Resource Management for Cross Layered Star and Mesh Networks

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ABSTRACT

Resource Management for Cross Layered Star and Mesh Networks
Ayda Basyouni, Ph.D.
Concordia University, 2008

The use of wireless services is rapidly spreading around the world and many of the world population no longer know how to cope without their cell phones; the feel of always being connected offers a great sense of flexibility and security. So far, voice has been the primary wireless application. However, with the Internet continuing to influence our daily lives, the demand for wireless data is extensively increasing. Already, in the countries that have cellular-data services readily available, the number of cellular subscribers taking advantage of data services has reached significant proportions.

In this thesis, we investigate resource management techniques for cross layered star and mesh wireless data networks. In particular, we investigate several aspects related to resource management techniques over the reverse packet data channel in cdma2000 1xEV star networks. We provide an upper bound for the reverse packet data channel throughput as a function of the number of mobile stations that are allowed to transmit instantaneously on each time slot. We also provide a lower bound for the average sector throughput based on the number of users per sector and propose several autonomous rate assignment, and scheduling techniques that provide a significant throughput improvement relative to other published techniques.

We also develop analytical models for lowest-rate-first, highest-rate-first priority scheduling techniques, and two round-robin fair scheduling techniques over
the reverse data channel in cdma2000 1xEV star networks. For these four scheduling techniques, the distribution of the mobile stations among the possible data rates is modelled as a Markov process. An analytical expression for the steady state system throughput is derived from the steady state distribution of the above Markov process. The above model is extended to evaluate the performance of the cross layered design between the hybrid ARQ, rate assignment, and time slot scheduling over the reverse packet data channel in cdma2000 1xEV. Expressions for the steady state system throughput and file transmission delay are derived from the steady state distribution of this model.

Fountain codes are a class of erasure codes with the property that a potentially limitless sequence of encoding symbols can be recovered from any subset of size equal to or slightly larger than the number of source symbols. In this thesis we relate the parameters of the higher layer Fountain codes to those of the physical layer codes to present a cross layered coding technique. Based on the bit error rate of physical layer codes, the upper layer Fountain code’s parameters are designed to obtain a prespecified performance. The performance of the proposed cross layered coding technique is found to be comparable to that of physical layer based Hybrid ARQ technique. As an application we studied the performance of WiMAX backhaul mesh networks where cashing is allowed at the service stations and proposed cross layered coding technique is used as its main coding scheme. In particular, the performance of the above network is modelled as a Markov process and analytical expressions for the steady state system performance are derived from its associated steady state distribution.

All the above analytical models are validated through simulations.
To my Father, who has been dreaming about this day since I was a child.

To my Mother, may Allah bless her.
Acknowledgement

I would like to express my sincere gratitude and thanks to my supervisors, Dr. Ahmed Elhakeem, and Dr. Anjali Agarwal for their valuable support, enlightening guidance, and sincere encouragement throughout the course of this work.

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Words fall short of expressing my love, appreciation, and gratitude for my parents in-law, family and friends for being always there for us, and giving us the hand whenever we needed it. Without their love, encouragement and support this work would have never been possible.

I also wish to acknowledge the financial support provided by the Natural Sciences and Engineering Research Council of Canada (NSERC), and by Concordia University.
TABLE OF CONTENTS

List of Tables xi

List of Figures xiii

1 Introduction 1

1.1 Motivations and Objectives 2

1.2 Outline 6

1.3 Standards 7

1.3.1 The cdma2000 1xEV 7

1.3.1.1 The cdma2000 1xEV-DO 8

1.3.1.2 The cdma2000 1xEV-DV 9

1.3.2 WiMAX 14

1.4 Literature Review 16

1.4.1 Rate Assignment 18

1.4.2 Scheduling Techniques 22

2 Resource Management Techniques for cdma2000 1xEV Star Networks 26

2.1 RoT Constraint 27

2.2 Theoretical Throughput Bounds 29

2.2.1 Upper Bound 30

2.2.2 Lower Bound 31

2.3 Proposed Rate Assignment Techniques 33
2.3.1 FPR Rate Assignment Technique .................................. 36
2.3.2 EPR Rate Assignment Technique .................................. 36
2.4 Proposed Scheduling Techniques ................................. 40
  2.4.1 Scheduling Process: Base Station Model .................. 40
  2.4.2 Scheduling Process: Mobile Station Model ............... 41
  2.4.3 Formulated Problem ......................................... 42
2.5 Analysis and Results .................................................. 43
  2.5.1 Proposed Rate Assignment Techniques ................... 45
  2.5.2 Proposed Scheduling Techniques ......................... 47
  2.5.3 Proposed Scheduling and Rate Assignment Techniques ... 50
2.6 Summary ................................................................. 53

3 Analytical Models for Scheduling Techniques in cdma2000 1xEV Star Networks ................................. 56
  3.1 Scheduling Techniques Models ................................... 57
    3.1.1 Lowest-Rate-First ....................................... 58
    3.1.2 Highest-Rate-First ...................................... 59
    3.1.3 Round-Robin Scheme 1 .................................... 60
    3.1.4 Round-Robin Scheme 2 .................................... 61
  3.2 Scheduling Process Model ......................................... 63
    3.2.1 Analysis and Simulation Results ........................ 67
  3.3 Cross Layered Design Model .................................... 70
1.1 Traffic to pilot ratio for 1xEV-DO data channel, channel BW = 1.25 MHz .......................... 9

1.2 Traffic to pilot ratio for 1xEV-DV R-PDCH, channel BW = 1.25 MHz 10

1.3 Fairness criterion .................................................. 14

1.4 Allowable modulation techniques ................................. 16

1.5 Wimax parameters [64] ............................................. 17

2.1 Values used for the simulation parameters ....................... 29

2.2 Rate distribution for case 1 ...................................... 34

2.3 Proposed file size thresholds in Kbyte for FTP traffic, FPR and EPR schemes (Case A) .................. 38

2.4 Proposed file size thresholds in Kbyte for FTP traffic, FPR and EPR schemes (Case B) .................. 39

4.1 HTTP traffic model parameters .................................. 102

A.1 Output punctured code for cdma2000 1X encoder ............. 132
A.2 States for the cdma2000 1X encoder ................. 138
<table>
<thead>
<tr>
<th>Figure</th>
<th>Description</th>
<th>Page</th>
</tr>
</thead>
<tbody>
<tr>
<td>1.1</td>
<td>An example of star network</td>
<td>4</td>
</tr>
<tr>
<td>1.2</td>
<td>An example of mesh network</td>
<td>4</td>
</tr>
<tr>
<td>1.3</td>
<td>Transmission process on R-PDCH</td>
<td>11</td>
</tr>
<tr>
<td>1.4</td>
<td>H-ARQ on the R-PDCH</td>
<td>11</td>
</tr>
<tr>
<td>1.5</td>
<td>Data time frame of $T_f$ msec</td>
<td>18</td>
</tr>
<tr>
<td>2.1</td>
<td>Upper and lower throughput bounds for cdma2000 1xEV-DV</td>
<td>32</td>
</tr>
<tr>
<td>2.2</td>
<td>Simulation flowchart for the results in Section 2.5</td>
<td>46</td>
</tr>
<tr>
<td>2.3</td>
<td>Rate assignment techniques, channel BW 1.25 MHz</td>
<td>48</td>
</tr>
<tr>
<td>2.4</td>
<td>Cumulative density function (Case A and B) with thirty MSs per sector</td>
<td>49</td>
</tr>
<tr>
<td>2.5</td>
<td>Scheduling techniques, channel BW 1.25 MHz</td>
<td>51</td>
</tr>
<tr>
<td>2.6</td>
<td>Cumulative density function (Case A and B) with thirty MSs per sector</td>
<td>52</td>
</tr>
<tr>
<td>2.7</td>
<td>EPR rate assignment technique (Case B) with different scheduling algorithms, channel BW 1.25 MHz</td>
<td>54</td>
</tr>
</tbody>
</table>
2.8 EPR rate assignment technique (Case B) (Cumulative density function with thirty MSs per sector) ........................................ 55

3.1 Round Robin Scheme 1 Fair Scheduler ..................................... 62

3.2 Performance Results for cdma2000 1xEV-DO reverse data channel .. 69

3.3 H-ARQ Probability Tree .......................................................... 72

3.4 Steady state link throughput for lowest-rate-first scheduling technique. ................................................................. 76

3.5 Steady state average file transfer time per user for lowest-rate-first scheduling technique. ........................................... 77

3.6 Steady state link throughput for highest-rate-first scheduling technique. ............................................................... 77

3.7 Steady state average file transfer time per user for highest-rate-first scheduling technique. ........................................ 78

4.1 An example of backhaul mesh network ..................................... 80

4.2 Delays due to propagation and queueing over number of links ... 87

4.3 Probability of file acceptance $p_f$ .......................................... 89
4.4 Efficiency of H-ARQ technique $\eta_{HA}$ .............................................. 90

4.5 Efficiency of cross-layer coding technique $\eta_{CL}$ ................................. 91

4.6 Delay of H-ARQ technique in number of required time frames to transmit a file $D_{HA}$ ................................................................. 91

4.7 Delay of cross-layered coding technique in number of required time frames to transmit a file $D_{CL}$ ......................................................... 92

4.8 Average BW per request. $R_{max} = 90, L_{max} = 4, p_{s1} = 0.8, \delta = 10^{-5}$,
and $Z_{SS} = 9$ ...................................................................................... 104

4.9 Average BW per request. $p_{miss} = 0.9, p_{s1} = 0.8, L_{max} = 4, \delta = 10^{-5}$,
and $Z_{SS} = 9$ ...................................................................................... 105

4.10 Average number of packets per request. $L_{max} = 4, \delta = 10^{-5}, p_{s1} =$
$0.8$, and $Z_{SS} = 9$ ........................................................................... 105

4.11 Average throughput per request. $R_{max} = 90, L_{max} = 4, \delta = 10^{-5}$,
and $Z_{SS} = 9$ ...................................................................................... 106

4.12 Average throughput per request for BS. $p_{miss} = 0.9, L_{max} = 4, \delta =$
$10^{-5}, p_{s1} = 0.8$, and $Z_{SS} = 9$ ....................................................... 106

4.13 Steady state distribution. $R_{max} = 90, L_{max} = 4, \delta = 10^{-5}, p_{s1} = 0.8$,
and $Z_{SS} = 9$ ...................................................................................... 107
4.14 Steady state distribution. \( p_{\text{miss}} = 0.9, L_{\text{max}} = 4, \delta = 10^{-5}, p_{s1} = 0.8, \)
and \( Z_{SS} = 9 \) .................................................. 108

A.1 cdma2000 1X turbo encoder .................................. 131
A.2 State diagram for cdma2000 1xEV encoder with \( R_c = 1/5 \) .... 137
A.3 State diagram for cdma2000 1xEV encoder with \( R_c = 1/3 \) .... 139
A.4 Lower bounds for the probabilities of packet acceptance .... 141
List of Acronyms

<table>
<thead>
<tr>
<th>Acronym</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>1xEV-DV</td>
<td>Evolution-Data Voice</td>
</tr>
<tr>
<td>1xEV-DO</td>
<td>Evolution-Data Optimization</td>
</tr>
<tr>
<td>1xRTT</td>
<td>Single Carrier Radio Transmission Technology</td>
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<tr>
<td>3G</td>
<td>Third Generation</td>
</tr>
<tr>
<td>4G</td>
<td>Forth Generation</td>
</tr>
<tr>
<td>3GPP2</td>
<td>Third Generation Partnership Project 2</td>
</tr>
<tr>
<td>8PSK</td>
<td>8 Phase Shift Keying</td>
</tr>
<tr>
<td>ACK</td>
<td>Acknowledgement</td>
</tr>
<tr>
<td>BW</td>
<td>Bandwidth</td>
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<tr>
<td>BS</td>
<td>Base Station</td>
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<tr>
<td>BPSK</td>
<td>Binary Phase Shift Keying</td>
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<tr>
<td>CDF</td>
<td>Cumulative Density Function</td>
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<td>CDMA</td>
<td>Code Division Multiple Access</td>
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<tr>
<td>CRC</td>
<td>Cyclic Redundancy Code</td>
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<tr>
<td>DL</td>
<td>Down Link</td>
</tr>
<tr>
<td>DRC</td>
<td>Data Rate Control Channel</td>
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<tr>
<td>DS/CDMA</td>
<td>Direct Sequence CDMA</td>
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<tr>
<td>FL</td>
<td>Forward Link</td>
</tr>
<tr>
<td>F-PDCH</td>
<td>Forward Packet Data Channel</td>
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<tr>
<td>F-SCH</td>
<td>Forward Supplemental Channel</td>
</tr>
<tr>
<td>F-GCH</td>
<td>Forward Grant Channel</td>
</tr>
<tr>
<td>F-ACKCH</td>
<td>Forward Acknowledgement Channel</td>
</tr>
<tr>
<td>F-RCCH</td>
<td>Forward Rate Control Channel</td>
</tr>
<tr>
<td>H-ARQ</td>
<td>Hybrid Automatic Repeat Request</td>
</tr>
<tr>
<td>HSDPA</td>
<td>High Speed Down-Link Packet Access</td>
</tr>
<tr>
<td>IS-95</td>
<td>Interim Standard 95</td>
</tr>
<tr>
<td>Kbps</td>
<td>kilo bit per Second</td>
</tr>
<tr>
<td>Mbps</td>
<td>Mega bit per Second</td>
</tr>
<tr>
<td>MHz</td>
<td>Mega Hertz</td>
</tr>
<tr>
<td>MS</td>
<td>Mobile Station</td>
</tr>
<tr>
<td>MAC</td>
<td>Medium Access Control</td>
</tr>
<tr>
<td>MSIB</td>
<td>Mobile Status Indicator Bit</td>
</tr>
<tr>
<td>NAK</td>
<td>Not Acknowledged</td>
</tr>
<tr>
<td>Acronym</td>
<td>Definition</td>
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<tr>
<td>OFDM</td>
<td>Orthogonal Frequency Division Multiplexing</td>
</tr>
<tr>
<td>OFDMA</td>
<td>Orthogonal Frequency Division Multiple Access</td>
</tr>
<tr>
<td>QoS</td>
<td>Quality of Service</td>
</tr>
<tr>
<td>QPSK</td>
<td>Quadrature Phase Shift Keying</td>
</tr>
<tr>
<td>QAM</td>
<td>Quadrature Amplitude Modulation</td>
</tr>
<tr>
<td>RL</td>
<td>Reverse Link</td>
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<tr>
<td>R-PDCH</td>
<td>Reverse Packet Data Channel</td>
</tr>
<tr>
<td>R-SCH</td>
<td>Reverse Supplemental Channel</td>
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<tr>
<td>R-PDCCH</td>
<td>Reverse Packet Data Control Channel</td>
</tr>
<tr>
<td>R-REQCH</td>
<td>Reverse Request Channel</td>
</tr>
<tr>
<td>RRI</td>
<td>Reverse Rate Indicator</td>
</tr>
<tr>
<td>RoT</td>
<td>Rise Over Thermal</td>
</tr>
<tr>
<td>SINR</td>
<td>Signal to Noise and Interference Ratio</td>
</tr>
<tr>
<td>TD</td>
<td>Time Division</td>
</tr>
<tr>
<td>TDM</td>
<td>Time Division Multiplexing</td>
</tr>
<tr>
<td>TDD</td>
<td>Time Division Duplex</td>
</tr>
<tr>
<td>TCP</td>
<td>Transmission Control Protocol</td>
</tr>
<tr>
<td>UMTS</td>
<td>Universal Mobile Telecommunications System</td>
</tr>
<tr>
<td>UDP</td>
<td>User Datagram Protocol</td>
</tr>
<tr>
<td>UL</td>
<td>Up Link</td>
</tr>
<tr>
<td>WiMAX</td>
<td>Worldwide Interoperability for Microwave Access</td>
</tr>
<tr>
<td>WCDMA</td>
<td>Wideband CDMA</td>
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Chapter 1

Introduction

Ever since people wanted to talk to other people who were out of earshot, there have been attempts at communication. These attempts have advanced from carrier pigeons, signalling towers, smoke signals through telegraph, telephone, radio and satellite. Technology has advanced to allow TV, Internet, high speed data transmission. During the past years, there has been a quickly rising interest in radio access technologies for providing mobile as well as fixed services for voice, video and data. The 3G/4G technologies such as cdma2000 1xEV, WCDMA, Wi-Fi and WiMAX enable network operators to offer users a wider range of more advanced services while achieving greater network capacity through improved resource management. The semi-annual wireless industry survey published by CITA at the end of December 2007 [1] shows the growth in the wireless industry. At the end of 2007, the number of wireless subscribers had reached almost 256 million with an increase of 22 million subscribers since the end of 2006. The total twelve-month revenues reached more than 138 billion dollars in 2007.
1.1 Motivations and Objectives

Mobile systems have recently evolved from being simple devices used for voice communications to systems that support new features such as data transmissions, video streaming, and Internet access. The capacity demands to handle these new services have led to a new era of wireless technologies, which resulted in a huge cellular market all over the world. The providers of this technology have to offer attractive services that are reasonably priced. Efficient resource management and standards that provide high data rates are the tools used by those providers to ensure quality of service (QoS) with affordable prices.

QoS could be satisfied by different resource management means such as, call admission control [2]-[12], power/rate control [13]-[31], and scheduling [32]-[45]. With the new era of wireless technologies, the need for dynamic resource allocation and optimized scheduling techniques arises in order to efficiently use the available bandwidth (BW). Packet delays, user and sector throughputs and fairness between users, are also important issues to investigate. The performance of the backhaul networks is also considered to be an important factor in the overall network performance. Recently the cross layer design [67]-[74] was adopted to fully optimize wireless networks. Rate, power and coding at the physical layer can be adapted to meet the requirements of the applications given the current channel and network conditions. In cross layer design knowledge is shared between layers to obtain the best performance.
In wireless networks a star topology consists of one central base station (BS), relay, or wireless modem and several end nodes such as mobile stations (MSs), and computers (see Fig. 1.1). All the end nodes are connected directly to the central hub. In mesh networks each node could be connected to other nodes directly or through multiple hubs (see Fig. 1.2). Fully connected mesh network is a network where all the nodes are connected to each other. In mesh networks a routing algorithm is applied to determine the best route from any node to any other node.

In the last few years wide range of wireless technologies were standardized. Throughout this thesis we consider CDMA-based standards as a star network. In particular, we studied cdma2000 1xEV-DV (evolution-data voice) revisions C and D [47] [48], and single carrier evolution data-optimization (1xEV-DO) [49] star networks. We also consider the IEEE 802.16e worldwide interoperability for microwave access (WiMAX) [50] as a backhaul mesh network. It is important to mention that WiMAX standard supports both star and mesh topologies, however in this thesis we only consider the mesh topology.

The objectives of this thesis is to improve the resource management techniques associated with the cdma2000 1x reverse link data channel. We also study the improvements in the performance of WiMAX backhaul network. In particular, we provide an upper bound for the reverse packet data channel throughput as a function of the number of mobile stations that are allowed to transmit instantaneously on each time slot. We propose several rate assignment, and scheduling
Figure 1.1: An example of star network

Figure 1.2: An example of mesh network
techniques that provide a significant throughput improvement relative to other published techniques in cdma2000 1x reverse link data channel. We also develop analytical models for lowest-rate-first, highest-rate-first priority scheduling techniques, and two round-robin fair scheduling techniques over the reverse data channel in cdma2000 1xEV star networks. For these four scheduling techniques, the distribution of the mobile stations among the possible data rates is modeled as a Markov process. An analytical expression for the steady state system throughput is derived from the steady state distribution of the above Markov process. The above model is extended to evaluate the performance of the cross layered design between the hybrid ARQ, rate assignment, and time slot scheduling over the reverse packet data channel in cdma2000 1xEV. We also investigate some techniques that could be used to improve the performance of backhaul mesh networks, specifically, we proposed cross layer coding technique in which we relate the parameters of the higher layer Fountain codes to those of the physical layer codes to present a cross layered coding technique. Based on the bit error rate of physical layer codes, the upper layer Fountain code's parameters are designed to obtain a prespecified performance. As an application we studied the performance of WiMAX backhaul mesh networks where cashing is allowed at the service stations and proposed cross layered coding technique is used as its main coding scheme.
1.2 Outline

The rest of the thesis is organized as follows. In the rest of this chapter we briefly describe the standards considered in this thesis and review the resource management techniques in the literature.

In Chapter 2 we consider a cdma2000 1xEV star network, for which we derive upper and lower bounds for the reverse packet data channel throughput as a function of the number of mobile stations that are allowed to transmit instantaneously on each time slot. We propose several autonomous rate assignment, and scheduling techniques that provide a significant throughput improvement relative to the other published schemes.

In Chapter 3 and based on the results of Chapter 2 we develop an analytical model for several scheduling techniques over the reverse data channel in cdma2000 1xEV. The models are extended to include the effect of the cross layer design between H-ARQ, rate assignment and scheduling. For these scheduling techniques, the distribution of the mobile stations among the possible data rates is modelled as a Markov process. An analytical expression for the steady state system throughput is derived from the steady state distribution of the above Markov process. The developed models are validated through simulations.

In Chapter 4 we proposed a cross layered coding technique. The performance of the proposed coding technique is found to be comparable to that of physical layer based Hybrid ARQ technique. As an application of the multi layer cod-
ing technique proposed, Markov process is used to model and analyze a WiMAX backhaul mesh network where caching is allowed at service stations. Throughout our analysis the proposed cross layered coding technique is used as the WiMAX coding technique. An analytical expressions for the steady state system performance are derived from the steady state distribution of the above Markov process. Results of the developed model is validated through simulations.

Finally, in Chapter 5 we discuss the conclusions and future work.

Some of the work presented in this thesis has been published in [101]-[108].

1.3 Standards

1.3.1 The cdma2000 1xEV

The IS-95A was the first CDMA-based wireless system deployed in the nineties [51]. However, the cdma2000 1xRTT (single carrier radio transmission technology) [53] is considered to be the first phase in CDMA2000 evolution. The cdma2000 1xEV-DO (evolution-data optimization) [54], which classified as a third generation (3G) system, is the third phase in CDMA2000 evolution. In this version of the standards, the data rates reached up to 2Mbps. The cdma2000 1xEV-DV (evolution-data voice) revisions C and D [47] [48] were real evolution on the forward and reverse links. By creating dedicated physical channels to carry high speed data from the base station (BS) to the mobile station (MS) and vice versa, the data rates on the reverse and forwarded links reached the third generation
set of goals. In what follows we briefly describe both cdma2000 1xEV-DO and 1xEV-DV standards.

1.3.1.1 The cdma2000 1xEV-DO

In the last few years, the third generation partnership project two (3GPP2) approved the single carrier evolution data-optimization (1xEV-DO) [49] standards to satisfy the demands for high data rate wireless networks.

In 1xEV-DO, each MS transmits on the reverse traffic channel which consists of a data channel, reverse rate indicator (RRI), pilot channel, data rate control channel (DRC), and an acknowledgement channel (ACK). Each channel in the reverse traffic channel is spread by an appropriate orthogonal Walsh function. A slot is a basic transmission unit, and is 1.666 ms long (2048 chips). A group of 16 slots is referred to as a frame.

The data channel supports five data rates (see Table 1.1) ranging from 9.6 kbps to 153.6 kbps with 26.66 ms time frame packets. The forward link supports data rates up to 2.4 Mbps. The RRI channel indicates the data rate of the associated data channel. The pilot channel is used for channel estimation and coherent detection. The DRC channel informs the access network of the best serving cell and the supportable data rate on the forward traffic channel. A DRC message is repeated over DRCLength (1, 2, 4, or 8 slots). The ACK channel informs the access network whether a packet transmitted on the forward traffic channel has been
<table>
<thead>
<tr>
<th>R</th>
<th>Data Rates in Kbps</th>
<th>$TPRD_k$ in dB</th>
</tr>
</thead>
<tbody>
<tr>
<td>$R_1$</td>
<td>9.6</td>
<td>3.75</td>
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<tr>
<td>$R_2$</td>
<td>19.2</td>
<td>6.75</td>
</tr>
<tr>
<td>$R_3$</td>
<td>38.4</td>
<td>9.75</td>
</tr>
<tr>
<td>$R_4$</td>
<td>76.8</td>
<td>13.25</td>
</tr>
<tr>
<td>$R_5$</td>
<td>153.6</td>
<td>18.5</td>
</tr>
</tbody>
</table>

Table 1.1: Traffic to pilot ratio for 1xEV-DO data channel, channel BW = 1.25 MHz received successfully.

1.3.1.2 The cdma2000 1xEV-DV

The cdma2000 1xEV-DV (evolution-data voice) revisions C and D [47] [48] were a real evolution on the forward and reverse links. By creating the forward packet data channel (F-PDCH) and the reverse packet data channel (R-PDCH), dedicated physical channels to carry high speed data from the BS to the MS and vice versa, the data rates on the reverse and forwarded links reached the third generation set of goals.

The BW of cdma2000 1xEV-DV is 1.25 MHz. On the R-PDCH, each MS can transmit using one of the eleven allowed rates (see Table 1.2). Two power levels are associated with each rate (normal or boosted). Throughout our thesis, we assume that only normal transmission power gain is used.

Fig. 1.3 shows the transmission process on R-PDCH. Cyclic redundancy check (CRC) bits and encoder tail bits are added to the information bits to form the en-
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<tr>
<th>R</th>
<th>Data Rates in Kbps</th>
<th>$TPRD_k$ in dB</th>
</tr>
</thead>
<tbody>
<tr>
<td>$R_1$</td>
<td>19.2</td>
<td>0.75</td>
</tr>
<tr>
<td>$R_2$</td>
<td>40.8</td>
<td>3.75</td>
</tr>
<tr>
<td>$R_3$</td>
<td>79.2</td>
<td>6.75</td>
</tr>
<tr>
<td>$R_4$</td>
<td>156.0</td>
<td>9.625</td>
</tr>
<tr>
<td>$R_5$</td>
<td>309.6</td>
<td>11.875</td>
</tr>
<tr>
<td>$R_6$</td>
<td>463.2</td>
<td>13.625</td>
</tr>
<tr>
<td>$R_7$</td>
<td>616.8</td>
<td>14.875</td>
</tr>
<tr>
<td>$R_8$</td>
<td>924.0</td>
<td>16.625</td>
</tr>
<tr>
<td>$R_9$</td>
<td>1231.2</td>
<td>18</td>
</tr>
<tr>
<td>$R_{10}$</td>
<td>1538.4</td>
<td>19.125</td>
</tr>
<tr>
<td>$R_{11}$</td>
<td>1845.6</td>
<td>21.25</td>
</tr>
</tbody>
</table>

Table 1.2: Traffic to pilot ratio for 1xEV-DV R-PDCH, channel $BW = 1.25$ MHz
coder packet. The encoder packet is fed into a turbo encoder with rate $1/5$ and then interleaved. Each encoded packet is divided into three sub-packets. The sub-packet symbol selector selects the symbols to be transmitted in each sub-packet such that the effective code rate becomes lower as more sub-packets are transmitted [55]. The sub-packet is modulated using BPSK, QPSK or 8-PSK depending on the user’s data rate. The hybrid automatic repeat request (H-ARQ) protocol is applied to sub-packets transmissions (Fig. 1.4). Each user is assigned four ARQ stop and wait channels, each of 10 ms. Each data packet will use one of these ARQ channels to transmit its sub-packets. All the ARQ channels could have the same data rate or different data rates. The ARQ channel keeps the same data
rate as long as it is transmitting the same packet. Incremental redundancy technique is used, where a subset of the turbo encoded packet is sent (sub-packet). This first sub-packet will have an encoding rate higher than the original encoder. As more redundancy (sub-packets) are transmitted, the received data packets reach the original encoding rate.

![Diagram](image)

Figure 1.3: Transmission process on R-PDCH

![Diagram](image)

Figure 1.4: H-ARQ on the R-PDCH

If the BS decodes the information correctly, an ACK will be transmitted and the MS will not send the rest of this packet. A NAK will be transmitted if the information is not decoded correctly and the MS sends the successive sub packets. The H-ARQ protocol has a 40 ms cycle (4 slots) from the time the MS starts transmitting the sub-packet until it receives the ACK/NAK and processes it. The 1xEV
turbo encoder has a final coding rate of $R_c = 1/5$. Each encoded packet is divided into three sub-packets. The first sub-packet has encoding rates of $R_c = 1/2$, the first and second sub-packets combined will have an effective lower encoding rate of $R_c = 1/3$, the three sub-packets combined will reach the final encoding rate of $R_c = 1/5$.

The R-PDCH is supported by the reverse packet data control channel (R-PDCCH) which is used to transmit control information that correspond to the current transmission on the R-PDCH. Amongst these, the size of the data packet, which sub packet is currently transmitted, whether or not the transmission is done using elevated power levels, and an indicator if the MS has enough power and data to transmit above the current rate. The reverse request channel (R-REQCH) transmits the MS's current application (FTP, video streaming, interactive gaming, etc.), the amount of available data, and its current power head room. The forward rate control channel (F-RCCH) allows the MS to change its current rate by one step (higher or lower). Forward grant channel (F-GCH) allows the MS to change its current rate to the granted rate. The granted rate could be any one of the allowed eleven rates. The forward acknowledgement channel (F-ACKCH) provides the MS with an ACK or NAK for the transmitted sub packet.

The rate assignment mechanisms supported by the 1xEV reverse data channel can be categorized as autonomous, differential, or absolute rate schemes [55].

In autonomous rate assignment schemes, the BS grants each MS a rate from the set of allowable rates. The MS is then free to transmit using this rate or any other
rate below it. This technique is considered the best option for delay sensitive applications and it has the least computational overhead.

Differential rate assignment allows the MS to change its rate by one step up or down. The BS can send one of three commands (UP, Down, or Hold). The problem with this technique is ramping up delays; it takes relatively long time to reach high data rate transmissions.

In absolute rate assignment schemes, the BS authorizes the MS to send on a certain rate. The grant message is initiated when the BS receives a change rate request from the MS.

The 3GPP2 [56] requires that all resource scheduling techniques, for 1xEV standards, to satisfy the fairness criterion. Fairness is defined as providing all mobile stations with a minimal level of throughput and is evaluated by determining the cumulative density function (CDF) of the normalized throughput, with respect to the average user throughput, for all users. In order to satisfy the fairness criterion, the standard specifies that the CDF shall lie to the right of the curve given by the three points in Table 1.3. The first point in the table requires that the normalized user throughput of at least 90% of the users exceeds 0.1. Similarly, the second and third points require that the normalized user throughput of at least 80% and 50% of the users exceeds 0.2 and 0.5 respectively.

Good surveys on the cdma2000 1xEV-DV system can be found in [57] [58] [13] [59] [55] [60]. In [57], Derryberry and Pi describe the reverse link (RL) physical layer enhancements introduced to support 1xEV-DV. The forward link (FL)
<table>
<thead>
<tr>
<th>Normalized throughput</th>
<th>Cumulative density function (CDF)</th>
</tr>
</thead>
<tbody>
<tr>
<td>0.1</td>
<td>0.1</td>
</tr>
<tr>
<td>0.2</td>
<td>0.2</td>
</tr>
<tr>
<td>0.5</td>
<td>0.5</td>
</tr>
</tbody>
</table>

Table 1.3: Fairness criterion

physical layer enhancements can be found in [58]. Kwon et al. [13] discuss the system performance evaluation for both FL and RL, describe how the evaluation was conducted and analyze the results. The hybrid automatic repeat request (H-ARQ) mechanism used in release D is described in [59]. This modified technique allows for variable retransmission power levels resulting in four modes of operation: normal, reduction, boost and boost reduction modes. In [55], Comstock et al. describe the design enhancements of the upper layers of the cdma2000 revision D, including basic channel operation, the multiplexing of traffic, H-ARQ and mechanisms for managing RL radio resources. Kim and Ti [60] look at the new system from practical prospective; they discuss the needs, services and market drivers and provide the market and revenue statistics for voice and data services.

1.3.2 WiMAX

The rapid increase in user demands for faster connections to the Web and VoIP services has led to faster developments in wireless communications technology.
WiMAX [50] is rapidly proving itself as a key player in the area of long-range, high-speed fixed and mobile wireless technology. It also provides fast wireless solution for backhaul systems.

The physical layer of WiMAX is based on orthogonal frequency division multiple access (OFDMA) [62]. The OFDMA can be thought of as a multi-user version of the orthogonal frequency division multiplexing (OFDM). In OFDM, the BW is subdivided into multiple frequency sub-carriers. Also, the input data stream is divided into several parallel sub-streams of reduced data rate (thus increased symbol duration) and each sub-stream is modulated and transmitted on a separate orthogonal sub-carrier. The increased symbol duration improves the robustness of OFDM to delay spread [64]. Multiple access in OFDMA is achieved by assigning subsets of sub-carriers to individual users, which allows simultaneous transmission from different users.

The WiMAX OFDMA structure consists of three types of sub-carriers: data sub-carriers for data transmission, pilot sub-carriers for estimation and synchronization purposes, and null sub-carriers as guard bands [64]. Data and pilot sub-carriers are grouped into subsets of sub-carriers called sub-channels, the sub channelization is supported in both down link and up link [50].

Each sub-carrier can be modulated using one of the following modulation and coding techniques shown in Table 1.4 where $SINR_k$ denotes the signal to interference and noise ratio required by each scheme to achieve a BER of $10^{-6}$ [50]. The parameters of WiMAX up-link for 5 and 10 MHz BW are shown in Table 1.5.
<table>
<thead>
<tr>
<th>Modulation Technique</th>
<th>Coding rate $C_k$</th>
<th>Data bits per sub-carrier $d_k$</th>
<th>$SINR_k$ in dB</th>
</tr>
</thead>
<tbody>
<tr>
<td>$M_1$ QAM4</td>
<td>1/2</td>
<td>1</td>
<td>5</td>
</tr>
<tr>
<td>$M_2$ QAM4</td>
<td>3/4</td>
<td>1.5</td>
<td>8</td>
</tr>
<tr>
<td>$M_3$ QAM16</td>
<td>1/2</td>
<td>2</td>
<td>10.5</td>
</tr>
<tr>
<td>$M_4$ QAM16</td>
<td>3/4</td>
<td>3</td>
<td>14</td>
</tr>
<tr>
<td>$M_5$ QAM64</td>
<td>2/3</td>
<td>4</td>
<td>18</td>
</tr>
<tr>
<td>$M_6$ QAM64</td>
<td>3/4</td>
<td>4.5</td>
<td>20</td>
</tr>
</tbody>
</table>

Table 1.4: Allowable modulation techniques

Both convolutional code and convolutional turbo code with variable code rate are supported. Hybrid Automatic Repeat Request (H-ARQ) is also supported by WiMAX to enhance the system performance.

The frame structure for a WiMAX time division duplex (TDD) implementation is divided into down link (DL) and up link (UL) sub-frames separated by transition gaps to prevent DL and UL transmission collisions. Each frame has its control information to ensure optimal system operation. As shown in Fig. 1.5, each UL and DL sub-frame is further divided horizontally to symbol durations and vertically to sub-channels and each frame is transmitted over $T_f$ time [64].

1.4 Literature Review

As we will cover more than one topic in this thesis, our literature review is divided into two parts. In the first part, we discuss the currently available rate
<table>
<thead>
<tr>
<th>Parameter</th>
<th>5 MHz BW</th>
<th>10 MHz BW</th>
</tr>
</thead>
<tbody>
<tr>
<td>Total number of sub-carriers ((N_{FFT}))</td>
<td>512</td>
<td>1024</td>
</tr>
<tr>
<td>Number of Null sub-carriers</td>
<td>104</td>
<td>184</td>
</tr>
<tr>
<td>Number of Pilot sub-carriers</td>
<td>136</td>
<td>280</td>
</tr>
<tr>
<td>Number of Data sub-carriers</td>
<td>272</td>
<td>560</td>
</tr>
<tr>
<td>Number of used sub-carriers ((N_{used}))</td>
<td>408</td>
<td>840</td>
</tr>
<tr>
<td>Number of sub-channels (N)</td>
<td>17</td>
<td>35</td>
</tr>
<tr>
<td>Number of sub-carriers (N_c) per sub-channel</td>
<td>16</td>
<td>16</td>
</tr>
<tr>
<td>Symbol duration ((T_s)) micro sec</td>
<td>102.9</td>
<td>102.9</td>
</tr>
<tr>
<td>Frame duration ((T_f)) msec.</td>
<td>5</td>
<td>5</td>
</tr>
<tr>
<td>Number of data symbols/frame ((N_{sym}))</td>
<td>44</td>
<td>44</td>
</tr>
</tbody>
</table>

Table 1.5: Wimax parameters [64]
assignment techniques. In the second part, we review the relevant scheduling schemes.

1.4.1 Rate Assignment

One should mention that the methodology by which the BS determines the appropriate data rate for each MS is not specified in the standards. In the rest of this section, we give a review of the current rate assignment methodologies.

Kwon et al. [13] [14] proposed a hybrid rate technique for the 1xEV-DV system. It is called rate control with quick start (RCQS) in which high data rates are assigned to the users at the beginning in order to avoid the delay of ramping up. However, they didn’t provide any specific details for the rate assignment scheme.

Shu and Niu [15] proposed a dynamic rate assignment based on computing the optimum number of simultaneous transmissions on one time slot with respect to
multiple access interference. In this scheme, the authors assume that all the users have the same packet size, each user is allowed to send more than one packet per slot, and the number of packets are computed as a function of the optimum number of transmissions and the buffer length of each user.

Lee, Yeo, and Cho [16] [18] [20] proposed rate assignment techniques for the 1xEV-DO system. The techniques are based on modelling the rate control as discrete Markov process and adopting the RoT as the traffic load measure. These techniques were proposed for a set of five data rates (1xEV-DO allowable data rates).

Vannithamby and Sousa [21] provided an analytical model for data rates on the forward link in WCDMA. Two cases are considered: only one user is transmitting per slot, and more than one user are transmitting at the same time. They compared more than one technique. Their techniques were mainly based on the signal to interference ratio required to provide QoS, transmitted power allocated for mobiles, and the normalized interference to each mobile.

Rodriguez and Goodman [22] addressed power and data rate allocations for each terminal, such that the network weighted throughput is maximized. The weights admit various interpretations, including levels of importance, "utility", and price. They stated that at least one terminal should operate at the highest available data rate. Lowering the highest available data rate increases the number of terminals which should operate at maximum data rate. They utilized a model which can accommodate many physical layer configurations of practical interest.
Rodriguez, Goodman, and Marantz [23] investigated power and data rate allocations that maximize the network weighted throughput. Each terminal has one of two possible weights, which admits various practical interpretations. In their work, they introduced a general procedure to seek a global optimizer allocation. Their analysis was based on classical optimization theory, and accommodate a wide variety of physical layer configurations.

Bjorklund et al. [24] studied the problem of forward link (FL) bandwidth allocation in WCDMA networks. They proposed an algorithm for assigning transmission rates to the users with the objective of optimizing some performance criterion such as the total throughput. They formulated the problem using one multiple-choice knapsack model per cell.

Price and Javidi [26] considered a WCDMA network with arbitrary but known layout and variable rate assignments, which is connected to a traditional wired network. They showed that by using an optimization framework it is possible to construct a distributed rate assignment algorithm which addresses issues like interference and congestion control. They formulated this as a maximization problem subject to interference and congestion constraints, and developed a distributed algorithms to solve this maximization problem.

Price and Javidi [27] addressed the QoS requirements and continually changing resource demands. In that paper, they examined such optimal resource allocation through non-uniform rate assignment in a CDMA system. In particular, they focused on reverse link rate assignment at the MAC layer, and showed that
a pricing structure can be used to de-centrally regulate each mobile's bandwidth consumption (transmission rate) based on the air-link's interference. Using an optimization formulation, they showed that unequal rate assignments perform better than equal-rate assignments, particularly as the density of mobiles at the boundary of two cells increases.

Fattah and Leung [29] proposed a load-based transmission rate (LTR) assignment scheme for non-real time data services in an integrated voice/data direct sequence code division multiple access (DS/CDMA) system. The LTR scheme optimally determines the transmission rate for each session according to its individual load to minimize the overall average packet transfer delay.

Ci and Guizani [30] [31] proposed an optimal rate assignment technique for the 1xEV-DV forward link. The scheme was proposed to support delay bounded multimedia services. In their analysis, they consider one user with mixed traffic per time slot. The scheme's objective is to maximize the one slot capacity subject to some constraints such as the number of available Walsh codes, service type, packet length, and maximum allowable data rate.

Kulkarni et al. [82] addressed the problem of assigning sub-carriers and bits to point-to-point wireless links in the presence of co-channel interference and Rayleigh fading in WiMAX networks. The objective is to minimize the total transmitted power over the entire network while satisfying the data rate requirement of each link. Simulation results show that the approach results in an efficient assignment of sub-carriers and transmitter power levels in terms of the energy

21
required for transmitting each bit of information.

In [74] the authors proposed a cross-layered resource allocation scheme over wireless relay networks with the objective of maximizing the relay network throughput subject to a given delay QoS constraint.

1.4.2 Scheduling Techniques

Efficient Scheduling algorithms are essential to guaranteeing QoS such as delay and throughput. The design of scheduling schemes for the wireless systems is challenging due to the high error rates in the wireless links. With the increasing demand for wireless data services, the demand for optimum scheduling techniques is also increasing. The perfect wireless scheduling technique should satisfy many aspects. In particular, it should maximize throughput, minimize the delay, and tolerate the high error rates on the wireless links. During the last few years, many scheduling techniques were proposed. Some of these proposals were inherited from the wired links such as the first come first serve scheduler, fair queuing technique, weighted fair queuing [35] [41]. In our review, we focused only on the ones designed specially for wireless systems. In the following we review some of the available wireless scheduling schemes.

Jalali, Padovani, and Pankaj [32] introduced the idea of proportional fair scheduling to improve the system throughput; the users at a certain time are given priorities based on their current rates. The priority metric for any user at time $t$
is a function of its current rate and its average throughput till time $t$. The proportional fair algorithm does not have delay control mechanism, which makes it unsuitable for delay-constrained applications.

Shakkottai and Stolyar [33] proposed an exponential scheduling algorithm that guarantees some delay constraints. Their proposal is based on the idea of proportional fairness but with different metric. The priority metric for any user at time $t$ is a function of its maximum allowable packet drop probability, its initial life time of each packet, and its head of line delay.

Shin and Lee [34] proposed a modification of the exponential fair algorithm where they identified a new fairness term which reflects the overall buffer status of the $i^{th}$ user at time $t$. The new priority metric for the modified exponential algorithm is a function of the weighted sum of the number of packets in the $i^{th}$ user's buffer at time $t$.

Almajano and Romero [36] presented scheduling algorithms that focus on giving QoS guarantee in terms of bit rate in a packet switched CDMA by taking into account the service classes that are defined in the UMTS system, i.e., conversational, streaming, interactive and background. This scheduler was based on the leaky bucket algorithm; they introduced a new parameter called “Service Credit” (SCr) which measures the difference between the bit rate requested by a user and the bit rate that the system has offered.

Lopez et al. [38] used energy transmitted per correct received bits as a cost function for evaluating the radio resource consumption of each user within a
UMTS network, and used a modified round robin scheme as their scheduler.

Malik and Zeghlache [39] presented a resource scheduling scheme to improve throughput and fairness for non real time data traffic on the FL WCDMA network. The proposed Fair Resource Scheduling (FRS) scheme takes into account link conditions to enhance throughput and provides fair resource sharing among traffic flows.

Al-Manhari et al. [43] proposed a priority packet scheduler algorithm for high speed down-link packet access (HSDPA). Users priorities are set based on their instantaneous channel conditions and their average throughputs.

Long et al. [44] found, through mathematical analysis, that TDM is more efficient than CDM for FL in high speed CDMA networks. Based on this finding, they proposed a new scheme for packet scheduling in high speed CDMA networks, namely Channel States Dependent Fair Service for CDMA (CSDFSC), which try to maximize channel throughput under fairness and transmission power constraints. This scheme assigns different queue weights to various traffic classes.

Wang et al. [45] proposed the channel adaptive fair queueing (CAFQ) algorithm for multicode TD-CDMA system. In each time slot, the CAFQ algorithm keeps on allocating codes in a fair manner until the whole bandwidth resource is used up (subject to the interference budget). The CAFQ algorithm guaranteed short-term fairness.

Viswanathan and Sayandev [79] developed a linear programming framework for determining an optimum routing and scheduling of flows that maximizes
throughput in a wireless WiMAX mesh network. They also discussed the application of mesh networking for load balancing of wired backhaul traffic under unequal access traffic conditions. Numerical results show a significant benefit for mesh networking under unbalanced loading.

Cao et al. [80] introduced a new fairness criteria based on the actual traffic demands and formulated a scheduling technique with the objective of maximizing the system throughput under the proposed fairness model in WiMAX networks. The scheduling algorithm is evaluated through simulations.

Cao et al. [81] developed a stochastic model for the distributed scheduler of the IEEE 802.16 mesh mode and analyzed the scheduler performance under various conditions.

A cross-layer packet scheduling scheme for video transmission over wireless downlink is presented in [73], a gradient based scheduling scheme is used in which user data rates are dynamically adjusted based on channel quality as well as gradients of a utility function. In addition user utilities are designed as a function of the distortion of the received video.
Chapter 2

Resource Management Techniques for cdma2000 1xEV Star Networks

In order to improve the BW efficiency, the 3GPP2 utilizes efficient mechanisms such as H-ARQ, adaptive modulation, and coding as mentioned in Section 1.3.1. While current standards fully specify several parameters, such as modulation schemes, encoder packet size, and coding techniques, resource management techniques remain unexplored.

In this chapter, and based on the developed rise over thermal (RoT) model, we derive an upper bound for the reverse packet data channel throughput as a function of the number of mobile stations that are allowed to transmit instantaneously on each time slot. We also provide a lower bound for the average sector throughput based on the number of users per sector. We propose several autonomous rate assignment, and scheduling techniques that provide a significant throughput improvement relative to the other published schemes.

The rest of the chapter is organized as follows. The rise over thermal model used throughout the thesis is developed in Section 2.1. In Section 2.2, the upper and lower bounds for the maximum instantaneous throughput over the R-PDCH is developed as a function of the number of active mobile stations per time slot.
The proposed rate assignment techniques are described in Section 2.3. The proposed scheduling techniques are described in Section 2.4. The analysis and simulation results are presented in Section 2.5. A summary of the chapter is given in Section 2.6.

2.1 RoT Constraint

The RoT [16] is defined as the ratio between the total power received to the thermal noise power at the base station. To ensure QoS, 3GPP2 [56] specifies that the RoT per slot should not exceed 7 dB more than 1% of the time.

In what follows we explain the model for the RoT to be used in our work. Let $P_T$ and $P_N$ be the total received power and the noise thermal power at the BS respectively. Then the RoT is given by $RoT = P_T / P_N$. Assume $M$ active mobiles in the sector under consideration. Therefore, $P_T = \sum_{i=1}^{M} P_i + P_N$, where $P_i$ is the received power of the $i^{th}$ mobile at the BS. Thus $RoT = P_T / (P_T - \sum_{i=1}^{M} P_i)$, which can be expressed as

$$RoT = (1 - \sum_{i=1}^{M} \frac{P_i}{P_T})^{-1}. \quad (2.1)$$

Let $\overline{P_i}$ be the pilot channel power for the $i^{th}$ mobile. Define the traffic to pilot ratio for the $i^{th}$ mobile as $TPR_i = P_i / \overline{P_i}$. Denote the targeted signal to noise and interference ratio for the pilot channel as $\tau_i = \overline{P_i} / (P_T - P_i)$, where $P_T - P_i$ is the total interference and the thermal power for the $i^{th}$ mobile. By substituting into
equation 2.1, we get

$$\text{RoT} = \left(1 - \sum_{i=1}^{M} \frac{TPR_i}{TPR_i + 1/\tau_i}\right)^{-1},$$

(2.2)

which can be rewritten as,

$$\text{RoT} = \left(1 - \sum_{k=1}^{r} N_k \frac{TPR_k}{TPR_k + 1/\tau}\right)^{-1},$$

(2.3)

where $r$ denotes the number of allowable rates, $N_k$ denotes the number of active mobiles transmitting with rate $R_k$ at the current time slot, and $\tau = \tau_i = 10^{0.1SINR}$, where $SINR$ is the targeted signal to noise plus interference ratio for the pilot channel (constant for all users). The total traffic to pilot ratio corresponding to rate $R_k, TPR_k$, is determined by

$$TPR_k = \begin{cases} 
1 + 10^{0.1TPRD_k} + 10^{0.1TPDRC}, & \text{1xEV-DO} \\
1 + 10^{0.1TPRD_k} + 10^{0.1TPRC} + 10^{0.1TPRQ}, & \text{1xEV-DV}
\end{cases}$$

(2.4)

where $TPRD_k$ denotes the traffic to pilot ratio (in dB) on the reverse data channel associated with rate $R_k$ (Tables 1.1 and 1.2). $TPDRC$, $TPRC$, and $TPRQ$ denote the traffic to pilot ratio (in dB) associated with the data rate control (DRC) channel (1xEV-DO), reverse control channel and reverse request channel (1xEV-DV) respectively. As specified by the standard, the values of $TPDRC, TPRC, TPRQ$, and $SINR$ are shown in Table 2.1.

To satisfy the RoT constraint, RoT must not be greater than 7 dB. Substituting
### Table 2.1: Values used for the simulation parameters

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>TPDRC</td>
<td>-5 dB</td>
</tr>
<tr>
<td>TPRC</td>
<td>-5 dB</td>
</tr>
<tr>
<td>TPRQ</td>
<td>-1 dB</td>
</tr>
<tr>
<td>SINR</td>
<td>-22 dB</td>
</tr>
</tbody>
</table>

In equation 2.3,

\[
(1 - \sum_{k=1}^{r} N_k \frac{TPR_k}{TPR_k + 1/\tau})^{-1} \leq 10^{0.7} \tag{2.5}
\]

Let

\[
c_k = \frac{TPR_k}{TPR_k + 1/\tau} \tag{2.6}
\]

be a constant for each rate \( R_k \). Therefore Eq. 2.5 can be re-written as

\[
\frac{1}{1 - \sum_{k=1}^{r} c_k N_k} \leq 10^{0.7}
\]

\[
\Rightarrow 1 \leq 10^{0.7} \left(1 - \sum_{k=1}^{r} c_k N_k\right)
\]

\[
\Rightarrow 10^{0.7} \sum_{k=1}^{r} c_k N_k \leq 10^{0.7} - 1
\]

\[
\Rightarrow \sum_{k=1}^{r} c_k N_k \leq \frac{10^{0.7} - 1}{10^{0.7}} = 0.8005
\]

#### 2.2 Theoretical Throughput Bounds

In this section, we develop bounds for the instantaneous throughput on the reverse packet data channel in cdma2000 1xEV as a function of the number of active MSs per time slot and subject to the rise over thermal constraint [56].
2.2.1 Upper Bound

On the reverse packet data channel, each encoded packet is divided into three sub-packets using incremental redundancy. Only one sub-packet is transmitted at a time. The BS uses a soft combining technique to decode the received packet (Section 1.3.1.2). In developing our upper bound, we assume that all the transmitted data are accepted after the first transmission.

In the following analysis we consider the two possible cases:

- Case 1: All mobile stations must be assigned a non zero rate from the set of data rates $R$ (Table 1.2) on each time slot.

- Case 2: Some mobile stations are allowed to be idle (i.e., not transmitting) on the given time slot.

Our objective is to maximize the instantaneous link throughput while satisfying the RoT constraint. Hence, finding an upper bound on the instantaneous throughput is equivalent to solving the following integer optimization problem:

Maximize $\text{Throughput} = \sum_{k=1}^{r} N_k R_k$

subject to

$C1. \sum_{k=1}^{r} N_k = M$ (for case 1)

$\hat{C}1. \sum_{k=1}^{r} N_k \leq M$ (for case 2)

$C2. \sum_{k=1}^{r} c_k N_k \leq 0.8005$

where the condition $C2$ is obtained from Eq. 2.7. The above optimization problem
is solved using the branch and bound optimization technique.

2.2.2 Lower Bound

In bounding the minimum achievable throughput we consider the worst case in which, each mobile station is assigned the lowest rate $R_1$ and each packet will be accepted after the third sub-packet transmission. The trivial case of all the users being idle is not considered. The minimum bound could be computed as follows,

$$\text{Throughput} = M \frac{R_1}{3},$$  \hspace{1cm} (2.8)

and it should satisfy,

$$c_1 M \leq 0.8005,$$  \hspace{1cm} (2.9)

where $c_1 = \frac{TPR_{R1}}{TPR_{R1} + 1/r}$ is constant for rate $R_1$, $M$ the number of active users, and $R_1/3$ is the effective rate after the retransmissions.

We apply the above optimization problem on cdma2000 1xEV-DV standards where $r = 11$ and the values for $TPRD_k$ are given in Table 1.2. Values of $TPRC$, $TPRQ$, and $SINR$ are shown in Table 2.1.

Fig. 2.1 shows the maximum theoretical link throughput for case 1 and case 2 respectively. Also the lower link throughput that could be achieved is demonstrated in the same figure. The developed lower and upper bounds are used to check our proposed resource management techniques.
From Fig. 2.1, it is clear that the maximum throughput in the first case is reached when three MSs are allowed to transmit. As the number of active users gets higher, we get a lower throughput. The second case confirms the same conclusion, i.e., if the system is allowed to choose the number of active users to transmit on each slot, only three users are chosen to transmit on each time slot.

Figure 2.1: Upper and lower throughput bounds for cdma2000 1xEV-DV

Let \((N_1, N_2, \cdots, N_{10}, N_{11})\) denote the rate distribution that achieves the maximum instantaneous throughput. Table 2.2 shows this distribution as a function of the number of active mobile stations, \(M\), for case 1. For case 2, the optimum
rate distribution is given by

\[(0, 0, 0, 0, 0, 0, 0, 0, 0, 0, 1), \quad M = 1\]

\[(0, 0, 0, 0, 0, 0, 0, 0, 2, 0), \quad M = 2\]

\[(0, 0, 0, 1, 0, 0, 0, 0, 2, 0), \quad M > 2.\]

Thus, for both cases, the maximum throughput is reached with 2 users assigned the rate \(R_{10}\) and one user assigned the rate \(R_5\). One should note that these values of the throughput may not be achieved in practice since we have some timing constraints on the H-ARQ channel. Also, rates should only be assigned if there is enough data in the buffer to form at least one packet with such rate, and packets may not be accepted from the first transmission.

From the above results, it is clear that we can achieve a better throughput if we allow only a subset of active users to transmit on each slot. One can also argue that the rate \(R_{11}\) should be used only if we have one active MS, otherwise it is better to use a combination of the other rates.

### 2.3 Proposed Rate Assignment Techniques

In this section, we present our proposed autonomous rate assignment schemes. For autonomous rate assignment, each MS is free to transmit using the autonomous rate \(R_k\) assigned to it by the base station or any other rate below it (Sec. 1.3.1).

In order to simplify our analysis, we assume that if the MS is assigned a rate \(R_k\), then it will always transmit using \(R_k\) and not a lower rate.
<table>
<thead>
<tr>
<th>$M$</th>
<th>$(N_1, N_2, N_3, N_4, N_5, N_6, N_7, N_8, N_9, N_{10}, N_{11})$</th>
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<tbody>
<tr>
<td>1</td>
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<td>2</td>
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</tr>
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<td>8</td>
<td>(5, 0, 0, 1, 0, 0, 0, 0, 1, 1, 0)</td>
</tr>
<tr>
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</tr>
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<td>28</td>
<td>(25, 2, 0, 0, 0, 0, 0, 1, 0, 0, 0)</td>
</tr>
<tr>
<td>29</td>
<td>(25, 2, 0, 1, 0, 0, 0, 1, 0, 0, 0)</td>
</tr>
<tr>
<td>30</td>
<td>(24, 5, 0, 0, 0, 0, 1, 0, 0, 0, 0)</td>
</tr>
</tbody>
</table>

Table 2.2: Rate distribution for case 1
For the trivial case of one active MS, the MS’ assigned rate is $R_{11}$. In what follows, we assume that we have more than one active MS.

For our schemes, the rate assigned to the MS is determined based on the user’s loaded file size.

Denote the loaded file size of the $i^{th}$ user by $F_i$. Let $T_{k-1}$ and $T_k$ be the minimum and maximum thresholds for rate $R_k$, then the $i^{th}$ user will be assigned rate $R_k$ if

$$T_{k-1} \leq F_i < T_k,$$  \hspace{1cm} (2.10)

where $T_k \in \mathbb{T}$ and $\mathbb{T}$ denote the set of predetermined thresholds. Based on the results of the previous section, we investigate the following two cases:

- Case A: $R = \{R_k, 1 \leq k \leq 11\}$

- Case B: $R = \{R_k, 1 \leq k \leq 10\}$

In the rest of this section, we propose two rate assignment schemes to determine the set of thresholds $\mathbb{T}$. Motivated by fairness amongst users, the first technique adapts to the file length, i.e file proportional rate assignment (FPR). In this technique fairness between users is provided in the sense of service time. The second technique is motivated by minimizing the fluctuation of the system load over time. In this technique each allowable rate is assigned equal number of times over a long interval of time, i.e equiprobable rates assignment (EPR).
2.3.1 FPR Rate Assignment Technique

In this scheme, the set $T$ is determined such that each MS is assigned a rate proportional to its file length and thus the differences between the number of packets in each MS's buffer is minimized. Hence, the number of time slots required to transmit any data file (not including the effect of time scheduling) is the almost the same for all MSs. Based on the traffic model, a minimum ($F_{\text{min}}$) and maximum ($F_{\text{max}}$) file size can be assumed. For case A, we have

$$
T_k = \begin{cases} 
F_{\text{min}}, & k = 0 \\
F_{\text{max}}, & k = 11 \\
(F_{\text{max}} - F_{\text{min}}) \times \frac{R_k}{R_{11}}, & 1 \leq k \leq 10.
\end{cases}
$$

Similarly, for case B, we have

$$
T_k = \begin{cases} 
F_{\text{min}}, & k = 0 \\
F_{\text{max}}, & k = 10 \\
(F_{\text{max}} - F_{\text{min}}) \times \frac{R_k}{R_{10}}, & 1 \leq k \leq 9.
\end{cases}
$$

2.3.2 EPR Rate Assignment Technique

In this scheme, the MSs are assigned rates in such a way that, over a long interval of time, each allowable rate is assigned equal number of times. This can be achieved by analyzing the pdf distribution of the file size curve. The set $T$ is determined by dividing the area under the pdf curve into (eleven for case A and
ten for case B) equal parts. Let \( f(x) \) denote the pdf of the file size, then the set \( T \) is determined such that we have
\[
\int_{T_{k-1}}^{T_k} f(x)dx = \int_{T_k}^{T_{k+1}} f(x)dx.
\]

The above two schemes were applied to the FTP traffic, the FTP file length distribution is modelled as a truncated lognormal distribution with file size between \( 0.5K \) bytes and \( 500K \) bytes, which is obtained from the following lognormal distribution

\[
f(x) = \frac{1}{\sqrt{2\pi}\sigma x} \exp \left( \frac{-(\ln x - \mu)^2}{2\sigma^2} \right), x \geq 0
\]

where \( \sigma = 2.0899, \mu = 0.9385 \) [56].

Table 2.3 shows the proposed set of file size thresholds \( T \) for FPR and EPR rate assignment schemes (case A).

Similarly, Table 2.4 show the proposed set of file size thresholds \( T \) for for FPR and EPR rate assignment schemes (case B).

Since each data packet is transmitted in 10 msec slot (Sec. 1.3.1), the number of bits required to construct a packet for rate \( R_k \) is equal to \( R_k \times (10 \times 10^{-3}) \) bits. Thus in both the schemes above, to ensure that there is enough data bits to construct at least one packet with data rate \( R_{k-1} \), the threshold \( T_{k-1} \) should also satisfy
\[
T_{k-1} \geq R_k/100.
\]

It is easy to verify that the threshold sets for both FPR and EPR techniques satisfy the above inequality.
<table>
<thead>
<tr>
<th>$T_i$</th>
<th>FPR Scheme</th>
<th>EPR Scheme</th>
</tr>
</thead>
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<td>$T_0$</td>
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<td>0.5</td>
</tr>
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<td>$T_1$</td>
<td>5.19</td>
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</tr>
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<td>$T_2$</td>
<td>11.04</td>
<td>1.2</td>
</tr>
<tr>
<td>$T_3$</td>
<td>21.43</td>
<td>1.8</td>
</tr>
<tr>
<td>$T_4$</td>
<td>42.22</td>
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<td>$T_5$</td>
<td>83.79</td>
<td>3.8</td>
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<td>$T_6$</td>
<td>125.362</td>
<td>5.5</td>
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<td>$T_7$</td>
<td>166.9</td>
<td>8.1</td>
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<td>$T_8$</td>
<td>250.07</td>
<td>12.4</td>
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<td>$T_9$</td>
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<td>416.35</td>
<td>50</td>
</tr>
<tr>
<td>$T_{11}$</td>
<td>500</td>
<td>500</td>
</tr>
</tbody>
</table>

Table 2.3: Proposed file size thresholds in Kbyte for FTP traffic, FPR and EPR schemes (Case A)
<table>
<thead>
<tr>
<th></th>
<th>FPR Scheme</th>
<th>EPR Scheme</th>
</tr>
</thead>
<tbody>
<tr>
<td>$T_0$</td>
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<td>0.5</td>
</tr>
<tr>
<td>$T_1$</td>
<td>6.23</td>
<td>0.8</td>
</tr>
<tr>
<td>$T_2$</td>
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<td>1.3</td>
</tr>
<tr>
<td>$T_3$</td>
<td>25.69</td>
<td>2.0</td>
</tr>
<tr>
<td>$T_4$</td>
<td>50.6</td>
<td>2.9</td>
</tr>
<tr>
<td>$T_5$</td>
<td>100.45</td>
<td>4.5</td>
</tr>
<tr>
<td>$T_6$</td>
<td>150.3</td>
<td>7</td>
</tr>
<tr>
<td>$T_7$</td>
<td>200.26</td>
<td>11.1</td>
</tr>
<tr>
<td>$T_8$</td>
<td>300.01</td>
<td>20.0</td>
</tr>
<tr>
<td>$T_9$</td>
<td>399.49</td>
<td>45.8</td>
</tr>
<tr>
<td>$T_{10}$</td>
<td>500</td>
<td>500</td>
</tr>
</tbody>
</table>

Table 2.4: Proposed file size thresholds in Kbyte for FTP traffic, FPR and EPR schemes (Case B)
2.4 Proposed Scheduling Techniques

Scheduling algorithms are one of the most important aspects to guarantee QoS parameters such as delay and throughput. As mentioned before, the standards left the door open for developing efficient scheduling algorithms.

Many scheduling techniques were proposed for wireless links over the last few years [32] [33] [34]. While the cdma2000 1xEV-DV resource scheduling techniques are only discussed in [14] [30], scheduling on the R-PDCH is barely touched.

In here, we investigate scheduling techniques for minimizing the mobile stations packet delays, and maximizing the link throughput. In these schemes, the base station uses a greedy optimization algorithm to dynamically determine the optimum subset of users that are allowed to transmit on each time slot. Several constraints, such as the rise over thermal, frame error rate, fairness criterion, and the constraints imposed by H-ARQ on the minimum allowed delay between the user sub-packets, must be satisfied.

We will also study the effect of reducing the set of available rates to ten (Case B) on the scheduling process.

2.4.1 Scheduling Process: Base Station Model

In here, we assume that the scheduling process is done over a prespecified time window (TW). Although, theoretically, no maximum time applies, in order to minimize the new users' delay, we assume that the scheduling window is one
time slot. On each slot, more than one user is allowed to transmit. The subset of users allowed to transmit will be chosen by the BS based on some optimization criterion using an efficient optimization technique. A slot will be allocated to the MS based on its current status reported on the R-PDCCH.

The optimization algorithm of the scheduling scheme can be based on different cost functions. The following schemes are considered:

*Congested-Buffer-First:* Minimize the number of packets accumulated in all active users’ buffers. Hence, fairness is provided between MSs in the sense of service times.

*Lowest-Rate-First:* Maximize the number of users transmitting on each slot. In order to maximize the throughput.

### 2.4.2 Scheduling Process: Mobile Station Model

A slot will be allocated to each MS based on its current status reported on the R-PDCCH. When the MS finishes transmitting, it leaves the system immediately. The MS decides which sub-packet to send on the allocated slot based on the H-ARQ constraints discussed earlier.

Let $s = S(i, l, j)$ be the number of the slot allocated for the $i^{th}$ user, where $l$ ($1 \leq l \leq B_i$) and $j$ ($1 \leq j \leq 3$) are the packet and sub-packet numbers the user decides to send on the allocated slot respectively. The H-ARQ operation Fig. 1.4 imposes the constraint that all sub-packets of the same packet have to be spaced
by four slots (40 ms), i.e., we must have \( S(i, l, j + 1) - S(i, l, j) \geq 4 \). The same user cannot send more than one sub-packet per slot, i.e., \( S(i, l + x, j + y) \neq S(i, l, j) \) for \( x \neq 0, y \neq 0 \).

2.4.3 Formulated Problem

The following notation will be used throughout this section:

- \( M \): the number of active mobiles in the current sector
- \( A(s) \): the number of users transmitting per slot
- \( B_i(s) \): the number of packets in \( i^{th} \) user buffer at slot number \( s \)
- \( N_k \): the number of active mobiles with data rate \( R_k \) transmitting at the current slot.
- \( RoT \): the rise over thermal defined in equation (2.3)
- \( \gamma \): the RoT threshold required in order to maintain a prespecified QoS
- \( S(i, l, j) \): the slot allocated for the \( i^{th} \) user, where \( 1 \leq l \leq B_i \) and \( 1 \leq j \leq 3 \) are the packet and sub packet numbers the MS uses to send on this slot.

Our proposed scheduling algorithm is equivalent to solving the following optimization problem on each slot:

\[
\text{Congested-Buffer-First: } \min \max B_i(s)
\]
Lowest-Rate-First: \( \max A(s) \)

Subject to the following constraints:

C1. \( \text{RoT} \leq \gamma, \gamma = 7dB. \)

C2. \( S(i, l, j + 1) - S(i, l, j) \geq 4. \)

C3. \( S(i, l + x, j + y) \neq S(i, l, j) \) for \( x \neq 0, y \neq 0. \)

C4. \( \sum_{k=1}^{11} N_k \leq M. \)

The above set of constraints can be re-stated as follows:

C1: The rise over thermal must be less than or equal to \( 7dB. \)

C2: All sub-packets of the same packet have to be spaced by four slots.

C3: The same user can not send more than one sub-packet per slot.

C4: The number of users transmitting on each slot must be less than or equal to the number of active users.

2.5 Analysis and Results

In our simulation, we consider a distributed wireless star network in which each sector performs scheduling for its active set of MSs without prior coordination with other sectors. Throughout our simulations, the MSs are assumed to be stationary, i.e., the effect of different mobility parameters (e.g., speed of MSs, multipath channel parameters), have not been investigated by our simulation models. In particular, in all cases, the MSs are assumed to have enough power to support reliable transmission of its assigned data rate.
Fig. 2.2 shows the simulation flow chart we used to generate the results for our rate assignment and scheduling techniques. The system is assumed to be always loaded with a prespecified number of users, i.e., whenever a MS leaves the system another one will join. As mentioned above, the file length of the FTP traffic is modelled by a truncated lognormal distribution (Eq. 2.11). Each MS is assigned a rate, using rate assignment scheme. All the scheduling techniques satisfy the H-ARQ and RoT constraints. The values of $TPRC$, $TPRQ$, and $SINR$ are given in Table 2.1. The values of $TPRD_k$ are given in Table 1.2. Average throughput is calculated as the number of bits successfully transmitted over the total time of transmission averaged over all users. The average service time is calculated as the total number of time slots the user takes to transmit its file (including the time slots the user is not transmitting on) averaged over all users. The results are divided into three categories, proposed rate assignment techniques, proposed scheduling techniques, and combined results.

As mentioned in Section 1.3.1.2 the 3GPP2 [56] requires that all resource management techniques, for 1xEV standards, to satisfy the fairness criterion. Fairness is defined as providing all mobile stations with a minimal level of throughput and is evaluated by determining the cumulative density function (CDF) of the normalized throughput, with respect to the average user throughput, for all users. In order to satisfy the fairness criterion, the standard specifies that the CDF shall lie to the right of the curve given by the three points in Table 1.3. The first point in the table requires that the normalized user throughput of at least 90% of the
users exceeds 0.1. Similarly, the second and third points require that the normalized user throughput of at least 80% and 50% of the users exceeds 0.2 and 0.5 respectively.

Throughout our results, we show that all our proposed techniques satisfy the fairness criterion.

### 2.5.1 Proposed Rate Assignment Techniques

To test our proposed rate assignment techniques the proportional fair scheduler [56] is used as the scheduling technique, and the subset of users chosen to transmit on each slot satisfies the RoT condition (Eq. 2.5) and H-ARQ constraints. The proposed techniques are compared with the highest rate assignment (HR) technique, which implies that each MS is assigned the highest rate its power and available data allow [56]. As we mentioned above the following two cases are investigated:

- **Case A:** \( R = \{R_k, 1 \leq k \leq 11\} \)

- **Case B:** \( R = \{R_k, 1 \leq k \leq 10\} \)

**Case A:** Figures 2.3(a), and 2.3(b) show that the proposed schemes outperform the highest rate scheme in terms of achieved throughput and service time. It is clear that the achieved system throughput is still far from the maximum achievable throughput bounds obtained in Section 2.2. From Figure 2.3(c), one can note that both the EPR rate assignment technique and highest rate schemes are
Figure 2.2: Simulation flowchart for the results in Section 2.5
not reaching the 7 dB RoT bound. This means that more improvement can be achieved for both schemes which will be further investigated.

Case B: From Figures 2.3(a), and 2.3(b), it is clear that a significant improvement is achieved when using the EPR rate assignment technique. As for FPR scheme, there is almost no improvement due to the RoT bound. Also it can be recognized that EPR technique outperforms the highest rate technique.

Figs 2.4(a), 2.4(b) show that both of the proposed techniques satisfy the fairness criterion required by the 3GPP2 (Section 1.3.1) as we can see that all the CDF curves are to the right of the fairness criterion line.

2.5.2 Proposed Scheduling Techniques

To fairly test our proposed scheduling techniques, highest rate assignment technique is used as the rate assignment technique. As mentioned above the two cases (A and B) are analyzed.

The proposed techniques are compared with the proportional fair scheduling technique and the subset of users chosen to transmit on each slot satisfies the RoT condition (Eq. 2.5) and H-ARQ constraints.

Case A: Fig 2.5(a) shows that the congested-buffer-first scheduling technique elevated the average sector throughput over the proportional fair scheduling algorithm and the lowest-rate-first scheduling technique. From Fig. 2.5(b), one can realize that the average service time for congested-buffer-first scheduling tech-
Figure 2.3: Rate assignment techniques, channel BW 1.25 MHz
Figure 2.4: Cumulative density function (Case A and B) with thirty MSs per sector
nique is better than the other two techniques. From Fig. 2.5(c), it is clear that the RoT did not reach the 7 dB threshold, which leaves the door open for more improvements.

**Case B:** Fig. 2.5(a) shows that the congested-buffer-first scheduling technique elevated the average sector throughput up to 900 kbps. Fig. 2.5(b), shows that the average service time for congested-buffer-first scheduling technique has been significantly improved over case A. From Fig. 2.5(c), it is clear that the RoT is still below the 7 dB threshold.

Figs. 2.6(b) and 2.6(a) show that the proposed scheduling techniques satisfy the fairness criterion required by 3GPP2.

The above results support the theory of using the highest rate R11 only if we have one active user per sector.

### 2.5.3 Proposed Scheduling and Rate Assignment Techniques

In this section we tested the affect of changing the rate assignment technique on the different scheduling algorithms. The EPR rate assignment technique (case B) is used as the rate assignment technique. From Fig. 2.7(a) we can see that the highest average throughput (1050 kbps) is reached using the lowest-rate-first scheduling technique. Fig. 2.7(b) shows that the average service time remains the same. From Fig. 2.7(c), it is clear that the RoT constraint is satisfied.

Fig. 2.8 shows that the proposed scheme satisfies the fairness criterion required
Figure 2.5: Scheduling techniques, channel BW 1.25 MHz
Figure 2.6: Cumulative density function (Case A and B) with thirty MSs per sector
2.6 Summary

In this chapter, the developed RoT model is used to determine a bound on the maximum theoretical throughput that can be achieved over the R-PDCH. Our analysis shows that the highest rate, $R_{11}$, is adequate only if we have one active user in the sector. It is shown that the maximum instantaneous throughput can be achieved if only a subset of active users are allowed to transmit on each time slot. We proposed two new rate assignment techniques. Our simulation results show that higher throughput and lower service times are achieved using the new proposed techniques. We also proposed two new scheduling techniques, our simulation results show that the proposed algorithm enhances the sector throughput compared to the widely used proportional fair scheduling algorithm. It should be noted that the highest average throughput and lowest service time are achieved with the EPR rate assignment technique (case B) and lowest-rate-first scheduling technique.

We also showed that all the proposed techniques satisfy the fairness criterion and the RoT constraint required by the 3GPP2.
Figure 2.7: EPR rate assignment technique (Case B) with different scheduling algorithms, channel BW 1.25 MHz
Figure 2.8: EPR rate assignment technique (Case B) (Cumulative density function with thirty MSs per sector)
Chapter 3

Analytical Models for Scheduling Techniques in cdma2000 1xEV

Star Networks

Results of the previous chapter show that the second proposed rate assignment technique (Section 2.3) and the second proposed scheduling techniques (lowest-rate-first) (Section 2.4) outperform the other techniques.

In this chapter we compare the performance of the second proposed scheduling technique with other scheduling techniques. In particular, we develop an analytical model for lowest-rate-first, highest-rate-first priority scheduling techniques, and two round-robin fair scheduling techniques over the reverse data channel in cdma2000 1xEV. For these scheduling techniques, the distribution of the mobile stations among the possible data rates is modelled as a Markov process. An analytical expression for the steady state system throughput is derived from the steady state distribution of the above Markov process. The developed model is validated through simulations. Also, the developed Markov model is extended to include the effect of the cross layer design between the H-ARQ protocol, rate assignment, and scheduling and general expressions for the steady state system throughput and file transmission delay are derived.

In our model we assume that each MS keeps its assigned autonomous rate
for the whole transmission session. The scheduler chooses a subset of MSs to transmit on each time frame based on its scheduling scheme, while satisfying the rise over thermal (RoT) constraint. Through out this chapter the second proposed rate assignment technique (Section 2.3.2) is used.

The rest of this chapter is organized as follows. The scheduling techniques models are developed in Section 3.1. The scheduling process and the cross layer design Markov models are developed in Sections 3.2 and 3.3 respectively. Finally a summary of the chapter is given in Section 3.4.

3.1 Scheduling Techniques Models

Scheduling algorithms are one of the most important mechanisms to guarantee QoS parameters such as delay and throughput. Many scheduling techniques were proposed for wireless links over the last few years [35] [41]. In this section we model and analyze the performance of the lowest-rate-first, highest-rate-first, and two round-robin scheduling techniques. We assume that all frames are accepted from the first transmission and the system is now loaded with $M$ MSs. The standard [49] guarantees that the BS knows the current number of MSs to be scheduled, which is done by transmitting initial preambles from the MS to BS indicating the imminent arrival of data packets.

The scheduling algorithm allows a subset of MSs to transmit on each time slot, while satisfying the RoT constraint given by Eq. (2.7). Let $x = (x_1, \ldots, x_r)$
denote the distribution of MSs among different data rates, where \( r \) is the number of allowable data rates. Let \( a_k(x) \) denote the number of mobile stations assigned a data rate \( R_k \) and allowed (by the scheduling technique) to transmit at the \( t^{th} \) time slot given \( x \). In what follows, we develop an analytical expression for \( a_k(x) \), \( k = 1, \cdots, r \) for all the above scheduling techniques.

### 3.1.1 Lowest-Rate-First

In this scheme, users with an assigned rate \( R_k \) are given higher priority to transmit over users with assigned rate \( R_{k'} \), \( k' > k \). Satisfying the RoT constraint (Eq. 2.7) with the lowest-rate-first scheduling algorithm implies that \( a_k(x) \) is given by

\[
a_k(x) = \begin{cases} 
    x_k, & 0 < \sum_{i=1}^{k} c_i x_i \leq 0.8005 \\
    \left[ \frac{0.8005 - \sum_{i=1}^{k-1} c_i x_i}{c_k} \right], & \sum_{i=1}^{k} c_i x_i > 0.8005, \text{ and } 0 < \sum_{i=1}^{k-1} c_i x_i < 0.8005 \\
    0, & \sum_{i=1}^{k} c_i x_i > 0.8005, \text{ and } \sum_{i=1}^{k-1} c_i x_i \geq 0.8005
\end{cases}
\]

(3.1)
where $c_i$, $1 \leq i \leq r$, are given by Eq. (2.6). Taking the floor of the term
\[
\frac{0.8005 - \sum_{i=k+1}^{r} c_i x_i}{c_k}
\]
in the above equation reflects the fact that the number of MSs should always be an integer.

3.1.2 Highest-Rate-First

Similarly, in this scheme, users with an assigned rate $R_k$ are given higher priority to transmit over users with assigned rate $R_{k'}$, $k' < k$. Satisfying the RoT constraints (Eq. 2.7) with the highest-rate-first scheduling algorithm implies that $a_k(x)$ is given by

\[
a_k(x) = \begin{cases} 
  x_k, & 0 < \sum_{i=k}^{r} c_i x_i \leq 0.8005 \\
  \left[ \frac{0.8005 - \sum_{i=k+1}^{r} c_i x_i}{c_k} \right], & \sum_{i=k}^{r} c_i x_i > 0.8005, \text{ and} \\
  0 < \sum_{i=k+1}^{r} c_i x_i < 0.8005 & (3.2) \\
  0, & \sum_{i=k}^{r} c_i x_i > 0.8005, \text{ and} \\
  \sum_{i=k+1}^{r} c_i x_i \geq 0.8005 
\end{cases}
\]
3.1.3 Round-Robin Scheme 1

In this scheme, users are divided into \( r \) groups according to their data rates. Depending on the distribution of MSs among different rates, for each time slot the scheduler loops \( z(x) \) times (the value of \( z(x) \) is determined below) through these groups starting from the one with the lowest data rate. Thus each scheduler loop is divided into \( r \) steps. During each step, the scheduler allows an additional user with rate \( R_k \) to transmit if this does not violate the RoT constraint (Eq. 2.7).

Let \( \Delta_{k,i} \in \{0, 1\} \) denote the increase, during the \( i^{th} \) loop of the scheduler, in the number of users with data rate \( R_k \) that are allowed to transmit. Satisfying the RoT constraint with this scheduling scheme implies that \( \Delta_{k,i} \) is calculated by the following recursion:

\[
\Delta_{k,i} = \begin{cases} 
1, & \sum_{j=1}^{r} c_j \left( \sum_{l=1}^{i-1} \Delta_{j,l} \right) + \sum_{j=1}^{k} c_j \leq 0.8005, \ i \leq x_k \\
0, & \text{otherwise} 
\end{cases}
\]

(3.3)

In the equation above, the term \( \left( \sum_{l=1}^{i-1} \Delta_{j,l} \right) \) denotes the number of rate \( R_j \) users that are added during the first \( i - 1 \) scheduler loops.

Recall that in the 1xEV system (Sec. 1.3.1), each user is allowed to transmit at most once during each time slot. Thus, for each time slot, the scheduler keeps on looping until all users are allowed to transmit, i.e., for \( \max_{k=1}^{x_k} \) times, or until the
following condition is satisfied during any \( k^{th} \) step of the scheduler loop

\[
\Delta_{k,i} = 0, \quad \Delta_{k-1,i} = 0, \ldots, \quad \Delta_{1,i} = 0
\]

and \( x_{k,i}' \neq 0, \, 1 \leq k \leq r, \)

where \( x_{k,i}' \) denotes the number of users that have not been allowed to transmit yet in the rate \( R_k \) group before the \( i^{th} \) loop of the scheduler, i.e.,

\[
x_{k,i}' = x_k - \sum_{j=1}^{i-1} \Delta_{k,j}.
\]

Eq. 3.4 means that the scheduler quits looping whenever no additional users can be added to the pool of users that are allowed to transmit due to the RoT constraint. Let \( i_{\min} \) denote the smallest positive integer \( i \) that satisfies the condition in Eq. 3.4. Then \( z(x) \) is given by

\[
z(x) = \min \{ \max_{k=1}^{r} x_k, i_{\min} \},
\]

and the number of users with data rate \( R_k \) and allowed to transmit is given by

\[
a_k(x) = \sum_{i=1}^{z(x)} \Delta_{k,i}
\]

3.1.4 Round-Robin Scheme 2

Similarly, in this scheme, users are divided into \( r \) groups according to their data rates. However, in this scheme, the scheduler starts looping from the group with the highest data rate. The scheduler keeps on looping until all users are allowed to transmit or until the following condition is satisfied.
Figure 3.1: Round Robin Scheme 1 Fair Scheduler

\[ \Delta_{k,i} = 0, \Delta_{k-1,i-1} = 0, \ldots, \Delta_{1,i-1} = 0 \]

(3.8)

and \( x_{k,i}' \neq 0, 1 \leq k \leq r \).

Eq. 3.8 means that the scheduler quits looping whenever no additional users could be added to the pool of users that are allowed to transmit (due to the RoT constraint).

Let \( i_{\text{min}} \) denote the smallest positive integer \( i \) that satisfies the condition in Eq. 3.8. Let \( z(x) \) denote the required number of iterations for each time slot. Satisfying the above constraints implies

\[ z(x) = \min \{ \max_{k=1}^{r} x_{k,i_{\text{min}}} \} \]

(3.9)
Also, satisfying the RoT constraint with the round-robin scheme 2 scheduling algorithm implies that \( a_k(x) \) is given by

\[
a_k(x) = \sum_{i=1}^{z(x)} \Delta_{k,i}
\]  
\[\text{(3.10)}\]

where \( \Delta_{k,i} \) is calculated by the following recursion:

\[
\Delta_{k,i} = \begin{cases} 
1, & \sum_{j=1}^{r} \sum_{l=1}^{i-1} c_j \Delta_{j,l} + \sum_{j=k}^{r} c_j \leq 0.8005, \ i \leq x_k \\
0, & \text{otherwise}
\end{cases}
\]  
\[\text{(3.11)}\]

### 3.2 Scheduling Process Model

Markov models are some of the most powerful tools available for analyzing communication systems. This analysis yields results for the steady state of the system. A Markov chain is said to be irreducible if it is possible to get to any state from any state. The states of a finite-state, irreducible Markov chains are all recurrent. A state \( x \) is said to have a period \( d \) if any return to itself must occur in some multiple of \( d \) time steps. An irreducible Markov chain is said to be aperiodic if \( d = 1 \) [84].

If the Markov model is irreducible, aperiodic and positive recurrent, the state probabilities reach steady state values that are independent of the initial state probabilities. The steady state probabilities \( \pi(x) \) for a state \( x \) could be obtained by solving the linear equations
\[ \pi(y) = \sum_{x} p_{xy} \pi(x) \]  
(3.12)

and

\[ \sum_{x} \pi(x) = 1 \]  
(3.13)

Let \( S_k(t) \) denote the number of mobile stations assigned a data rate \( R_k \) at the \( t^{th} \) time slot. We define a state vector \( S(t) = (S_1(t), S_2(t), \cdots S_r(t)) \). Note that a MS with an assigned rate \( R_k \) may not necessarily be allowed (by the scheduling algorithm) to transmit at a given time slot. The assumption that the system is always loaded with \( M \) MSs implies that \( \sum_{k=1}^{r} S_k(t) = M \) and hence the number of valid states is given by \( \binom{M+r-1}{r-1} \) [84].

Let \( F_k(t+1), k = 1, \cdots, r \) denote the number of mobile stations assigned a data rate \( R_k \) and leaving the system (after finishing their file transmission) at the end of the \( t^{th} \) time slot, i.e., at the beginning of slot \( t + 1 \). Similarly, let \( J_k(t+1), k = 1, \cdots, r \) denote the number of MSs that join the system at time slot \( t + 1 \) with an assigned rate \( R_k \). Thus

\[ S(t+1) = S(t) - F(t+1) + J(t+1). \]  
(3.14)

Let \( x = (x_1, \cdots, x_r) \), and \( y = (y_1, \cdots, y_r) \) be the sample of \( S(t) \) and \( S(t+1) \) respectively. Similarly let \( f = (f_1, \cdots, f_r) \), and \( j = (j_1, \cdots, j_r) \) be the sample of \( F(t+1) \) and \( J(t+1) \) respectively.

Then the state transition probability \( p_{xy} = p(S(t+1) = y|S(t) = x) \) is given by
\[ p_{xy} = \sum_{i,j} p(S(t+1) = y|f, j, x) \times p(J = j|f, x) \times p(F = f|x). \]  

(3.15)

The conditional probability \( p(S(t+1) = y|f, j, x) \) is given by

\[ p(S(t+1) = y|f, j, x) = \prod_{k=1}^{r} p(y_k = x_k - f_k + j_k) \]  

(3.16)

where,

\[ p(y_k = x_k - f_k + j_k) = \begin{cases} 
1, & y_k = x_k - f_k + j_k, \\
0, & y_k \neq x_k - f_k + j_k.
\end{cases} \]  

(3.17)

Example:

Let the total number of users \( M = 2 \), number of allowable rates \( r = 3 \). If \( x = (1, 1, 0) \), \( f = (0, 1, 0) \), and \( j = (1, 0, 0) \), then the new state \( y = (2, 0, 0) \).

Again, the assumption that the system is always loaded with \( M \) MSs implies that \( \sum_{k=1}^{r} f_k = \sum_{k=1}^{r} j_k \).

Example:

Let \( M = 2 \), \( r = 3 \), \( x = (2, 0, 0) \), and \( y = (2, 0, 0) \). Then the possible sets of \( f \) and their corresponding sets of \( j \) are

\( \{f = (0, 0, 0), j = (0, 0, 0)\}, \{f = (1, 0, 0), j = (1, 0, 0)\}, \{f = (2, 0, 0), j = (2, 0, 0)\} \).

Using the rate assignment technique described in section 2.3.2 implies that the new mobile stations joining the system at time slot \( t + 1 \) will be distributed uniformly over the possible rates. Then \( p(J = j|f, x) \) is given by
\[ p(J = j | f, x) = \frac{1}{\sum_{k=1}^{r} \frac{f_k + r - 1}{r - 1}} \]  

(3.18)

where the denominator in the equation above denotes the number of possible choices for \( j \) given \( f \).

Example:

Let \( r = 3 \), and \( f = (1, 1, 0) \). Then

\[ j \in \{(2, 0, 0), (1, 1, 0), (1, 0, 1), (0, 2, 0), (0, 1, 1), (0, 0, 2)\}. \]

Let \( p(F_k = f_k | x) \) denote the probability that a MS with an assigned rate \( R_k \) will finish its transmission at time slot \( t + 1 \). This probability is equal to the probability that the number of packets left at this MS buffer at the beginning of time slot \( t \) is equal to one and the scheduling algorithm allows this MS to transmit it.

Let \( B_{u,k} \) and \( L_u \) denote the number of packets and the file size in bits for the \( u^{th} \) MS with rate \( R_k \) respectively. Hence

\[ B_{u,k} = \frac{L_u}{b_k} \]

(3.19)

where \( b_k \) is a constant that denotes the number of bits per packet associated with rate \( R_k \). Then \( p(f_k | a_k(x)) \) can be approximated by

\[ p(f_k | a_k(x)) \approx \begin{cases} 
0, & a_k(x) < f_k \\
\left( \frac{a_k(x)}{f_k} \right)^{\frac{1}{b_k}} f_k \left( 1 - \frac{1}{b_k} \right)^{(a_k(x)-f_k)}, & f_k \leq a_k(x),
\end{cases} \]  

(3.20)

66
where
\[ \overline{B}_k = \frac{1}{b_k} E[L_k], \]  
(3.21)

where \( E[\cdot] \) denotes the expected value of the enclosed argument and \( L_k \) is determined assuming the rate assignment technique discussed in section 2.3.2. Thus,
\[ p(F = f|x) = \prod_{k=1}^{r} p(f_k|a_k(x)) \]  
(3.22)

where \( a_k(x) \) denote the number of mobile stations assigned a data rate \( R_k \) and allowed (by the scheduling technique) to transmit at the \( t^{th} \) time slot given \( S(t) = x \) as shown in Section 3.1.

By noting that the above Markov model is irreducible, aperiodic, and positive recurrent [85], there will be a unique steady-state probability \( \pi(x) \) for a state \( x \).

The steady state system throughput is given by
\[ \sum_{\forall x} \sum_{k=1}^{r} a_k(x) R_k \pi(x) \]  
(3.23)

where \( \pi(x) \) denotes the steady state probability distribution of \( S \).

3.2.1 Analysis and Simulation Results

The above model is applied to 1xEV-DO system, where \( r = 5 \). The values of \( TPDRC \), and \( SINR \) are given in table 2.1. The values of \( TPRD_k \) are given in Table 1.1. The above scheduling and rate assignment techniques were applied to the FTP traffic in which the file length distribution is modelled as a truncated lognormal distribution Eq. 2.11.
The simulation and analytical model results are shown in Fig. 3.2(a). Both scheduling and rate assignment techniques are adapted in simulation and analysis. One can realize that for $M < 5$ most of the users (with different rate distributions) can transmit in the same time slot while satisfying the RoT constraints, hence the achieved average throughput for $M < 5$ are the same for all four schemes.

In the course of computing the steady state probability vector $\pi(x)$ for Eq. 3.30, we noted that the states with large number of MSs and high data rates (waiting to transmit) dominate in the case of lowest rate first scheduling. This can be explained by noting that this scheduling technique allows a large number of low rate MSs to transmit first, which lowers the probability that MSs with higher rates can satisfy the RoT constraints, consequently they stay longer within the system waiting for service. This results in lowering the overall system throughput as compared to the highest-rate-first scheduling technique.

The same explanation also applies to the round-robin scheme 1 scheduling technique. Although it allows a combination of low rate MSs and high rate MSs to transmit, the priority in each round is given to low rate MSs.

The average RoT obtained by simulation for different values of $M$ is shown in Fig. 3.2(b). It is clear that the RoT constraints are satisfied for all four schemes. One should also note that all the considered scheduling techniques do not reach the 7 dB RoT bound.

By evaluating the second largest eigenvalue for the transition matrices $\{p_{xy}\}$,
Figure 3.2: Performance Results for cdma2000 1xEV-DO reverse data channel
we have also analyzed the rate of convergence [85] of the Markov chain for all schemes. None of the schemes shows any clear advantage over the other with regard to this criterion.

3.3 Cross Layered Design Model

In this section we extend the model proposed in Section 3.2 to model and evaluate the overall performance of the hybrid ARQ, rate assignment, and time slot scheduling over the reverse packet data channel in cdma2000 1xEV. Our model is based on the analysis of the cdma2000 1xEV turbo encoder (Appendix A) and the probability that a mobile station is scheduled for transmission. Expressions for the steady state system throughput and file transmission delay are derived from the steady state distribution of the Markov process. The developed models are validated through simulations. The obtained results show how the adaptation of scheduling to the status of the physical layer and H-ARQ operation constitutes a cost effective cross layer design.

In order to determine a lower bound for the probabilities of sub-packets acceptance, it is inevitable to examine the details of the turbo decoding process. Let $ps_{1,k}$ and $ps_{2,k}$ denote the probabilities that a packet, transmitted with rate $R_k$, is accepted from the first or second transmissions respectively. Let $b_k$ denote the number of data bits per packet associated with rate $R_k$. Thus
\[ p_{s_{i,k}} = (1 - (P_b(E))_{i,k})^{h_k} \]  

where \( i = 1, 2, 3 \) correspond to the first, second and third sub-packets respectively, and \( (P_b(E))_{i,k} \) denotes the BER for the \( i^{th} \) sub-packet with data rate \( R_k \). In Appendix A, based on the analysis of the cdma2000 1XEV turbo encoder with code rates \( R_{c_i} = 1/2, 1/3, 1/5 \), \( (P_b(E))_{i,k} \) is determined (Eq. A.9), and hence the values of \( p_{s_{i,k}} \) are obtained. We also show how different FEC coding rates \( R_{c_i} \) associated with the H-ARQ process affect the BER Fig. A.4.

As mentioned in section 1.3.1, the H-ARQ protocol has a 4 time slot cycle from the time the MS starts transmitting the sub-packet until it receives the ACK/NAK and processes it. Therefore, while the first sub-packet is transmitted in one time slot, the time elapsed until the second and third sub-packets are transmitted is 5 and 9 time slots respectively (see Fig. 1.4). Therefore, from these delay constraints, the average number of time slots (without the affect of scheduling Fig. 3.3) to transmit a packet is given by

\[ p_{s_{1,k}} + 5(1 - p_{s_{1,k}})p_{s_{2,k}} + 9(1 - p_{s_{1,k}})(1 - p_{s_{2,k}}), \]  

where \( p_{s_{1,k}} \) and \( p_{s_{2,k}} \) are the probabilities that the packet is accepted from the first or second retransmissions respectively. The probabilities \( p_{s_{1,k}} \) and \( p_{s_{2,k}} \) are evaluated by Eq. (3.24). When allowed by the scheduling algorithm, the MS can transmit different packets during the 40 msec cycle. Therefore the effective number of
time slots to transmit a file for a MS with data rate $R_k$ including retransmissions due to H-ARQ $T_k$ is given by

$$T_k = \overline{B}_k(p_{s1,k} + 2(1 - p_{s1,k})p_{s2,k} + 3(1 - p_{s1,k})(1 - p_{s2,k}))$$ (3.26)

where $\overline{B}_k$ denotes the number of packets associated with rate $R_k$ by Eq. 3.21.

Thus, the Mrakov model developed in section 3.2 can be extended to include the H-ARQ effect by changing Eq. 3.20. Hence $p(f_k | a_k(x))$ can be approximated by

$$p(f_k | a_k(x)) \approx \begin{cases} 
0, & a_k(x) < f_k \\
\binom{a_k(x)}{f_k} \left(\frac{1}{f_k}\right)^{f_k} \left(1 - \frac{1}{f_k}\right)^{(a_k(x)-f_k)}, & f_k \leq a_k(x),
\end{cases}$$ (3.27)
Given the steady-state probability $\pi(x)$ for a state $x$, the steady state average number of MS that is assigned data rate $R_k$ is given by

$$\bar{x}_k = \sum_{x \in \mathcal{X}} x_k(x)\pi(x)$$

and the steady state average number of MSs that is assigned data rate $R_k$ and is scheduled for transmission is given by

$$\bar{a}_k = \sum_{x \in \mathcal{X}} a_k(x)\pi(x)$$

Let $\bar{I}_k = \frac{\bar{a}_k}{\bar{x}_k}$ denote the steady state probability that a MS assigned data rate $R_k$ is scheduled for transmission. One can show that the number of time slots needed to transmit the first sub-packet is

$$\bar{I}_k + 2(1 - \bar{I}_k)\bar{I}_k + 3(1 - \bar{I}_k)^2\bar{I}_k + \cdots = \sum_{l=0}^{\infty} (1 + l)(1 - \bar{I}_k)^l \bar{I}_k$$

$$= \frac{1}{I_k}$$

Similarly, number of time slots needed to transmit the second and third sub-packets are given by

$$\sum_{i=0}^{\infty} \sum_{t=0}^{\infty} (2 + l + i)(1 - \bar{I}_k)^{t+i+i} \bar{I}_k^2 = \frac{2}{I_k}$$

and

$$\sum_{l=0}^{\infty} \sum_{t=0}^{\infty} \sum_{j=0}^{\infty} (3 + l + i + j)(1 - \bar{I}_k)^{i+l+j} \bar{I}_k^3 = \frac{3}{I_k}$$

respectively.

Let $D_k$ denote the number of slots needed to transmit a file by a MS with an assigned rate $R_k$, including the effect of scheduling and H-ARQ. From Eq. (3.26)
and by noting the number of time slots required for each transmission (obtained above), we obtain

\[
\bar{D}_k = \frac{B_k}{I_k} (p_{s_{k,1}} + 2(1 - p_{s_{k,1}})p_{s_{k,2}} + 3(1 - p_{s_{k,1}})(1 - p_{s_{k,2}}))
\]

Therefore, the steady state average file transfer delay, including both transmission and access, is given by

\[
\frac{1}{r} \sum_{k=1}^{r} \bar{D}_k T_s
\]

where \( T_s \) is the length of each time slot, \( r \) is the number of allowable rates, and \( B_k \) is the average number of packets as defined in Eq. 4.27.

Due to the H-ARQ constraints, each packet with data rate \( R_k \) needs more than one time slot to be transmitted. Hence, the effective data rate is given by

\[
R_{k(f)} = \frac{R_k}{p_{s_{1,k}} + 2(1 - p_{s_{1,k}})p_{s_{2,k}} + 3(1 - p_{s_{1,k}})(1 - p_{s_{2,k}})}
\]

and the steady state system throughput is given by

\[
\sum_{k=1}^{r} \bar{\sigma}_k R_{k(f)}
\]

3.3.1 Analysis and Simulation Results

We consider a system where the number of allowable rates \( r \) is set to seven \((R_1, \cdots , R_7)\). The values of \( TPRC \), \( TPRQ \), and \( TPRD_k \) are given in Tables 2.1 and 1.2. Also in all the results \( W = 1.25 \text{ MHz} \) which implies single-carrier cdma2000, the extension of this work to multi-carrier system is a straightforward
computation. The probability of sub-packet acceptance for different values of $\tau$ are determined based on the values $E_b/N_0$ (Eq. A.22).

The above models were applied to the FTP traffic in which the file length distribution is modelled as a truncated lognormal Eq. 2.11. The system is assumed to be always loaded with a prespecified number of users, i.e., whenever a MS leaves the system another one will join. The rate assignment is performed whenever a new MS joins the system. Each MS keeps its assigned rate for the duration of its transmission. The simulation results presented in this section are obtained by simulating the system, for each point on the curves, over 500,000 time slots.

The simulation and analytical results for the steady state throughput and file transfer delay of the lowest rate first scheduling technique are shown in Figs. 3.4 and 3.5, respectively, for different values of $\tau$. The corresponding results for the highest rate first scheduling technique are shown in Figs. 3.6 and 3.7. In the case of well-conditioned channel, packets are assumed to be accepted from the first sub-packet transmission, i.e., $p_{sk,1} = 1$. In the case of an ill-conditioned channel, $p_{sk,1} \leq 1$, and the H-ARQ technique is applied.

For well-conditioned channel, it is clear that lower values of $\tau$ allow higher throughput and lower file transfer delay. This can be explained by noting that the scheduling technique allows more MSs to transmit on each time slot for lower values of $\tau$. Basically, since the channel is well-conditioned, the throughput will not be affected because there would be no re-transmission. On the other hand, for ill-conditioned channel, lower values of $\tau$ result in a large degradation in both the
Figure 3.4: Steady state link throughput for lowest-rate-first scheduling technique.

system throughput and file transfer delay which can be explained by the required increase of the sub-packets retransmissions. In other words, higher values of $\tau$ help bring up the performance of the ill-conditioned channel to reach that of the well-conditioned channel but it will reduce the overall system throughput in order to satisfy the scheduling constraints.

3.4 Summary

In this chapter, four scheduling techniques for the reverse data channel of the 1xEV star network were modelled. The distribution of the MSs among the possible data rates is modelled as a Markov process. The effects of scheduling and
Figure 3.5: Steady state average file transfer time per user for lowest-rate-first scheduling technique.

Figure 3.6: Steady state link throughput for highest-rate-first scheduling technique.
Figure 3.7: Steady state average file transfer time per user for highest-rate-first scheduling technique.

rate assignment techniques are included in the Markov model. The validity of the obtained models was confirmed with simulations. The results show that highest-rate-first scheduling technique outperforms all the other scheduling techniques in the sense of achieved throughput. The simulation results are fairly close to their analysis counterparts, which validates the assumptions of the analysis.

We extended our work to develop analytical expressions for the steady state link throughput and file transfer delay that are based on modelling the cross layer design involving the physical layer H-ARQ, network layer rate assignment, and MAC layer time slot scheduling techniques used over the reverse data channel. The results obtained show how the choice of the targeted signal to noise and interference ratio affect the overall system performance.
Chapter 4

Caching Effects on Cross-layered WiMAX Mesh Networks

The IEEE 802.16 backhaul mesh network consists of a base station (BS) and multiple service stations (SSs). The BS serves as a gateway for the SSs to the Internet, and each SS serves as a base station collecting aggregated traffic from end users in different star networks (see Fig. 4.1). In previous chapters we considered different rate assignment and scheduling techniques for star connected networks, the rate assignment and inherent cross layer design techniques that we adapted for star networks are naturally extended to backhaul mesh oriented networks.

In this chapter we investigate some techniques that could be used to improve the performance of backhaul mesh networks. A common characteristic of such networks is the use of mesh multi-hop networking to improve efficiency. IEEE 802.16 supports two modes of operation, point to multi-point (PMP) and mesh mode. In PMP each SS directly communicates with the BS through a single hop link, which requires all SSs to be within clear line of sight to the BS. In contrast, in mesh mode, the SSs can communicate with mesh BS and with each other through multi hop routes via other SSs. Mesh topology not only extends the network coverage and increases the capacity in non-line of sight environments, it also provides higher network reliability and availability when node or link failures occur,
or when channel conditions are poor [80].

In the last few years, the importance of cross layer design resulted in many proposals. In [67] the author surveyed the cross layer designs based on how the layers are coupled. The layer coupling is divided into four categories: creation of new interfaces [68] and [69], merging of adjacent layers, design coupling without new interfaces [70], and vertical calibration across layers [71].

In this chapter, we provide cross layer techniques to improve the effects of forward error correction (FEC) coding amongst other measures on the overall network performance. Coding is one of the important measures to improve network performance. An opportunistic coding approach increases the throughput of wireless networks by minimizing the transmission delay. Turbo coding [89] is
one of the important coding techniques used on the physical layer. While turbo
codes have been employed for some time, recently upper layer FE codes such as
LT codes [98], Raptor codes [97], and Fountain codes [96] have been introduced.
Fountain codes are a class of erasure codes with the property that a potentially
limitless sequence of encoding symbols can be recovered from any subset of size
equal to or slightly larger than the number of source symbols.

In this chapter we relate the parameters of the higher layer Fountain codes
to those of the physical layer codes to present a cross-layered coding technique.
Based on the bit error rate of physical layer codes, the upper layer Fountain codes
parameters are designed to obtain a prespecified performance. The performance
of the proposed cross-layered coding technique is found to be comparable to
that of the physical layer based Hybrid-ARQ technique. As an application, we
study the performance of a WiMAX backhaul mesh network where caching is al-
lowed at the service stations and the proposed coding technique is used instead
of H-ARQ. In particular, the performance of the above network is modelled as
a Markov process and analytical expressions for the steady state system perfor-
mance are derived from its associated steady state distribution. The developed
model is validated through simulations.

The rest of this chapter is organized as follows. In Section 4.1 the proposed
cross layered coding technique is described, and its performance compared with
that of the H-ARQ technique. In Section 4.2 we model and analyze the effect of
caching on WiMAX backhaul mesh network using Markov process. An analytical
expression for the steady state system transmission delays and throughput are derived from the steady state distribution of the above Markov processes. The developed model is validated through simulations. Finally a summary is given in Section 4.3.

4.1 A Cross-Layered Coding Technique

In this section we relate the parameters of the higher layer Fountain code to those of the physical layer turbo code to present a cross-layered coding technique. In our proposed technique we assume that each data packet is encoded using turbo code with coding rate 1/2. Based on the bit error rate of physical layer turbo code, the upper layer Fountain code parameters are designed to reach the required performance. The performance of the proposed cross-layered coding technique is compared with that of physical layer H-ARQ technique [56]. In what follows we describe both coding techniques.

4.1.1 Proposed Cross-Layered Code

Upper layer forward error correction (FEC) codes such as LT codes [98], Raptor codes [97], and Fountain codes [96] are mainly designed for erasure channels. Fountain code was proposed by Mackay [96]. Basically it produces packets that are random functions of the whole file. The transmitter sends these packets without any knowledge of the packets been received. If the original file size is K
packets the receiver needs $N_R$ packets, slightly larger than $K$ to restore the data file.

Let $N_R = K + E$ where $E$ is the number of excess packet needed to restore a file of $K$ packets. For any file size, the probability that the receiver fails to recover the file is bounded by $\delta(E) \leq 2^{-E}$ [96]. Hence, the probability that the file is recovered correctly is given by $p_f \geq 1 - 2^{-E}$ and the number of extra packets required at receiver is $E = \log_2 \frac{1}{1-p_f}$. Thus

$$N_R \approx K + \log_2 \frac{1}{1-p_f}$$ (4.1)

In the proposed cross-layered approach we assume that a turbo encoder with code rate $R_{c_1} = 1/2$ is used to encode each packet on the physical layer, this collaborates with the higher layer error concealing Fountain code. Let $N_{CL}$ and $\overline{N}_R$ denote the number of packets transmitted and the average number of packets decoded correctly at the receiver. Thus, $\overline{N}_R$ is given by the mean of a binomial distribution i.e.,

$$\overline{N}_R = p s_1^L N_{CL},$$ (4.2)

where $p s_1^L$ is the probability that a packet is accepted after travelling $L$ links, $p s_1$ is the probability that a packet is successfully decoded using rate 1/2 turbo code (i.e. $i = 1$ in Eq. A.19). Therefore the number of packets that should be sent in order to satisfy the required $p_f$ is given by
\[ N_{CL} = \left\lceil \frac{K + \log_2 \frac{1}{(1-p_{el})}}{ps_1} \right\rceil, \]  

(4.3)

The Fountain code parameter \( N_{CL} \) is designed based on information received from the lower layer turbo decoder. We assume the existence of a mechanism at the lower layer that estimates \( ps_1 \), for example using some CRC attached to each packet and counting the statistics of CRC checking. It is interesting to see in Eq. (4.3) that the numerator represents the Fountain code functionality, while the dominator is the merit of turbo coding.

Fountain codes are suited between the application and transport layers. In this regard we recommend utilizing UDP as the transport vehicle for its simplicity and resilience to mobility. The utilization of Fountain codes on top of UDP more than compensates for occasional loss or error of UDP datagrams. On the other hand, intermittent and channel burst errors are better handled by turbo codes at the physical layer.

Let \( B_{CL} \) denote the average number of transmitted bits per file for the proposed technique. Hence \( B_{CL} \) is calculated based on details of turbo code,

\[ B_{CL} = \frac{\Gamma N_{CL}}{R_{c1}}, \]  

(4.4)

where \( \Gamma \) denotes the number of data bits per packet, and \( R_{c1} = 1/2 \) is the coding rate of the physical layer turbo encoder. Define the efficiency as the number of information bits over the total number of transmitted bits. Therefore, the efficiency
of the proposed cross-layered code is given by

\[ \eta_{CL} = \frac{rK}{B_{CL}} = \frac{K}{N_{CL}} \]  

(4.5)

and the average number of time frames at the physical layer required to transmit a file considering the cumulative effect of Fountain and turbo code,

\[ D_{CL} = N_{CL} + L - 1 + \sum_{i=1}^{L-1} N_{Q_i}, \]  

(4.6)

where \( L - 1 \) reflects the arrival of the first packet at destination after travelling \( L \) links, one should note that all the other packets arrive in a pipeline after the first one. \( N_{Q_i} \) is the number of packet delay per queue as given in Eq. (4.12).

We are studying only one possible turbo code rate i.e. 1/2 and then the upper layer Fountain code is designed accordingly, but there is nothing in principle against adjusting the lower layer turbo code to other code rate i.e. 1/3. In this case the upper layer Fountain code design would automatically be adjusted.

4.1.2 Hybrid ARQ

H-ARQ protocol is usually applied in order to improve the efficiency of the transmission channel (see Chapter 1). In this section we consider a turbo encoder with a final coding rate of \( R_c = 1/5 \). Each encoded packet is divided into three sub-packets. The first sub-packet has encoding rates of \( R_{c_1} = 1/2 \), the first and second sub-packets combined will have an effective lower encoding rate of \( R_{c_2} = \)
1/3, the three sub-packets combined will reach the final encoding rate of $\Gamma_{c_3} = 1/5$.

Let $ps_1$, $ps_2$ and $ps_3$ denote the probabilities that a packet is decoded correctly after the first, second and third transmissions respectively. Let $p_p$ denote the probability that a packet is finally decoded correctly (see Fig 3.3). Thus,

$$p_p = ps_1 + (1 - ps_1)ps_2 + (1 - ps_1)(1 - ps_2)ps_3$$  \hspace{1cm} (4.7)

where $ps_i, i = 1, 2, 3$ is obtained from Eq. (A.19).

Hence, the end-to-end probability that the requested file is decoded correctly is given by

$$p_f = p_p^{LK}$$  \hspace{1cm} (4.8)

where $K$ and $L$ denote the number of packets per data file and the number of links each packet has to travel respectively.

Eq. (4.8) effectively says that the total file is successfully decoded at the receiver if all packets successfully propagate through all links to destination.

Let $B_{HA}$ denote the average number of bits required to transmit a file if the H-ARQ technique is used. Hence, $B_{HA}$ is given by

$$B_{HA} = \Gamma K \left( \frac{ps_1}{R_{c_1}} + \frac{(1-ps_1)ps_2}{R_{c_2}} + \frac{(1-ps_1)(1-ps_2)}{R_{c_3}} \right);$$  \hspace{1cm} (4.9)

where $ps_1$, $ps_2$ denote the probability that the packet is decoded correctly after first, and second sub-packet transmission respectively. $\Gamma/R_{c_1}, \Gamma/R_{c_2},$ and $\Gamma/R_{c_3}$
Figure 4.2: Delays due to propagation and queueing over number of links
denote the effective number of bits transmitted if a packet is accepted after the first, second, and third transmission respectively.

Thus, the efficiency of the H-ARQ technique is given by

\[
\eta_{HA} = \frac{\Gamma K}{B_{HA}}
\]

\[
= \frac{\frac{1}{R_{e1}}}{\frac{1}{R_{e2}} + \frac{1}{R_{e3}}} - \left(\frac{1 - p_{s1}}{R_{e1}} + \frac{1 - p_{s2}}{R_{e2}}\right)\frac{1}{R_{e3}}
\]

As shown in Fig. 1.4 the packets are sent in a pipe. Assume that each sub-packet is transmitted over one time frame. Therefore, the average number of time frames required to transmit a file over one link, while using the H-ARQ technique is \(K(p_{s1} + 2(1 - p_{s1})p_{s2} + 3(1 - p_{s1})(1 - p_{s2}))\) and the end-to-end average number of time frames required to transmit a file taking into account the number of links and queueing effect (see Fig. 4.2) is given by
\[ D_{HA} = K(ps_1 + 2(1 - ps_1)ps_2 + 3(1 - ps_1)(1 - ps_2)) \\
+ (L - 1)(ps_1 + 5(1 - ps_1)ps_2 + 9(1 - ps_1)(1 - ps_2)) \]
\[ + \sum_{i=1}^{L-1} N_{Qi}, \] \hspace{1cm} (4.11)

where the term \( (L - 1)(ps_1 + 5(1 - ps_1)ps_2 + 9(1 - ps_1)(1 - ps_2)) \) reflects the average delay in time frames at which the first packet arrives at destination (Assuming the same transmission policy used by 1xEV standards Section 1.3.1). Remaining packets of the file arrive sequentially at the receiver in a time equal to the first term. \( N_{Qi} \) denotes the delay in packets (time frames) associated with each queue and the \( \sum_{i=1}^{L-1} N_{Qi} \) represents the total queueing delay over all links.

### 4.1.3 Analysis Results

In this section efficiency and file transmission delays are obtained for both H-ARQ and the proposed cross-layered coding techniques. For analysis purpose, we consider M/D/1 queues at all SSs. The expected number of packets at the \( i^{th} \) SS queue is given by [99]

\[ N_{Qi} = \frac{\rho_i^2}{2(1 - \rho_i)} \] \hspace{1cm} (4.12)

where \( \rho_i = \frac{\lambda_i}{\mu_i} = 0.5 \) denotes utilization factor (the ratio between the arrival \( \lambda_i \) and service rate \( \mu_i \)) for the \( i^{th} \) SS and it is assumed constant for all the queues.

We assume that the original file size is one thousand packets. The results are obtained for \( E_b/N_0 = 0, 1, \cdots, 4 \) dB and the number of links \( L = 1, 2, \cdots, 5 \).
Figure 4.3: Probability of file acceptance $p_f$

For the H-ARQ case Fig. 4.3 shows the probability of file acceptance $p_f$ in Eq. (4.8) for different $E_b/N_0$ and different number of links. The same $p_f$ is used as an input parameter to conduct the analysis of the cross-layered coding technique so that we have a fair comparison. The figure shows that $p_f$ improves as $E_b/N_0$ gets higher and the effect of the number of links almost vanishes for high $E_b/N_0$.

Figs. 4.4 and 4.5 show the efficiency of the H-ARQ and cross-layered coding techniques as in Eqs. (4.10 and 4.5) respectively. It is clear that for $E_b/N_0 \geq 3$ dB the efficiency of the cross-layered coding technique is almost the same as this of the H-ARQ.

The effective number of time frames required to transmit a file for both H-ARQ and cross-layered coding techniques as in Eqs. (4.11 and 4.6) are shown in Figs. 4.6 and 4.7 respectively. One can notice that for $E_b/N_0 \geq 1$ dB the performance of

89
the cross-layered coding technique is comparable with that of the H-ARQ technique even for large number of links. Note that the cross-layered FEC approach is an end to end forward approach that does not suffer the ARQ dialog necessary at each link as in the case of H-ARQ. Thus, by using the cross-layered FEC we can reach almost the same performance of H-ARQ, with the privilege that the overhead of the H-ARQ technique is removed.

It is also important to mention that the proposed cross-layered coding technique fits better for files with large number of packets otherwise the efficiency will get lower. As an application, the proposed cross-layered FEC technique is used to encode data files while we study the effects of caching data on WiMAX backhaul mesh network.
Figure 4.5: Efficiency of cross-layer coding technique $\eta_{CL}$

Figure 4.6: Delay of H-ARQ technique in number of required time frames to transmit a file $D_{HA}$
Figure 4.7: Delay of cross-layered coding technique in number of required time frames to transmit a file $D_{CL}$

4.2 Network Performance Model

In this section we model and study the effect of caching on the performance of WiMAX backhaul mesh network. As mentioned above the proposed cross-layered coding technique is used as the WiMAX coding scheme.

We study a distributed WiMAX backhaul mesh network where each local SS has cached some of the information required by its end users including actual web-pages, files, network topology, and other control parameters. Each end user node has the option to contact the BS directly or a SS, it will select either route based on the required QoS. If the request is processed through a SS, the SS tries to locate the required data in its cache or in other SSs caches along its route to the BS. Hence, the delay is reduced. Typically, the data rate assigned to an end user
node served directly through the BS is lower than the one assigned to an end user
node served by a SS. An example of the used setting is given in Fig. 4.1.

4.2.1 Caching Effect Model

Let $p_{\text{miss}}$ denote the probability that a file requested by the end user node is
not found within its local SS cache, $Z_{SS}$ denote the total number of SSs in the
mesh network. Assume a network with a maximum depth of $L_{\text{max}}$ links, where
$2 \leq L_{\text{max}} \leq Z_{SS} + 1$. Let $K_{BS}$ and $K_{SS}$ denote the number of data packets per
file if the end user node is served by BS and SS respectively. Let $N_{BS}$ denote the
total number of packets required to transmit a file if the end user node is served
directly by the BS. We can also notice that if the user is served by the BS $L = 1$,
hence substitute in Eq. (4.3)

$$N_{BS} = \left\lfloor \frac{K_{BS} + \log_2 \frac{1}{\delta}}{ps_1} \right\rfloor,$$

(4.13)

where $(1 - \delta = p_f)$ denote the required probability of successful file recovery.

As for the requests served by a SS the total number of packets depends on the
number of the links the requested file has to traverse until the end user node. Let
$N_{SS}(l)$ denote the total number of packets if the file travels $l$ links.

$$N_{SS}(l) = \left\lfloor \frac{K_{SS} + \log_2 \frac{1}{\delta}}{ps_1^l} \right\rfloor,$$

(4.14)

Hence, from Fig. 4.1 on average $N_{SS}$ is given by
\[ N_{SS} = \left[ \sum_{i=2}^{L_{\text{max}}} \frac{N_{SS}(i)p_{\text{miss}}^i + \sum_{j=1}^{i-1} N_{SS}(j)p_{\text{miss}}^j}{L_{\text{max}} - 1} \right] \tag{4.15} \]

To clarify the above equation, assume the case when \( i = 3 \), i.e. the end user node is located at distance of three links from the BS. Hence, the request may travel only one link with a probability \((1 - p_{\text{miss}})\) (the requested file is cached at the first SS along its route to the BS). The other scenario is the file is not cached at the first SS but it is cached at the second SS along its route, therefore the file travels two links with probability \(p_{\text{miss}}(1 - p_{\text{miss}})\). Finally, if the file is not cached at any of the SSs along its route the request is served by the BS and the file has to travel three links with a probability \(p_{\text{miss}}^2\).

Define \( x = \{x_1, x_2\} \), where \( x_1 \) and \( x_2 \) denote the number of requests directly served by the BS or by all SSs respectively. \( W_{BS}(x) \) and \( W_{SS}(x) \) denote the available BW in packets per time frame for each request if it is processed by the BS or by a SS respectively. In a simple policy the end user node should decide to connect to the BS or SS based only on the availability of the requested file in the cache of the SS. In a more elaborate policy the end user node estimates the total file transmission time if it hooks to the BS or a SS and make its decision accordingly. Let \( T_{BS}(x) = \frac{N_{SS}}{W_{BS}(x)} \) and \( T_{SS}(x) = \frac{N_{SS}}{W_{SS}(x)} \) denote the number of time frames required to transmit a file if the request is served directly by the BS or a SS respectively. The end user node selects the shorter of the two with a certain probability. In this case the probability of the file being processed by BS is given by
\[ p_{BS}(x) = \frac{T_{ss}(x)}{T_{BS}(x)+T_{ss}(x)} \] (4.16)

The above equation implies that if \( T_{BS}(x) < T_{ss}(x) \), the probability that a request is served by BS is greater than the probability that it is served by a SS.

Let \( W \) denote the maximum WiMAX BW in packets per time frame for BS or any SS, hence \( W_{BS}(x) \) is given by

\[
W_{BS}(x) = \begin{cases} 
\frac{W}{x_1 + x_{BS_{miss}}} & x_1 + x_{BS_{miss}} \geq 1, \\
W & x_1 + x_{BS_{miss}} < 1,
\end{cases} \quad (4.17)
\]

The term \( x_{BS_{miss}} \) represents the average number of requests that are assigned to SSs and reached the BS due to cache missing. Assume that all SSs are distributed uniformly over \( L_{max} - 1 \) links and \( x_2 \) is distributed uniformly over all SSs (see Fig. 4.1). Hence, \( \frac{x_2}{L_{max} - 1} \) requests could reach the BS with probability \( p_{miss} \), another \( \frac{x_2}{L_{max} - 1} \) requests could reach the BS with probability \( p_{miss}^2 \) and so on. Therefore \( x_{BS_{miss}} \) is given by

\[ x_{BS_{miss}} = \sum_{k=1}^{L_{max} - 1} \frac{x_2}{L_{max} - 1} p_{miss}^k \] (4.18)

Assume that each SS will serve on average \( \frac{x_2}{Z_{SS}} \) requests, where \( Z_{SS} \) denote the total number of SSs, plus the number of requests assigned to other SSs and reach it due to \( p_{miss} \) denoted by \( x_{SS_{miss}} \). Let \( W_{ss}(x, l) \) denote the average available BW
at any SS distant \(l - 1\) links from BS. Therefore

\[
W_{SS}(x, l) = \begin{cases} 
\frac{W}{x_{SS} + x_{SS_{miss}}(l)} & \frac{x_{SS}}{Z_{SS}} + x_{SS_{miss}}(l) \geq 1 \\
W & \frac{x_{SS}}{Z_{SS}} + x_{SS_{miss}}(l) < 1;
\end{cases}
\]  

(4.19)

Similar to Eq. 4.18, \(x_{SS_{miss}}(l)\) is given by,

\[
x_{SS_{miss}}(l) \approx \sum_{k=1}^{L_{max} - l} \frac{x_{2}}{L_{max} - 1} p_{k}^{l}
\]

(4.20)

Similar to Eq. 4.15, if the requested node is located at a distance of 3 links from the BS, the available BW for this request is \(W_{SS}(x, 3)\) with probability \((1 - \text{p}_{\text{miss}})\), \(W_{SS}(x, 2)\) with probability \(\text{p}_{\text{miss}}(1 - \text{p}_{\text{miss}})\) or \(W_{BS}(x)\) with probability \(\text{p}_{\text{miss}}^{2}\). Hence, the average BW available for any request served by a SS is given by

\[
W_{SS}(x) = \sum_{i=2}^{L_{max}} \frac{W_{BS}(x)p_{i-1}^{i-1} + \sum_{j=2}^{i} W_{SS}(x, j)p_{j}^{i-j}(1 - \text{p}_{\text{miss}})}{L_{max} - 1}
\]

(4.21)

### 4.2.2 Transmission Process Model

As discussed previously, Markov models are among the most powerful tools available for analyzing communications systems. If the Markov model is irreducible, aperiodic and positive recurrent, the state probabilities reach steady state values that are independent of the initial state probabilities. The steady state probabilities \(\pi(x)\) for a state \(x\) could be obtained by solving the linear equations
\[ \pi(y) = \sum_{x} p_{xy} \pi(x) \]

and

\[ \sum_{x} \pi(x) = 1 \]

where \( p_{xy} \) is the transition probability from state \( x \) to state \( y \).

In this section, using Markov models we derive the steady state distribution of the number of requests served by either the BS or SSs in a WiMAX backhaul network where caching is allowed at SSs. Hence, the steady state performance of the network is determined.

Define a state vector \( R(t) = (R_1(t), R_2(t)) \), where \( R_1(t) \) and \( R_2(t) \) are the number of requests served directly by the BS and by other SS nodes respectively at time \( t \). Without loss of generality, we assume that the system can handle \( R_{\text{max}} \) requests which implies that \( R_1(t) + R_2(t) = R_{\text{max}} \) at any time \( t \). Hence, the number of valid states is given by \( R_{\text{max}} + 1 \).

The distance between the embedded points of the considered Markov chain is one frame time. Let the vector \( F(t+1) = (F_1(t+1), F_2(t+1)) \), denote the number of requests that are already finished downloading at the end of the \( t \)th time slot, i.e., at the beginning of slot \( t + 1 \). Similarly, Let \( A(t+1) = (A_1(t+1), A_2(t+1)) \) denote the number of requests that arrived at time slot \( t + 1 \). Thus

\[ R(t+1) = R(t) - F(t+1) + A(t+1). \] (4.22)

Let \( x = (x_1, x_2) \), and \( y = (y_1, y_2) \) be the sample of \( R(t) \) and \( R(t+1) \) respectively.
Similarly let \( f = (f_1, f_2) \), and \( a = (a_1, a_2) \) be the sample of \( F(t + 1) \) and \( A(t + 1) \) respectively.

Therefore the state transition probability \( p_{xy} = p(R(t + 1) = y | R(t) = x) \) is given by

\[
p_{xy} = \sum_{f, a} p(R(t + 1) = y | a, f, x) \times p(A = a | f, x) \times p(F = f | x)
\]

(4.23)

The conditional probability \( p(R(t + 1) = y | f, a, x) \) is given by

\[
p(R(t + 1) = y | f, a, x) = \prod_{i=1}^{2} p(y_i = x_i - f_i + a_i)
\]

(4.24)

where,

\[
p(y_i = x_i - f_i + a_i) = \begin{cases} 
1, & y_i = x_i - f_i + a_i, \\
0, & y_i \neq x_i - f_i + a_i.
\end{cases}
\]

(4.25)

where the above equation represents the legitimate transitions.

Let \( p_a \) denote the probability that a new arrival is served by the BS. As mentioned before the end user node chooses to be served by the BS or by SS based on the estimated downloading time. Hence, refer to Eq. (4.16) \( p_a \) is given by

\[
p_a = p_{BS}(x - f)
\]

(4.26)

which depends on the current distribution at the beginning of time slot \( t + 1 \) once \( f = \{f_1, f_2\} \) requests finished downloading. Hence, the probability \( p(A = a | f, x) \) is given by the following binomial distribution,
\[ p(A = a|f, x) = \left( \frac{a_1 + a_2}{a_1} \right) p_a^{a_1} (1 - p_a)^{a_2} \] (4.27)

which is the probability that \( a_1 \) requests choose to be directly served by BS, and \( a_2 \) requests choose to be served by all SSs.

Let \( p_1(x) \approx \frac{1}{T_{BS}(x)} \) and \( p_2(x) \approx \frac{1}{T_{SS}(x)} \) denote the probability that the number of packets left to be downloaded for a request served by the BS or a SS respectively at the beginning of time slot \( t \) is equal to the available BW. Where, \( T_{BS}(x) \) and \( T_{SS}(x) \) denote the required number of time frames to transmit a file if it is served by the BS or by a SS respectively as defined in Section 4.2. Let \( p(f_k|x) \) denote the probability that \( f_k \) requests that are served by either BS \((k = 1)\) or SSs \((k = 2)\) finishes downloading at time slot \( t + 1 \). The above probabilities are modelled as binomial distribution i.e.,

\[ p(f_1|x) = \binom{x_1}{f_1} (p_1(x))^{f_1} (1 - p_1(x))^{(x_1 - f_1)}, \] (4.28)

and,

\[ p(f_2|x) = \binom{x_2}{f_2} (p_2(x))^{f_2} (1 - p_2(x))^{(x_2 - f_2)}, \] (4.29)

The probability \( p(F = f|x) \) is therefore obtained by multiplying the above probabilities

\[ p(F = f|x) = \prod_{k=1}^{2} p(f_k|x) \] (4.30)

By noting that the above Markov model is irreducible, aperiodic, and positive
recurrant [85], there will be a unique steady-state probability $\pi(x)$ for a state $x$ and the steady state average available BW if the request is served by BS and by a SS is given by

$$W_{BS} = \sum_{x} W_{BS}(x)\pi(x)$$

$$W_{SS} = \sum_{x} W_{SS}(x)\pi(x)$$

The average effective steady state throughput (in bits per frame time) is given by

$$\overline{TH}_{BS} = \frac{F}{D_{BS}}$$

$$\overline{TH}_{SS} = \frac{F}{D_{SS}}$$

where $F$ is the number of information bits per file. $D_{BS} = \frac{N_{BS}}{W_{BS}}$ and $D_{SS} = \frac{N_{SS}}{W_{SS}}$ denote the steady state average file transfer delay in time frames $T_f$, if the request is served by BS or a SS respectively.

4.2.3 Analytical and Simulation Results

In our analysis we consider a 5 MHz WiMAX backhaul mesh network with one BS and nine SSs. We assume each end user node has enough power to reach $ps_1 = 0.8$. The probability that the receiver fails to recover the file is set to $\delta = 10^{-5}$.
and hence $p_f = 1 - 10^{-5}$. The values of $W_{BS}$ and $W_{SS}$ are allowed to be fractions which reflect the effect of time scheduling.

The above analysis is applied to end users downloading web pages. Let $F_{main}$ and $F_{emb}$ denote the number of bits per main object and embedded object respectively. Let $N_{emb}$ denote the number of embedded objects per page as shown in Table 4.1 [56]. Hence, the total page size $F$ in bits is given by

$$F = F_{main} + N_{emb}F_{emb} \tag{4.31}$$

We average for the purpose of analysis, i.e.

$$F = \overline{F}_{main} + \overline{N}_{emb}\overline{F}_{emb} \tag{4.32}$$

We assume that each packet occupies one sub-channel for the whole frame duration (one row in Fig. 1.5). Since the request served directly by the BS it is assigned lower data rate than that served through a SS, thus we assume that $K_{BS} = 2K_{SS} = \frac{F}{2N_cN_{sym}}$ data packets, where $N_c$ is the number of sub-carriers per sub-channel, and $N_{sym}$ is the number of symbols for the whole frame duration (Table 1.5).

Throughout the simulation different network topologies that are generated randomly were considered. For each topology the simulation runs for a large number of time frames (100000). The results are averaged over all topologies and time frames.

Figs. 4.8 and 4.9 show the average steady state BW per request with different
<table>
<thead>
<tr>
<th>Component</th>
<th>Distribution</th>
<th>Parameters</th>
<th>PDF</th>
</tr>
</thead>
<tbody>
<tr>
<td>Main object</td>
<td>Truncated</td>
<td>Min=100 bytes</td>
<td>$g(x) = \frac{1}{\sqrt{2\pi}\sigma} \exp \left( \frac{-(\ln x - \mu)^2}{2\sigma^2} \right), \quad x \geq 0$</td>
</tr>
<tr>
<td></td>
<td>lognormal</td>
<td>Max=2 Mbytes</td>
<td>$\sigma = 1.37, \mu = 8.35$</td>
</tr>
<tr>
<td></td>
<td></td>
<td>mean=10710 bytes</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td>Std. dev=25032 bytes</td>
<td></td>
</tr>
<tr>
<td>Embedded object</td>
<td>Truncated</td>
<td>Min=50 bytes</td>
<td>$g(x) = \frac{1}{\sqrt{2\pi}\sigma} \exp \left( \frac{-(\ln x - \mu)^2}{2\sigma^2} \right), \quad x \geq 0$</td>
</tr>
<tr>
<td></td>
<td>lognormal</td>
<td>Max=2 Mbytes</td>
<td>$\sigma = 2.36, \mu = 6.17$</td>
</tr>
<tr>
<td></td>
<td></td>
<td>mean=7758 bytes</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td>Std. dev=126168 bytes</td>
<td></td>
</tr>
<tr>
<td>Number of Embedded</td>
<td>Truncated</td>
<td>mean=5.64</td>
<td>$g(x) = \frac{k^k x^{k-1}}{\Gamma(k)}$, $k \leq x &lt; m$</td>
</tr>
<tr>
<td>objects</td>
<td>Pareto</td>
<td>Max=53</td>
<td>$g(x) = \left( \frac{k}{m} \right)^\alpha$, $x = m$</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>$\alpha = 1.1, k = 2, m = 55$</td>
</tr>
</tbody>
</table>

Table 4.1: HTTP traffic model parameters
$p_{\text{miss}}$ and $R_{\text{max}}$ respectively. The results are obtained for both BS and SS. It is clear that the size of cache per SS which defines $p_{\text{miss}}$ and the total number of requests have a great impact on the average available BW per request. Studying the effect of the cache size on the available BW per request we notice that the average BW for a request served directly by BS is smaller than that of a request served by a SS. This reflects the fact that BS receives missed requests from all SSs in the network, as for the SSs the missed requests and the number of users are distributed among them. One can also recognize that as $p_{\text{miss}}$ gets larger the average BW per request served by a SS approaches that of a request served by BS, this simply happens because for smaller cache sizes requested files are not found at SSs and hence most of the requests reach the BS. Therefore the effective BW per request is constrained with that of the BS (see Eq. 4.21).

Fig. 4.10 shows the average number of packets for a request served by BS or SS (see Eqs. 4.13 and 4.15). One may notice that while $p_{\text{miss}}$ has no effect on $N_{BS}$, the number of packets $N_{SS}$ grow as $p_{\text{miss}}$ gets larger. This can be clarified by noting that $N_{SS}$ is a function of the number of links and as $p_{\text{miss}}$ gets larger the number of links each packet has to travel gets higher. As for the average number of packet for a request served by BS it is only a function in $p_{S1}$.

In Fig. 4.11 we can see the effect of the $p_{\text{miss}}$ and on the average throughput per request if served by BS and SS. The impact of caching is clearly seen through these results. Fig. 4.12 shows the change of the average throughput per request with the total number of requests if served by BS or SS. It is obvious that the through-
Figure 4.8: Average BW per request. $R_{\text{max}} = 90$, $L_{\text{max}} = 4$, $p_{sl} = 0.8$, $\delta = 10^{-5}$, and $Z_{SS} = 9$

put gets higher for higher number of requests, however the average throughput per user is reduced.

Figs. 4.13 and 4.14 show the steady state distribution of the requests with different $p_{\text{miss}}$ and $R_{\text{max}}$ respectively. It should be noted that when $p_{\text{miss}} = 0$ some of the requests will still be served by the BS, even though this requires longer downloading time because of the probabilistic decision (Eq. 4.16), and hence will stay longer in the system.
Figure 4.9: Average BW per request. $p_{miss} = 0.9$, $p_{s1} = 0.8$, $L_{max} = 4$, $\delta = 10^{-5}$, and $Z_{SS} = 9$

Figure 4.10: Average number of packets per request. $L_{max} = 4$, $\delta = 10^{-5}$, $p_{s1} = 0.8$, and $Z_{SS} = 9$
Figure 4.11: Average throughput per request. $R_{max} = 90$, $L_{max} = 4$, $\delta = 10^{-5}$, and $Z_{SS} = 9$

Figure 4.12: Average throughput per request for BS. $p_{miss} = 0.9$, $L_{max} = 4$, $\delta = 10^{-5}$, $p_{s1} = 0.8$, and $Z_{SS} = 9$
Figure 4.13: Steady state distribution. $R_{\text{max}} = 90, L_{\text{max}} = 4, \delta = 10^{-5}, p_{s1} = 0.8,$ and $Z_{SS} = 9$

4.3 Summary

In this chapter we proposed a cross-layered coding technique in which upper layer Fountain codes is applied on top of physical layer turbo code. The results showed that the performance of the proposed technique is comparable to that of the H-ARQ technique. We also considered a WiMAX backhaul mesh network in which caching is allowed at SSs and UDP is used as its transport protocol. The network is modelled and analyzed using Markov process. An analytical expression for the steady state system transmission delays and throughput are derived from the steady state distribution of the above Markov process. The developed model is validated through simulations.
Figure 4.14: Steady state distribution. \( p_{\text{miss}} = 0.9, L_{\text{max}} = 4, \delta = 10^{-5}, p_{\delta} = 0.8, \)

and \( Z_{SS} = 9 \)
Chapter 5

Conclusion

In this thesis, the developed RoT model is used to determine a bound for the maximum theoretical throughput that can be achieved over the reverse data channel in cdma2000 1xEV star network. It is shown that the maximum instantaneous throughput can be achieved if only a subset of active users are allowed to transmit on each time slot. We proposed two new rate assignment techniques. Our simulation results show that higher throughput and lower service times are achieved using the new proposed techniques. We also proposed two new scheduling techniques; our simulation results show that the proposed algorithm enhances the sector throughput compared to the widely used proportional fair scheduling algorithm.

We also modelled four scheduling techniques for the reverse data channel of the 1xEV star network. The distribution of the MSs among the possible data rates is modelled as a Markov process. The effects of scheduling and rate assignment techniques are included in the Markov model. The validity of the obtained models was confirmed with simulations. The results show that highest-rate-first scheduling technique outperforms all the other scheduling techniques in the sense of achieved throughput. The simulation results are fairly close to
their analysis counterparts, which validates the assumptions of the analysis.

We extended the above model to develop analytical expressions for the steady state link throughput and file transfer delay over the reverse data channel in cdma2000 1XEV where the H-ARQ protocol is applied. These expressions are based on modelling the cross layer design involving the physical layer H-ARQ, network layer rate assignment, and MAC layer time slot scheduling technique used over the reverse data channel. The results obtained show how the choice of the targeted signal to noise and interference ratio affect the overall system performance.

Finally, we proposed a cross layered coding technique in which upper layer Fountain code is applied on top of the physical layer turbo encoder. The results showed that the performance of the proposed technique is comparable to that of the H-ARQ technique. We also considered a WiMAX backhaul mesh network in which caching is allowed at SSs. The network is modelled and analyzed using Markov process. An analytical expression for the steady state system transmission delays and throughput are derived from the steady state distribution of the Markov process. The developed model is validated through simulations. The results shows that, both the cash size at each SS and the total number or requests being handled have a great effect on the network performance.
5.1 Contributions

Throughout this thesis we considered resource management for 3G-4G technology in star and mesh networks. In particular we have

- obtained maximum theoretical throughput that can be achieved over the reverse packet data channel in cdma2000 1xEV star network,
- shown that the maximum instantaneous throughput can be achieved if only a subset of the active users is allowed to transmit on each time slot,
- proposed two new rate assignment techniques,
- proposed two new scheduling techniques,
- derived an analytical models for four scheduling techniques,
- modelled the transmission process over the reverse data channel as Markov process,
- analyzed the cdma2000 1xEV turbo encoder and obtained values for the probability of packet acceptance after first, second and third sub-packet transmissions,
- extended the transmission process model to include the effects of the cross layer design between H-ARQ, rate assignment, and scheduling,
- proposed new cross layered coding technique which allows the use of UDP with a prespecified performance,
• introduced the idea of caching at service station in WiMAX backhaul mesh networks,

• modelled the performance WiMAX backhaul mesh networks as a Markov process,

• provided expressions for the steady state performance for all the models developed throughout the thesis,

• and validated all the developed models through simulation.

It should be noted that while, in this thesis, we considered specific standards, the techniques used throughout our work are generic and can be applied to other types of star and mesh networks.

5.2 Future Work

In a typical cellular system, the maximum allowable transmit power of the mobile stations limit its maximum permissible data rates. It would be interesting to consider how this limitation would affect the overall system throughput and service time.

The rise over thermal model in this thesis considers only the intra-cell interference. The effects of the inter-cell interference on the performance of data transmissions should be further investigated.

The probability of cache missing has a great impact on the performance of
WiMAX backhaul mesh network. This probability mainly depends on the caching policy used at the service station. Studying the effect of different caching policies on the WiMAX backhaul mesh networks should be explored.

During the course of this thesis the impact of handovers are not been considered. For future research it is important to study and model the impact of handovers on all the proposed techniques.
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127


Appendix A

Bit-Error Bounds for cdma2000 1x Turbo Encoder

In what follows, based on the properties of the 1xEV turbo encoder [48], lower bounds on the probabilities of data sub-packets acceptance are derived.

A.1 The cdma2000 1xEV Turbo Encoder

As described in section 1.3.1, each encoder packet is fed into a turbo encoder with code rate 1/5. The structure of cdma2000 1xEV turbo encoder is shown in Fig. A.1. It consists of two parallel concatenated constituent encoders, each of the encoders has a transfer function which is given by

\[
\mathbf{G}(D) = [1 \quad \frac{n_1(D)}{d(D)} \quad \frac{n_2(D)}{d(D)}],
\]

where

\[
d(D) = 1 + D^2 + D^3,
\]

\[
n_1(d) = 1 + D + D^3,
\]

\[
n_2(D) = 1 + D + D^2 + D^3.
\]

Each encoded packet is divided into three sub-packets. The first sub-packet has an encoding rate of 1/2. As the second redundancy sub-packet is transmitted, the combined received data packets will have a higher encoding rate of 1/3.
Figure A.1: cdma2000 1X turbo encoder
Table A.1: Output punctured code for cdma2000 1X encoder

As the third sub-packet is transmitted, it reaches the original encoding rate of 1/5.

As shown in Table A.1 the total codeword with encoding rate 1/5 has a codeword structure \{x, y_0, y_1, y'_0, y'_1\}. The first sub-packet with code rate \( R_c = 1/2 \) is considered to be a punctured output of the encoder shown in Fig. A.1 with codeword structure \{x, y_0 \} \{x, y'_0 \}. The first and second sub-packets combined are treated as a punctured output of the encoder in Fig. A.1 with codewords structure \{x, y_0, y'_0\} [48].
A.2 BER Upper bound

In what follows, we briefly review the procedure for determining upper bounds for the BER of parallel concatenated constituent codes (PCCCs) as outlined in [89], [94].

For a PCCCs with large interleaver of size $L$ and $(n, 1, v)$ systematic feedback constituent encoder, an approximate BER upper bound is given by [89], [94]

$$ P_b(E) \approx \sum_{w_{\text{min}} \leq w \leq L} \frac{w}{L} \left| W^w A_w^{PC}(Z) \right|_{W^w_{\text{min}} = e^{-R_c \epsilon_k \sqrt{N_0}}}, $$ \hspace{1cm} (A.2)

where $R_c$ is the code rate, $\frac{\epsilon_k}{N_0}$ is the SNR per information bit, $w$ is the codeword weight, $w_{\text{min}}$ is the minimum codeword weight, and $A_w^{PC}(Z)$ is the codeword conditional enumerating function for parallel concatenated turbo encoder which is given by

$$ A_w^{PC}(Z) = \frac{A_w^{C_1}(Z) A_w^{C_2}(Z)}{\binom{L}{w}}, $$ \hspace{1cm} (A.3)

where $A_w^{C_1}(Z)$ and $A_w^{C_2}(Z)$ are the codeword conditional enumerating function for the first and second constituent encoders respectively.

Let $h$ denote the number of error events. For large $L$, we can approximate the number of codewords containing $h$ error events by $\binom{L}{h}$. Therefore $A_w(Z)$ can be defined as

$$ A_w(Z) \approx \sum_{1 \leq h \leq h_{\text{max}}} \binom{L}{h} A_w^{(h)}(Z), $$ \hspace{1cm} (A.4)
where \( A_w^{(h)}(Z) \) is the h-error enumerator for input weight \( w \), and \( h_{\text{max}} \) is the maximum number of error events. Substituting into Eq. (A.3), we get

\[
A_w^{PC}(Z) \approx \sum_{1 \leq h_1 \leq h_{\text{max}}} \sum_{1 \leq h_2 \leq h_{\text{max}}} \frac{L}{h_1} \binom{L}{h_2} A_w^{(h_1)}(Z) A_w^{(h_2)}(Z).
\]  

(A.5)

For \( L \gg h \), the approximation \( \binom{L}{h} \approx \frac{L^h}{h^h} \) can be used. Hence Eq. (A.5) can be rewritten as

\[
A_w^{PC}(Z) \approx \sum_{1 \leq h_1 \leq h_{\text{max}}} \sum_{1 \leq h_2 \leq h_{\text{max}}} \frac{w!}{h_1! h_2!} L^{(h_1+h_2-w)} A_w^{(h_1)}(Z) A_w^{(h_2)}(Z).
\]  

(A.6)

A further approximation can be achieved by considering only the most significant terms in the double summation. Let \( h_1 = h_2 \approx h_{\text{max}} \). therefore

\[
A_w^{PC}(Z) \approx \frac{w!}{h_{\text{max}}!^2} L^{2h_{\text{max}}-w} [A_w^{(h_{\text{max}})}(Z)]^2
\]  

(A.7)

By substituting into Eq. (A.2) we get

\[
P_b(E) \approx \sum_{w_{\text{min}} \leq w \leq L} \frac{w!}{h_{\text{max}}!^2} L^{(2h_{\text{max}}-w-1)} W^w [A_w^{(h_{\text{max}})}(Z)]^2 \big|_{W=Z=e^{-r c E_b / N_0}}.
\]  

(A.8)

For \((n, 1, v)\) systematic feedback constituent encoders, \( w_{\text{min}} = 2 \), and \( h_{\text{max}} = \left[ \frac{w}{2} \right] \). Therefore the term \( L^{2h_{\text{max}}-w-1} = L^{-2} \) for odd \( w \), and \( L^{2h_{\text{max}}-w-1} = L^{-1} \) for even \( w \). Thus for large \( L \) the term \( L^{-2} \) can be neglected, and only the terms of the type \( A_2^{(w)}(Z) \) are considered. Further, for any \((n, 1, v)\) systematic feedback constituent encoder \( A_2^{(w)}(Z) = [A_2^{(1)}(Z)]^w \). By substituting into (A.8) we get
\[ P_b(E) \approx \sum_{1 \leq w \leq \left\lfloor \frac{L}{2} \right\rfloor} 2^w L^{-1} W^{2w} [A_2^{(1)}(Z)]^{2w} \mid_{W=Z=e^{-R_c E_b/N_0}} \]  

(A.9)

where, \( A_2^{(1)} \) is the parity weight enumerator for single error event with input weight 2.

From the above cdma2000 1xEV encoder description, it is clear that a codeword with code rate \( R_c = 1/3 \) and 1/5 can be generated using constituent encoder with a transfer functions \( G(D) = [1 n_1(D)/d(D) n_2(D)/d(D)] \), and \( G(D) = [1 n_1(D)/d(D)] \) respectively. Also one can recognize that for code rates \( R_c = 1/3 \) and 1/5 the parity outputs from both constituent encoders are the same, and hence Eq. (A.9) can be applied. As for code rate \( R_c = 1/2 \) half of the codewords are generated only by the first convolutional encoder \( \{x, y_0\} \), and the other half is generated by puncturing the output of the \( R_c = 1/3 \) turbo encoder \( \{x, y'_0\} \).

Applying Eq. (A.5), we get

\[ A_{\text{FC}}^{(1)}(Z) \approx \sum_{1 \leq h_1 \leq h_{\text{max}}} \sum_{1 \leq h_2 \leq h_{\text{max}}} \binom{L}{h_1} \binom{L}{h_2} A_{\text{w}}^{(h_1)}(Z) A_{\text{w}}^{(h_2)}(Z) \{x, y_0\} \{x, y'_0\}. \]  

(A.10)

By inspecting the turbo encoder (see Fig. A.1), we can see that for the codeword \( \{x, y_0\} \), there is no output from the second convolutional encoder. As for the codeword \( \{x, y'_0\} \), the output from the turbo encoder is the same as the output of first convolutional encoder. Note that the first part of Eq. (A.10) already takes
care of the effect of the interleaver. Thus

\[ (A_w^{(h_1)}(Z)A_w^{(h_2)}(Z))_{\{x,y_0\}} = (A_w^{(h_1)}(Z)A_w^{(h_2)}(Z))_{\{x,y_0\}} = (A_w^{h_1}(Z)) \]  

(A.11)

and Eq. (A.10) can be rewritten as

\[ A_w^{PC}(Z) \approx \sum_{1 \leq h_1 \leq h_{max}} \sum_{1 \leq h_2 \leq h_{max}} \frac{L}{h_1} \frac{L}{h_2} \frac{L}{w} [A_w^{(h_1)}(Z)] \]  

(A.12)

Following the steps as in Eqs. A.5-A.7, one can show that

\[ A_w^{PC}(Z) \approx \frac{w!}{(h_{max})^2} L^{(2h_{max}-w)} [A_w^{(h_{max})}] \]  

(A.13)

Thus the BER for code rate \( R_c = 1/2 \) is given by

\[ P_b(E) \approx \sum_{1 \leq w \leq \left\lfloor \frac{1}{2} \right\rfloor} 2^w \left( \frac{2^w}{w} \right) L^{-1} W^{2w} [A_2^{(1)}(Z)]^w |_{W = Z = e^{-R_c E_b/N_0}} \]  

(A.14)

where, \( A_2^{(1)} \) is the parity weight enumerator for single error event with input weight 2 for code rate \( R_c = 1/3 \).

In what follows, we determine the weight enumerator functions of the cdma2000 1xEV encoder with different code rates. Table A.2 shows the states of cdma2000 1xEV encoder and their corresponding input/output bits for constituent encoder. Figs. A.2 and A.3 show the state diagram for cdma2000 1xEV constituent encoder with \( R_c = 1/5 \), and \( R_c = 1/3 \) respectively.

Form Table A.2 and Figs. A.2 and A.3 one can notice that the shortest path for single error event with input weight 2 is given by: \( S_0 \rightarrow S_1 \rightarrow S_2 \rightarrow S_5 \rightarrow S_3 \rightarrow \)
Figure A.2: State diagram for cdma2000 1xEV encoder with $R_c = 1/5$

$S_7 \rightarrow S_6 \rightarrow S_4 \rightarrow S_0$ with one cycle around the loop ($S_4 \rightarrow S_1 \rightarrow S_2 \rightarrow S_5 \rightarrow S_3 \rightarrow S_7 \rightarrow S_6 \rightarrow S_4$). Therefor,

$$A_2^{(1)}(Z) = \begin{cases} 
Z^{12} + Z^{20} + Z^{28} \ldots \ldots = \frac{Z^{12}}{1-Z^8}, & R_c = 1/5 \\
Z^6 + Z^{10} + Z^{14} \ldots \ldots = \frac{Z^6}{1-Z^4}, & R_c = 1/3 
\end{cases}$$

A.3 Probability of Packet Acceptance

In this section we derive an approximate expression for the probability of packet acceptance that is a function of coding rate $R_c$ and $E_b/N_0$.

Let $p_{S_1,k}$ and $p_{S_2,k}$ be the probabilities that the packet is accepted after the first or second transmissions respectively. Let $b_k$ denote the number of bits per packet associated with data rate $R_k$. Hence,
<table>
<thead>
<tr>
<th>State</th>
<th>$R_e = 1/5$</th>
<th>$R_e = 1/3$</th>
<th>Next State</th>
</tr>
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<tr>
<td>$S_0$</td>
<td>000</td>
<td>0/00</td>
<td>000 ($S_0$)</td>
</tr>
<tr>
<td>$S_0$</td>
<td>000</td>
<td>1/11</td>
<td>100 ($S_1$)</td>
</tr>
<tr>
<td>$S_1$</td>
<td>100</td>
<td>0/01</td>
<td>010 ($S_2$)</td>
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<td>100</td>
<td>1/100</td>
<td>110 ($S_3$)</td>
</tr>
<tr>
<td>$S_2$</td>
<td>010</td>
<td>0/010</td>
<td>101 ($S_5$)</td>
</tr>
<tr>
<td>$S_2$</td>
<td>010</td>
<td>1/101</td>
<td>001 ($S_4$)</td>
</tr>
<tr>
<td>$S_3$</td>
<td>110</td>
<td>0/001</td>
<td>111 ($S_7$)</td>
</tr>
<tr>
<td>$S_3$</td>
<td>110</td>
<td>1/110</td>
<td>011 ($S_6$)</td>
</tr>
<tr>
<td>$S_4$</td>
<td>001</td>
<td>0/000</td>
<td>100 ($S_1$)</td>
</tr>
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<td>001</td>
<td>1/111</td>
<td>000 ($S_0$)</td>
</tr>
<tr>
<td>$S_5$</td>
<td>101</td>
<td>0/011</td>
<td>110 ($S_3$)</td>
</tr>
<tr>
<td>$S_5$</td>
<td>101</td>
<td>1/100</td>
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</tr>
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<td>$S_6$</td>
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<td>$S_6$</td>
<td>101</td>
<td>1/101</td>
<td>101 ($S_3$)</td>
</tr>
<tr>
<td>$S_7$</td>
<td>111</td>
<td>0/001</td>
<td>011 ($S_6$)</td>
</tr>
<tr>
<td>$S_7$</td>
<td>111</td>
<td>1/110</td>
<td>111 ($S_7$)</td>
</tr>
</tbody>
</table>

Table A.2: States for the cdma2000 1X encoder
\[ p_{s_{i,k}} = (1 - (P_b(E))_{i,k})^{b_k} \]
\[ = 1 - \binom{b_k}{1}(P_b(E))_{i,k} + \binom{b_k}{2}(P_b(E))_{i,k}^2 - \cdots + (P_b(E))_{i,k}^{b_k} \]  
(A.15)

where \((P_b(E))_{i,k}\) is the probability of BER (Eqs. A.9 and A.14) associated with the \(i^{th}\) transmission \((i = 1, 2)\) and \(L \approx b_k\). By noting that \(P_b(E)_{i,k} \leq 1\), we consider only the first few dominant terms in Eq. A.15. Thus

\[ p_{s_{i,k}} \approx 1 - \binom{b_k}{1}(P_b(E))_{i,k} + \binom{b_k}{2}(P_b(E))_{i,k}^2 \]  
(A.16)

By applying the binomial coefficient approximation used in Eq. (A.6), we get

\[ p_{s_{i,k}} \approx 1 - b_k(P_b(E))_{i,k} + \frac{b_k^2}{2!}(P_b(E))_{i,k}^2 \]  
(A.17)

Eqs (A.9) and (A.14), \((P_b(E))_{i,k}\) can be rewritten in the following form:
\[ (P_b(E))_{i,k} \approx \frac{c_i}{L_k}, \quad i = 1, 2, 3. \] (A.18)

where, \( L_k \approx b_k \) is the interleaver size corresponds to data rate \( R_k \) and

\[
c_1 = \sum_{1 \leq w \leq 10} 2w \binom{2w}{w} W^{2w} [A_2^{(1)}(Z)]^{2w} |_{W = Z = e^{-1/2E_b/N_0}},
\]

\[
c_2 = \sum_{1 \leq w \leq 10} 2w \binom{2w}{w} W^{2w} [A_2^{(1)}(Z)]^{2w} |_{W = Z = e^{-1/3E_b/N_0}},
\]

and

\[
c_3 = \sum_{1 \leq w \leq 10} 2w \binom{2w}{w} W^{2w} [A_2^{(1)}(Z)]^{2w} |_{W = Z = e^{-1/5E_b/N_0}},
\]

By substituting into Eq. A.17, we get

\[ p_{s_{1,k}} \approx 1 - c_i + \frac{1}{2!} (c_i)^2 \overset{def}{=} p_{s_i}, \quad i = 1, 2, 3. \] (A.19)

which is function of only the coding rate \( R_c \) and \( E_b/N_0 \).

Based on the above analysis, Fig. A.4 shows how the lower bounds for the probabilities of sub-packets acceptance vary with \( E_b/N_0 \). It is clear that, even for low \( E_b/N_0 \), the probability of sub-packets acceptance is well above 90\%, and for \( E_b/N_0 \geq 4dB \), the data packet is practically accepted from the first transmission.
Figure A.4: Lower bounds for the probabilities of packet acceptance

A.4 Eb/No

In this section we show the relation between $E_b/N_0$ and the required signal to noise and interference ratio for the pilot channel $\tau$.

Let $P_T$ be the total power received at the BS. Then $\frac{E_b}{N_0}$ is given by

$$\frac{E_b}{N_0} = \frac{P_i}{(P_T - P_i)} \times \frac{W}{R_i} \quad (A.20)$$

where $P_i$ and $R_i$ are the received power at the BS, and the data rate assigned to the $i^{th}$ mobile respectively. $W$ is the available bandwidth.

Let $\overline{P_i}$ be the pilot channel power for the $i^{th}$ mobile. Define the traffic to pilot ratio for the $i^{th}$ mobile as $TPR_i = P_i/\overline{P_i}$. Denote the targeted signal to noise and interference ratio for the pilot channel as $\tau = \overline{P_i}/(P_T - P_i)$, where $(P_T - P_i)$ is the
total interference and the noise power for the $i^{th}$ mobile. Therefore Eq. (A.20), can be rewritten as,

$$\frac{E_b}{N_o} = \frac{P_i}{(P_i^{-1}P_i)} \times \frac{P_i}{P_i} \times \frac{W}{R_i}$$

$$= \frac{TPR_k}{R_i} \times \frac{W}{R_i}$$

$$= \tau \times TPR_k \times \frac{W}{R_i}$$

(A.21)

The traffic to pilot ratio $TPR_k$ corresponding to data rate $R_k$, is given by Eq. 2.4. Hence, the $E_b/N_o$ corresponding to data rate $R_k$ is given by,

$$\left(\frac{E_b}{N_o}\right)_k = \tau \times TPR_k \times \frac{W}{R_k}$$

(A.22)

One can notice that $\left(\frac{E_b}{N_o}\right)_k$ depends on the value of $\tau$ as $TPR_k$, $W$, and $R_k$ have prespecified values (see Section 1.3.1).