2.8 above for discussion) then it can be assumed that every packet that enters a buffer at the wireline-wireless interface is eventually delivered to the mobile terminal. In this case, the only losses seen by TCP will be the buffer overflow at this point. Hence the buffer B overflow probability should be no more than $1/(Ct)^2$ when the utilisation of the wireless link is significant;
- Any DLC small block ARQ implemented (again, see Section 4.4.7 below and Section 2.8 for discussion) must deliver the TCP packets in sequence to the higher layers; and
- The end-to-end round trip timers must be conservative enough to allow time for DLC retransmissions to allow recovery on the wireless link. In practice this is a given as the source calculates the round trip time as the measured time plus a conservative factor proportional to the measured standard deviation for a TCP connection [Com96].

4.4 Solutions to Non-congestion Performance Issues in TCP/IP

4.4.1 Introduction

There are two main categories of solution to the problem of non-congestion losses in TCP. The first is to hide the 'lossy' parts of the heterogeneous network so that only congestion losses are detected at the source (the focus of this chapter). The second type of solution is to enhance TCP with some mechanisms to help it to distinguish between different types of losses [BPK97] [Ray97] [CLM99].

An outline of the specifics of the second solution is provided in [BAD99] and proposals have included the split connection approach [BB94] and the snoop protocol. Because the packet size (and for the case of Go Back *N* transmission, the window size), will be large for any TCP recovery mechanism, Figure 19 shows that an investigation of the first solution, where smaller block ARQ and FEC can be implemented, is more the useful of the two.

4.4.2 Split Connection Schemes

4.4.2.1 Split protocol

In [BB97], Bakre and Badrinath further develop their split protocol scheme, an indirect TCP protocol that emulates the mobile for the purpose of defining the socket. This means that the TCP connection is set up between the wireline-to-wireless gateway and the source, the eventual delivery of the data to the mobile terminal being handled by a protocol internal to the wireless system, usually a different wireless link optimised transport protocol.

There are several drawbacks in such a system as outlined in [Ban00]. The most important is that the system suffers from the lack of end-to-end semantics. An ACK arriving at the remote host only guarantees delivery of the data packet to the wireless gateway (and not necessarily to the mobile) requiring the possible addition in higher layers of a session layer capable of performing its own end-to-end integrity check.

4.4.2.2 Mobile Host Protocol (MHP)

Yavatkar and Bhagawat [YB95] propose a different split protocol scheme which calls for a new session layer protocol (that they call the MHP) to sit below the TCP layer between the wireless gateway and the mobile host. The MHP fragments wireline data segments into smaller wireless link segments to provide greater efficiency over the lossy link, also overseeing exchange of transport layer state information during handoffs. The authors study the effect of MHP with two different transport protocols: ordinary TCP, and SACK.

4.4.2.3 Mobile-end Transfer Protocol (METP)

METP [WT98] is another split protocol scheme. Here, again, the TCP/IP connection terminates at the base station, but this time the authors develop a different relay mechanism for the transfer of data between the base station and the mobile. Although more robust in some circumstances, this protocol features the common drawback of all split connection schemes described in Section 4.4.2.1.

4.4.3 Network Layer Schemes

In [HH98] the problem of random loss over wireless link was addressed by having TCP work on top of an end-to-end reliable ATM class of service called SSCF-TADA5 augmented with Specific Connection Oriented Protocol (SCOP) headers [Ban00]. The SSCF-TADA5 protocol maintains error-free in-order data delivery, preventing TCP from going into its rate back-off mode.

There are several issues with this and similar schemes:

- As stated in [HH98], the proposal adds considerable overhead, resulting in lesser goodput, and lesser throughput except in high BER (>10^{-4}) cases;
- It is not clear whether a network layer end-to-end delivery scheme can work well in presence of frequent handoff – the issue of handoff was not addressed in [HH98]; and
- A complicated network layer is required for implementation, which is, in general, unsuitable for commercial purposes (the scheme was originally designed for military satellite applications).

4.4.4 End-to-End Schemes

4.4.4.1 EBSN

One way of combating the above issue with Link Layer schemes is to alter the TCP round trip timer's value to reflect the bad channel state by using an Explicit Bad State Notification (EBSN). The EBSN method, when combined with Link Layer retransmission, has the advantage that it can (in theory) eliminate TCP rate backoff if the timer is increased so as not to expire while the Link Layer tries to retransmit the lost packet.

The scheme suffers from the obvious disadvantage of requiring changes in the TCP implementation that can, if improperly used, lead to network deadlock. The study in [BKV97] also tries to determine the optimum packet size for a particular wireless channel. There appears to be further work to be done.

4.4.5 Random Packet Loss Detection

Any approach seeking to determine whether a packet loss is due to random bit errors or complete loss in transmission or whether it is due to congestion-related buffer overflow must solve the sophisticated problem of distinguishing between the two in real-time. Described below is some promising research in this area.

4.4.5.1 Interarrival times

Biaz and Vidya [BV99] distinguish between the two types of loss using a scheme in which the average interval between packets is measured and then an algorithm used to determine whether an out-of-order packet arrival indicates wireless loss or not. The algorithm tests for packet loss due to error by testing whether a packet arrives in place of the packet that should have immediately preceded it. A mobile detecting random loss sends a message to the remote host so that the congestion window is not reduced.

This simple scheme has been shown to work well in simulations using moderate AWGN channels when the wireless link capacity is smaller than the wireline link capacity but there are several limitations. The scheme assumes packets travel through internet space as a regularly spaced ordered scheme even though the IP datagram service makes no such guarantee. Further, modelling results for the scheme assume that the source has enough data to send at a constant rate, an assumption that is not true in general. Finally, the simulations require the use of a long warm up period (100sec) for training the average arrival time – in practice very few sessions with continuous data transmission will last that long.

4.4.5.2 Intermediate acknowledgement

Cobb and Agrawal [CA95] advocate an entirely different approach to wireless packet loss detection. Their method uses an intermediate acknowledgement back from the last hop router (i.e. the one directly in communication to the mobile) to the remote host to indicate that the packet has traversed the wired portion of the link without any congestion. If this were the case, the standard TCP ACK from the mobile terminal indicates that the packet was received without an error.

The scheme is essentially a Link Layer conjugate scheme for the wireline Internet and, as such, it has the following drawbacks:

- The scheme assumes that the remote host will be aware of whether it is supposed to receive intermediate ACKs or not (i.e. whether talking to a mobile or not). This implies setting up a different kind of TCP connections when it comes to communicating with a mobile;
- Additional ACKs may have to traverse several hops spanning continents that may not take the same path followed by the original packets, an example being a triangle route for MobileIP. For this reason it would be unclear whether these intermediate ACKs really indicate congestion in the original packet path, or rather the return path; and
- The most likely place for a congestion to occur will be at the wireless link whose capacity will typically be less than that of a wireline link. Hence the scheme fails to detect congestion in its most probable location.

4.4.6 Application Level Schemes

This family of solutions attempts to address TCP issues without changing TCP itself. One example solution [All99] calls for the establishment of many parallel TCP connections for the same transfer. This accelerates the growth of the resultant window but increases the aggressiveness of the transfer, hence increasing losses in the system. An adaptive solution to this difficulty is currently under study [All99].

4.4.7 *Link Layer ARQ Schemes*

In addition to the more complex schemes that have been considered so far, there are two well known mechanisms for the improvement of link quality at lower levels and hence the solution of inter-layer effects. These are retransmissions (ARQ) and FEC and they can be utilised effectively in Link Layer schemes. ARQ is efficient when losses are not frequent and when the propagation delay is not significant, however it may interfere with TCP mechanisms. FEC, on the other hand, utilises additional redundant transmission to allow for the reconstruction of a message at the receiver. However, when that redundant information is not required, there is a waste of bandwidth. There are further CPU processing penalties and memory issues.

A common disadvantage of all Link Layer schemes is that multiple retransmissions due to a long fading period typically causes expiry of the TCP timer. Almost all such schemes depend on 500ms granularity in the timer for justifying non-expiry during Link Layer retransmissions. There are proposals (unrelated to the wireless TCP investigation) to improve this granularity to 100ms, which can affect these schemes considerably if independently implemented at the source.

4.4.7.1 Link Shaping

Chaskar, Lakshman and Madhow [CLM99] develop an analytical model for TCP operation using a Link Layer retransmission. They model a Rayleigh fading channel as a two-state (good and bad) Markov model with transition probabilities derived in the paper. Assuming a certain Link Layer buffer size, they take the system to be a finite Markov chain whose states correspond to all combinations of channel states and buffer occupation levels. Transition probabilities are computed assuming that the TCP packet arrival at the wireless gateway is a Bernoulli process, and that a packet is never delivered in the bad channel state while it is delivered with a certain probability in the good channel state. The buffer overflow probability (which is also the random loss probability after the Link Layer scheme is used) is then computed and the buffer size is chosen such that it becomes equal to $1/(Ct)^2$ (see Section 4.3 above), which is a threshold between congestion-limited and random loss-limited operation.

Another important finding of that work is that the Link Layer buffer size required for satisfying this condition scales logarithmically to the bandwidth delay product, while the buffer size required improving congestion probability for a error free link scales linearly to the bandwidth-delay product. An issue, for any link level scheme, however, is that in order for these performance figures to be utilised, a TCP-aware Link Layer Protocol is required.

4.4.7.2 Wireless TCP (WTCP)

A specific Link Layer scheme termed WTCP is proposed in [RM98] whereby a modified TCP protocol is used for Link Layer retransmissions between the wireless gateway and the mobile. Although this protocol uses a TCP-TAHOE like window decrease, it uses an aggressive window increase when packets start flowing again, effectively bypassing the slow start and congestion avoidance phases. This leads to significant increase in throughput for longer fade duration, but it achieves this throughput at the cost of reducing wireless goodput (not reported).

An improvement in the base WTCP suggestion uses a finer granularity timer for the wireless link. The protocol also uses time stamp updating on RFC 1323 compliant TCP timestamps by subtracting the queuing time at the base station. This allows it to detect and respond to wireline congestion more effectively. The process notably increases probability of timeouts at the remote host while the Link Layer retransmissions are in progress – a possibility that was not considered by [RM98].

4.4.7.3 Snoop Protocol

Another specific proposal for a Link Layer scheme is the Snoop Protocol developed at Berkeley. The performance of the Snoop [BSK95] protocol was experimentally verified over multi-slot GSM data networks. It was found that although Snoop improves performance over a low round trip time (25ms) WaveLAN network about 20 times, it improves performance on the high round trip time (2.4s) GSM network by about 4 times in high BER (3.5×10^{-5}) cases, lower in lower BER cases.

The particular problem of Snoop is that it violates the e2e semantics of the congestion control scheme found in most implementations of TCP. The cause of the violation is the suppression of duplicate ACKSs (which filters out a congestion signal) as well as the inclusion of a proposed NACK scheme. When sending *to* the mobile host, packets dropped at a bottleneck link between the wireless link and the mobile host are mistaken for damage loss by the TCP-aware cache. The congestion signal is not propagated back to the sender. For packets sent *from* the mobile host, the Negative ACK (NACK) scheme causes a problem. If the wireless link itself (or any other link between the mobile host and the wireless link) becomes the bottleneck, packets lost due to congestion cannot be discriminated from those lost due to damage. A NACK is sent in either case, and the sender again relies on external means to get the congestion signal (e.g., the infamous source quench).

4.4.8 Analysing Link Layer Solutions

4.4.8.1 Motivation

An option considered in [NBJ97] suggests the implementation of an additional small block ARQ in a specific Wireless Data Link Control Layer (rather than end-to-end ARQ) to deliver a better quality channel over the wireless link to the higher layer protocols. The small block, low level ARQ has the effect of 'fooling' higher layers into believing that the physical layer is a higher quality than it is, thus avoiding inefficiencies in retransmission of larger blocks at higher levels. This approach indirectly encapsulates the efficiency benefit of smaller block size retransmission for multi-media wireless networks identified in Section 3.5, above.

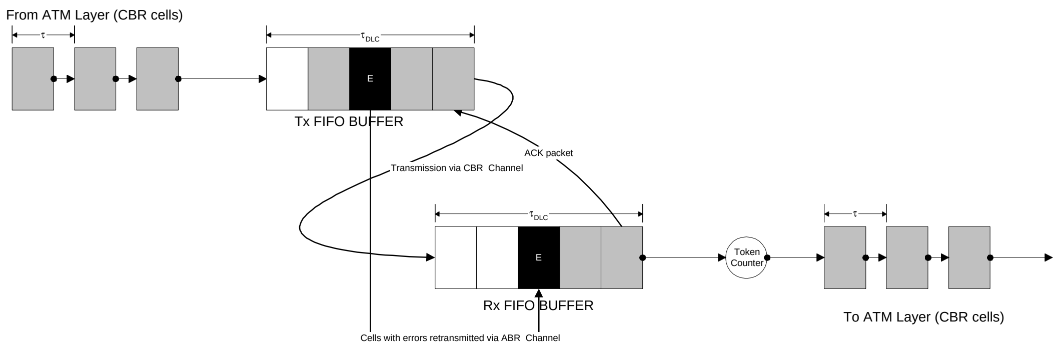

Figure 28. Block diagram of the DLC ARQ system.

An example of a DLC system for CBR VCs is given in Figure 28. Cells arrive in sequence from the ATM layer. A FIFO buffer is maintained with size bounded by acceptable delay and observations from Section 4.3 (t_{DLC}), providing a window for cell based error recovery. Functions of the DLC at the remote station are the same as those for the base, except the remote DLC does not send requests for the retransmission of lost cells. It was found that such an approach could have a significant effect on the efficiency of the system, almost doubling it [NBJ97].

The reasons for such an efficiency gain are clear. If C is the available capacity for the connection (service rate of the queue in Section 0 above), BER_{FEC} is the BER after FEC rate R coding, and d^* the point at which the benefit of utilisation versus mean delay is the highest in the wireless ATM network (see [ZHG98] for detailed discussion), then optimum traffic arrival for a set capacity link is given by:

$$l_{CUS} = \max_R \left(d^* CR \left(1 - BER_{FEC}\right)^N \right), \qquad (18.)$$

which is largest for small N since the capacity penalty due to random errors is a decreasing function of N, where N is the data block size for error checking. Hence the performance increase resulting from the addition of error checking in the DLC is explained by the fact that correction on a per cell basis (N=53*8 bits) has a smaller data block than using TCP packets (N=53*8*S bits where S is the TCP window size factor). The above analysis does not take into account the penalty of greater overhead for smaller block size (a penalty for the small block ARQ), nor does it take into account round trip delays for retransmission requests. The former is neglected as small by comparison to the efficiency gain from the smaller block size and the latter is neglected and small for the case of the localised ARQ (but note that it would be much larger if TCP is required to provide the error correction). These simplifying assumptions are avoided in the fuller analysis presented in the next section.

4.4.8.2 Optimising block size

[Sch96] gives the optimal packet size to be used by a data link protocol when a perfect retransmission scheme (one that only retransmits packets that are in error and can continuously retransmit new packets as long as no errors occur, e.g. selective repeat ARQ) is used. This is

$$N_{opt} = \frac{-h\ln(1-e(G,R)) - \sqrt{-4h\ln(1-e(G,R)) + h^2\ln\left(1-e(G,R)^2\right)}}{2\ln(1-e(G,R))} ,$$

(19.)

where N_{opt} is the optimal packet size to be used by the data link protocol, $e(G,R)$ the known bit error rate of the link after all correction other than ARQ, and h is the number of overhead bits in the packet.

There are two methods of determining in real-time what packet size should be used. These are use of channel feedback provided from lower layers to access a look up table [AJ00], or the study of previous retransmissions to determine future packet size [Mod99]. Each method must be tailored to take into account the speed of fading in the channel in order to maximise efficiency.

Of particular interest here is the approach adopted in [Mod99] because of its ease of use and simplicity. Avoiding the inferior solution of choosing a single optimal packet size based on knowledge of the *average environment*, a multi-level block size ARQ is considered. Repeating again the requirement for an 'optimal' ARQ, the efficiency of any scheme with payload size k ($N=k+h$) is given by

$$E = \left(\frac{k}{k+h}\right)(1-e(G,R))^{(k+h)} , \qquad (20.)$$

where k is the number of information bits in the ARQ block. Given an efficiency approximation (that disregards channel delay) and the number of retransmitted packets (D) out of the last L packet transmissions, then a measure of the expected efficiency is:

$$E_D(k) = \int_{BER} \frac{k(1-e(G,R))^{k+h}}{k+h} P(e(G,R)|D) \quad . \qquad (21.)$$

$P(BER\,|\,D)$ is the conditional probability of $e(G,R)$ given that D out of the last L packets required retransmission. This leads to an expression for the expected efficiency of the protocol given E, the probability that a packet contains errors which is just

$$E = 1-(1-e(G,R))^{N_L} \quad ,$$

where N_L is the block size of the ARQ for the previous L transmissions. Since

$$P(D\,|\,e(G,R)) = \binom{L}{D} E^D (1-E)^{L-D} \quad ,$$

then Eq. (21) can be given as

$$E_D(k) = \int_{e(G,R)} \left[\frac{k(1-e(G,R))^{k+h} \binom{L}{D} E^D (1-E)^{L-D}}{(k+h) \int_{e(G,R)} \binom{L}{D} E^D (1-E)^{L-D}} \right] \quad . \qquad (22.)$$

The size of the next packet is found by finding the value of k that maximises Eq. (22). In general, it is likely that any ATM Adaptation

Layer (AAL) protocol will be capable of only a small set of k values, making the maximisation search quite simple.

Even so, one issue with the above approach would be determining an efficient size of L. The solution might be ineffective in a channel that is characterised by fast fading, so that the optimisation of the channel packet size could trail the channel characteristics, never attaining an optimal ARQ block size. A simpler heuristic solution would be would use Eq. (19) together with frame-by-frame BER feedback. As long as the duration of fades on the channel were not small by comparison with the frame size, a simple lookup table could be devised using Eq. (19) to give payload sizes k for ARQ blocks during that frame. This approach is computationally more efficient and avoids problems of lag in the control algorithm.

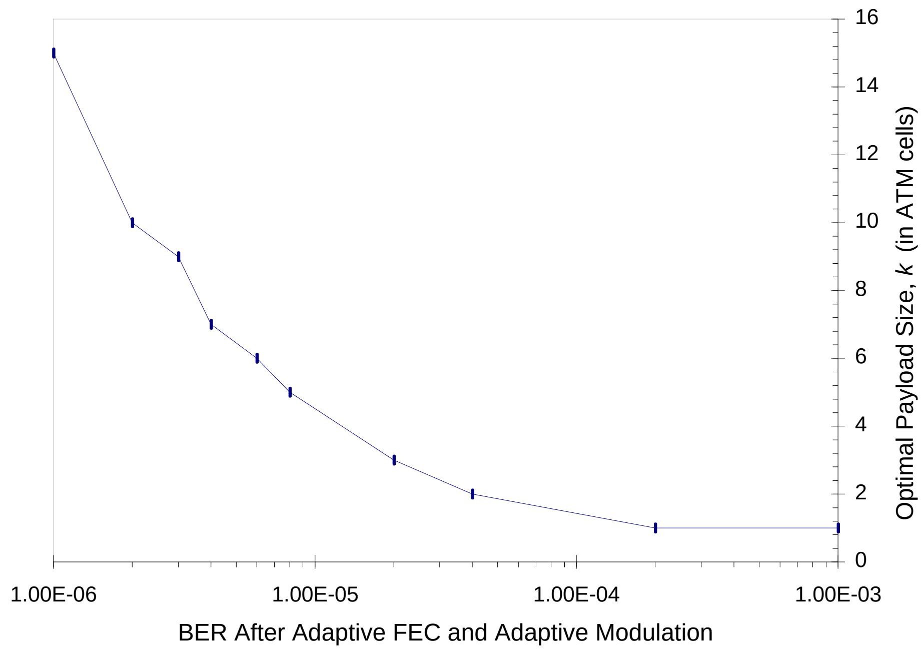

Figure 29. Look up diagram for payload lengths, k as a function of given BER after all adaptive FEC and adaptive Modulation. The chart suggests four primary ARQ payload lengths: 15, 10, 5 and 1 cells.

Using the simulation model that outlined in Chapter 3, this approach is implemented utilising the control packet format in Figure 30.

Figure 30. Representative frame structure for AAL small block ARQ system with variable payload.

The control part of the packet contains a commencement tag (1 byte), followed by a length indicator for the payload (2 bytes). Although not specifically marked, there is also a trailing octet to indicate the end of the frame. Further, the frame has a sequence number to indicate the identity of the block being transmitted. This field is 2 bytes long. CRC-16 is used for error detection. Given this structure of frame and a selective repeat ARQ, the look-up chart (Figure 29) was derived from the simulations of the previous chapter and gives a value for *k*.

4.5 Conclusion

This Chapter has provided critical review of a number of suggestions for solving issues of inter-layer interaction between the physical error process of a wireless access network and the error recovery systems of the higher layer TCP/IP. It recommends the use of an adaptive DLC (AAL) ARQ where payload sizes are set dynamically using a lookup chart (Figure 29).

The addition of such a subsystem has the dual advantages of: an efficiency increase to the overall system regardless of the nature of the higher OSI layers; as well as providing a solution to the 'slow start' problem in TCP/IP over wireless. It does so, however, at the cost of an additional layer of complexity.

In the absence of a reliable AAL layer below TCP that performs end-to-end ARQ, the results suggest that, for the fixed modulation case, reduction in ARQ block size for fixed and adaptive modulation is of benefit in increasing efficiency due to severe threshold effect in utilisation for larger ARQ block sizes. Figure 19 shows the SNR decrease required for constant EBE for the fixed modulation case of almost 5 dB achieved as a result of reduction of ARQ block size from 25 ATM cells to 1 cell. Any TCP packet would be much larger than 25 ATM cells.

5. CONTENTION RESOLUTION ALGORITHMS

5.1 Introduction

The concept of CRAs and their use in both the random assignment environment and hybrid demand assigned MACs was introduced in Chapter 2. In Section 2.7, considerations in the design of a Demand Assigned MAC were discussed and a proposal for a demand assigned MAC with central control was outlined. Vital to that proposal was the use of a CRA to regulate bandwidth reservation by the mobile terminals in the R-B Region of the hybrid TDMA (Section 2.7.5.2). In Chapter 6, new models are developed to study their performance under deadlock conditions – a key consideration in determining the survivability and hence real world efficiency of the MAC structure proposed. In this chapter, a number of CRAs are specified that will be further studied in Chapter 6. Efficiency statistics for each algorithm are reviewed.

5.2 General Uses and Definition of CRAs

In many wireless MAC protocols, random access techniques are used at some point of time (see Section 2.7.1 above). They are either used at during the initialisation (power up) of new stations, or during an “initial transmission” of a station (or in both cases).

The use of random access techniques is dictated by the presence of a large number of stations, most of which are expected to be silent at any single moment, and all of which have no means to co-ordinate directly with each other in their transmission attempts. A means must be dictated for choosing a time to speak and a means to resolve a situation where more than one station chooses to speak at one time – that means is the CRA. A collision occurs in a mini-slot (MS) of an upstream frame when two or more MTs transmit request messages in the same MS. Collisions are detected at the base station and status messages are transmitted back to the mobile terminals (MTs). A CRA is employed to resolve the collision.

A CRA can be defined as an algorithm (distributed in space and time) that organises the retransmission of colliding packets in such a way that every packet is eventually transmitted successfully with finite delay [MF85]. The theory of CRAs for medium access in communications networks has been maturing for well over a quarter of a century, and roots spread several decades into the past [Bis96a]. Capetankis first introduced detailed investigation of CRAs in 1977 followed by Tsybakov and Mikhailov who later, but independently, furthered the use of CRAs and advanced their analysis [MF85].

A number of CRAs have been suggested, studied, analysed and compared in open literature and are now widely deployed in commercial products. CRAs are of particular importance to demand assigned MAC protocols and this is the motivation for their study in this thesis. An adaptive MAC protocol dynamically adjusts the capacity of the contention channel according to contention slot allocator (CSA) algorithm. Different MAC protocols have been studied in the literature with different delay and frame structure. Since the overall system efficiency depends mainly on the slot structure and assignment, it is very important that the CRA does not act as a barrier or bottleneck to allocation of contention free data slots in a hybrid MAC by the CSA.

Studies of the dynamics of adaptive mechanisms of MAC protocols show that the performance of the overall system depends on the following main factors [SL98]:

- Number of minislots assigned to the contention channel and the frequency in time of appearance of the contention channel.
- Capability of the CRA to use CMS efficiently.
- Efficiency of the CRA.

This study investigates all of these factors.

5.3 CRAs – Definition of Specific CRAs and Analysis of Throughput

5.3.1 Slotted ALOHA Algorithm

The first data network to be based upon a random access protocol was the ALOHA network, which went into operation throughout the state of Hawaii in early 1970s [Abr70]. That system was optimised by the introduction of slotted time periods or Slotted ALOHA in 1972.

The algorithm for ALOHA is quite straightforward and the simplicity of the algorithm has ensured its continued use despite efficiency drawbacks. Each node simply transmits newly arriving requests in the immediate CMS following the arrival of the request, thus risking occasional collisions but achieving very low delay if collisions are rare. If a collision occurs, each transmitting node queues that collided request, then retransmitting it following the passing of a random number of slots. Each node can have at most one outstanding request at any given time.

5.3.1.1 Infinite Number of Stations Analysis

With an infinite node assumption, the number of new requests transmitted in a CMS is a Poisson random variable with arrival rate l. If the retransmissions from back-logged nodes are sufficiently random, the total number of new arrivals and retransmissions in a given CMS can be approximated by a Poisson random variable, X with attempted transmission rate $G > l$. Hence

$$X \xrightarrow{d} Pn(G) \; . \tag{23.}$$

Under this approximation, the probability of successful transmission, P_{succ}, in any given CMS is,

$$P_{succ} = P(X = 1) \; .$$

Hence,

$$P_{succ} = \left. \frac{e^{-G} G^{x}}{x!} \right|_{x=1} = G e^{-G} \tag{24.}$$

and Eq. (24) gives the channel throughput S of slotted ALOHA under Poisson arrivals [Pro98].

The main disadvantage of this conventional Slotted ALOHA algorithm is its instability. Indeed, the Slotted ALOHA algorithm is unstable for all arrival rates under the infinite node assumption. As the number of backlogged requests increases, the system experiences an ever larger load, increasing the number of collisions in the system and hence further decreasing the departure rate. With time, the departure rate will gradually drop to zero, after which no successful transmissions are possible within the system, a pattern of catastrophic behaviour that is independent of the arrival rate.

To construct a more precise model, assume that each back-logged request is retransmitted with a fixed probability p in each successive slot until it is transmitted successfully. Then, the number of slots from a collision until a given MT involved in the collision retransmits is a geometric random variable X with parameter p,

$$X \xrightarrow{d} G(p) \ . \tag{25.}$$

This allows slotted ALOHA to be described as a discrete-time Markov chain. If n is the number of backlogged requests at the beginning of a given CMS, then the probability that i of these are retransmitted in the forthcoming CMS, $P(i, n)$ is given by,

$$P(i,n) = \binom{n}{i} p^i (1-p)^{n-i} \ . \tag{26.}$$

The system contains of m terminals, so the number of non-backlogged MTs will be m–n and the probability that i of these transmit new requests in this particular slots, $Q(i,n)$ is given by,

$$Q(i,n) = \binom{m-n}{i} q^i (1-q)^{n-i} \ , \tag{27.}$$

where, q is the probability of transmitting a new request, which is given by,

$$q = 1 - P(X = 0) = 1 - e^{-\frac{l}{m}} \ . \tag{28.}$$

Thus, the state transition probabilities of going from state n to n + i is given by

$$P_{n,n+1} = \begin{cases} Q(i,n) & \text{if } 2 \le i \le (m\text{-}n) \\ Q(1,n)[1-P(0,n)] & \text{if } i = 1 \\ Q(1,n)P(0,n) + Q(0,n)[1-P(1,n)] & \text{if } i = 0 \\ Q(0,n)P(1,n) & \text{if } i = \text{-}1 \end{cases} \quad . \qquad (29.)$$

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Figure 31. Markov chain for slotted ALOHA.

This Markov chain is illustrated in Figure 31. The expected change in backlog over one CMS starting from state *n* is given by

$$D_n = (m-n)q - P_{succ} \ , \qquad (30.)$$

where, for small *q* and *p*, the attempted transmission rate can be approximated by

$$P_{succ} = Q(1,n)P(0,n) + Q(0,n)P(1,n) \approx G(n)e^{-G(n)} . \qquad (31.)$$

The attempted transmission rate *G*(*n*) is defined as

$$G(n) = (m-n)q + np \ . \qquad (32.)$$

If the no-buffering assumption is replaced with the infinite node assumption, the attempt rate *G*(*n*) becomes $l + np$, and the undesirable stable point disappears. Therefore, once the state of the system passes the unstable equilibrium, it tends to increase without bound. In this case, the corresponding infinite-state Markov chain has no steady state distribution and the expected backlog increases without bound as the system continues to run [Pro95].

5.3.1.2 Stabilising Slotted ALOHA

It is well known in the literature that:

- Under a No-Buffering assumption, the system discards a large number of arriving requests and has a very large but finite delay. The system has well defined steady state behaviour for all arrival rates.
- With a Infinite-Node assumption, no arriving requests are discarded. In this case, there is no steady state distribution and the expected delay grows without bound as the system continues to run.

Therefore, the objective is to achieve finite delay for a given arrival rate under the infinite node assumption, thus preventing the necessity for discarding requests to achieve stability.

It was shown above that the channel throughput of Slotted ALOHA under Poisson arrivals is given by $G(n)e^{-G(n)}$, which is maximised at $G(n)=1$. From this, it follows that if p can be dynamically adjusted to maintain an attempted transmission rate $G(n)$ of 1, the system will remain stable for all arrival rates less than or equal to $1/e$. There are many strategies for estimating the number of backlogged requests n, or alternatively, the appropriate retransmission probability p. One such strategy is the p Persistence Algorithm that is presented next.

5.3.2 *p* Persistence

5.3.2.1 Introduction

A NEC proposal for WATM operation uses S-ALOHA as the packet contention protocol for the R-B Control section of the frame. This section is used to provide bandwidth demand information from all the remote stations in the cell to the S-MAC situated at the base station. S-ALOHA has the advantage of operating with little or no central co-ordination, but often makes inefficient use of the transmission medium (maximum utilisation is 37%). Further, when too many terminals attempt to speak at once, throughput goes down and transmission delay increases rapidly. As foreshadowed, *p* persistence offers a form of solution.

5.3.2.2 The *p* Persistence Algorithm

The *p* persistence algorithm differs from the conventional Slotted ALOHA algorithm in that new arrivals are not distinguished from backlogged ones, but rather regarded as backlogged immediately on arrival. Unlike in Slotted ALOHA where they are transmitted with certainty in the next CMS, in *p* persistence they are transmitted independently of each other with the probability *p* in the same way as requests involved in previous collisions. Therefore, MTs apply *p* persistence on the first attempt as well as subsequent attempts.

If there are *n* backlogged requests (including new arrivals) at the beginning of a given CMS, the attempted transmission rate is $G(n)=np$, and the probability of a successful transmission is

$$P_{succ} = np(1-p)^{n-1} \ .$$

For unstabilised slotted ALOHA, this modification would not make much difference, since *p* has to be relatively small. However, for stabilised slotted ALOHA, *p* can be as large as 1 when the backlog is negligible, so that the new requests are held up only when the system is congested.

5.3.2.3 Ideal *p* Persistence Algorithm

Let the number of backlogged requests (note that these include new arrivals as well in this context) at the beginning of a CMS be *n*. Then the optimal usage of the channel is achieved if *p* is taken to be [Pro98]

$$p = \min\left(1, \frac{1}{n}\right).$$

The problem, however, is that it is not possible to know the exact value of *n* in real-time in a real system, meaning that an estimator must be implemented. There are several proposals for estimation techniques, and one representative example is given here – *Pseudo-Bayesian Estimation*.

5.3.2.4 Adaptive *p* Persistence Algorithm (Pseudo-Bayesian Estimator)

In 1985, Rivest proposed the pseudo-Bayesian estimation method for stabilising the *p* persistence algorithm [Riv85]. According to this method, the estimated number of backlogged requests, n_{k+1}, at the beginning of slot $k+1$ can be estimated using the following equation:

$$n_{k+1} \approx \begin{cases} \max\{l, n_k + l - 1\} & \text{if idle or success} \\ n_k + l + (e-2)^{-1} & \text{if collision} \end{cases} . \qquad (33.)$$

This algorithm has been shown to achieve a stable maximum throughput of e^{-1} under Poisson arrivals, but it requires that all MTs use the same retransmission probability, $p_k = 1/n_k$. Since the arrival rate l is not known, this too has to be estimated [Bis96a].

The system is stable up to a maximum throughput of 0.368 for Poisson traffic although maximum throughput for real traffic will vary according to the nature of the stream. After reaching the maximum stable throughput, the algorithm starts to collapse gradually due to the increasing number of collisions occurring within the system. Mean access delay for both traffic streams starts to rise at a very high rate, once the system pasts its point of maximum stable throughput. It is also important to note that real traffic stream suffers more delay compared with the Poisson traffic stream due to its fractal characteristics.

5.3.2.5 Implementation Complexity

Adaptive schemes require that two additional control signals be transmitted from the S-MAC to the MTs; the value of *p*, and an indicator to distinguish between contention and data mini-slots. With these parameters, the S-MAC (see Section 2.7.5.3 above) gains complete control over the MTs resulting in a highly flexible system. Under these circumstances, the MTs can send their requests only in certain designated mini-slots and at a rate specified by the S-MAC. Generalising slot specification such that different types of mini-slots (such as prioritised) can further extend this flexibility, immediate access mode and entry level mini-slots can be incorporated into the system [SLK97].

Three primary functions performed by the S-MAC: contention resolution; contention slot allocation; and scheduling, are largely independent of one another and suggest an architecture where very high-speed performance can be obtained by implementing each function in a separate module operating with minimum amount of coupling [SLK97].

5.3.3 Tree based CRAs

In order to overcome the instability of the conventional Slotted ALOHA algorithm, a set of more sophisticated contention resolution techniques known as Splitting Algorithms (including Tree or Stack) have been proposed. This set of CRAs requires MTs to take retransmission decisions collectively, based on continuous observation of the feedback from the S-MAC. As well as maintaining system stability without complex estimation procedures, this class of algorithms are capable of considerably increasing achievable throughput. Algorithms presented in Sections 5.3.1 and 5.3.2 will be contrasted in Chapter 6 with tree-based CRAs, and this section introduces the operation and performance of this form of CRA.

Many modern multi-service MAC protocols use a collision based capacity request-signalling channel as part of a hybrid but most propose a p persistence based approach. msSTART is proposed for use in IEEE 802.14 hybrid fibre/coaxial network and Section 6.5.3 of this thesis will study the performance of Q-ary tree CRA using a new Basic and Binomial Deadlock model. The effect of 'piggybacking' will be considered on the above research.

Tree CRAs differ from p persistent type algorithms in many ways. During the contention resolution phase of a splitting algorithm, each MT that transmitted a collided request splits itself into one of Q (usually Q is 2 or 3) subgroups by flipping a fair dice with Q sides. The MTs that fell into the i^{th} subgroup (where, i = 0, 1,…, Q-1) only retransmit after ensuring that all MTs in subgroups $j<i$ have successfully retransmitted their requests. As the process can be easily visualised through an Q-ary Tree or a Stack, the cases where Q = 2 and 3 are known as Binary and Ternary Tree algorithms respectively.

In general, the tree algorithms are classified into two main categories:

- Blocked tree algorithms; and
- Unblocked tree algorithms.

5.3.4 *Blocked Tree Algorithms*

5.3.4.1 Introduction

In a seminal paper, John Capetanakis introduced a new class of high speed, high throughput, stable multiple access algorithms known as tree algorithms [Cap79]. One type of that class of CRAs is the blocked tree algorithm.

In blocked tree algorithms, all new requests are queued and stopped from entering the system by the MTs while a contention resolution is in progress and there is a first transmission rule for allowing the blocked packets into the network. The quarantining of new arrivals during any CRA resolution means that the system adopts a batch process, characterised by the release in certain slots of a large number of contending requests.

A number of different first transmission rules have been suggested in the literature as the rule significantly affects the efficiency and properties of the resolution scheme [GSP96]. There are two major types of first transmission rules: obvious and non-obvious.

- The obvious first transmission rule states that all blocked MTs with new request arrivals during the current CRI can transmit immediately after the current CRI terminates.
- The non-obvious first transmission rule states a range of contention slots during which the blocked MTs can transmit, thereby spreading the transmissions apart to reduce collisions.

It is well known in the literature that the maximum throughput possible for the static binary tree algorithm under Poisson arrivals is 0.347, whereas for the optimal dynamic binary tree algorithm (the case in which the tree is binary everywhere except at the root node whose degree of branching depend on the estimated number of accumulated requests) throughput can be as high as 0.43 [Cap79].

5.3.4.2 Basic Blocked Tree Algorithms

The basic blocked tree algorithm is based on the observation that a contention among several active stations (i.e., stations with requests to transmit) is resolved if, and only if, all the sources are somehow subdivided into groups such that each group contains at most one active source.

The algorithm operates as follows: when a collision occurs in the i^{th} CMS, all MTs involved in the collision split into two subsets, represented by the simple test by flipping a fair *Q* sided coin for the case of a *Q-ary* tree.

```
MTs in a first subset retransmit in slot i+1, and if that
slot is idle or successful, the second subset transmits in
slot i+2.

However, if another collision occurred in slot i+1, MTs in
the first of the two subsets have to divide themselves into
two subsets again and retransmit as before.

The MTs in the second of the first two subsets have to wait
until the collision in the first subset has been completely
resolved before retransmitting.

All the new arrivals at MTs during a collision resolution
period are delayed from transmission while a CRI is in
progress.
```

The resulting tree structure is shown in Figure 32. We assume immediate binary feedback from the S-MAC or MT sensing (either an indication of successful transmission or contention).

Using the information available to all of the nodes from the S-MAC, each MT can construct this tree (or implement it as a stack) and keep track of its own subset in that tree. Therefore, each MT can determine when to transmit without system wide knowledge.

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Figure 32. Collision resolution phase.

The transmission order above corresponds to that of the stack that is created internally by each of the MTs. When a collision occurs, the subset involved is split and each resulting subset is pushed onto the stack. Then the subset at the head of the stack is pushed off the stack and transmitted. Viewing Figure 32 on a time axis, each slot marked on the diagram corresponds to an event in time from Slot time 1 to Slot time 8. The MTs colliding in Slot 1 are able to transmit requests in Slot 5, Slot 8 and Slot 9 respectively.

5.3.4.3 Non-obvious First Transmission Rule (Unblocking New Requests)

One improvement that can be incorporated into the basic binary tree algorithm arises from the possible estimation of new arrivals during a CRI. For small arrival rates, the division of new arrivals into two or three subsets then starting a new CRI may suffice. For larger arrival rates, the number of waiting requests may be so large that there will be a large number of colliding sets in the CRI until subsets get smaller towards the leaves of the trees.

The solution is to divide the set of MTs with new arrivals into *j* subsets initially after a CRP, where *j* is chosen such that the expected number of requests per slot is slightly greater than 1. The resulting algorithm is known as the dynamic blocked tree algorithm. Since the dynamic blocked tree uses a non-obvious first transmission rule it requires a mechanism to determine the optimal number of splitting, or in other words, the range, *R*, over which each station is allowed to transmit their new requests at the end of a CRI [Cap79].

This parameter *R* can be estimated as follows [GSPM96].

$$R(j+1) = \max\left[\min\left\{ R(j) - MS(j) + col(j) * \frac{e-1}{e-2} + \frac{MS(j)}{e} \right\}, MS(j+1) \right] \quad (34.)$$

where *n* is the number of stations, $MS(j)$ is the number of contention minislots in the j^{th} frame, and $col(j)$ is the number of collided minislots in the j^{th} frame.

5.3.4.4 Blocked Tree Algorithm with Collision Avoidance

Further extensions to the CRAs can bring additional gains in efficiency. Consider the situation given in Figure 33.

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Figure 33. Collision resolution phase - Avoiding guaranteed collisions.

Here, in slots 4 and 5, an idle slot follows a collision. This means that all the requests involved in the collision were assigned to the second subset, therefore guaranteeing a collision in the transmission of the second subset. Omitting the transmission of such sets, and splitting them beforehand can optimise the algorithm. The maximum throughput possible with this improvement is shown to be 0.46 requests/slot [Mas80].

5.3.5 Unblocked Tree Algorithm

5.3.5.1 Introduction

In unblocked tree algorithms, new requests can enter the system at any time regardless of whether a collision resolution is taking place or not. The collision resolution period in this case is defined as the time span from the slot where an initial collision occurs up to and including the slot from which all transmitters (or at least all those to which this relevant) recognise that all packets involved in the above initial collision have been successfully transmitted. It is worth noting that in tree-search algorithms, the splitting in subgroups can be done not only by flipping dice, but also by appropriately splitting the enabled interval. This (much harder to implement) version of the tree search algorithm can achieve absolute FIFO packet transmission ordering in contrast to the Quasi-FIFO service of the dice splitting tree search and Quasi-Last In First Out (LIFO) service of the RAAs. Nonetheless, ordering depends mainly on the policy used to scan the tree structure [Bis96a].

5.3.5.2 Basic Unblocked Tree

One disadvantage of the blocked tree algorithm that is solved by the Basic Unblocked Tree is that the blocked tree algorithm requires each MT to monitor the channel feedback in order to keep track of when each CRI ends. This is a disadvantage if the receivers at MTs are turned off in the absence of packets to transmit. One way to overcome this while preserving other features of the tree algorithm is to allow new arrivals to join the subset of requests at the head of the stack. In this way, only the currently backlogged MTs need to monitor the channel feedback, thus saving power and preventing unnecessary operation. Algorithms of this type are known as Unblocked Tree algorithms.

Unfortunately, this power advantage comes at a cost: the rate of collapse of the algorithm at high loads is much larger than the blocked tree. This is due to the possibility of new arrivals transmitted jointly with split subsets, thus increasing the chance of collisions. This is compensated, however, by a lack of necessity to estimate new transmissions during CRIs and also by the comparatively smaller delay during the stability region. Moreover, only binary feedback is needed for its operation.

5.3.5.3 The msSTART Algorithm

The Multi-slot *Q*-ary Stack Random Access algorithm is an evolution of the unblocked stack algorithm and is an example of a more advanced CRA under consideration for IEEE 802.14 use. It is an easy to implement, high-efficiency, free-access, robust, CRA [Bis96b]. For the implementation of the msSTART algorithm, each MT is required to maintain an msSTART counter for each request it want to transmit. This counter exists only while an MT has something to transmit (i.e., while the station is active), and has no use during the inactive period.

This CRA is often analysed as a pointer to a stack [Bis96a] and the msSTART counter update algorithm is given below. When an inactive station becomes active, initialise $COUNT = random\{0, \dots, M-1\}$ where M is the number of CMSs in the block following the activation of the station. Q is defined as before (in Section 5.3.3) as the number of subsets into which a colliding set may divide;

```
If (COUNT < M) then
```

```
Station transmits in CMS number COUNT;
Upon receiving feedback information:
       If (did not collide) then
Done;
       Elseif (collide)
Set: COUNT ← Q*col{M}+random{0, … , Q-1};

Elseif (COUNT > M–1) then
Station does not transmit in current block;
Upon receiving feedback information:
Set: COUNT ← Q*col{M}+COUNT–M;
```

Col{M} is given as the number of collisions during a block. New arrivals after the start of a block have to wait until the start of the next block before they can transmit. It is worthwhile to note that for high loads, it has been shown that the msSTART algorithm does not collapse as fast as the unblocked tree algorithm [Bis96]. This result follows from the temporary blocking of new arrivals until the start of the next block followed by random distribution into tree nodes. The algorithm is stable under sudden bursts in the traffic stream, and can clear backlogged requests in a very short time period.

5.4 Comparison – User perspective

Figure 34 and Figure 35 compare the extent of user delay (measured in CMSs) against the average request arrival rate. The CRAs are subjected to a real traffic trace and the chart demonstrates sensitivity to errors over the wireless link.

Efficiency statistics for each of the CRAs are provided by the simulation described in Chapter 6 and verified against the expected efficiency curves provided by the analysis in this chapter. Errors are added to the simulation results via a simple binomial slot error distribution. For simplicity it is assumed that no error correction is performed.

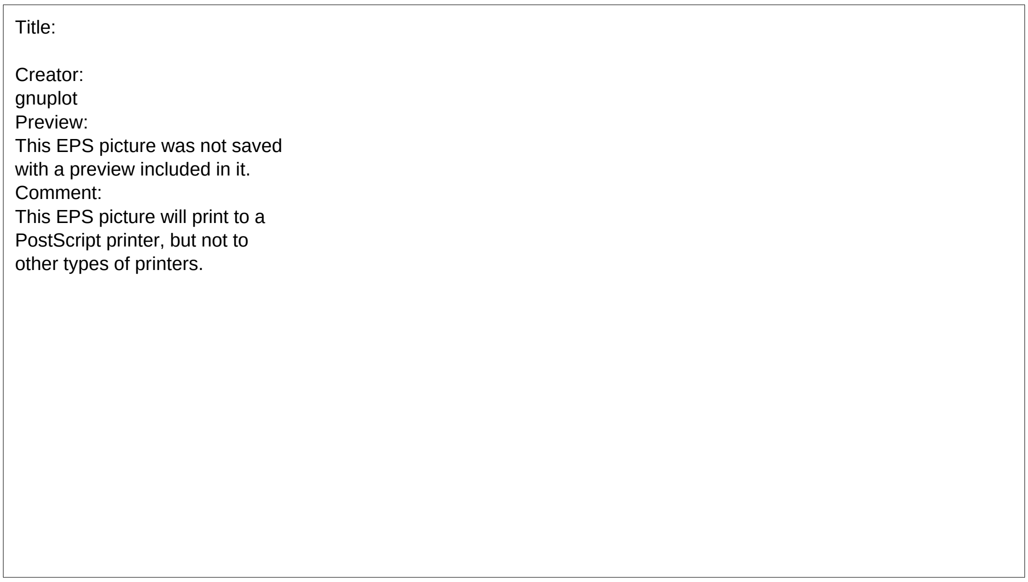

Figure 34. Relationship between channel throughput and mean arrival rate for a range of schemes with a selection of error rates.

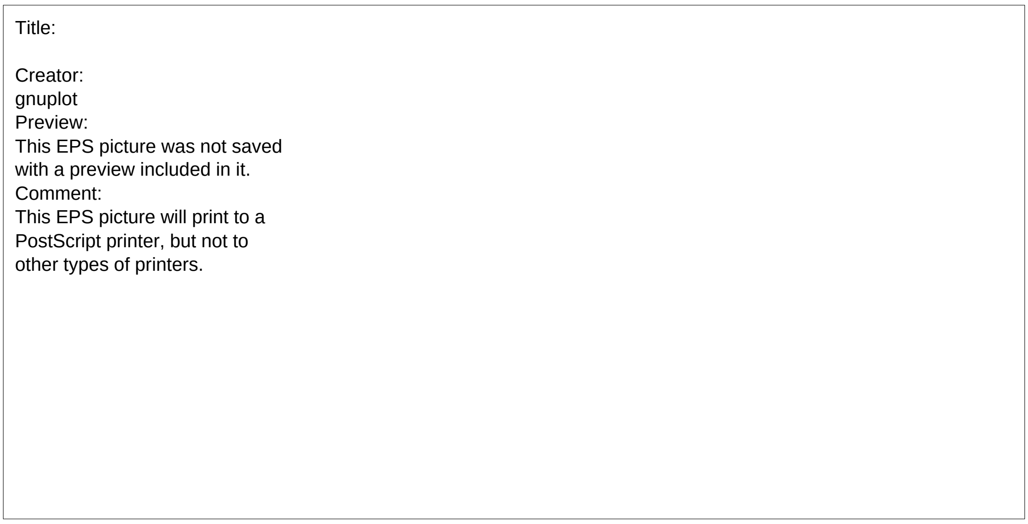

Figure 35. Relationship between channel throughput and mean arrival rate for a range of schemes with a selection of error rates.

Note that even unrealistically high slot error rates of 0.1 causes at worst a marginal drop in the achievable signalling throughput level and that the relative performance of the CRAs remains the same under error stress. Note also that, although Figure 34 and Figure 35 demonstrate mean access delay from a user perspective, our studies in Chapter 6 will focus on performance statistics from a system perspective.

5.5 *Summary*

This chapter serves as an introduction to the function and performance of a selection of CRAs. Describing both the tree and *p* persistence families of CRA, their importance to the hybrid MDR-TDMA MAC proposed in Chapter 2 is highlighted and established measures of their performance are presented in preparation for the disaster scenario investigation in the next Chapter.

6. Deadlock Models for a Multi-Service MAC Protocol: Comparing *p* Persistence and Tree Based CRAs

6.1 Introduction

As introduced in Chapters 2 and 5, many modern multi-service MACs use a collision based capacity request-signalling channel as part of a hybrid demand assigned TDMA frame structure. msSTART is proposed for use in IEEE 802.14 hybrid fibre/coaxial network and will be highly relevant for the S-MAC development of evolving WATM and other wireless access network MAC specifications. *p* persistence is proposed as the CRA of choice in the hybrid TDMA MACs of many current WATM testbeds.

This chapter investigates the performance of a number of CRAs in the wireless environment using a range of novel models that we will introduce in this thesis. The rationale for doing this is provided by the vital nature to system performance of the efficiency of the CRAs proposed for use in the MACs outlined in Section 2.7.5.3. Much of the chapter is contained in the author's joint [IZC00] and sole [CZI00] research publications. [IZC00] is represented in part in Sections 6.2 - 6.4 and incorporates some of [Iva97] which deals with wireline HFC research. [IZC00], on the other hand, is represented all of Sections 6.2 – 6.5.9 and follows the novel theme of this chapter – the contrasting of simulation and analytical results for msSTART performance, obtained by simulation under extreme inter-station correlation, with simulation results for the more popular *p* persistence CRA used in several testbed WATM implementations. Indeed, in this chapter, three signalling channel schemes designed to:

- Provide support for increased system stability;
- Implement priority in the wireless MAC; and
- Maximise efficiency

provide comparative results for evaluation of msSTART and *p* persistence ALOHA under what the IEEE 802.14 working group has termed the ‘disaster scenario’.

According to current developments, a wireless network access MAC is likely to rely on the broad principles of the (wired) IEEE 802.14 standard for HFC and use a dynamic TDMA/TDD protocol with a centralised control radio protocol for the wireless link. A S-MAC in such a scheme is operated at the base station or head end to receive reservations from the member stations in the shared area and allocate bandwidth dynamically. A C-MAC is used to multiplex and de-multiplex arriving and departing traffic. Since there is no centralised control between arrivals for transmission into the network at the MTs, nodes contending for access to the network must compete with each other to inform the HC of their need for a share of the bandwidth available to the population and distributed by the HC (see Chapter 2, especially Section 2.7.5 for further detail).

There are many possible ways for the MTs to transmit data to the HC. It is most efficient to ‘piggyback’ requests for future bandwidth on the end of previously scheduled data sent by the MT to the HC, however this cannot be accomplished in all situations, for example in cases where the MT is new to the cell, or has been idle for a period. In such situations, the most efficient means of allocation is for a request to be made during a R-B Control region in the demand assigned TDMA frame (see Figure 7). This reserves a slot in a later portion of the overall hybrid TDMA where there is no contention and the MT can transmit the data free of collisions.

Authors often refer to the contention period in which MTs reserve bandwidth as containing CMSs and the remainder of a frame as containing DSs. While CMSs are subject to collision, the DSs are collision free, each station in the population is informed of its transmission rights by the HC using a scheduling algorithm in the S-MAC.

Such a scheme requires the use of an efficient CRA in the R-B Control Region so that the MTs can make use of subsequent scheduled data slots. Deadlock in the CRA (a situation in which the throughput of request signals decreases dramatically despite increasing requests arriving) will mean that no (or very restricted amounts of) remote – base data can be transmitted in the system.

A sample frame structure is as follows: downlink information including control information and ATM cells are transmitted at the start of the TDMA frame containing the preamble, frame header, wireless network control signals, and data. The uplink subframe consists of the uplink control region for remote to base control information transmission, followed by allocated ABR, VBR, and CBR data slots in positions assigned by the S-MAC. Total overhead from frame headers and preamble could make the maximum utilisation of the channel before consideration of retransmission, modulation constellation and forward error correction as low as 75%.

The importance of the R-B Control region requires the use of a high performance, stable CRA for the resolution of transmission requests from the remote stations to the S-MAC. In determining the performance and hence suitability of any potential candidate for use in WATM, it is necessary to investigate its 'survivability'. This term, used in [CMT97], demonstrates the necessity of this analysis of the CRA using 'disaster' type rather than Poisson traffic conditions to measure useful performance.

Further rationale for this research is provided by the requirement that the WATM MAC cope with the ever increasing variability of traffic arrival characteristics of different classes of connection, and that the MAC be robust enough to recover from failure or other limiting effects of the wireless link. Hence, a central issue in the consideration of candidates for any multi-service medium access protocol for a wireless network is the effect of atmospheric and other interference effects on the error prone link. A further issue, related to both wired and wireless networks, is the transient or nonstationary congestion period triggered by the backlog of packets due to retransmission after a failure or sustained period of low transmission capability.

Indeed, the findings of Cotter, Medhi and Tipper [CMT97] of the impact of designing for survivability in wide area VP-based ATM networks is relevant to the study of wireless access MAC contenders under transient traffic patterns. Their analysis of virtual-path level survivability addresses questions of network design by introducing a transient time threshold to clear backlog as one of the QoS parameters. They find that significant additional network allocation is required to guarantee manageable transient threshold times after network failure and propose an analytic framework for the determination of additional network dimensioning required for network survivability under transient traffic patterns following failure.

The focus in this chapter is similar – but candidate CRAs for a generic wireless access MAC will be examined with a view to minimising the effect of such transient traffic patterns, rather than providing excess provisioning guidelines. The study of MAC performance during periods of extreme load and traffic correlation is concentrated on, as numbers of remote stations attempt to retransmit after the failure of the wireless link. Hence the 'survivability' of the contention resolution region of the MAC protocol is investigated. In particular, the performance of simple *p* persistence CRAs is compared with the more complex *Q*-ary tree variant msSTART in an attempt to minimise additional network provisioning required for survivability. It is clear that the performance of the CRA is a key indicator of overall MAC performance since in many situations reservation of a future DS by a MT is the only method of R-B transmission (see Section 6.5.5).

The remainder of the chapter is organised as follows: in Section 6.2 the basic analytical deadlock model for extreme traffic is defined and analysed, taking into account upstream channel errors. Key assumptions are stated. In addition, a "background traffic" model (which represents an enhancement of the basic deadlock model) is defined and analysed. In Section 6.3 several alternative CMS signalling capacity management schemes are proposed for use within the framework of each of our models. In Section 6.4 results and conclusions of the *p* persistence analysis are presented, providing a comprehensive set of numerical results for both deadlock models, and exploring the performance of each of the signalling capacity management schemes. Section 6.5 treats tree based CRAs to the same analysis and investigates the use of piggybacking on the MAC structure via the introduction and analysis of a further deadlock model incorporating background traffic.

6.2 Deadlock Models - Definition & Analysis

Under the Slotted ALOHA approach, the worst-case traffic situation is obtained when the inter-station correlation of the CMS traffic is maximised. That is, when all stations in the system generate simultaneous signalling request “storms”. The deadlock models defined in this section incorporate a Markov-chain analysis of such a situation. The reason to study extreme scenarios arises from the desire to be able to provide design guidelines for a multi-service MAC protocol, which helps to avoid signalling congestion collapse in almost all conceivable cases.

6.2.1 Assumptions and Relationship to Previous Models

In analyses of the classical Slotted ALOHA multi-access approach (e.g. [BG92]) it has often been stated that the assumption of arrivals modelled by a Poisson process is unrealistic for real traffic and the main reason that such traffic is used is to enable analytical tractability. Real systems will definitely be prone to errors due to noise, making the perfect reception assumption in classical ALOHA analysis quite idealised. In this section a more realistic Slotted ALOHA-based model is devised which takes into account channel errors and extremely correlated inter-station traffic. In particular, the concern is to model worst-case traffic conditions for wireless systems’ signalling channels capable of leading to deadlock.

Channel errors aside, the principal difference between deadlock models presented in this section and the Slotted ALOHA analysis presented in [BG92], is that Poisson arrivals are not assumed. Table 6.1 summarises commonality between assumption sets of the two models.

Classical Slotted ALOHA Assumption	Deadlock Model
Slotted System	✓
Poisson Arrivals	✗ (use simultaneous arrivals).
Collision or Perfect Reception	✗ (allow for errors).
Immediate Feedback	✓
Retransmission of Collisions	✓
No Buffering	✓
Infinite Set of Nodes	✗

Table 6.1. Similarities Between Classical Slotted ALOHA and Deadlock Model.

Instead of using Poisson arrivals, extreme inter-station correlation is modelled through arrivals of batches of capacity requests. In Section 6.2.3 the size of these batches represent a subset of the total station population and in Section 6.2.2 the batches represent the entirety of the offered traffic. In all the deadlock models proposed, the station population is finite and so the notoriously unstable nature of the Slotted ALOHA family of multiaccess algorithms is not a factor by itself (since Slotted ALOHA is theoretically stable for a finite population). The concepts of theoretical and practical stability are worlds apart and this paradigm forms a major part of this investigation into deadlock models.

The inter-station correlation described is the “disaster scenario” of the IEEE 802.14 committee [SHL96a] [Bis96], where it is assumed that the totality of stations (say, 200) are all simultaneously powered-up (for example after a power failure) and transmit a request signal in the same slot. In particular, the Basic Deadlock model (Section 6.2.2) is an enhancement of the model investigated in [SHL96a], including a provision for signalling channel errors. However, the focus here is different, considering the operation of the normal signalling channel under “extreme stress”, manifested by inter-station correlated traffic conditions rather than solely targeting the initial station registration process after a simultaneous power-up event. With this in mind, Section 6.2.3 proposes a further deadlock model based on the Basic Deadlock Model that goes on to extend the analysis presented in this chapter to cases where, aside from the arrivals of large capacity request batches, the Slotted ALOHA signalling channel also experiences so-called *background traffic* noise. In a final and very significant model, the case of the Basic Deadlock model with background ‘piggybacked’ traffic is also considered.

Both of the models allowing background traffic have physical significance because they can capture scenarios where one set of the access network stations are still operating normally (and hence providing background traffic), while other stations have failed, powered-up, re-registered, and are simultaneously wanting first-time access to the signalling channel. This has been known to happen in some neighbourhoods where power outages take down only part of the network. When the affected part comes back online, a scenario such as the one described may occur.

Note that the particular attention we pay to modelling and exploring deadlock is warranted because the reservation feature of most wireless MACs (which rely on the use of collision free DMSs) will ensure relatively “uneventful” and efficient operation of both the signalling and data channel under most normal conditions. The bottleneck, and point of interest, then becomes the set of scenarios where it is impossible to use this reservation feature and the entire station population burdens the signalling channel directly.

6.2.2 *Basic Deadlock Model*

Let N be the number of active stations, meaning that these N stations request capacity, each for a single-cell message, all at the same time – at the beginning of the CRI. This creates the worst-case scenario from a signalling collision point of view. All stations attempt to write into one CMS simultaneously, so that a batch of N capacity request signals arrives at once for contention resolution. The concept of *batch size* is henceforth used to describe the number of simultaneously arriving requests. The aim is to derive the mean of the CRI duration, measured in the number of CMSs, f. This mean is designated as $E[f]=T_C(Scheme, N, p, P_{err})$, or T_C for short, where *Scheme* identifies the signalling capacity allocation scheme used for that wireless system (see Section 6.3), and P_{err} quantifies the probability of a CMS minislot being erroneous.

As stated earlier, in modelling the signalling channel, it is assumed that time is divided into fixed-length intervals. In order to justify this assumption in the context of a WATM or other wireless access based MAC protocol, note that each of these fixed-length intervals represents one CMS minislot. In practice, depending on the number of CMS minislots associated with each upstream data slot, there may or may not be variable length gaps between consecutive CMS minislots. This means that, by using the notion of the number of CMS minislots elapsed, time is being measured non-linearly. However, due to the cyclic nature (with a small cycle) of this non-linear relationship in a practical MAC protocol, we can assume linearity. In other words, if T_C is measured over a large number of CMSs, and if the CMS cycle is short, then T_C gives the mean inter-CMS time, which will be very close to the CRI measured in linear time units (e.g., seconds).

Any given CMS has the possibility of suffering a random error, with probability P_{err}. Whether one or multiple errors hit a given CMS minislot, the effect will be the same meaning only single errors need be considered. Since it is assumed that there is an independent, constant probability of an error during each CMS, the error process is one with geometrically distributed inter-occurrence times (parameter P_{err}).

The variable t represents the number of elapsed CMS minislots from the arrival of the batch of size N. At time t=0, a batch of size N arrives. Let $P(j, t)$ be the probability of having j contending requests at the end of time interval t (t = 1, 2, ... and j=1, 2, …, N). $P(0, t)$ is defined as the probability of having no contending requests at time t, for the first time. In other words, $P(0, t)$ is the probability that T_C=t. Since all N stations try to access a CMS during the first time interval, and they all collide, there remain N outstanding requests at the end of the first interval. Hence, $P(N, 1)$=1 and $P(j, 1)$=0 for all j=0, 1, 2, …, N-1.

After the first time interval, all N stations try to access with probability p, but there is also the potential for an error to happen in any given timeslot with probability P_{err}. The probability of a successful transmission is therefore given by $(1-P_{err})\cdot Np(1-p)^{N-1}$. In general, when there are *j* stations contending for transmission, the probability of successful transmission (i.e. a reduction of one in the number of requests "waiting" for resolution) will be given by

$$P_S(j)=(1-P_{err})\cdot jp(1-p)^{j-1}. \qquad (35.)$$

On the other hand, the failure outcome when *j* stations are contending is defined as the event when the number of outstanding requests does not decrease by one. The probability of failure is given by

$$P_F(j)=(1-jp(1-p)^{j-1})+P_{err}\cdot jp(1-p)^{j-1}=1-P_s(j). \qquad (36.)$$

The case $j = N$ is an upper bound on the system occupancy and hence a special boundary condition exists: $P(N, 2)= P_F(N)$, $P(N, 3)= P_F(N)^2$, $P(N, 4)= P_F(N)^3$, and in general, $P(N, t) = P_F(N)^{t-1}$. The general case arises when j=1,2, …, N-1, and the state *j* could have been entered from a higher state, or it remains unchanged

$$P(j,t)=P(j+1,t-1)\cdot P_S(j+1)+P(j,t-1)\cdot P_F(j). \qquad (37.)$$

Finally, the probability of zero outstanding requests at time *t* is given by

$$P(0,t)=p\cdot P(1,t-1)\cdot(1-P_{err}). \qquad (38.)$$

The model described has a bounded state-space, and for any given state *j*, the probability of an increase in state is zero. It is therefore trivial to show that the absorption probability of such a system must be unity. This intuitive result suggests that, as expected, the mean CRI duration T_C will always be finite, regardless of the choice of model parameters. Employing the “summation of steps” technique from [KT75], the analytical expression for the mean CRI duration is obtained. Given an initial state $j=N$,

$$T_C = \sum_{j=1}^{N} \frac{1}{P_S(j)} . \tag{39.}$$

The model described is a pure death process, with no possibility of increase from any state j. This explains the very simple form of Eq. (37) – it is merely a sum of the average sojourn times in each of the states the system descends down through, from $j=N$ to $j=1$ [IZC00], [CZI00], [Iva97].

A slightly more useful method of obtaining T_C is numerical recursion: a numerical recursive solution of the set of Eqs (33) – (36) will yield the probabilities $P(0,t)$, $(t=1,2,3,\ldots)$. After obtaining the $P(0,t)$ probabilities, and recalling that T_C is the mean time to absorption in the state $j=0$, by definition, T_C can be obtained as follows:

$$T_C = E[f] = \sum_{t=1}^{\infty} t \cdot P(0,t) . \tag{40.}$$

Note that in all the numerical solutions, the recursion termination condition: $t \cdot \mathrm{P}(0,t) < 10^{-9}$ is used. The method of numerical recursive solution of the model’s state transition equations is extremely useful because it yields the series of exact system occupancy probability distributions, from timeslots 1 through to ∞ (in theory). This then allows other important statistics to be calculated, such as the average background traffic offered during the mean CRI (important in the more complex model considered in Section 6.2.3), and the entire discrete probability density function for f. However, if the sole interest were to calculate the mean CRI length, T_C, adopting an analytical solution approach would have been simpler.

Having determined T_C, the maximum achievable throughput of the Basic Deadlock model is now known: it is merely the ratio of N request messages to the T_C slots it takes to clear them from the contention state. The ratio is defined as L_{crit}, signifying *critical load*. This load is considered critical because; if the N requests arrive with a period less than T_C, the arrival rate exceeds the system's service rate and the system becomes 'practically unstable'. In this situation, the number of outstanding requests increases towards infinity. Note the term 'theoretically unstable' is not used since it has already been shown analytically that T_C is always finite. In this sense 'practical instability' signifies a scenario where the uncontrolled increase in the number of backlogged requests causes T_C, although theoretically finite, to become so large that it tends to infinity for practical purposes. In situations where L_{crit} is very small, it may be desired to increase it by some other means. This is where signalling capacity allocation schemes play a role, describing how to manage the CMS minislots. Three schemes are proposed in Section 6.3.

6.2.3 A Deadlock Model with Background Traffic – BIN

This section turns to another important model that, in addition to erroneous CMSs, takes into account the possibility of new request arrivals during the contention resolution phase of an initial batch of N outstanding message requests. It is important for a model to capture the complete set of circumstances that may be part of a deadlock situation. In the model that is presented throughout this and the following sections, both the impact of collided retry requests using the *p* persistence CRA, and the effect of new request arrivals on stability and average length of the CRI are captured.

The base model considered has Binomial (BIN) background traffic arrival processes, and thence derives its name. The description of the base properties of the model and the notation used, are the same as those in Section 6.2.2. The main difference is that now there is a possibility of multiple additional arrival(s) during the current CMS minislot, with a probability P_{arr} (in addition to any given CMS having the possibility of suffering a random error, with probability P_{err}). The new arrival probability P_{arr} is state-dependent. In this section, the background traffic arrival process for the model is described first, progressing to a tabulation of the analytically derived expressions for the model's key metrics (e.g. state increase/decrease probabilities, mean CRI length, critical load).

The following scenario describes the BIN model. Consider the case of a limited set of stations, L. Initially, N out of the L stations are subject to failure; they are powered-up simultaneously generating simultaneous CMS requests. The remaining L-N stations are generating background traffic, each with a fixed probability. The current system state j represents the number of active stations with one outstanding request each. This system state increases with collisions of the additional background CMS requests and decreases with successful CMS request transmission. Hence a state-dependent rather than constant probability of an additional single arrival is present during a timeslot, denoted by $P_{arr}(x, j)$. The existence of only two states in which the stations could be in is assumed: *Active* and *Inactive* in terms of generation of CMS requests. In other words, a scenario of short messages and no use of piggybacked slots is considered. The Binomial model is so named because, during a unit timeslot, the probability distribution of the number of newly arriving message requests is Binomial. The limit for j is L, determined by total station population L, and the expression for $P_{arr}(x, j)$ is thus

$$P_{arr}(x,\, j) = \begin{cases} \binom{L-j}{x} \cdot \left(\frac{l}{L}\right)^{x} \cdot \left(1 - \left(\frac{l}{L}\right)\right)^{L-j-x}, & j = 0,1,2,..(L-1);\ x = 0,1,...,L-j \\ 0, & j = L \end{cases} \quad (41.)$$

where l/L represents the probability of a single station changing from the Inactive to the Active State during a single timeslot, assumed independent of any other event. As more stations migrate to the Active State, the subsequent reduction in the number of potential request generators causes a linearly proportional drop in the mean request arrival rate. This type of parameter matching suggests that

$$E[arr, t\,]^{\mathrm{BIN\_model}} = \sum_{j=1}^{L} p(\,j, t\,) \cdot E[arr,\, j\,]^{\mathrm{BIN\_model}} \quad < \quad l\ , \quad (42.)$$

with $E[arr,\, j]^{\mathrm{BIN\_model}}$ averaged at some arbitrary timeslot $t=t$, over an arbitrarily occupied transient state space $[1,L]$. The relationship $E[arr, t\,]^{\mathrm{BIN\_model}} < l$ will clearly hold, due to the fact that for all $j > 0$,

$$E[arr,\, j\,]^{\mathrm{BIN\_model}} = (L - j) \cdot \left(\frac{l}{L}\right) < l\ . \quad (43.)$$

6.2.3.1 Derived Expressions

The following table summarises important expressions for the BIN model derived by discrete-time Markov chain analysis.

Finite State Space	✓
Single or Multiple Arrivals per Timeslot	Multiple (Binomially Distributed)
Probability of State Increase in State *j*	$P_{Inc}(x,j)=\begin{cases}P_{arr}(x,j)\cdot\begin{pmatrix}(1-(1-p)^{j})\\+P_{err}\cdot(1-p)^{j}\end{pmatrix} & x=1\\ P_{arr}(x,j), & x>1\end{cases}$
Probability of State Decrease in State *j*	$P_S(j)=(1-P_{err})\cdot(1-P_{arr}(x,j))\cdot jp(1-p)^{j-1}$
Theoretical Probability of Absorption from initial State *N*	1
Practical Probability of Absorption from initial State *N*	$PPA\vert_N=\sum_{t=1}^{T_{pr}}P(0,t)\vert_{P(N,1)=1}$
Analytical Mean CRI Length, T_C from Initial State *N*	$w_N=\sum_{i=1}^{N}x_{i,i-1}$ with $x_{i,i-1}=\left(\frac{1}{P_S(i)}\right)+\sum_{x=1}^{L-i}\left(\frac{P_{Inc}(x,i)}{P_S(i)}\cdot\sum_{k=1}^{x}x_{i+k,i+k-1}\right)$
Numerical (*practical*) Mean CRI Length, T_C	$\sum_{t=1}^{T_{pr}}t\cdot P(0,t)$
Mean number of background arrivals during T_C	$E[arr\_during\_T_C]^{\mathrm{BIN\_model}}=T_C\cdot l$
Critical Load (*Maximum Practically Stable Throughput*)	$L_{crit}=\frac{N+E[arr\_during\_T_C]^{BIN\_model}}{T_C}$

Table 6.2. Derived Analytical Expressions for the BIN Background Traffic Deadlock Model.

Key concepts of theoretical and practical stability emerge from Table 6.2. Recall that a classical discrete-time birth-death process with an absorbing state and a finite state space is always theoretically stable

[KT75] and regardless of its initial state, is guaranteed to be ultimately "pulled into" the absorbing state. That is, the probability of absorption (PA) is equal to one. Hence, the BIN model is referred to as 'theoretically stable'. On the other hand, if absorption will not, on average, take place for an exceedingly long time, which can for practical purposes be considered infinity, the system is labelled as 'practically unstable'. This paradigm tends to be more useful to us, since the background traffic model can be assigned a Practical Probability of Absorption (PPA) value.

The PPA directly quantifies the probability that, for a given deadlock model and set of system parameters, absorption will take place, on average, by some reasonable time, as shown by the appropriate equations in Table 6.2. From its definition in the table, we see that PPA is the probability that absorption into state 0 has already taken place by some very large value of $t=T_{pr}$, which would be equivalent to infinity for the practical purposes of wireless contention resolution algorithms.

An example is the value of T_{pr}=12,500,000 timeslots, which represents roughly 10 minutes of real-time in an implementation of the F-CPR protocol for HFC access networks, with 60-byte upstream slots and an upstream channel speed of 10 Mbit/s. Any situation where all outstanding messages cannot be cleared by T_{pr} (i.e. absorption into state 0 does not take place by such time) would undoubtedly be considered a catastrophic system deadlock for any contention resolution algorithm. A system considered practically stable has

$$PPA \geq 1-e , \qquad (44.)$$

where e is the threshold of our numerical solutions, and is taken throughout this chapter for convenience as 10^{-9}.

6.3 Signalling Channel Schemes

The maximum (practical) stable signalling throughput considered so far, L_{crit}, relied solely on the concept of all stations contending in a single collision minislot. Three different schemes to manage and access CMS signalling slots are now considered. Let M be the number of CMSs associated with each upstream data slot (ATM cell). The first scheme, where all stations may access any of the M slots, is termed Full CMS sharing with multiple CMSs per data slot (FCS), with a critical load given by,

$$L_{crit} = \frac{MN}{T_c\left(FCS, N, p, P_{err}\right)} \quad . \tag{45.}$$

The critical load, L_{crit}, is obtained as the ratio of the amount of work arriving in a batch, namely N, and the average time in *upstream super-slots or frames* (defined as super-slots containing one data slot and the M associated CMSs shown in Figure 7) during which the contention resolution of these N requests is completed, which is equal to T_C/M upstream slots. In considering all of these CMS management and allocation schemes, an underlying extreme-case model of a batch of N single cell messages arriving periodically and generating N simultaneous requests is assumed.

Under the second scheme, the N stations are subdivided into M groups, so that each group must access a different CMS associated with the data slot. In this fashion, the effective load accessing any given

CMS is reduced to N/M. Implementation-wise, the station counts passing CMSs and is allowed to access every M^{th} CMS, which occurs once per data slot since we have M CMSs for every one data slot. This scheme is best termed Cyclic CMS sharing with multiple CMSs per data slot (CCS_M), with a critical load given by

$$L_{crit} = \frac{N}{T_c\left(CCS\_M, \frac{N}{M}, p, P_{err}\right)} \quad . \tag{46.}$$

Lastly, the third scheme, similar to the multi-CMS Cyclic sharing, is its *single-CMS* variant (and is termed CCS_S). Using only one CMS per data slot, if we divide the N stations into k groups, whereby each group may only access every k^{th} CMS, the critical load is given by

$$L_{crit} = \frac{N}{kT_c\left(CCS\_S, \frac{N}{k}, p, P_{err}\right)} \quad . \tag{47.}$$

Recalling the definition of the average CRI duration, T_C, there are four common factors which affect T_C and L_{crit}:

- The CMS scheme used and if appropriate, the number of CMSs per timeslot;
- The batch size, N;
- The p persistence probability p; and
- The probability of CMS error, P_{err}.

Note that if single-CMS Cyclic Sharing is used, there is a fifth factor – the number of separate contention resolution groups, k. Both variants of Cyclic CMS sharing, (CCS_M and CCS_S), may in some sense be viewed as the p persistence algorithm's non-adaptive version of the collision-slot grouping concept from *START-n* [BMN96].

6.4 Results and Implications – p Persistence

In this section comprehensive analytical results that give insight into performance and dimensioning issues that support effective operation of the MAC protocol are provided. Section 6.4.1 considers the impact of various system parameters on the signalling channel performance for the Basic Deadlock model. Afterwards, additional techniques for alleviating deadlock problems are explored. Section 6.4.2 then gives more insight into possible deadlock scenarios by providing a numerical analysis of the BIN model.

6.4.1 The Basic Deadlock Model and Signalling Capacity Allocation Schemes

Numerical results are presented here for the Basic Deadlock model derived in Section 6.2.2, and for the three proposed signalling allocation schemes from Section 6.3. The aim is to study the effect on F-CPR protocol resiliency, of:

- The parameter p;
- The number of CMSs per timeslot for the basic FCS scheme;
- The probability of CMS errors, P_{err}; and
- The number of separate contention resolution groups, k, under the CCS_S scheme.

In addition, the FCS and CCS_M schemes are compared to determine whether full or partial resource sharing is more efficient, given that the number of CMS minislots is kept identical.

The section begins by observing the behaviour of the critical load as a function of the number of stations for different values of p, without considering the presence of signalling channel errors (i.e. P_{err}=0). The analysis is initially unconcerned with the particular signalling capacity allocation scheme being implemented, so the simplest and default scheme is used: the Full CMS sharing scheme, with only one CMS minislot (FCS with M=1). In the ensuing figures and discussion, note that unless there is a specific focus on the performance of a named signalling scheme, the FCS scheme is assumed as a default.

In Figure 36 we show the expected performance of the *p* persistence algorithm for a selection of *p* values from the analysis in Section 6.2.2. We next show results for the same scenario in Figure 37 that are the output of a simulation of the described system created by the author using C++. Simulation results confirm the analysis and small variations in the plot of Figure 37 are the result of restricted simulation run times.

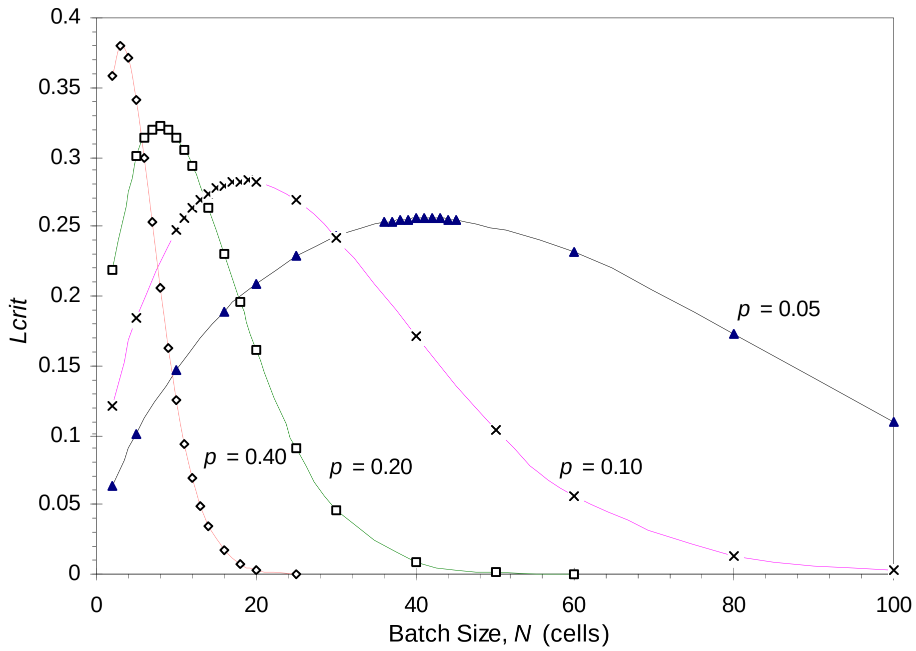

Figure 36. Finding an Optimum Batch Size for a given *p*, FCS (M=1) Scheme. This Figure shows the expected results from the analysis in Section 6.2.2 and are to be compared with results in Figure 37 from the C++ simulation of the system.

As shown in Figure 36 and Figure 37, aggressive *p* persistence (large values of the parameter *p*) allows a greater critical load (L_{crit}) when the batch size (the number of simultaneously and periodically contending stations) is small. On the other hand, a low level of p (e.g. p=0.05) provides reasonable protection against deadlock (notice that 50 contending stations transmitting small messages can achieve utilisation of 0.25 without deadlock), but achieves quite a low L_{crit} for a small number of contending stations. Nevertheless, in a scenario where a small number of simultaneously contending stations are heavily loaded, a pre-reserved DMS is more likely to be used and higher overall signalling throughput can be achieved, due to the reduced pressure on the contention-based signalling channel. This observation leads to the 'piggybacking' model introduced later in this chapter.

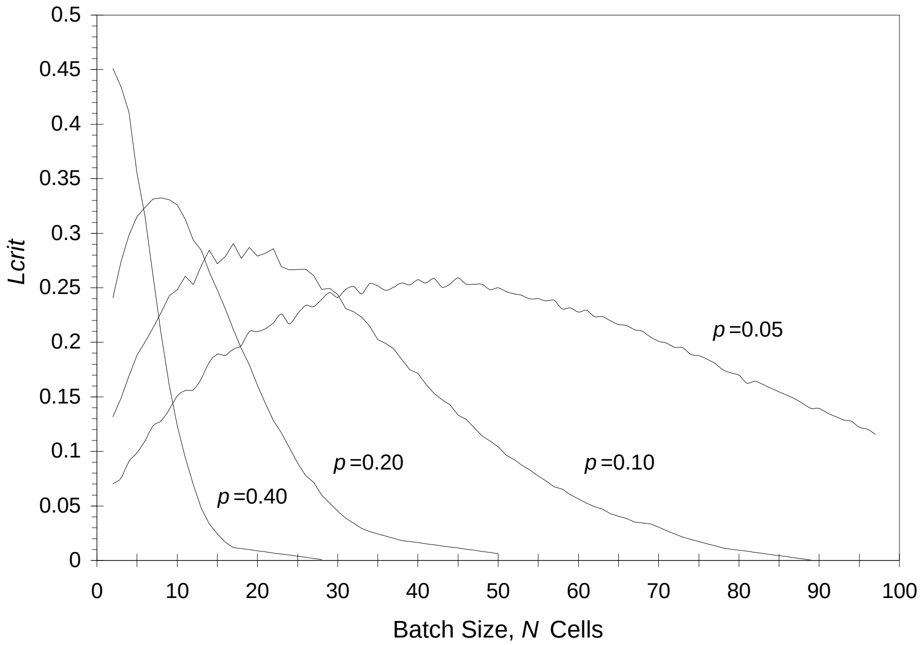

Figure 37. Finding an Optimum Batch Size for a given *p*, FCS (*M*=1) Scheme. Analytic results in Figure 36 are compared the results in this figure. Simulation verification of the analysis in this figure (using C++ to model the system) is based on a 95% confidence interval of ±10%. All results are within confidence intervals.

These results suggest that where it is impossible to vary p in real-time wireless access system operation, a relatively high level of p should be used, given the presence of the pre-reserved DMS contention-free minislots. The issue of stability is an interesting one in this setting. Recall that if the assumptions of this model were slightly augmented, to apply to an infinite population generating Poisson arrivals, instability would result for any non-zero arrival rate. However, the arrivals are not Poisson - they are periodic batches of N requests. Also, there is no infinite pool of stations - rather, just N stations. Hence, a system is created where the mean CRI length is theoretically finite but can, under certain conditions, be so large that it is for all practical purposes infinite. A practically infinite T_C leads to a near-zero throughput level. From Figure 36 and Figure 37 we see that this undesirable scenario can happen when the fixed value of p is far from its optimal value: if p is too low, a small batch will take unnecessarily long to be cleared; if p is too high, a large batch of messages will result in repeated collisions that maintain the backlog at a high level for a long time.

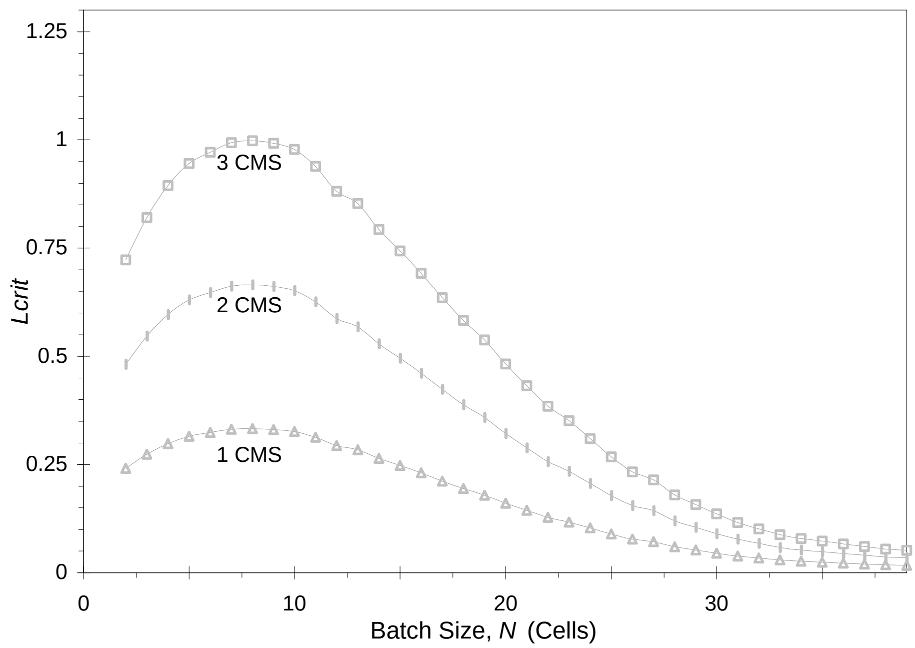

Figure 38. Effect of Number of CMSs – FCS (M=1,2 and 3) Scheme: p=0.20. Data points are obtained from simulation of p persistence under the Basic Deadlock Model using C++.

In Figure 38 the effect of the number of CMSs per data slot for the Full CMS sharing scheme is demonstrated (once again ignoring the probability of CMS error). The important message of Figure 38 is that increasing the number of CMSs does not provide the desired protection against deadlock for the case of a large number of contending stations.

In any case, the cost of increasing the number of CMSs is significant. Given the various protocol overheads for the wireless MAC proposed in Chapter 2 of the thesis, we estimate that increasing the number of CMSs from 1 to 3 would add about another 10% of signalling overhead and decrease the "actual user data" throughput capability of the system, although the exact numbers may be different depending on frame format. Although this method would triple the critical load (L_{crit}) for any number of stations and *may* provide efficient operation and protection against deadlock in some instances, those instances are limited to relatively small numbers of contending stations, as seen in Figure 38. When the number of contending stations is large, and the critical load approaches zero, these results demonstrate that tripling the critical load is not beneficial.

In Figure 39 (with P_{err}=0 once again), it is demonstrated that the parameter p not only has a significant effect on the critical load (and on the resiliency of the protocol), but that it can be optimised for maximum load, for each combination of other parameters. A well-known fact is that the optimal value of the p persistence algorithm is $1/N$ if the system currently has N contending stations [Riv85]. As explained in [SHL96a], the optimal p for the contention resolution of N initial backlogged requests would be $1/N < p_{opt} < 1$, since, during the contention resolution process, the system spends some amount of time in each of the states $\{N, N\text{-}1, N\text{-}2 \ldots 1\}$ prior to absorption into state zero. This inequality for p_{opt} is clearly illustrated in Figure 39. A different way of looking at the inequality is provided in Figure 37: for a given fixed value of p, the optimal N is always slightly larger than $1/p$.

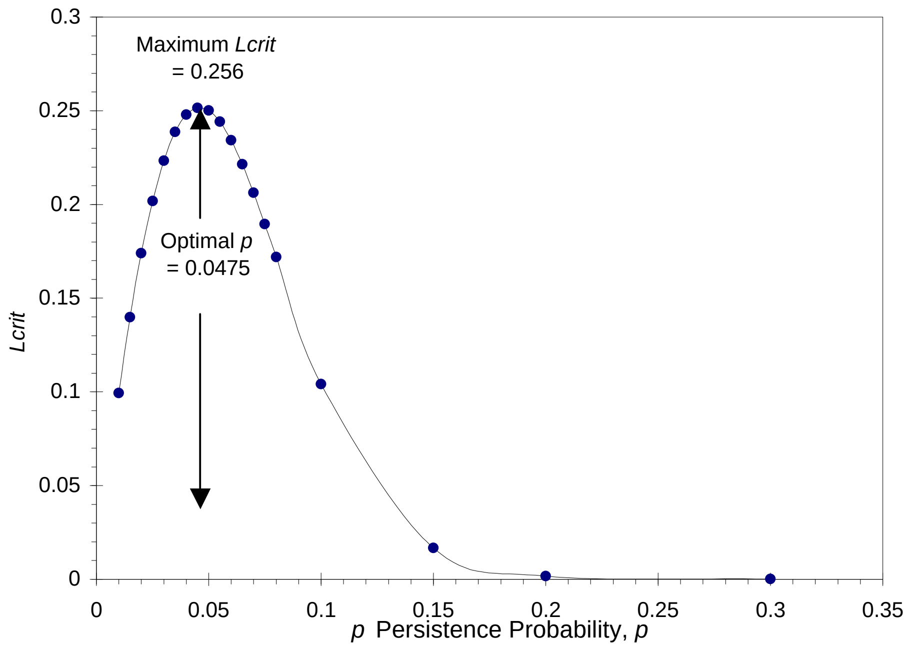

Figure 39. Optimising the *p* persistence Algorithm – FCS (*M*=1) Scheme. Batch size, *N*, is set at 50 and there is 1 CMS. Data points in this Figure are obtained via a numerical recursive solution to Eq. (38).

Turning now to look at the effect of errors on the signalling channel, the plots in Figure 40 highlight the fact that the critical signalling load is largely unaffected by the presence of CMS errors. At CMS error levels of 10^{-2} or less, the critical load versus batch size behaviour is identical to that of error-free systems, as can be seen from the Figure. It requires an unrealistically high CMS error level of 0.1 to cause a significant (≈10%) decrease in the achievable signalling throughput level.

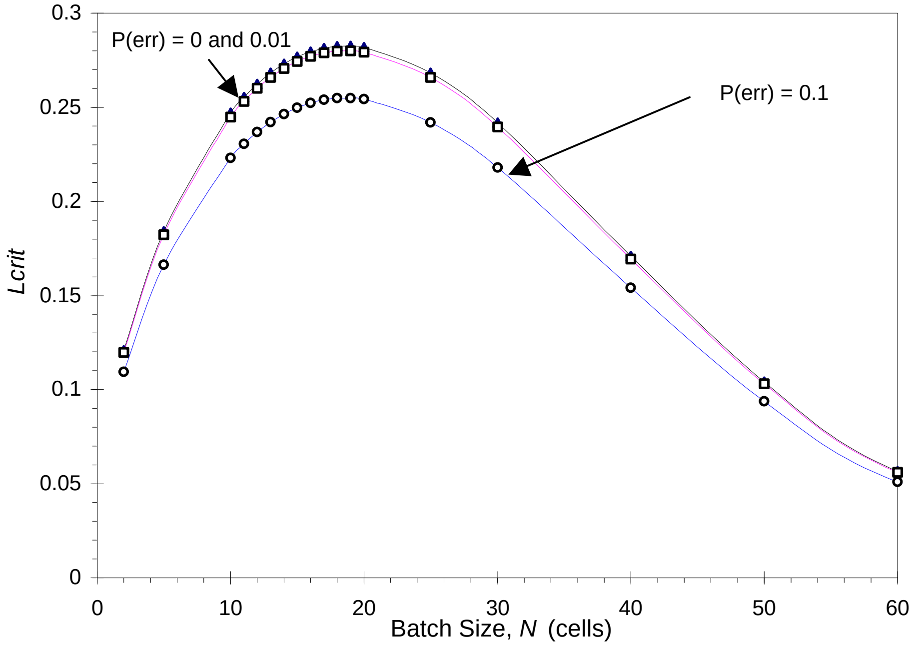

Figure 40. CMS Error Sensitivity - FCS (*M*=1) Scheme. Sensitivity to error is similar for other values of *p*. 1 CMS, *p* = 0.10.

Multi-CMS Cyclic CMS sharing, CCS_M, can be thought of as "circuit switching" and it involves more complexity at the station than its FCS counterpart. In particular, with CCS_M, we need the ability for the Head-End to randomly assign the stations to sub-groups; these groups then use only one of the multiple CMSs available. However, as Figure 41 shows, the CCS_M scheme yields better performance than FCS since the critical load (L_{crit}) is maintained at a significantly higher level for a larger number of contending stations.

Note, however, as expected, below a certain threshold when the batch size is relatively small (approximately N = 13 in Figure 41), collisions upon retries are less likely, and so it is more efficient to implement Full CMS sharing ("packet switching") and not to waste CMS slots by cyclically reserving them in TDM-like fashion. This result can be seen via comparison of Figure 38 and Figure 41. For N<13 and 3 CMSs in both cases, FCS exhibits better performance for the same value of p

(shown in Figure 38). For N>13 CCS_M, again with 3 CMSs and p=0.2 exhibits superior performance (shown in Figure 41).

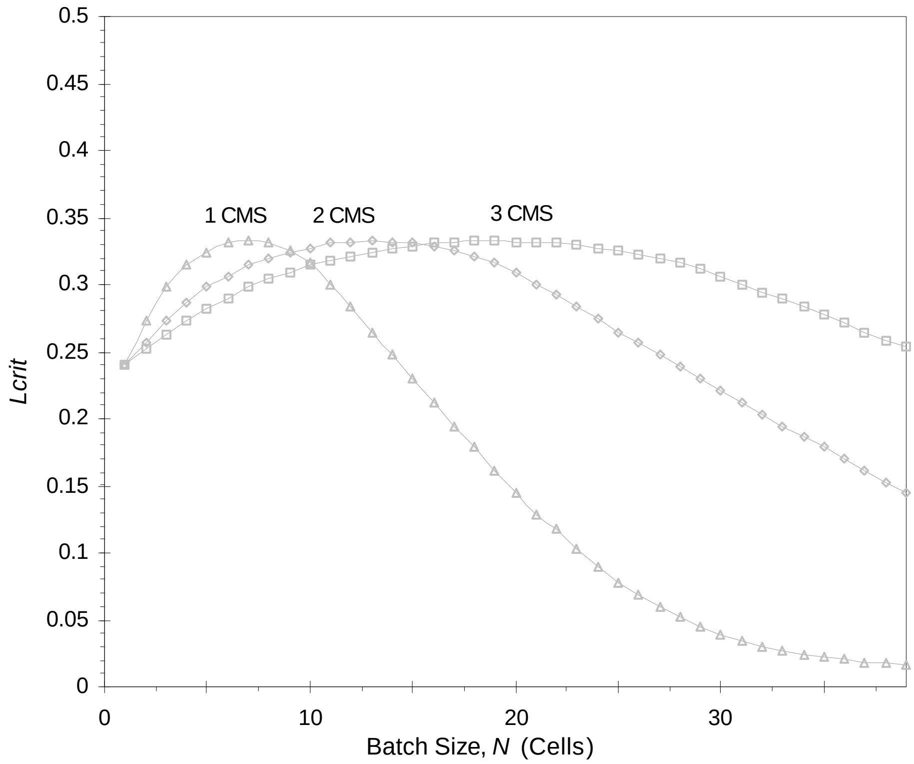

Figure 41. Cyclic (CCS_M) with 1, 2, and 3 CMS minislots, without CMS Errors. p=0.20.

Both simulation and analytical results show that both of these observations apply not only to error-free systems, but also to those with severe CMS error rates. Indeed, even an extreme CMS error level of P_{err}=0.1 causes, for both schemes, a maximum decrease in L_{crit} of only about 10%. However it does appear that the CCS_M scheme is slightly more affected by error over a wider range of N values. In general, the maximum decrease in L_{crit} is of little impact since a CMS error level of one in every ten will be extremely unlikely to ever happen (or be tolerated) in practice. A final point to note is that the introduction of errors does not change the shapes of the curves, nor the conclusions drawn earlier about the better overall performance of the Cyclic Multi-CMS Sharing scheme.

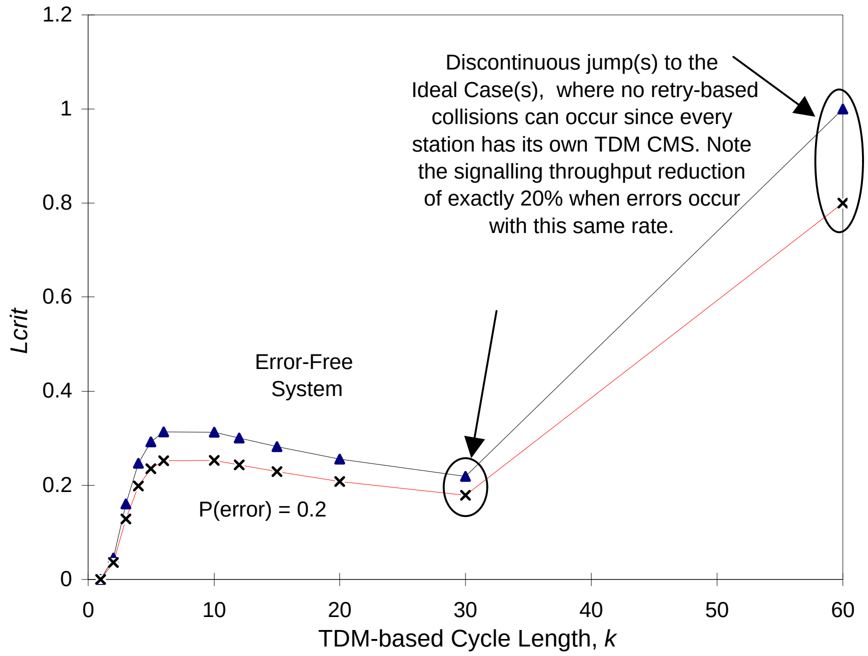

Figure 42. Cyclic Single-CMS Sharing (with and without Errors) – TDM-based access to CMSs. Data points are provided by C++ simulation of the model. Batch size, *N*=60, *p*=0.20, 1 CMS.

In Figure 42, the effect of the number of groups k on the critical load for the CCS_S scheme is demonstrated. The synchronous mode of access implicit in this scheme has similarities with TDM systems. It is therefore sensible to think of k as the TDM Cycle Length. Although there is only one CMS per data slot, access to this CMS is regulated in a TDM-like cyclic manner, so that each station is assigned membership to a group that has a certain position in the cycle ($TDM_{position}$), and is allowed to access a CMS only at that position and every k (i.e. cycle length) timeslots (notice that k is both the period length and the number of groups). That is, access is allowed at time T, only if $(T-TDM_{position}) \bmod k=0$. Once again, greater complexity would be required within the MAC protocol, both at the stations and Head-End, in order to implement this scheme.

Nevertheless, the CCS_S scheme shows some promise as a method to avoid deadlock since, as Figure 42 highlights, for a large batch size (N=60), and a relatively aggressive p persistence probability, p=0.20, a TDM cycle as short as 6 slots is enough to maximise L_{crit} (except of course for the discontinuity at $k = N$, when $L_{crit} = 1.0$). It has been found that the TDM technique implicit within CCS_S, is good for the alleviation of the signalling congestion created by the extreme inter-station correlation (i.e. large and periodic batches of simultaneous arrivals) which are being studied. However, under normal conditions (when the requests are not generated by single-cell messages, and so no longer deterministically arriving in correlated periodic batches), one can see that such a TDM technique introduces an amount of unnecessary increase in the average access delay. Hence, a trade-off between cost and benefit exists, the balance of which depends strongly on the traffic profile.

As with Figure 40 exploring the error-sensitivity of signalling performance, Figure 42 illustrates the relatively small effect of the presence of errors in the CMS signalling channel, where we once again see no more than a 20% worst-case drop in maximum achievable signalling throughput (L_{crit}), even when the CMS error rate is an unrealistically high 0.2. Also of interest in this graph, is the 20% difference in L_{crit} which may be observed between the very last pair of points when k = 60. While the dotted curve, representing the error-free case, shows an L_{crit} of exactly one (collision-free operation), the solid curve L_{crit} drops to 0.8, with the error level at 0.2.

This observation is now accounted for: the CMS minislots are segmented in a TDM-like fashion and, since $N=k=60$, there can never be any interference between the request arrivals which would cause a collision. The only outcome to require a retry (for any given slot) is an erroneous CMS and, given the probability of a CMS error is denoted by P_{err}, the time to successfully clear a single request is geometrically distributed with an average value of $1/(1\text{-}P_{err})$ CMS minislots. Therefore, the mean time to clear all the slots is given by $N/(1\text{-}P_{err})$, giving a critical load of $N/\{N/(1\text{-}P_{err})\} = 1\text{-}P_{err} = 0.8$ as seen in Figure 42.

6.4.2 Deadlock Model with Background Traffic (BIN)

Thus far, the performance of a Basic Deadlock model under traffic conditions leading to signalling channel congestion has been investigated. That model accounted for the worst-case periodic burst arrivals, with and without the impact of erroneous CMSs. The study was focused on the impact of the *p* persistence and error probability parameters.

CMS Access Scheme	Full CMS sharing with one CMS per upstream data slot
Number of CMS Minislots per Upstream Data Slot	One
Probability of CMS Minislot being Erroneous	10^{-3}
p persistence parameter, *p*	0.1

Table 6.3. System Parameters Common to the Models under Consideration.

In addition to the study of the Basic Model, Section 6.3 also evaluated different schemes for accessing a varying number of CMS minislots per upstream data slot. For this reason, it is not necessary to study these parameters and schemes again, so in the ensuing results a common scenario is assumed. The common parameters used for the remainder of this paper, are given in Table 6.3.

6.4.2.1 Average Contention Resolution Interval, T_c

In Figure 43 the value of T_C is calculated from Table 6.2 for each fixed average number or requests per CRI using different values of l. Of course, to fix the average number or requests per CRI, it is necessary to reduce the value of N whenever l is increased.

The higher the proportion of total traffic made up of "background" requests during a CRI (i.e. the higher the l), the smaller the T_C, and hence the better the performance of the system. Figure 43 shows that a request load comprising smaller initial batch sizes, N, with some background traffic is resolved faster than its counterpart with the same overall request load, but comprised of a larger N and less newly arriving traffic. Taken to the extreme, the system where the request load is made up only of the initial unresolved batch and no background traffic performs the worst, having the longest T_C under all request load conditions.

There is, however, an exception in the case of heavy background traffic. Figure 43 exhibits a 'crossover point' for l=0.3. Before this point the higher traffic BIN case bucks the trend by exhibiting a longer average CRI length than other models with lower background traffic. This peculiarity is due to the fact that there is a point in the BIN model where the background traffic actually dominates the collision resolution process that is exhibited in this case.

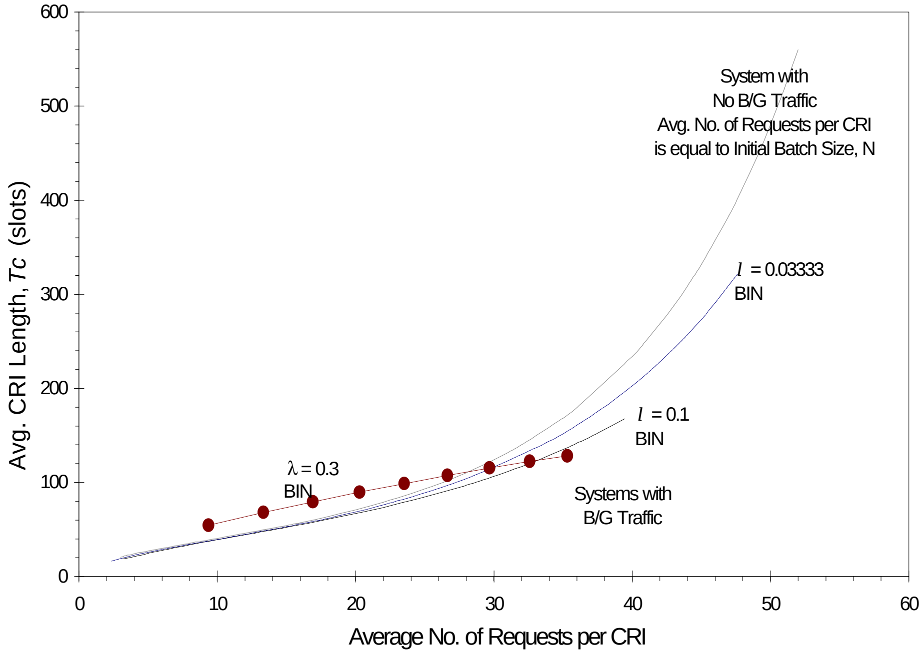

Figure 43. BIN Model – Impact of Traffic Composition on T_C for Different l Values. The simulation verification is indistinguishable when plotted on the same chart [IZC00] [IVA97].

If N is fixed at values 10, 30 and 50 (and for each of those values l is varied to obtain the correct average number or requests per CRI), an identical finding is made (Figure 44). Curves for which the initial batch size N is large show a longer T_C. Once again, the system where the overall request load lacks any newly arriving background traffic and consists only of the initial unresolved batch of requests, performs the worst, having the longest average CRI length.

Figure 43 and Figure 44 emphasise that the average CRI length is dominated by the size of the initial unresolved batch (i.e. T_C depends *exponentially* on N), for a fixed p persistence CRA and system parameters as given in Table 6.3.

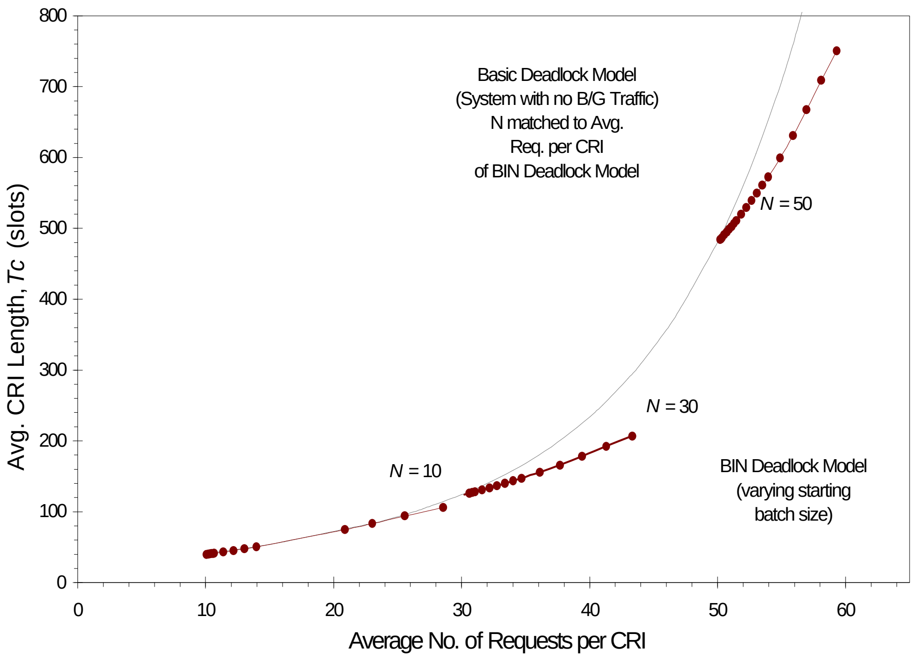

Figure 44. BIN Model - Impact of Traffic Composition on T_C for Different N Values [IZC00] [IVA97].

If, on the other hand, a scheme which dynamically adjusted p to its optimal value was used, the dependence of the CRI length on N would tend to become linear (with gradient e) as N ® ¥ [SHL96a]. This is significant because use of such an adjusted p persistence algorithm, together with the background traffic models proposed here, would result in the background traffic intensity becoming a dominant factor in the behaviour of average CRI length. At this point, there are no known algorithms for dynamically optimising p in conjunction with the BIN background traffic model.

6.4.2.2 Critical load, *Lcrit*

Figure 45, where L_{crit} is the ratio between the average number of carried requests during T_C and T_C itself, illustrates the same effect as noted in Figure 43. In this instance, the greater the proportion of the request load made up of newly arriving traffic during T_C, the greater is the maximum achievable signalling throughput, L_{crit}, as a direct result of the shorter CRI lengths discussed earlier.

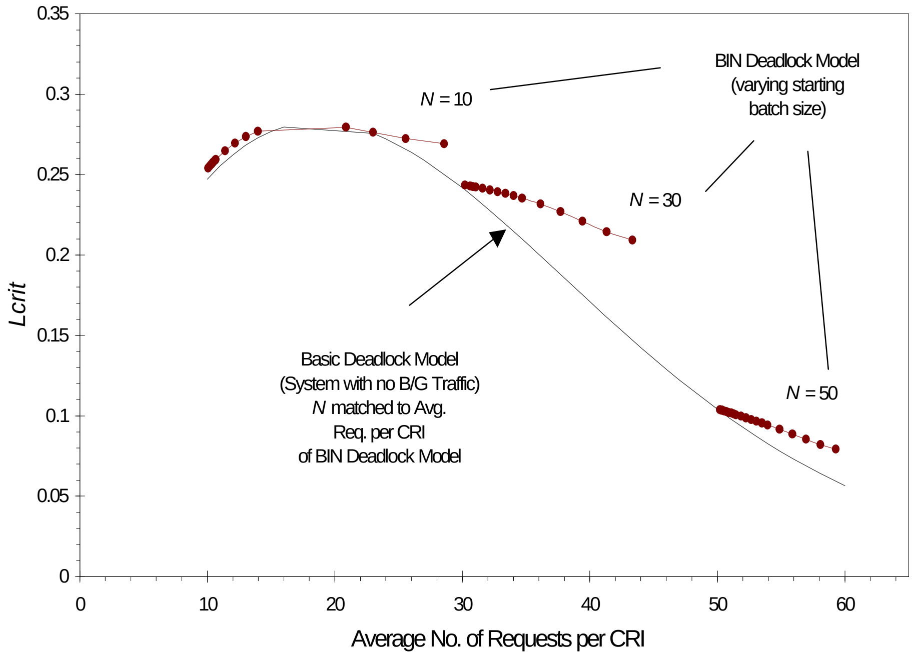

Figure 45. BIN Model - Impact of Traffic Composition on Critical Load L_{crit} , with N as parameter [IZC00] [IVA97].

A second observation is that the systems where the request load is made up of more background traffic have L_{crit} curves which are 'flatter' – thus more insensitive to changes in request load. In particular, once the peak L_{crit} value has been reached, systems with larger l tend to show a markedly smaller rate of decrease in L_{crit} as the request load increases. This last point suggests that a system with an overall request load made up of more background traffic and smaller initial unresolved batches, will experience a growth in T_C which is linearly proportional to any growth in the request load per CRI. On the other hand, systems with less background traffic and larger initial batches are more adversely affected, so that growth in T_C occurs at an exponentially increasing rate as the request load per CRI increases.

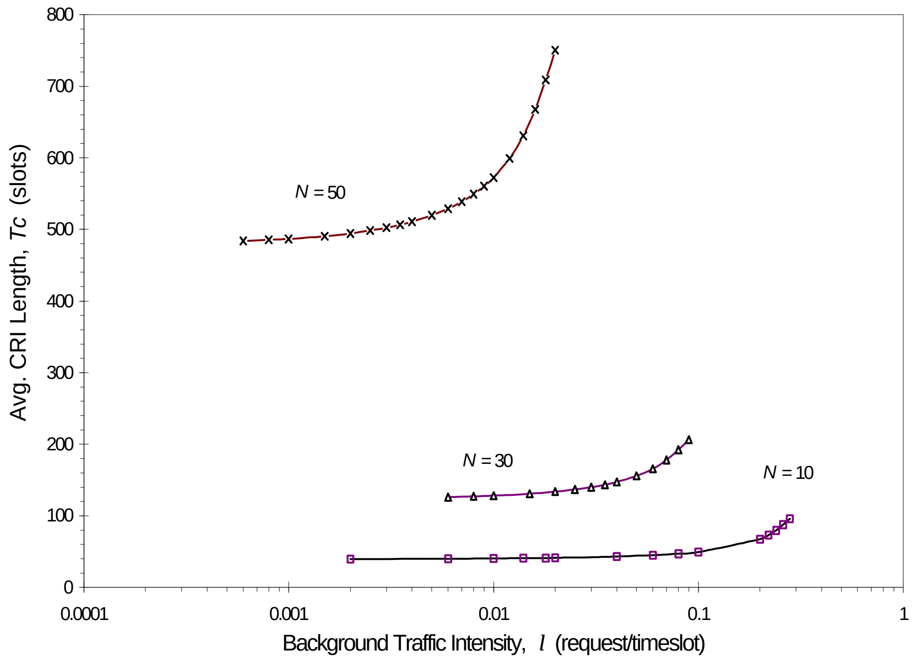

Figure 46. Comparison of T_C for different values of N under the BIN Model [IZC00] [IVA97].

This finding is further reinforced by Figure 46, which shows that for sufficiently small initial batch size, say N=10, T_C actually increases slower than linearly with more request load, and thus causes a positive gradient for the L_{crit} vs. load curve (recall that L_{crit} is the ratio of carried request load to T_C).

6.4.2.3 Probability of Absorption (Obtaining a finite T_C)

Figure 47 and Figure 48 show the impact of traffic on absorption probability with l and N respectively as parameters. Note that the system without any background traffic always has finite T_C as expected, since the rate of increase of unresolved requests is invariably zero and hence the upper bound of the state space is always finite, and equal to N. Figure 46 shows that the heavier the background traffic, the smaller the initial batch size, N, allowable for the system to still have a practically finite mean CRI length.

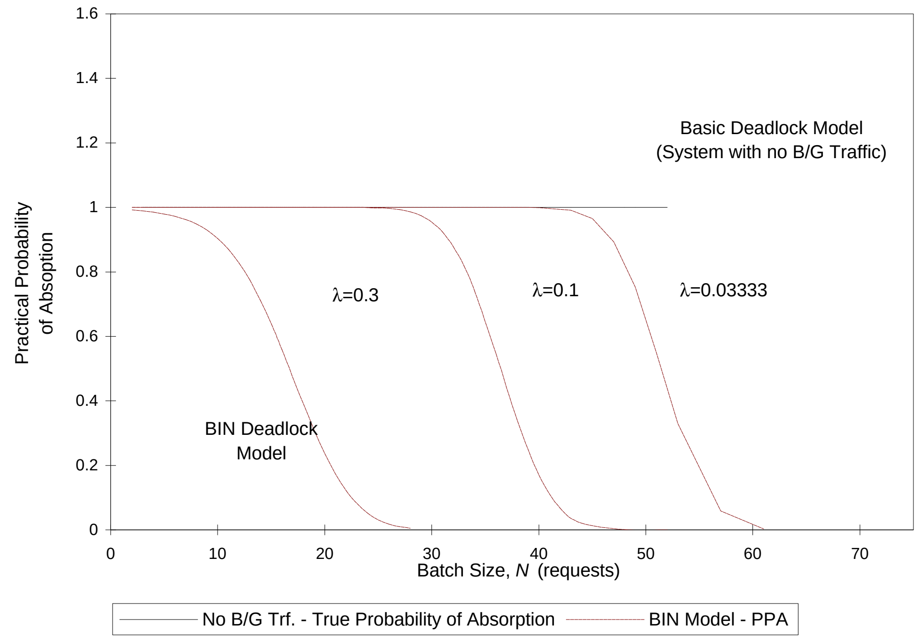

Figure 47. BIN Model - Impact of Traffic Composition on Absorption Probability, with *l* as parameter [IZC00] [IVA97].

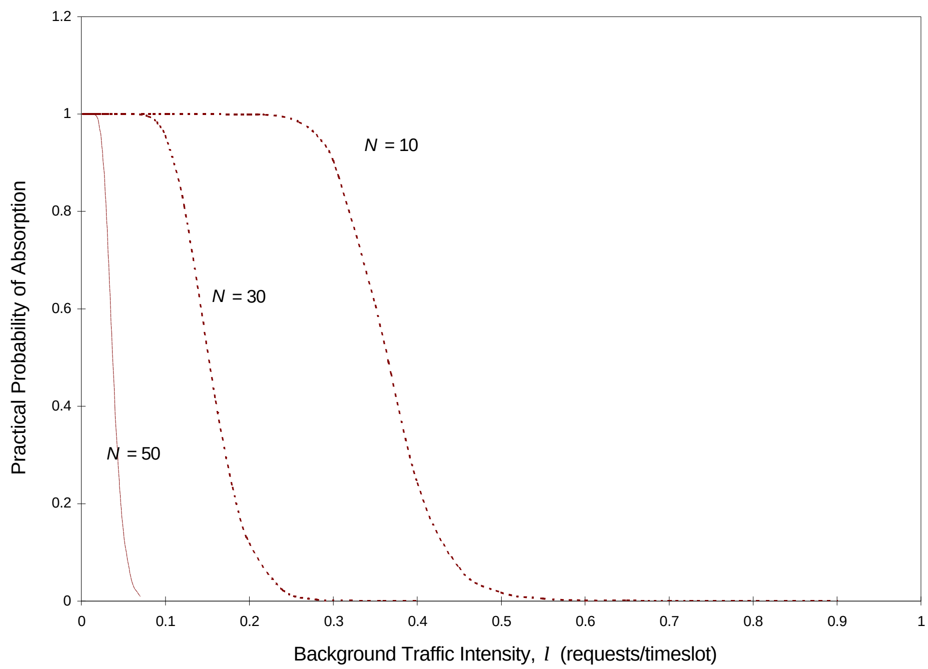

Figure 48. BIN Model - Impact of Traffic Composition on Absorption Probability, with *N* as parameter. Analytic results and results obtained by the author's C++ simulation are indistinguishable [IZC00] [IVA97].

Looking from a different perspective, the larger the initial batch size, the smaller the allowable background traffic needed to force the system out of the region of practical stability. As an aside, Figure 48 highlights the excellent match between the analytical and simulation methods of obtaining the Probability of Absorption. The plot shows the simulation curves however the analytic solution gives identical results.

6.4.3 Conclusions

So far, focus has been placed on an analysis of the performance of a *p* persistence CRA subject to extreme inter-station correlation. Deadlock conditions have been studied where the time to collision resolution becomes unacceptably high and two models in which it is assumed that many stations are powered-up simultaneously, hence generating multiple simultaneous capacity requests, have been considered. One of the models assumes no background traffic whereas this is included in the other model.

As well as introducing the concept of contrasting between practical and theoretical stability, the chapter has provided a detailed set of conditions that have shown to lead to practical instability and deadlock. These depend on such factors as: signalling channel error probability; the profile of signalling traffic (in particular the proportion of background traffic versus the initial batch); and the way the request signals access CMSs.

In assessing the schemes introduced, the best performing of these three schemes, in terms of extending the protocol's practically usable load region the furthest, is identified as CCS_M. It is demonstrated that the TDM-like property of this scheme is the key to its success, especially under the disaster scenario conditions simulated via the deadlock models. However, this scheme has also been shown to require the most additional Head-End and station intelligence, as well as some extra signalling bandwidth, to implement. The CCS_S scheme has also shown to be able to avoid deadlock, but under normal traffic conditions it introduces high average access delay.

Finally, the chapter so far has demonstrated that the p persistence parameter is the most significant in reducing the likelihood of deadlock. Increasing the number of CMSs does not provide the desired protection against deadlock for the case of a large number of contending stations. The study of error prone signalling channel conditions has highlighted that the maximum practically stable signalling throughput is largely unaffected by the presence of single or multiple signalling channel errors, so that even in extremely noisy or otherwise error prone environments, the limiting performance factor of the CRA is still the collision rate which is mostly affected by the parameter p. it has further been demonstrated that the parameter p not only has a significant effect on the critical load (and on the resiliency of the protocol), but that it can be optimised for maximum load, for each combination of other parameters and we have confirmed that the optimal value of the p is close to $1/N$ from above.

6.5 Results and Implications – Other CRAs

6.5.1 Introduction

The analysis is now developed for other candidate CRAs using both the Basic Deadlock Model (BDM) from Section 6.2.2 and further new models. Rather than continuing with an analysis of the BIN model, we introduce a new disaster model (presented in Section 6.4.2) to capture the effect of background traffic using *piggybacking* to transmit sequences of data packets. Hence the chapter implements a model in which multiple arrivals as well as correlated arrivals in subsequent time slots are captured. This is denoted the Piggybacking BDM model or BDM_P.

The same arrival process used in the first half of this chapter is adopted, where all *N* stations in a population transmit simultaneously into one CMS at a particular instant, using the parameter L_{crit} as a measure of worst case performance for the 'disaster scenario'. As before, the rationale for the research is the design and dimensioning of a multi-service MAC protocol robust to all traffic scenarios that might arise. In so doing this work provides dimensioning guidelines so the MAC presented in Chapter 2 is capable of avoiding signalling congestion collapse in all cases.

6.5.2 CRA Types

In this section, the CRAs to be studied are re-introduced and a number of terms are defined. The following CRAs, introduced in Section 5.3, will be considered and compared:

- ***Simple p persistence protocols***: New packets are retransmitted with fixed or adapted probability in a slot following initial contention. This method is used in the R-B Control region of a recent NEC proposal for a WATM MAC [IZ98]. The basic *p* persistence scheme is simple to implement, however variants with higher performance such as estimators for adaptive optimisation of *p* for the system create significant extra control complexity;
- ***Blocked-access Q-ary tree protocols* (BAP)**: New packets arriving are transmitted in the first slot after all previous conflicts are resolved, i.e. packets are blocked at their transmitters until the current CRI terminates. This requires each MT to monitor channel

feedback so that they can keep track of when each CRI ends which is a disadvantage since MTs cannot be turned off in the absence of packets to transmit;

- ***Free-access Q-ary tree protocols* (FAP)**: New packets are transmitted immediately at the beginning of the next slot following their arrival. New arrivals join the subset of requests at the head of the stack so that only currently backlogged MTs need to monitor channel feedback. This saves power and prevents unnecessary complexity. The rate of collapse of the algorithm at high loads, however, is much higher than for the blocked tree; and
- ***FAP derivatives*: (msSTART)**: msSTART is an evolution of the unblocked stack algorithm. It is an easy to implement high-efficiency, free-access, robust, stable CRA optimised to operate in an environment where CMS could be changed dynamically [Bis96a]. It does not collapse as fast as the basic unblocked tree algorithm at high loads as a result of the temporary blocking of new arrivals until the start of the next block. The algorithm can be easily interleaved and requires only binary feedback for operation.

The rest of this section will focus particularly on the msSTART algorithm contrasting it with the simple p persistence considered in detail earlier.

6.5.3 The Basic Deadlock Model and Q-ary Tree CRAs

6.5.3.1 Performance of simple BAPs under the BDM

Section 6.2.2 derives an analytical solution for p persistence CRAs subject to the correlated traffic requests of the BDM. The performance of basic Q-ary trees subject to BDM traffic and analysis for FAPs or FAP derivatives is reserved for this section. Again, the focus is on the mean and the distribution of the Q-ary CRA CRIs rather than considering an individual user's signalling delay, and it is shown that a more complex CRA can deliver in practice significantly higher stability and throughput than p persistence, requiring lesser dimensioning for 'survivability'.

In defining the performance of tree CRAs, it is assumed that:

- Time is slotted;
- For the Q-ary tree case, each 'node' in the tree is split into Q *concurrent* slots after a contention occurs;
- There is immediate feedback to MTs of collision or no collision, but (for the tree case) the action taken in any slot of a group of Q concurrent related slots cannot depend on the contents of other slots within the same group.
- No advantage is taken of efficiency gains available as a result of tree collapse made available by ternary feedback; and
- There is retransmission of collisions and no buffering.

The principal difference between this study and other analyses is that Poisson arrivals are not assumed. Instead, as noted before, the correlated arrivals are modelled by having each of the N stations in the population transmit periodic batches of capacity requests to the HC.

For the simple BAP CRAs it is possible to derive a probability generating function for f_N, the CRI length, given N stations. It is assumed using the deadlock model that each of the N stations in the population transmit a request at the same time and that all other prior contention resolutions are complete.

For the case of the blocked or unblocked basic tree CRA with no optimisation, N packets will initially collide under the Basic deadlock traffic conditions. The reader's interest is drawn to the first moment of

f which is denoted $T_c = E[f]$ in order to determine the average CRI length. For the case where we do not re-order the tree,

$$f_N = 1 + \sum_{j=1}^{Q} f_{I_j} \tag{48.}$$

Where $N > 1$, and initial values are

$$f_0 = f_1 = 1. \tag{49.}$$

The Probability Generating Function (PGF) of f_N is defined as

$$G_N(s) \equiv \sum_{k=0}^{\infty} P(f_N = k) s^k = E\left[s^{f_N}\right] = E\left[s^f \middle| N\right] \; . \tag{50.}$$

Using the technique of further conditioning the expectation on the right hand side of the above [MF85], gives

$$G_N(s) = E\left[E\left[s^f \middle| I_1,...,I_Q, N\right] \middle| N\right] \; . \tag{51.}$$

Substituting Eq. (46) into (51) and realising that the size of the j^{th} subtree depends only on I_j, it can be seen that the relationship between PGFs for the CRA under study is (for $N > 1$):

$$G_N(s) = s \sum_{i_1 \cdots i_Q}^{N} \binom{N}{i_1 \cdots i_Q} \cdot \prod_{j=1}^{Q} p_j^{i_j} G_{i_j}(s), \tag{52.}$$

where the first sum is defined as the sum over all possible combinations of $i_1 \ldots i_Q$ (with i_j a positive integer) such that:

$$\sum_{j=1}^{Q} i_j = N$$

$\begin{pmatrix} N \\ i_1 \cdots i_Q \end{pmatrix}$ is defined as a multinomial coefficient

$p_j = P\{\text{the } j\text{th value of the random test}\}$

$$\sum_{j=1}^{Q} p_j = 1 \tag{53.}$$

The initial conditions translate as

$$G_0(s) = G_1(s) = s \tag{54.}$$

To obtain T_c it is necessary to take the derivative of Eq. (52) with respect to s and evaluate at $s=1$:

$$T_c = 1 + \sum_{j=1}^{Q} \sum_{i_j=0}^{N} \begin{pmatrix} N \\ i_j \end{pmatrix} p_j^{i_j} \left(1 - p_j\right)^{N-i_j} L_{i_j} \quad , \tag{55.}$$

which can be solved recursively to obtain T_c. So, after the first time interval, all N of the stations try to access the channel and are forced into contention resolution using for an average CRI given by T_c.

The above solution can be presented in a non-recursive form so that the mean value of the CRI length, given N transmitters initially collided, is easily calculable. Again, this is denoted L_{crit}, and for $N > 1$,

$$L_{crit} = 1 + \sum_{k=2}^{N} \binom{N}{k} \frac{(-1)^k Q(k-1)}{1 - \sum_{j=1}^{Q} p_j^k} \quad . \tag{56.}$$

Remember from Section 6.2.3 that L_{crit} signifies the critical load. This load is considered to be critical because if the N requests arrive with a period less than T_c, the arrival rate exceeds the system's service rate, and the system becomes practically unstable with the number of outstanding requests increasing towards infinity. This level does not denote theoretical instability, since it has already been shown that T_c is always finite, however, we observe that practical instability defines a scenario in which the number of backlogged requests, although finite, causes T_c to become so large that it is infinite for practical purposes.

6.5.3.2 Performance of advanced FAPs under the BDM

The chapter now turns to the performance of the CRAs under our deadlock models. The first model considered is the BDM.

As described in Section 5.3.5.3, msSTART operates as a tree (or stack) based Q-ary CRA. As for the case of the BAP considered in the previous section, after a collision each transmitter flips a 'Q-sided die' with values 1, 2, … ,Q (it is assumed that the Q-sided dice are fair). This splits the set of contending transmitters into Q subsets and an index is assigned to each of these subsets indicating the result of the 'Q sided die' that was 'flipped'.

If N is the number of active stations, then all N stations request capacity for a single cell message at the same point of the beginning of the CRI. From a signalling contention perspective, this extreme correlation is once again the worst-case scenario. We use batch size to describe the number of simultaneously arriving requests and the aim is to derive the mean of the CRI duration, measured in the number of CMSs (f_N). This mean is designated as

$$E[f_N] = T_c(\text{Scheme}, N, p, P_{err}) \quad , \tag{57.}$$

which is denoted T_c for short. *Scheme* stands for the form of CRA scheme used in the analysis. P_{err} denotes the probability of an error in a CMS mini-slot. Any given CMS has the possibility of suffering a random error and it is assumed that, during each CMS, errors are independent. Hence the error process is one with geometrically distributed inter-occurrence times (parameter P_{err}).

msSTART is often analysed as a pointer to a stack [Bis96a] and the msSTART counter update algorithm is given as follows: when an inactive station becomes active, initialise $COUNT = random\{0, \ldots, M-1\}$ where M is the number of CMSs in the block following the activation of the station. Q is defined as the number of subsets to split into:

```
If (COUNT<M) then
Station transmits in CMS number COUNT;
Upon receiving feedback information:
       If (did not collide) then
Done;
       Elseif (collide)
Set: COUNT ← Q*col{M}+random{0, … , Q-1};

Elseif (COUNT > M–1) then
Station does not transmit in current block;
Upon receiving feedback information:
Set: COUNT ← Q*col{M}+COUNT–M;
```

$Col\{M\}$ is given as the number of collisions during a block.

T_c results for the msSTART algorithm are approximated using a computer simulation written in C++. The simulation is designed to mimic the operation of the MAC layer of a multi-service wireless network, providing utilisation, delay and stability statistics for a range of system specifications. The user can control all of: input traffic characteristics, CRA type, CMS number and type per frame, as well as error distribution and probability. This simulation is an extension of that utilised to verify analytical results presented for the *p* persistence CRA.

6.5.4 Performance of simple BAP schemes under BIN

The relevance of the BIN model discussed in Section 6.2.3 to this analysis of BAP CRAs is now considered. As well as taking into account the possibility of errors in the transmission of requests, the BIN considers the situation where new request arrivals during the contention phase of an initial batch of N requests. If there are L stations in a given cell, then initially, out of the L stations, N transmit simultaneously. The remaining L-N stations are generating background traffic. The limited population of stations L-N experience a state-dependent probability of an additional arrival during a timeslot.

Recalling the description of the BAPs in Chapter 5, it is clear that, for the BAP case, those of the L-N who receive requests during the CRI must wait until the resolution of the current batch before they can inject their excess traffic into the CRA. It is only once the BAP CRI is complete that any new arrivals can be allowed into the system. At this point the MTs can send their new packet in the first slot following the resolution of *all* collisions that had occurred previous to the arrival of the packet.

This observation operates to limit the usefulness of the BIN model for BAP systems. Each MT can inject only one request per CRI, and each CRI must complete before any background traffic can enter the CRA. However, under the BIN, *N* stations transmit initially and the remaining *L-N* stations receive traffic in a proababilistic fashion. If *H* of those stations receive traffic during a CRI, then the *BIN model reduces to the BDM model for a station population of N+H under the BIN*. Hence, for the BAP system, the BIN yields the BDM where the number of simultaneously transmitting stations is determined as the expected number of arrivals during the average CRI plus the *N* simultaneously transmitting stations.

This observation is used in the development of the BDM_P model for tree based CRAs later in this chapter. *p* persistence is also studied under BDM_P conditions.

6.5.5 Performance of BAPs and FAPs Under BDM_P

The failure of the BIN model to capture adequately background traffic effects for all forms of tree CRA requires the development of the BDM_P. The BDM_P considers the case where *N* stations initially power up. Based on a traffic burstiness parameter P(*fin*), some stations receive further requests for bandwidth during any given CRI before their previous request for bandwidth has been successfully transmitted over the R-B Control portion of the hybrid TDMA MAC.

The MTs may either make a request for bandwidth during a R-B Control Region (from Section 2.7 above), or they may add their request to the end of a previously allocated data frame – thus ‘piggybacking’ the request. These processes reserve the MT’s transmission space in a contention free portion of the overall hybrid TDMA frame so that the MT can transmit upstream data free of collisions. A MT with data in a recent previous slot need not participate in the R-B CRA due to the piggybacking of a request for additional bandwidth on the back of a previous data slot. However, any MT that has just entered the cell, been switched on, or was been silent in a previous frame will need to compete with the other stations in the B-R region to reserve bandwidth for transmission of data in the later collision free DMSs.

As stations are successful in transmitting their requests for bandwidth during R-B Control, and they are allocated one or more collision free DS by the S-MAC, stations having received further requests for bandwidth before transmission in their allocated DSs are able to piggyback requests. Hence, from a cell population of N stations, there will be N initial participants in the CRA. If B stations receive further requests for bandwidth before the transmission of upstream data on their allocated DSs, there will be only N-B participants in the next contention round since the other B stations have already had their requests for the round successfully received via piggybacking. Hence the BDM model is now capable of taking into account the ability of MTs experiencing background traffic to avoid participation in the R-B Control CRA.

It is assumed that MTs exist within a 3 state system – they must either be:

1. *Free* and awaiting traffic which will arrive in correlated 'waves'; or
2. *Competing for access* in the R-B region; or
3. Following success in *State 2*, now involved in collision free transmission of data, *piggybacking* further requests for bandwidth in each MDR-TDMA frame.

MTs move through the states in strictly ascending order and return from a sojourn in State 3 (whose length depends on the burstiness and volume of data to send) back to State 1 during any period T_c (defined as the average time for the N competing stations to all submit their requests without collision for a given CRA) with probability P(*fin*).

Under the BDM_P model, correlated bursts of traffic are sent by the MTs. All of the MTs will thus enter *State 2* in the second CMS (aside from the case that the MT population is less than two). When an MT is able to send its request for bandwidth in a contention free CMS (and hence the R-B request is resolved), it then enters *State 3* and transmits its data in the collision free DMS, piggybacking a request for further bandwidth as required to the end of the DMS. When the MT becomes idle, it ceases to request DMSs in future TDMA frames and returns to *State 1*.

The speed with which the MT moves from *State 3* to *State 1* depends on the volume of traffic appearing at the MT for transmission to the HC, as well as the burstiness of that traffic. As discussed above, variability is modelled using P(*fin*) – the probability that an MT will return back to State 1 during the time period T_c.

A value of P(*fin*) close to one indicates either extremely bursty, or very low levels of background traffic, whereas a value of P(*fin*) close to zero indicates both non-bursty CBR type traffic and a substantial volume of background traffic. In this instance the MTs will rarely leave *State 3*.

Note that low values of P(*fin*) imply a small number of MTs participating in the R-B Control CRA and hence good throughput as well as low CRIs. The separate problem of DS capacity dimensioning is not considered here, as it relates to link capacity and allocation strategies raising link dimensioning issues (Chapters 3 and 4 of this thesis) and CAC issues (considered by the author in [LZC99]).

6.5.6 Signalling Channel Schemes

As in the preceeding parts of this Chapter, three schemes to manage and access CMS signalling slots are considered. The first and most simple scheme is FCS. Here the critical load is given by

$$L_{crit} = \frac{MN}{T_c\left(FCS, N, p, P_{err}\right)} \quad . \tag{58.}$$

This defines the Critical Load L_{crit} as the ratio of work arriving in a batch, namely N, and the average time in upstream super-slots (containing a data and M associated CMSs during which the resolution of these N requests is completed).

Under the second scheme, the N stations are subdivided into M groups, so that each group must access a different CMS associated with the data slot. Implementation-wise, the station counts passing CMSs and is allowed to access every M^{th} CMS, which occurs once per data slot since there are M CMSs for every one data slot. Hence the effective load accessing any CMS is reduced to N/M. This scheme is denoted by CCS_M and has a critical load of

$$L_{crit} = \frac{N}{T_c\left(CCS\_M, \frac{N}{M}, p, P_{err}\right)} \quad . \tag{59.}$$

Finally, the third scheme, which is similar to the CCS_M, is its single-CMS variant, CCS_S. Using only one CMS per data slot, if the N stations are divided into k groups, then each group may only access every k^{th} CMS. We obtain a critical load of

$$L_{crit} = \frac{N}{kT_c\left(CCS\_S, \frac{N}{k}, p, P_{err}\right)} \quad (60.)$$
.

Note that if CCS_S is used, there is an additional fifth factor - the number of separate contention resolution groups, *k* (as defined earlier). This scheme corresponds to the separate collision resolution for different priorities of traffic. Stations initially transmit in slots exactly matching their priority level so the headend knows that all stations participating in a particular collision are of the same priority level. Requests collide only with other requests of the same priority preventing other priorities from interfering with them.

6.5.7 Results and Implications – Comparing msSTART with p Persistence

This section presents numerical results using the BDM and BDM_P models and the proposed signalling schemes. The aim is to compare the resilience of the newly defined tree based CRAs with the already analysed *p* persistence case.

The section commences by verifying analytic results with the result of a computer simulation of the system under study. All tree results are for $Q = 3$ unless otherwise stated. In Section 6.4.1, the results of an analytic evaluation of the performance of simple *p* persistence CRA are presented for comparison with a C++ simulation of the system created by the author. That simulation, written to validate the mathematical results of the thesis is the same simulation is used here and Figure 49 repeats the comparison. In that section, and again here, the results from the two methods of evaluation of L_{crit} are similar, thus verifying the analytical models over a range of *p* values.

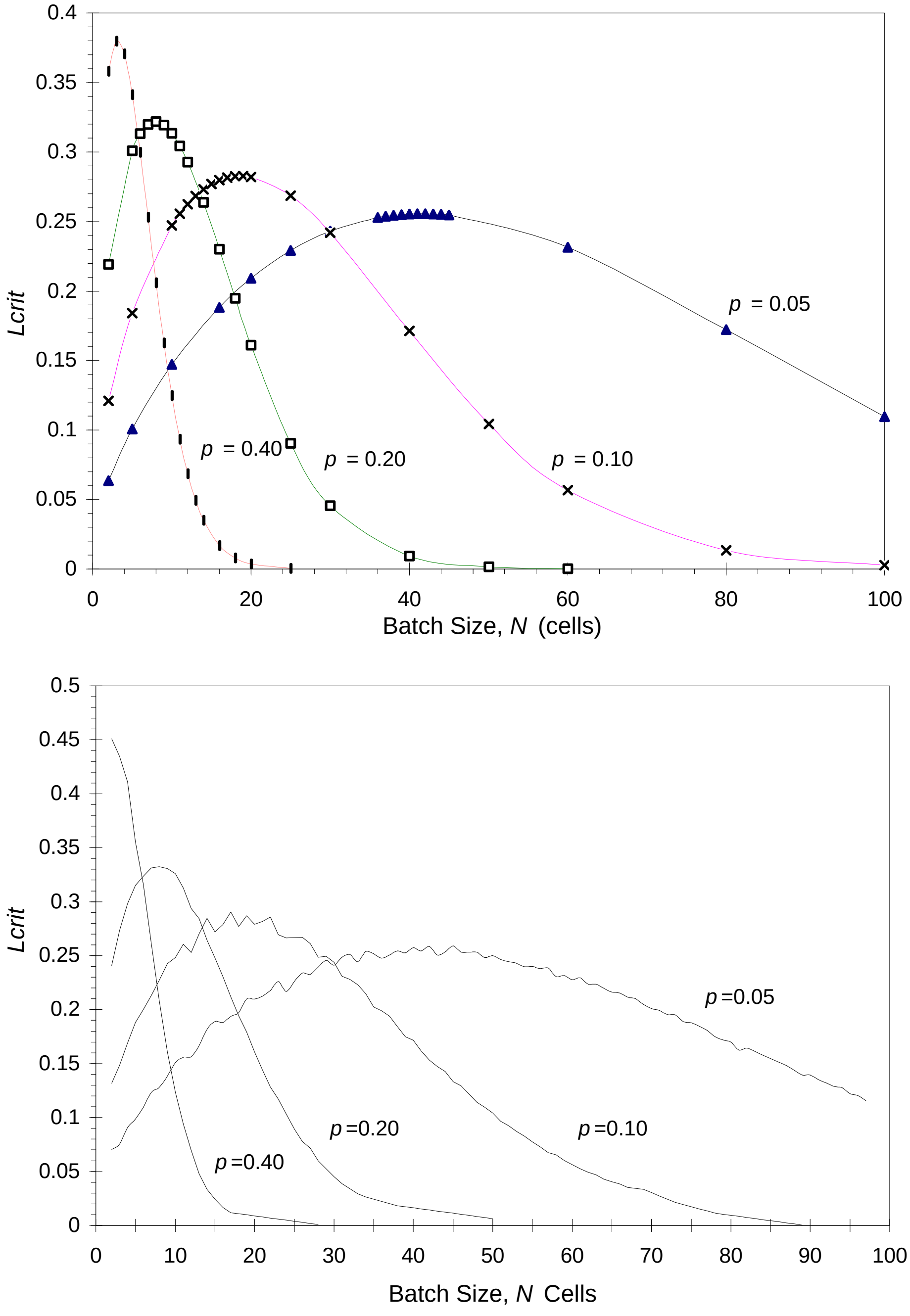

Figure 49. Comparison of analytic results (above) and simulation approximation (below) for several values of p for the slotted p persistence CRA. As was observed in Section 6.4.1, the simulation results confirm our analytical model.

Unless otherwise stated in this section, the analytic model is used in the figures for the *p* persistence algorithm and simulation approximations are used for msSTART and the other tree cases. Any differences between the simulation and analytical models are considered insignificant. In essence, the only discerning feature of the simulation results, as against the analytical results, remains small fluctuations, present in the simulation results. These show the effect of small error in the simulation caused by insufficient data sets passed through the computer environment. Ability to smooth these small variations is limited by the computing resources available and the imperative of reasonable delay between experiment design and results.

A similar agreement between analytical and simulation results is observed for the tree based CRAs in Figure 52. In that figure both the average CRI length (measured in CMSs) and L_{crit} are measured against the number of stations in the cell. Both the analytical expectation from Eq. (54) and the simulation results from a C++ simulation of the environment are given and the results compared. It is clear that, aside from small inconsistencies caused by restrictions on computer power, the results are similar, thus validating the analytical model for the tree based CRAs over the range of Batch Sizes that are relevant for this chapter.

The simulation used for validation of the tree based CRAs in Figure 50 is the same as that used for Figure 49, save only for the replacement of the *p* persistence CRA module with another designed to mimic the operation of a stack or tree based algorithm. The *Q*-ary CRA reported on in Figure 50 is msSTART.

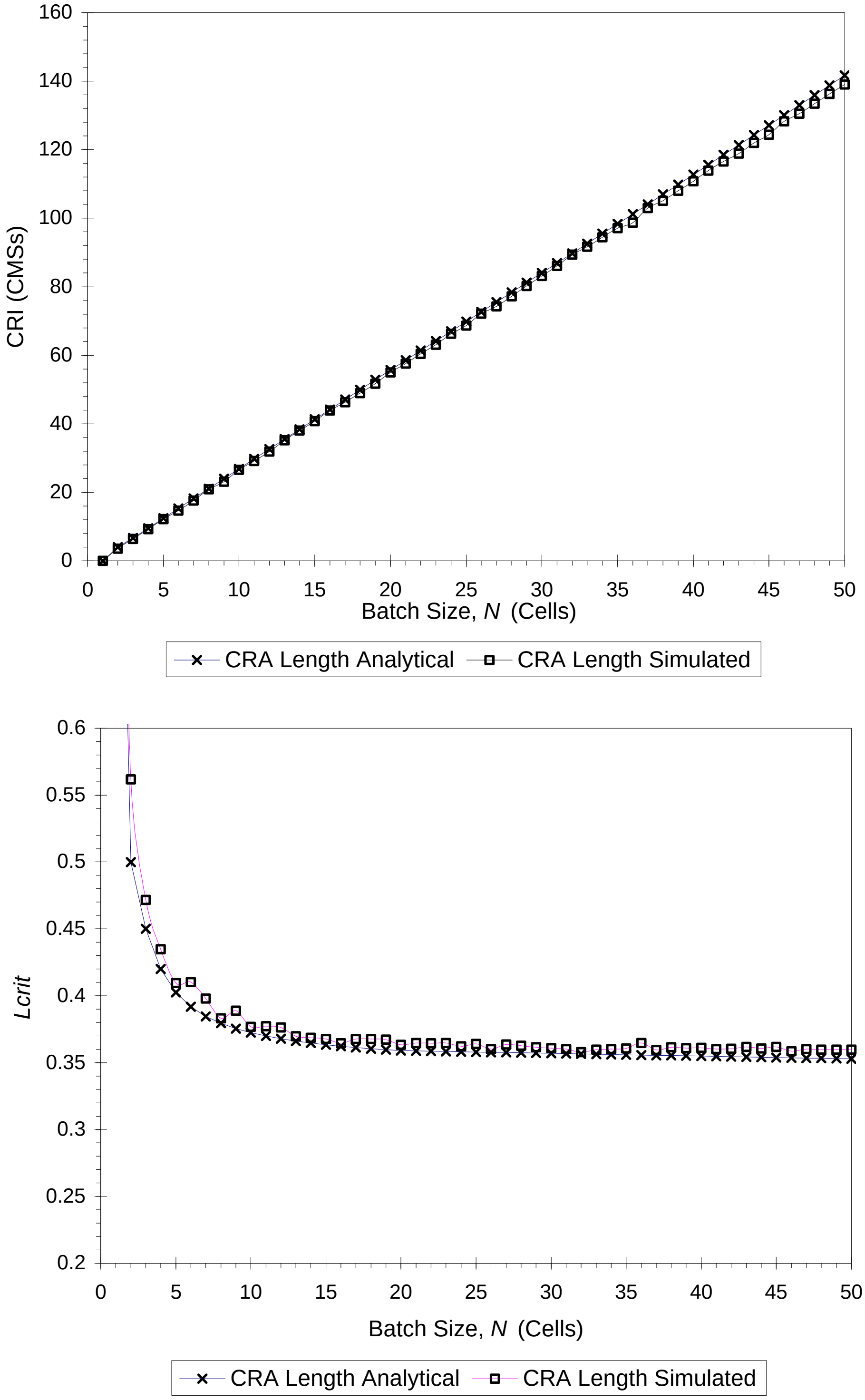

Figure 50. Comparison of analytic results (above) and simulation approximation (below) for an example of a *Q*-ary CRA. As with the case of *p* persistence, the simulation results confirm our analytical model.

It is the variability of L_{crit} with N for the p persistence case that motivates the comparative study of other CRAs. One of the main aims of the development of alternate models in the previous sections has been to offer a comparison of CRAs under stress conditions and Figure 51 compares network performance after failure for four schemes including:

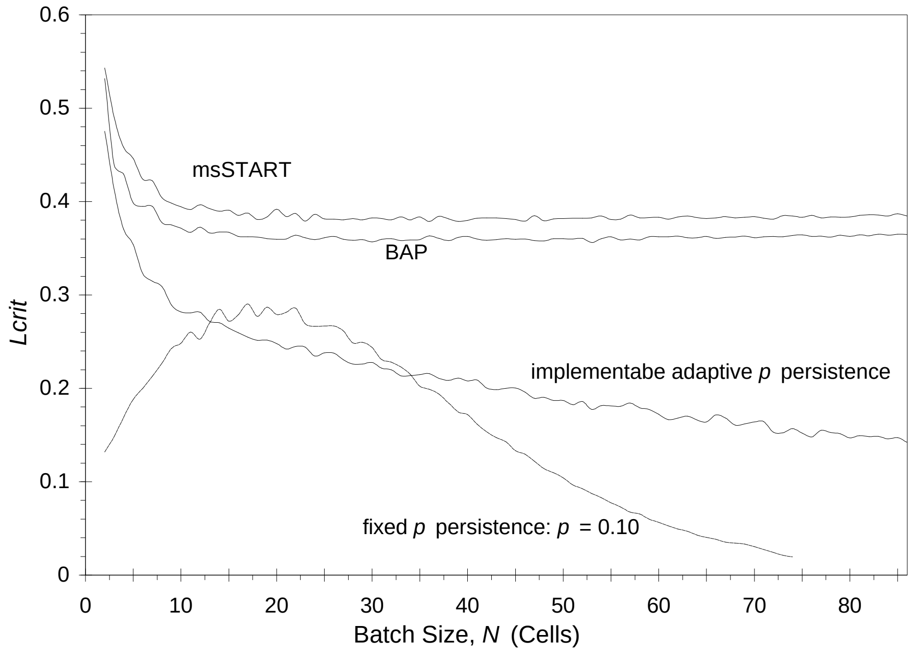

Figure 51. Comparison of simulated results for Adaptive *p* persistence, fixed *p* persistence, the basic binary tree algorithm and msSTART. The adaptive *p* persistence can never reach the maximum *Lcrit* theoretically obtainable by fixed any *p* persistence curve, as its estimation method is non-ideal. All results use the FCS signalling scheme with 1 CMS and all results are from a C++ simulation of the model.

- A fixed *p* persistence slotted ALOHA scheme with moderate *p*;
- An implementable adaptive *p* persistence algorithm using a pseudo-Bayesian estimator (see Section 5.3.2.4 above). The method used here was introduced in 1985 by Rivest for stabilising the conventional slotted ALOHA algorithms and seeks to estimate the number of backlogged requests. It can achieve a stable maximum

throughput of e^{-1} under Poisson arrivals. Note, however, that the method requires all the MTs to use the same retransmission probability. Since the arrival rate is not known, this must be estimated as well, causing increased control complexity in the wireless system [Riv85]. Such complexity is to be avoided since errors in the multicast of the value of *p* to be used from the S-MAC to the MTs can cause severe estimation errors (so non-optimal performance), and the requirement that the MTs monitor the channel to determine the current value of *p* to be able to transmit is wasteful of power;

- MsSTART; and
- BAP.

Figure 51 displays the effect of the disaster scenarios on CRA performance in the R-B control section and gives an indication of overall wireless multi-access system R-B performance for the four CRAs on start-up or after failure.

The performance of the *p* persistence case is strongly dependent on the number of MTs competing for bandwidth in the cell. On the other hand, implementation of the adaptive *p* persistence algorithm gives reduced sensitivity to the number of MTs in the system although there is a clear trend toward lower efficiency and hence higher risk of deadlock with increased numbers of MTs. Note also that the estimation process employed by the adaptive *p* persistence is sub-optimal. This means that peak L_{crit} performance of any of the fixed *p* persistence system curves cannot be achieved by the adaptive system even in the presence of an identical number of stations. In other words, because of errors in estimating the optimal *p* in the adaptive system for a given station population, *N*, the L_{crit} obtained for optimal *p* at that *N* is not achieved by the adaptive system as a different value of p has been incorrectly estimated as optimal. Nevertheless, for most N values, the adaptive system performs better than the fixed p persistence system.

Most striking is the superior performance of msSTART together with its BAP cousin: not only do these CRAs achieve significantly higher values of L_{crit} for all studied values of N, they are also far less sensitive to changes in N except at very small station populations. msSTART and the more simple BAP do not require any of the additional control complexity involved in estimating optimal p, and MTs do not require periodic update from the S-MAC of any optimal retransmission probability. Since MTs can calculate when to retransmit using a single stack implemented within each MT, there is no possibility of synchronisation error and hence (unlike the case for adaptive p persistence), no risk of incorrect adaptive estimation leading to deadlock.

L_{crit} is important in determining stability (Section 6.4.1). If this model were slightly augmented to apply to an infinite population generating Poisson arrivals, instability for any non-zero arrival rate would result. In this case, however, the arrivals are not Poisson, rather they are periodic batches of N stations. Hence it is possible to envisage a situation where the mean CRI length is theoretically finite, but for all practical purposes leads to a near-zero throughput level. From Figure 51 it is clear that the advantage of the tree based scenarios is that there is no need to alter the value of any parameter (as is required for p persistence) as the batch size N within the cell changes.

It is also vital to consider the importance of the choice of tree order in the Q-ary tree based CRAs. Figure 52 shows the effect of changing the value of Q in the BAP. Values of Q=2 or Q=3 are optimal under the

BDM, with greater splitting in the tree following contention leading to suboptimal CRIs. Degraded performance is caused by waste in the leaves of the trees caused when a collision between a small number of contending stations in a single slot leads to the provision of *Q* additional slots for resolution. Where the number of contending stations is as low as two in the collision slot and the number of additional leaves generated on the tree by the collision is as large as six, there is significant waste of CMSs that might have been better used via the generation of a lesser number of additional leaves for each collision.

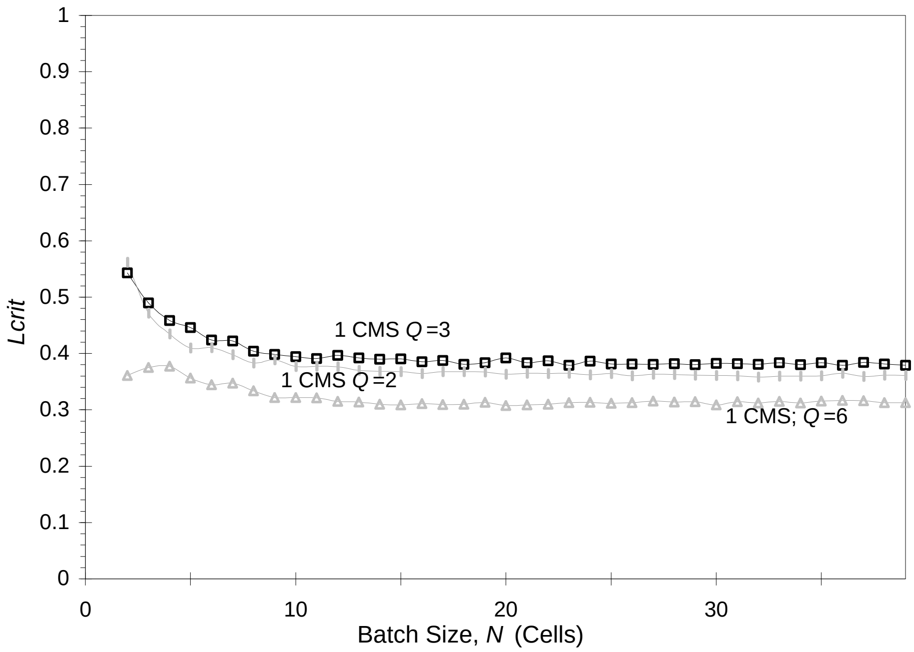

Figure 52. Effect of changing the value of *Q* in the BAP. Values of *Q*=2 or *Q*=3 are optimal, with greater splitting in the tree following contention leading to suboptimal CRIs.

Notably, this effect is offset somewhat by a lesser number of collisions required for an initial set of colliding MT requests to reach the leaves of the tree. Since there is more splitting at an earlier stage, it is clear that there will be fewer points of collision before resolution. Although

Figure 52 presents results only for the BAP case, the results presented in this dissertation and the wider literature [Bis96] show that this effect is common to all tree based CRAs.

6.5.7.1 Effect of Channel Signalling Schemes

In this section an example from the *Q*ary tree family is used to contrast efficiency with earlier presented *p* persistence results. msSTART was chosen as a readily available and implementable *Q*ary tree.

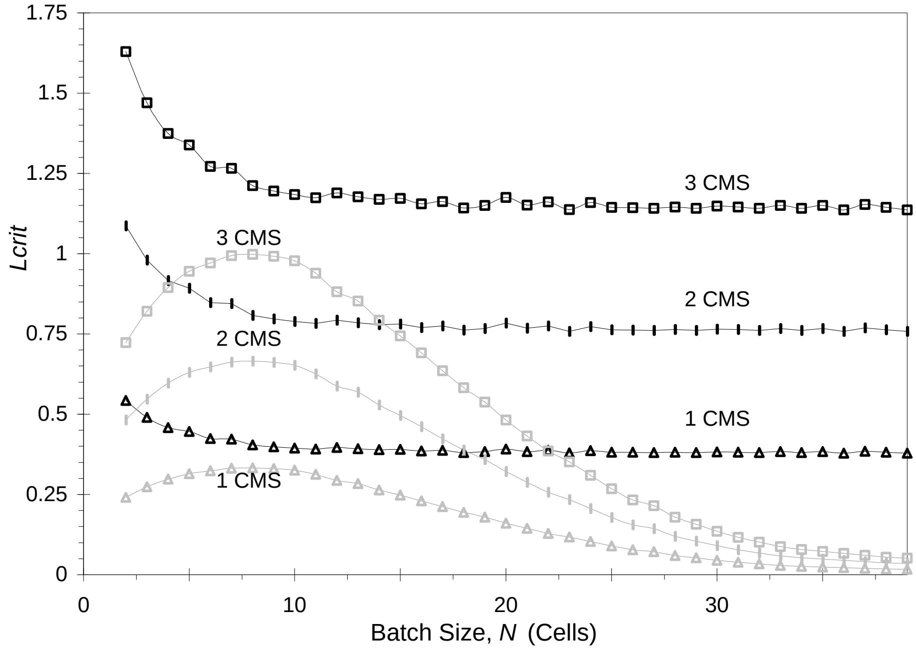

Figure 53. Effect of the number of CMSs – full CMS sharing scheme. The *p* persistence results are in grey and are all for p = 0.2. The msSTART results are in black.

Figure 53 demonstrates the effect of the number of CMSs per data slot using the full sharing CMS scheme for both *p* persistence (Eqs (38) and (56)) and msSTART (Eqs (54) and (56)). Increasing the number of CMSs dramatically increases the 'survivability' of the system but further reduces the maximum utilisation of the MAC protocol, decreasing the actual user data throughput capability.

In fact, doubling the number of CMSs has the effect of doubling L_{crit} for any number of MTs for all signalling schemes. Again it can be seen that for the fixed *p* persistence case, this signalling channel scheme is advantageous only for low numbers of MTs in a cell. For higher numbers of MTs, there is no significant stability increase. MsSTART, on the other hand, exhibits significant additional stability for any number of contending stations.

Figure 54 presents a comparison of performance results for msSTART and the BAP for the FCS signalling scheme (Eqs (54) and (56)). The relative performance of msSTART and the BAP remains unchanged for each signalling scheme studied in this chapter.

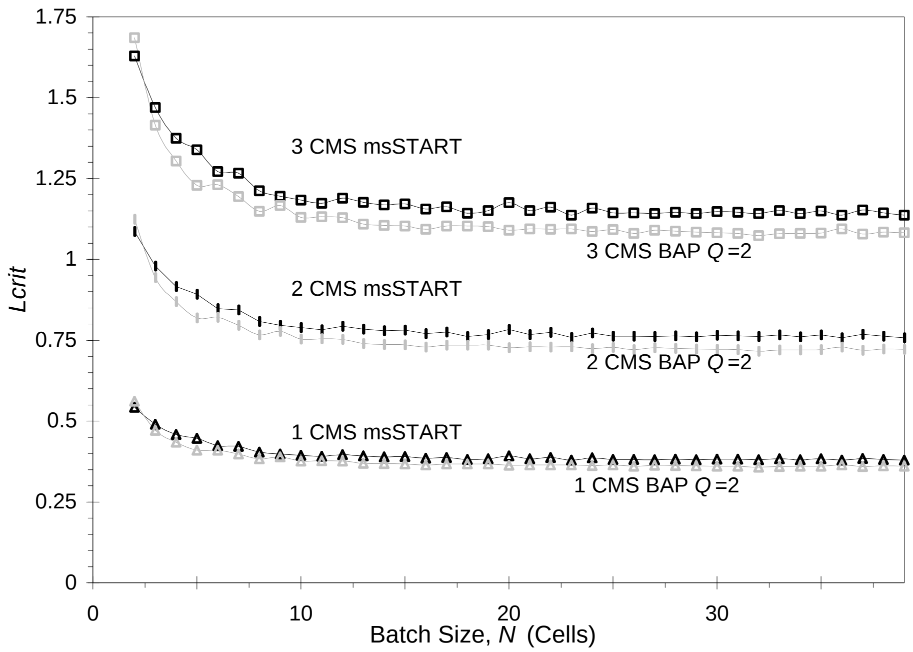

Figure 54. Comparison of performance results for msSTART and the BAP for the FCS signalling scheme. The relative performance of msSTART and the BAP remains unchanged for each signalling scheme studied in this chapter.

Multi-CMS Cyclic CMS sharing is considered next (Eqs (54) and (57) for msSTART and Eqs (38) and (57) for *p* persistence). This scheme requires greater control complexity than its fully shared counterpart because the S-MAC must assign the stations to subgroups. Despite this drawback, Figure 55 (with P_{err}=0 once again) shows that the *p* persistence CRA shows significant additional stability as a result. L_{crit} is maintained at a higher level for a larger number of contending stations, however, as described in Figure 39, performance can still be optimised by altering *p* with batch size *N*.

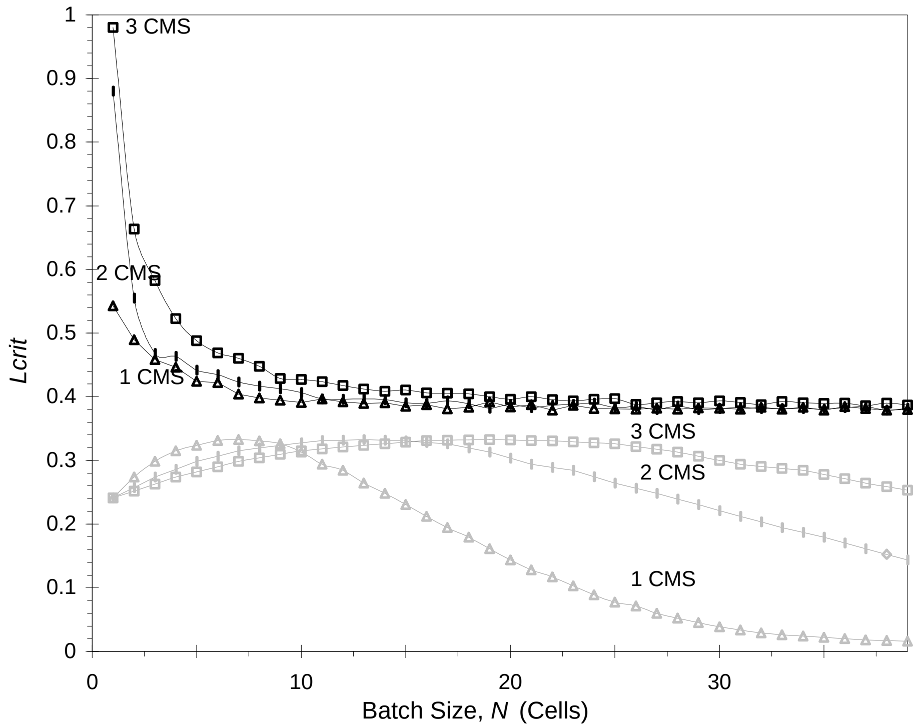

Figure 55. Effect of the number of CMSs in the msSTART CRA using the cyclic multi-CMS sharing scheme. Performance of the *p* persistence cases are shown in grey.

msSTART, on the other hand, is largely unaffected by the CCS_M signalling scheme due to its relative insensitivity to the initial number of contending stations in any contention group (see Figure 51). Improvements are marginal for this CRA and exist meaningfully only for situations in which there are small numbers of stations in a given cell. Gains correspond to the greater chance that, for the situation of a small number of stations, each station will be assigned a different CMS for transmission and hence will not require contention resolution at all. Again, the relative performance of msSTART, and its insensitivity to the number of MTs in the cell are a significant benefit.

Turning to a performance analysis of Single CMS Cyclic Sharing (Priority Access) it is important firstly to consider issues of priority implementation in a wireless MAC. In order to implement an effective priority access for a WATM MAC, it is necessary to use two mechanisms: first the head end or S-MAC must use a pre-emptive scheduler when allocating bandwidth to stations with different priorities; and second the MAC protocol must regulate collisions so that high priority stations are able to transmit bandwidth requests without interference from lower priorities. Some proposed multi-service MACs seek to merely add a Queue Identifier (QI) field in the CMS to indicate traffic priority level. The S-MAC then uses the QI as a priority scheduler for stations indicating high priority so a station transmitting a successful request for its priority traffic to the head end gains immediate access to the channel.

Noting that the time from MT request to channel allocation must be kept low, even during times of high congestion or following failure, it is possible to find serious drawbacks inherent in the use of a QI. One issue is that during contention resolution under the preceding signalling schemes, traffic requests from all stations are treated equally without regard for their priority but yet any bandwidth request must be received at the base station before the S-MAC can assign the request queueing priority in the allocation of the DSs. This situation means that newcomers can easily be blocked in the request phase for extended periods of time – a problem that may result in unacceptable delays for high priority traffic at peak periods.

It is clear that, if a high priority MT request is blocked from accessing the channel or suffers a high number of collisions from lower priority traffic, it cannot rely on the pre-emptive scheduler to receive low access delays. What is required in the multi-service wireless MAC is a means of giving higher traffic immediate access to the R-B Channel so requests can be rapidly received by the S-MAC at peak periods. The CCS_S signalling allocation scheme can achieve this effectively and efficiently in the multi-service multi-media environment.

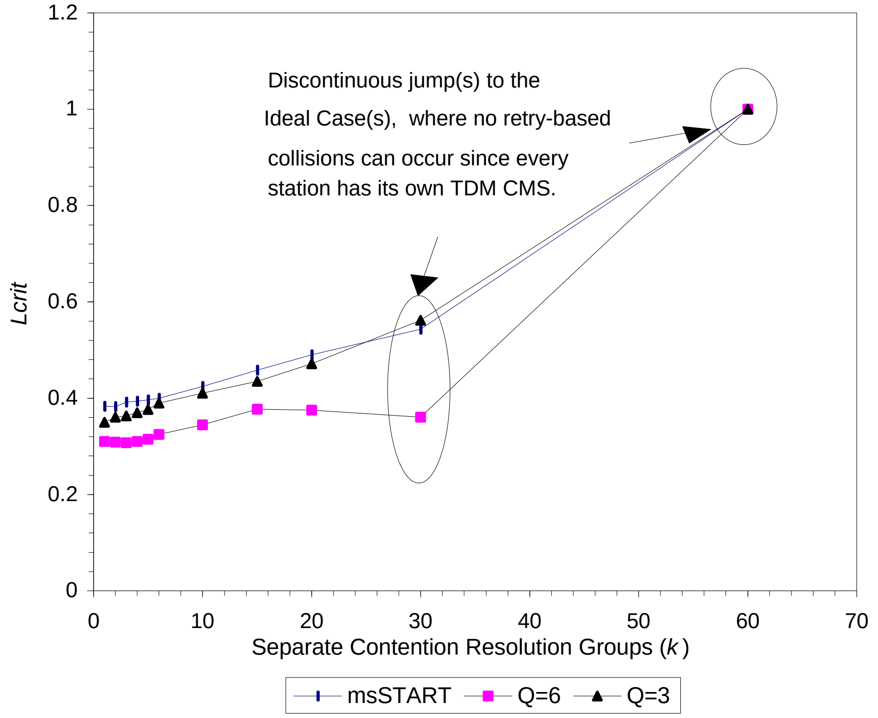

Figure 56. Simulation approximation of the performance of msSTART using single CMS cyclic sharing for a variety of separate contention resolution groups *k*. Performance is compared with BAP cases for *Q*=3 and *Q*=6.

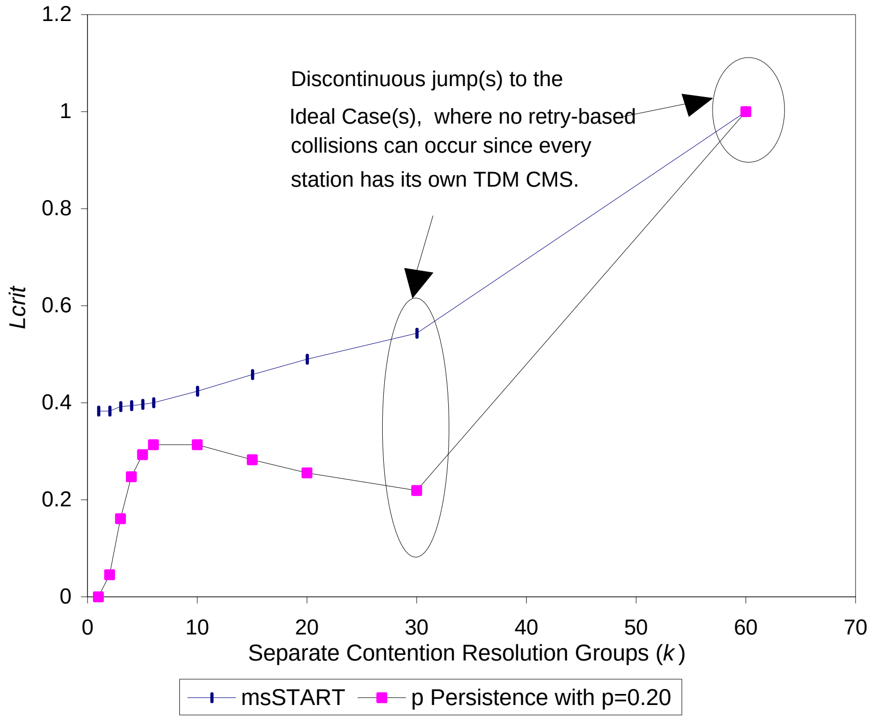

Figure 57. Simulation approximation of the performance of the BAP with Q=3 using single CMS cyclic sharing for a variety of separate contention resolution groups k, N=60, 1 CMS. Performance is compared with the p persistence case which is the lower line, p=0.20, all other details unchanged.

Figure 56 and Figure 57 show the effect of of the number of groups on the critical load for the CCS_S scheme. Although there is only one CMS per data slot, the access to this CMS is regulated in a TDMA-like manner so that k is similar to a TDM cycle length. Each station is given membership of a group with a certain position in the cycle and is allowed to access a CMS only at that position. Once again, it is worth noting the additional complexity required at the S-MAC and MTs to implement this scheme.

The example of a population of 60 MTs is taken, showing the effect of the CCS_S signalling channel scheme on both msSTART, a range of BAP CRAs, and the p persistence CRA. The stations transmit only in the group to which they are assigned membership and can only transmit in every k^{th} frame. It is assumed that stations are distributed in equal numbers in $k_1 \ldots k_{60}$ however it is important to note that in a real priority scheme, station population would be weighted toward the lower priority groups. Alternately, priority can be implemented by awarding different numbers of FCS channels to each of the k separate contention resolution groups to provide lesser user delay periods and greater stability for greater priority traffic.

In practice, very large values of k introduce unacceptable individual user delay despite the additional stability of the system. From Figure 56, it is clear that the best case BAP scheme of Q=3 performs in a similar fashion to msSTART using our simulation approximation. We note, however, that other choices of the parameter Q, with the exception of Q=2, can cause significant performance degradation.

It is clear from Figure 56 and Figure 57 that the CCS_S scheme shows promise as a method to avoid deadlock under correlated requests in the p persistence case. For an aggressive value of p=0.2 and a large number of stations, implementation of this TDM-like division of the single CMS in the frame produces reasonable stability for a small number of contention resolution groups. This result in the p persistence case is heavily dependent on the even distribution of the 60 MTs between the k contention resolution groups. msSTART, on the other hand, provides significantly superior performance over all k and is far less dependent on the distribution of the N contending stations.

As discussed in Section 6.4.1, it has been found that the TDM technique implicit within CCS_S is good for alleviating the signalling congestion created by the extreme inter-station correlation (large and periodic batches of simultaneous arrivals) which we are studying. However, under normal conditions (when the requests are not generated just by single-cell messages, which in turn are not deterministically arriving in correlated periodic batches any more, and are more spread in time), one can see that such a TDM technique introduces an amount of unnecessary increase in the average access delay. Hence, a trade-off between cost and benefit exists, whose balance depends strongly on the traffic profile.

6.5.7.2 Effect of Errors

The effect of errors on the performance of the signalling schemes and CRAs studied in this chapter is now considered. In all cases the effect of errors on the system is small as the systems do not distinguish between collision and error. Further, it is concluded that relative performance of the CRAs is not changed by the introduction of errors to the channel – even where those errors are increased to levels capable of causing system failure.

In the case of the Q-ary tree, errors are significant only when they occur in a slot in which there would otherwise have been a successful transmission. In this situation, the performance penalty is an additional node in the tree that would have been absent without the error. The difference corresponds to an additional Q slots for the resolution of the tree.

For p persistence on the other hand, error is also significant only in situations where the absence of error would otherwise have allowed a successful transmission (i.e. there would not have been a collision had there been no error). In this situation, the performance penalty is an additional q slots where q depends on the probability of collision with another MT on a repeat transmission attempt. The two CRAs show similar relative performance up to unrealistically high levels of error.

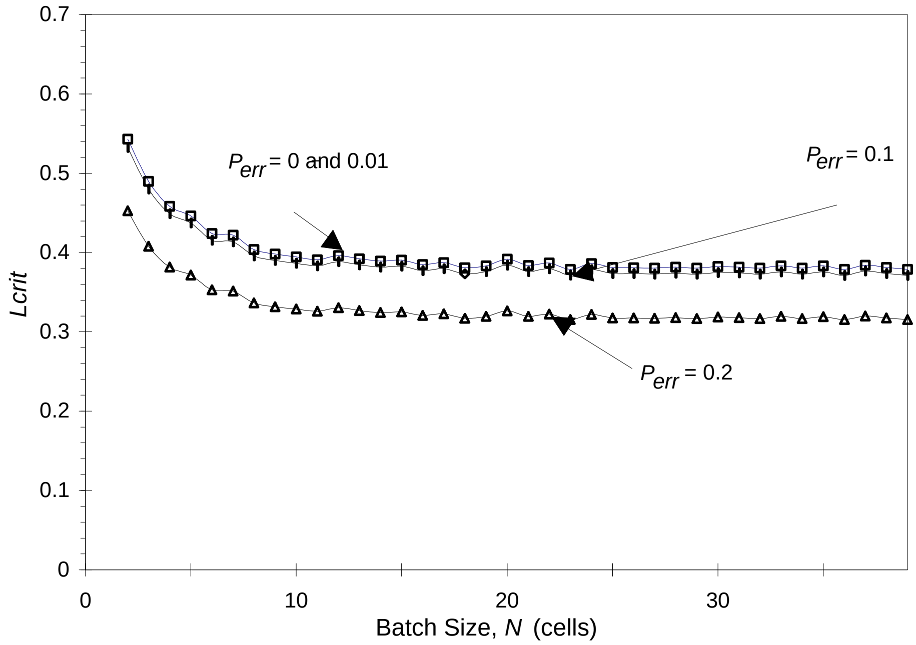

Figure 58. CMS error sensitivity for the msSTART single CMS scheme (Q = 3).

Figure 40 presents error sensitivity for the *p* persistence scheme and Figure 58 gives the same for msSTART. Both CRAs are quite insensitive to reasonable levels of noise in the wireless channel. Error probabilities as high as 0.1 give only marginal (15%) reductions in signalling channel performance. Moreover, the CRAs are relatively insensitive to the signalling scheme used.

6.5.8 *Effect of Piggybacking*

In Section 6.5.5, methods for piggybacking requests in the BDM_P model were outlined. In communicating bandwidth requirements to the S-MAC, the MTs may either make a request for bandwidth during the R-B Control which employs a CRA, or they may add the request to the end of a previously allocated data frame – thus 'piggybacking' the request. A MT with data in a recent previous slot need not participate in the R-B CRA, however, any MT that has just entered the cell, been switched on, or was been silent in a previous frame, will need to compete with the other stations in the B-R region to reserve bandwidth for transmission of data in the later collision free DMSs.

Under the BDM_P model, correlated bursts of traffic are sent by the MTs. All of the MTs will thus enter State 2 (Section 6.5.5) in the second CMS except where the MT population in the cell is less than 2. When an MT is able to send its request for bandwidth in a contention free CMS (and hence the R-B request is resolved), it enters State 3 and transmits its data in the collision free DMS, piggybacking requests for further bandwidth as required to the end of the DMS. When the MT becomes idle, it ceases to request DMSs in future TDMA frames and returns to State 1.

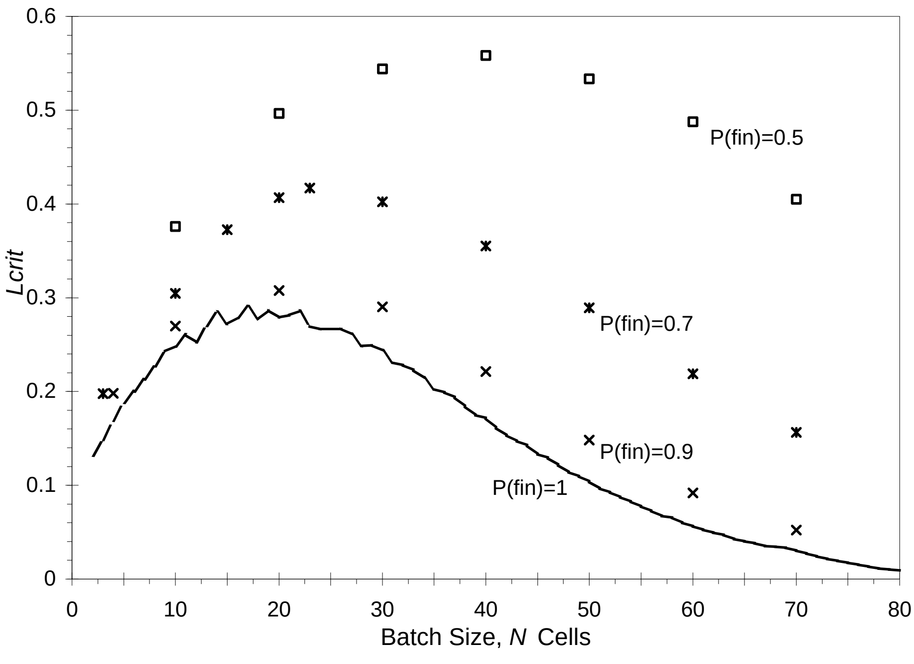

Figure 59. Effect of piggybacking on the performance of *p* persistence using the BDM_P.

The speed with which the MT moves from State 3 to State 1 depends on the volume of traffic appearing at the MT for transmission to the HC as well as the burstiness of that traffic. As foreshadowed earlier, we model this variability using P(*fin*) – the probability that an MT will return back to State 1 during the time period T_c. A higher P(*fin*) indicates burstier traffic or lesser background traffic.

Figure 59 demonstrates the gain in L_{crit} as a result of including piggybacking, particularly for MTs with less bursty traffic flows. Note that the burstiness of the traffic flowing following the initial MT request changes the point of optimum L_{crit} for the algorithm, meaning that efficiency of a MAC using *p* persistence CRA is heavily dependent both on the number of stations in the system and *also on the characteristics of the traffic offered by the MTs*. This later point is important. The difficulty in optimising the *p* persistence algorithm is quantified in Figure 59, demonstrating that a two dimensional real-time adjustment to the *p* parameter is required to maximise system efficiency. The dimensions of change are: the burstiness of presented traffic; and the number of MTs present in any cell at any time.

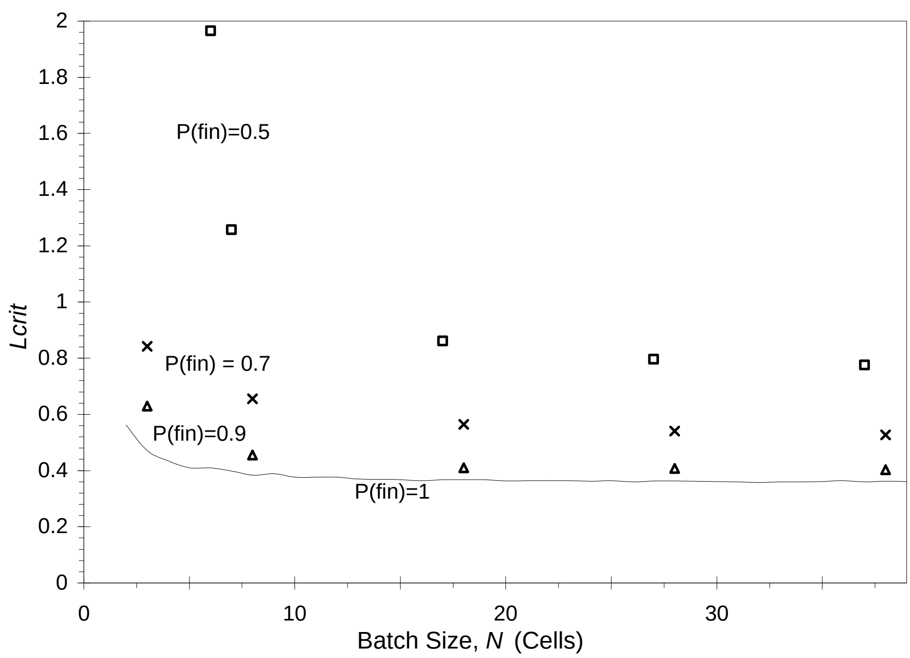

Figure 60. Effect of piggybacking on the performance of msSTART using the piggybacking enhancement to the BDM.

For the case of msSTART, it is clear that piggybacking gives higher relative efficiency gains for less bursty traffic for the overall performance of the MAC under BDM_P than that offered by *p* persistence (Figure 60). Importantly, it is further noted that this gain is (with the exception of a high peak for L_{crit} for very low numbers of MTs) relatively independent of the number of stations in the cell. Hence msSTART is less sensitive to the dimension of presented traffic than *p* persistence, as well as being insensitive to the number of MTs present in the cell except at low numbers.

6.5.9 Conclusions

Although not exhaustive, Chapter 6 has studied contenders for the R-B Control region of a modern multi-service MAC protocol. Motivated by the need to consider the performance of networks under failure recovery conditions and to minimise the impact on the network of transient traffic patterns caused by such events, the chapter evaluates the performance of msSTART, *p* persistence and other CRAs, using the novel Basic Deadlock Model, the Binomial Deadlock Model as well as the Basic Deadlock Model with alterations allowing for the piggybacking of requests.

The chapter provides a comprehensive set of analytical and simulation results which are used to demonstrate that msSTART provides superior performance in the wireless medium to both the fixed and dynamic *p* persistence cases. Further, it is demonstrated that the introduction of an effective priority scheme does not have a significant impact on the stability of the *Q*-ary Tree based CRA.

The three signalling channel schemes provide insight into the stability of the wireless MAC after the implementation of priority for different traffic classes and comparative results for evaluation of msSTART and *p* persistence are provided under what the IEEE 802.14 working group has termed the 'disaster scenario'. Of the three schemes evaluated, the FCS sharing scheme employing multiple CMSs per data region extends the protocol's usable load region the furthest.

The overall result is that *Q*-ary tree CRAs (in particular msSTART) are better adapted to the wireless environment, providing better guarantees of non-case sensitive performance than all other considered methods.

7. CONCLUSIONS

This dissertation is an investigation of traffic engineering issues in the design and development of multi-media wireless networks systems. Through comprehensive modelling of the operation of these systems under a variety of traffic flows, guidelines have been provided for multi-media wireless networks network provisioning and survivability. Using research results presented here, the chapters have established principles of multi-media wireless networks network dimensioning, leading to the development of specific guidelines on how to perform cost effective design and dimensioning of future wireless multi-service access-networks. The work has addressed a fundamental question of the best strategy to minimise the amount of bandwidth required for transmission over an access multi-media wireless networks link, subject to meeting specified Quality of Service (QoS) requirements, for realistic traffic conditions.

Specifically, the following areas have been examined in detail:

- Layering issues in the design and performance specification of wireless access networks with particular focus on inter-layer interactions;
- MAC architectures and multi-access techniques;
- A model for performance specification across layers in a model wireless access scheme for both transparent and non-transparent transmission;
- Suitability of CRAs for the R-B Control region of a prototype wireless MAC; and
- Disaster models for analysis of stability and efficiency under startup conditions for a range of CRAs, complete with analysis of the effects of CMS error, traffic changes and signalling scheme alterations on the models.

The following sections highlight the major results obtained in each part of the dissertation.

7.1 Layering Issues

Although layering has remained very important as a simplifier of telecommunications processes, a more inclusive approach is essential to optimise globally transmission parameters and link dimensioning. This observation is used throughout the dissertation to consider performance-related interactions between the different layers: physical, MAC, as well as data link, and transport layers. This approach has complemented other research on physical layer coding, particularly Turbo Coding, as well as work on ARQ enhancements. In each of these two bodies of research, a local optimisation has been used to improve the performance of an OSI layer separately, however it is demonstrated here that optimising one layer may interfere with the performance of the other. Hence this work has provided dimensioning guidelines in a system able to capture the global effects of change at any OSI layer level.

7.2 MAC Design

Another important part of the thesis centres on the general area of analysis and synthesis of MAC wireless protocols. A wireless MAC protocol is synthesised using the above mentioned global optimisation results to optimise transmission resource allocation.

MAC structures capable of dynamically varying the amount of capacity allocated to each user were analysed keeping in mind a dynamic alteration in net capacity required by the real-time adjustment of the physical layer HLM/FEC as well as higher layer ARQ retransmission. Further, later chapters of the thesis provided important stability and efficiency results to be consulted when determining the CRA for use in the R-B Control region of a MDR-TDMA MAC.

7.3 Inter-layer Performance Optimisation

In Chapter 3 simulations were used to develop a model designed to show a comparison between FEC, ARQ and Modulation Gain in an error prone environment so that a limit on the use of FEC could be established. The results from that work show that, in many instances, an adaptive modulation system utilising small block ARQ over the wireless link exhibits optimal EBE without FEC over wide SNR ranges.

Findings from the chapter determine the level of preference that should be given to systems which adaptively change their parameters according to the instantaneous level of interference and noise. It was demonstrated that optimum performance under many conditions can be delivered by a system using packet by packet adaptive Modulation Optimisation rather than focussing on FEC based solutions.

Chapter 4 provided critical review of a number of suggestions for solving issues of inter-layer interaction between the physical error process of a wireless access network and the error recovery systems of the higher layer TCP/IP. It recommended the use of an adaptive DLC (AAL) ARQ where payload sizes are set dynamically using a lookup chart. Results showed that the addition of this subsystem has the dual advantages of: an efficiency increase to the overall system regardless of the nature of the higher OSI layers; as well as providing a solution to the ‘slow start’ problem in TCP/IP over wireless. It does so, however, at the cost of an additional layer of complexity.

7.4 CRA Models

In any wireless network, atmospheric and other interference effects lead to transitory periods in which the system cannot transmit data. Hence the recovery of the system after network ‘failure’ is a central issue. A related issue is the transient or nonstationary congestion period triggered by the backlog of packets due to retransmission after a failure or sustained period of low transmission capability. The dissertation has focussed attention on the survivability of proposed systems under deadlock conditions, including periods of extreme load and traffic correlation as numbers of remote stations attempt to retransmit after failure of the wireless link.

Adopting an approach to analysis incorporating three novel deadlock models, the thesis has reported on the performance of a tree based and p persistence CRAs subject to extreme inter-station correlation. Using three signalling channel capacity allocation schemes and tree based algorithms for the resolution of contention in the remote to base control section of the MAC, Chapter 6 develops a comprehensive CRA dimensioning guide for multi-service medium access protocols employing a form of multi-slot Q-ary tree algorithm. The results of Q-ary tree CRAs under the various disaster scenarios suggest a strong preference for their inclusion in the R-B Control signalling mechanisms of future wireless multi-media multi-access MACs.

7.5 Further Work

There is a significant amount of work remaining in the comprehensive fields that are touched upon in this thesis. By taking a global approach to performance modelling and attempting to capture the interactions of optimisations across system layers, important questions of quantitative analysis are opened in many remaining areas of research. Indeed, it is this breadth of techniques and protocols throughout different OSI layers, each interacting with overall system performance, that will remain the main challenge of this form of traffic engineering. A further understanding of interactions studied in this dissertation has the potential to bring great gains to efficiency in multi-service wireless access networks.

In the specific approach that this thesis takes, there is further work remaining in the development of more comprehensive extensions of the deadlock and inter-layer efficiency models presented in the preceding chapters, models that are able to capture even more of the nuances of the wireless system.

It is my hope that the lasting broader aspects of this dissertation will remain the adoption of the worst-case deadlock models developed in this work for 'survivability' analysis, as well as a continuing focus applied to globalising optimisation strategies for wireless networks.

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