Miguel Ramos Pereira

# Time Diversity Solutions to Cope With Lost Packets

A dissertation submitted to Departamento de Engenharia Electrotécnica
of Faculdade de Ciências e Tecnologia of Universidade Nova de Lisboa
in partial fulfilment of the requirements for the degree of Doctor of
Philosophy in Engenharia Electrotécnica e de Computadores.

Adviser: Professor Luís Filipe Lourenço Bernardo
Co-Adviser: Professor Rui Miguel Henriques Dias Morgado Dinis

President of Jury: Doctor Paulo da Costa Luís da Fonseca Pinto
First Reader: Doctor Atílio Manuel da Silva Gameiro
Second Reader: Doctor Paulo Rogério Barreiros d'Almeida Pereira
Third Reader: Doctor Rui Jorge Morais Tomaz Valadas

Universidade Nova de Lisboa
Faculdade de Ciências e Tecnologia
Departamento de Engenharia Electrotécnica

Lisbon
January 2013

Time Diversity Solutions to Cope With Lost Packets

To my parents, my sister

and my girlfriend

# Acknowledgments

My presence at the Faculdade de Ciências e Tecnologia (FCT) has been fulfilled with a wealth of memorable moments and I have been privileged to be surrounded by truly inspirational people. Foremost amongst these people has been my advisor, Prof. Luís Bernardo, who has been a mentor and a friend. I thank him for his tremendous guidance and for encouraging me to pursue this PhD. His advices impacts my whole life. I would also like to thank my co-advisor Prof. Rui Dinis, who has always been a valuable source of perspective and guided me through physical layer issues in wireless communications with his invaluable feedback. The telecommunications group, Prof. Rodolfo Oliveira, Prof. Paulo Pinto and Prof. Paulo Montezuma just to name a few, have always been a source of motivation and extraordinary collaborators - I thank you all. I will continue to endeavour towards the standards of excellence that you all have taught me.

I would like to thank all my friends and members of the lab who supported me even in the hardest times. They have always kept me smiling and made the lab such a wonderful place to work. Thank you to: Michael Figueiredo, Francisco Ganhão, Miguel Luís, Miguel Luzio, Fábio Coelho, and so many other who I am sure I have left out. The life in FCT will always be filled with excellent memories.

Most importantly, I thank my family, especially my parents, my sister and my girlfriend, for their tireless and tenacious support, love and encouragement over the last four years. Without them none of this would have been possible.

I am grateful to Fundação para a Ciência e Tecnologia for awarding me a PhD grant (SFRH/BD/41515/2007), which made this PhD journey a reality. I am also thankful to all the financial support that helped me through this journey (travel funds, conference payments, etc.): CTS multi-annual funding project PEst-OE/EEI/UI0066/2011, IT pluri-annual funding project PEst-OE/EEI/LA0008/2011, U-BOAT project PTDC/EEA-TEL/67066/2006, MPSat project PTDC/EEA-TEL/099074/2008 and OPPORTUNISTIC-CR project PTDC/EEA-TEL/115981/2009.

Miguel Ramos Pereira

January 2013

# Abstract

Modern broadband wireless systems require high throughputs and can also have very high Quality-of-Service (QoS) requirements, namely small error rates and short delays. A high spectral efficiency is needed to meet these requirements. Lost packets, either due to errors or collisions, are usually discarded and need to be retransmitted, leading to performance degradation. An alternative to simple retransmission that can improve both power and spectral efficiency is to combine the signals associated to different transmission attempts.

This thesis analyses two time diversity approaches to cope with lost packets that are relatively similar at physical layer but handle different packet loss causes. The first is a low-complexity Diversity-Combining (DC) Automatic Repeat reQuest (ARQ) scheme employed in a Time Division Multiple Access (TDMA) architecture, adapted for channels dedicated to a single user. The second is a Network-assisted Diversity Multiple Access (NDMA) scheme, which is a multi-packet detection approach able to separate multiple mobile terminals transmitting simultaneously in one slot using temporal diversity. This thesis combines these techniques with Single Carrier with Frequency Division Equalizer (SC-FDE) systems, which are widely recognized as the best candidates for the uplink of future broadband wireless systems. It proposes a new NDMA scheme capable of handling more Mobile Terminals (MTs) than the user separation capacity of the receiver. This thesis also proposes a set of analytical tools that can be used to analyse and optimize the use of these two systems. These tools are then employed to compare both approaches in terms of error rate, throughput and delay performances, and taking the implementation complexity into consideration.

Finally, it is shown that both approaches represent viable solutions for future broadband wireless communications complementing each other.

# Keywords

Diversity Combining Hybrid ARQ, Network Diversity Multiple Access, Medium Access Control Protocols, Analytical Models, Frequency-Domain Processing

# Resumo


Os actuais sistemas sem fios de banda larga apresentam elevados requisitos a nível de débito e de Qualidade de Serviço, nomeadamente baixas taxas de erros e de atraso. De forma a cumprir estes requisitos é necessário obter uma elevada eficiência espectral. Os pacotes perdidos, quer sejam por erros de transmissão quer por colisão, são normamente eliminados originando a sua retransmissão, que invariavelmente leva à degradação do desempenho do sistema. A combinação da recepção de sinais associados a diferentes tentativas de transmissão é uma alternativa à simples retransmissão, que possibilita uma melhoria de resultados tanto a nível de potência como a nível de eficiência espectral.

Esta dissertação analisa duas abordagens distintas de diversidade temporal com o propósito de lidar com perdas de pacotes. Apesar de serem ambas semelhantes ao nível do canal físico, cada uma destas abordagens lida com causas diferentes de perdas de pacotes. A primeira abordagem é um esquema ARQ (pedido de retransmissão automático) com DC (combinação de diversidade) de baixa complexidade implementado numa arquitectura TDMA (acesso múltiplo por divisão no tempo) onde cada canal é reservado para um utilizador. A segunda abordagem é um esquema NDMA (acesso múltiplo com diversidade assistida pela rede) que consiste num sistema de detecção multi-pacote que é capaz de separar múltiplas transmissões simultâneas, usando métodos de diversidade temporal. Ambas as abordagens são aplicadas em sistemas que usam SC-FDE (portadora comum com equalização por divisão na frequência), largamente reconhecido como um dos melhores candidatos para os canais para a estação base em futuros sistemas sem fios de banda larga. Também é proposto um novo esquema NDMA capaz de lidar com um número de terminais móveis superior ao número máximo de transmissões em simultâneo suportado pelo receptor. No decorrer desta dissertação é proposto ainda um conjunto de ferramentas analíticas que são usadas no estudo e optimização dos dois sistemas analisados. Estas ferramentas possibilitam uma comparação entre as duas abordagens em termos de taxas de erro no receptor, débito do sistema e atrasos nas transmissões, tendo em consideração a complexidade de implementação dos receptores.

Por último, é possível concluir que as duas abordagens apresentam características complementares e representam soluções viáveis para os futuros sistemas de comunicação sem fios de banda larga.

# Palavras-Chave

DC Hybrid ARQ (ARQ Híbrido com Combinação de Diversidade), NDMA (Acesso Múltiplo com Diversidade assistida pela Rede), Protocolos de Controlo de Acesso ao Meio, Modelos Analíticos, Processamento no Dominio da Frequência

# Contents

**Contents** **xiii**

**List of Tables** **xvii**

**List of Figures** **xix**

**1 Introduction** **1**

1.1 Motivation . . . . . 2

1.2 Research Goals . . . . . 6

1.3 Thesis Outline . . . . . 6

**2 Diversity Methods and Error Control Schemes** **9**

2.1 Diversity Techniques . . . . . 9

2.1.1 Spatial Diversity . . . . . 10

2.1.2 Frequency Diversity . . . . . 11

2.1.3 Time Diversity . . . . . 12

2.2 Multiple Access Schemes . . . . . 13

2.2.1 Time Division Multiple Access . . . . . 13

2.2.2 Code Division Multiple Access . . . . . 13

2.2.3 Space Division Multiple Access . . . . . 16

2.2.4 Frequency Division Multiple Access . . . . . 16

2.2.4.1 Orthogonal Frequency Division Multiplexing . . . . . 17

2.2.4.2 Single Carrier with Frequency Division Equalizer . . . . . 18

2.3 Hybrid ARQ architectures . . . . . 22

2.3.1 The Early Ages . . . . . 22

2.3.1.1 Basic retransmission strategies . . . . . 23

2.4 Hybrid-ARQ Schemes . . . . . 26

2.4.1 Code Combining . . . . . 29

2.4.2 Diversity Combining . . . . . . . . . . . . . . . . . . . . . . . . . . . . 33
2.4.3 Performance Studies . . . . . . . . . . . . . . . . . . . . . . . . . . . . 39

**3 An Hybrid-ARQ SC-FDE System** **41**
3.1 Hybrid-ARQ System Design . . . . . . . . . . . . . . . . . . . . . . . . . . . . 42
3.1.1 Receiver Characterization . . . . . . . . . . . . . . . . . . . . . . . . . . 42
3.1.2 Coping With Static Channel Conditions . . . . . . . . . . . . . . . . . . 46
3.1.3 Receiver Performance . . . . . . . . . . . . . . . . . . . . . . . . . . . . 49
3.2 Hybrid-ARQ Performance on a TDMA System . . . . . . . . . . . . . . . . . 52
3.2.1 System Characterization . . . . . . . . . . . . . . . . . . . . . . . . . . . 52
3.2.2 Generating Function of the System Size . . . . . . . . . . . . . . . . . . 53
3.2.3 Mean System Size . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . 57
3.2.4 Delay Analysis . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . 58
3.2.4.1 Poisson Arrival Process . . . . . . . . . . . . . . . . . . . . . . 59
3.2.4.2 Geometric Arrival Stream . . . . . . . . . . . . . . . . . . . . . 59
3.2.5 Goodput Analysis . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . 59
3.3 Performance Results . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . 60
3.4 Conclusions . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . 66

**4 Collision Resolution Techniques** **67**
4.1 MAC-Layer Solutions . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . 68
4.1.1 ALOHA and its Evolutions . . . . . . . . . . . . . . . . . . . . . . . . . 68
4.1.2 Tree Algorithms . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . 69
4.1.3 Carrier Sensing . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . 73
4.1.4 Reservation Schemes . . . . . . . . . . . . . . . . . . . . . . . . . . . . 74
4.2 PHY-Layer Solutions . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . 75
4.2.1 Multiple Packet Reception . . . . . . . . . . . . . . . . . . . . . . . . . 75
4.2.2 Interference Cancellation: Conceptual Schemes . . . . . . . . . . . . . 77
4.3 PHY-MAC Joint Resolution . . . . . . . . . . . . . . . . . . . . . . . . . . . 79
4.3.1 SICTA - Successive Interference Cancellation in a Tree Algorithm . . 84
4.3.2 Network-assisted Diversity Multiple Access Protocols . . . . . . . . . . 87
4.3.2.1 A Training Based Collision Resolution . . . . . . . . . . . . . 89
4.3.2.2 Blind NDMA: a Blind Signal Separation Method . . . . . . . 91
4.3.2.3 Different x-NDMA Evolutions . . . . . . . . . . . . . . . . . 92

**5 A Network Diversity Multiple Access protocol** **95**
5.1 System Overview . . . 97
5.1.1 MAC Protocol . . . 98
5.1.2 Receiver Characterization . . . 101
5.2 Performance Analysis and Optimization . . . 104
5.2.1 Detection process . . . 105
5.2.2 Epoch Analysis . . . 106
5.2.3 Network's Steady State . . . 109
5.2.4 Delay Analysis . . . 111
5.2.5 Optimization of the transmission probability . . . 113
5.3 Performance Results . . . 114
5.4 Conclusions . . . 125

**6 Performance comparison** **127**
6.1 Scenario Characterization . . . 127
6.2 Packet Detection Performance Results . . . 128
6.3 Medium Access Protocol and System Behaviour . . . 135
6.3.1 Diversity Combining Hybrid ARQ . . . 135
6.3.2 Network-assisted Diversity Multiple Access . . . 135
6.4 DC Hybrid ARQ vs NDMA . . . 138
6.5 Conclusions . . . 141

**7 Conclusions and Discussion** **143**
7.1 Contributions and Future Work . . . 145

**Bibliography** **149**

# List of Tables


2.1 Main contributions on CC schemes of type-II Hybrid ARQ. . . . . . . . . . . 34
2.2 Main contributions on DC schemes of type-II Hybrid ARQ. . . . . . . . . . . 39

3.1 Simulations setups. . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . 50

4.1 Different interference cancellation techniques with $K$ number of users and $P$ stages. . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . 79

5.1 Simulations setups from Figure 5.3 to Figure 5.7. . . . . . . . . . . . . . . . . 115
5.2 Simulations setups from Figure 5.8 to Figure 5.13. . . . . . . . . . . . . . . . 116
5.3 Adopted power delay profile. . . . . . . . . . . . . . . . . . . . . . . . . . . 117

6.1 Simulations setups from Figure 6.1 to Figure 6.5. . . . . . . . . . . . . . . . . 129
6.2 Simulations setups from Figure 6.6 to Figure 6.12. . . . . . . . . . . . . . . . 130

# List of Figures


2.1 Spatial diversity examples. . . . . . . . . . . . . . . . . . . . . . . . . . . . . . 10
2.2 Illustration of Multiple Input Multiple Output (MIMO) solutions with spatial multiplexing (LST (a) and SDMA (b)). . . . . . . . . . . . . . . . . . . . . . . 12
2.3 Transmitter side of a typically Code Division Multiple Access (CDMA) system. 15
2.4 Illustration of the near-far problem. . . . . . . . . . . . . . . . . . . . . . . . 15
2.5 SC-FDE with linear equalization. . . . . . . . . . . . . . . . . . . . . . . . . . 19
2.6 Block processing in frequency domain equalization. . . . . . . . . . . . . . . . 19
2.7 Complexity comparison of time and frequency domain linear equalizers [Falconer *et al.*, 2002]. . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . 20
2.8 SC-FDE decision feedback equalizer. . . . . . . . . . . . . . . . . . . . . . . . 21
2.9 Coexistence of SC-FDE and OFDM: uplink and downlink asymmetry. . . . . 21
2.10 Stop-and-wait ARQ scheme. . . . . . . . . . . . . . . . . . . . . . . . . . . . 24
2.11 Go-back-N ARQ scheme. . . . . . . . . . . . . . . . . . . . . . . . . . . . . . 25
2.12 Selective-repeat ARQ scheme. . . . . . . . . . . . . . . . . . . . . . . . . . . 26
2.13 Sindhu's scheme for Hybrid Automatic Repeat reQuest (Hybrid ARQ) architectures. . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . 29
2.14 Mapping sequence of Code Combining systems. . . . . . . . . . . . . . . . . . 31
2.15 Sindhu's proposal for Hybrid ARQ architectures. . . . . . . . . . . . . . . . . 35
2.16 Block diagram of Benelli's proposal for Hybrid ARQ. . . . . . . . . . . . . . . 37

3.1 Receiver process. . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . 43
3.2 Structure of the iterative receiver for the first packet reception (the shadowed part corresponds to a linear Frequency Domain Equalization (FDE)). . . . . . 44
3.3 Structure of the iterative receiver for the $l$ packet reception (the shadowed part corresponds to a linear FDE). . . . . . . . . . . . . . . . . . . . . . . . . 44
3.4 Example of a possible channel response for two different transmissions employing Shifted Packet (SP) strategy. . . . . . . . . . . . . . . . . . . . . . . . 48

3.5 Concept illustration of the SP strategy. . . . . . . . . . . . . . . . . . . . . . 48
3.6 Packet Error Rate (PER) performance for the linear FDE receiver. . . . . . . 51
3.7 PER performance for an iterative FDE receiver with 4 iterations. . . . . . . . 51
3.8 Goodput as function of $E_b/N_0$ with $R = 5$ and saturated load. . . . . . . . . 61
3.9 Packet delay with $E_b/N_0$ for Poisson traffic, $R = 5$ and $1/\lambda$=20 slots/packet. 61
3.10 Goodput as function of $E_b/N_0$ with $R = 5$. . . . . . . . . . . . . . . . . . . 62
3.11 Packet delay with $E_b/N_0$ for Poisson traffic and $R = 5$. . . . . . . . . . . . . . 62
3.12 Packet delay with $1/\lambda$ for Poisson traffic and $R = 5$. . . . . . . . . . . . . . . 63
3.13 Packet delay with $1/\lambda$ for Geometric traffic and $R = 5$. . . . . . . . . . . . . . 64
3.14 Packet delay for $1/\lambda$ and $E_b/N_0$ with packet combining for Poisson traffic and $R = 5$. . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . 64
3.15 Packet delay for $1/\lambda$ and $E_b/N_0$ without packet combining for Poisson traffic and $R = 5$. . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . 65

4.1 Tree splitting algorithm behaviour. . . . . . . . . . . . . . . . . . . . . . . . . 70
4.2 Example of a non-persisting Carrier Sense Multiple Access (CSMA) protocol in a three node wireless communication without hidden node problem. . . . . 73
4.3 Example of a CSMA/Collision Detection (CSMA/CD) protocol in a three node wireless communication with hidden node problem. . . . . . . . . . . . . 74
4.4 Example of the tree concept with Successive Interference Cancellation in a Tree Algorithm (SICTA) protocol. . . . . . . . . . . . . . . . . . . . . . . . . 85
4.5 Example of the deadlock problem. . . . . . . . . . . . . . . . . . . . . . . . . 86

5.1 Medium Access Control (MAC) protocol example. . . . . . . . . . . . . . . . 100
5.2 Network Diversity Multiple Access receiver . . . . . . . . . . . . . . . . . . . 102
5.3 PER after 4 iterations, for $Q = 1$ (without collision), 2, 4 and for the Hybrid ARQ scheme where four transmissions of the same packet are combined. 118
5.4 Throughput with $p$ and $Q_{max}$ for $J = 8$ MTs, CB, $E_b/N_0$ =7dB and $1/\lambda \in \{5(sat), 90, 250\}$ ms/packet. . . . . . . . . . . . . . . . . . . . . . . . . . 120
5.5 Throughput with $p$ and $Q_{max}$ for $J = 8$ MTs, EB, $E_b/N_0$ =7dB and $1/\lambda \in \{5(sat), 90, 250\}$ ms/packet. . . . . . . . . . . . . . . . . . . . . . . . . . 120
5.6 Throughput with $p$ and $E_b/N_0$ for $J = 8$ MTs, CB or EB, $Q_{max}$ =4 and $1/\lambda \in \{5(sat), 90, 250\}$ ms/packet. . . . . . . . . . . . . . . . . . . . . . . . 121
5.7 Throughput with CB, EB, $Q_{max}$ and $E_b/N_0$ for $J = 8$ MTs and saturation load. . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . 121

5.8 Delay with $p$ and $Q_{max}$ for $J = 8$ MTs, Constant Backoff (CB) or Exponential Backoff (EB), $E_b/N_0 = 7$dB and $1/\lambda = 150$ ms/packet. . . . . . . . . . . . . . 122
5.9 Delay with $p$ and $E_b/N_0$ for $J = 8$ MTs, CB or EB, $Q_{max} = 4$ and $1/\lambda = 150$ ms/packet. . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . 123
5.10 $G^{ndma}_{max}$ with $J$, CB or EB and $Q_{max}$ for $E_b/N_0 = 1$dB and saturation load. . . 124
5.11 $D_{min}$ with $Q_{max}$, CB or EB and $J$, $E_b/N_0 = 7$dB and $1/(\lambda J) = 150$ ms/packet. 124
5.12 $D_{min}$ with $Q_{max}$, CB or EB and $1/\lambda$ for lossless scenario and $J = 8$ MTs. . . 125
5.13 $D_{min}$ with $E_b/N_0$, CB or EB and $1/\lambda$ for $Q_{max} = 4$ and $J = 8$ MTs. . . . . . 126

6.1 BER performance of the DC-HARQ receiver with 4 iterations when in the presence of Uncorrelated Channel (UC) and Equal Channel (EC) channel conditions. . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . 132
6.2 BER performance of the DC-HARQ receiver with 4 iterations when in the presence of UC channel condition and employing SP strategy. . . . . . . . . . 132
6.3 PER performance of the DC-HARQ receiver under the same conditions of Figure 6.1. . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . 133
6.4 PER performance of the DC-HARQ receiver under the same conditions of Figure 6.2. . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . 133
6.5 BER performance of the NDMA receiver with 4 iterations. . . . . . . . . . . . 134
6.6 PER performance of the NDMA receiver with 4 iterations. . . . . . . . . . . . 134
6.7 Diversity Combining Hybrid ARQ behaviour. . . . . . . . . . . . . . . . . . . 136
6.8 MAC behaviour for a Network-assisted Diversity Multiple Access protocol. . . 137
6.9 Saturation throughput over $E_b/N_0$. . . . . . . . . . . . . . . . . . . . . . . . . 138
6.10 Saturation throughput over number of mobile terminals. . . . . . . . . . . . . 139
6.11 Delay over $E_b/N_0$. . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . 140
6.12 Delay over number of nodes. . . . . . . . . . . . . . . . . . . . . . . . . . . . 140

# Acronyms

| | | |
|---|---|---|
| **1G** | First-Generation | 17 |
| **2G** | Second-Generation | 2 |
| **3G** | Third-Generation | 2 |
| **3GPP** | 3rd Generation Partnership Project | 3 |
| **4G** | Fourth-Generation | 3 |
| **ACK** | Positive Acknowledgement | 99 |
| **ADC** | Analog-to-Digital Converter | 78 |
| **ARQ** | Automatic Repeat reQuest | 90 |
| **ATM** | Asynchronous Transfer Mode | 38 |
| **AWGN** | Additive White Gaussian Noise | 86 |
| **BEB** | Binary EB | 86 |
| **BER** | Bit Error Rate | 131 |
| **BNDMA** | Blind NDMA | 92 |
| **BS** | Base Station | 143 |
| **CB** | Constant Backoff | 99 |
| **CC** | Code-Combining | 41 |
| **CDMA** | Code Division Multiple Access | 95 |
| **CDPD** | Cellular Digital Packet Data | 28 |
| **CIR** | Channel Impulse Response | 11 |
| **CPC** | Complementary Punctured Convolutional | 33 |
| **CSI** | Channel State Information | 83 |
| **CSMA/CA** | CSMA/Collision Avoidance | 74 |

| | | |
|---|---|---|
| **CSMA/CD** | CSMA/Collision Detection | 74 |
| **CSMA** | Carrier Sense Multiple Access | 73 |
| **CTS** | Clear to Send | 74 |
| **DCF** | Distributed Coordination Function | 74 |
| **DC** | Diversity-Combining | 143 |
| **DFE** | Decision Feedback Equalization | 77 |
| **DFT** | Discrete Fourier Transform | 103 |
| **DMT** | Discrete Multi-Tone | 17 |
| **DS-CDMA** | Direct Sequence - Code Division Multiple Access | 11 |
| **DS-CSMA** | Direct Sequence - Carrier Sense Multiple Access | 40 |
| **DS-SSMA** | Direct Sequence - Spread-Spectrum Multiple Access | 33 |
| **DSSS** | Direct Sequence Spread Spectrum | 14 |
| **DTMC** | Discrete Time Markov Chain | 109 |
| **EB** | Exponential Backoff | 99 |
| **EC** | Equal Channel | 131 |
| **FCFS** | First Come First Served | 72 |
| **FDE** | Frequency Domain Equalization | 101 |
| **FDMA** | Frequency Division Multiple Access | 69 |
| **FDM** | Frequency Division Multiplex | 16 |
| **FEC** | Forward Error Correction | 22 |
| **FF-NDMA** | Feedback Free - NDMA | 92 |
| **FFT** | Fast Fourier Transform | 128 |
| **FH-CDMA** | Frequency Hoping - Code Division Multiple Access | 14 |
| **FH-NDMA** | Frequency Hopping - NDMA | 92 |
| **FHSS** | Frequency Hopping Spread Spectrum | 14 |
| **GSM** | Global System for Mobile communication | 11 |
| **Hybrid ARQ** | Hybrid Automatic Repeat reQuest | 143 |
| **HAP** | High Altitude Platform | 145 |
| **HSDPA** | High-Speed Downlink Packet Access | 40 |

| | | |
|---|---|---|
| **HSPA** | High-Speed Packet Access | 3 |
| **HSUPA** | High-Speed Uplink Packet Access | 40 |
| **HoL** | Head-of-Line | 53 |
| **IB-DFE** | Iteractive Block-Decision Feedback Equalization | 101 |
| **ICA-CoopNDMA** | ICA - Cooperative NDMA | 92 |
| **ICA-NDMA** | Independent Component Analysis - NDMA | 91 |
| **IDFT** | Inverse Discrete Fourier Transform | 103 |
| **IFFT** | Inverse Fast Fourier Transform | 21 |
| **IMT-Advanced** | International Mobile Telecommunications - Advanced | 3 |
| **IP** | Internet Protocol | 40 |
| **IR-ARQ** | Incremental Redundancy Automatic Repeat reQuest | 93 |
| **ISI** | Inter-Symbol Interference | 77 |
| **ITU** | International Telecommunication Union | 3 |
| **LDPC** | Low-Density Parity-Check | 33 |
| **LLR** | LogLikelihood Ratios | 44 |
| **LST** | Layered Space-Time | 11 |
| **LTE** | Long Term Evolution | 40 |
| **MAC** | Medium Access Control | 144 |
| **MDS** | Maximum Distance Separable | 32 |
| **MIMO** | Multiple Input Multiple Output | 146 |
| **MISO** | Multiple Input Single Output | 11 |
| **MPR** | MultiPacket Reception | 144 |
| **MRC** | Maximum Ratio Combining | 46 |
| **MT** | Mobile Terminal | 143 |
| **NACK** | Negative Acknowledgement | 46 |
| **NDMA** | Network-assisted Diversity Multiple Access | 144 |
| **OFDMA** | Orthogonal Frequency Division Multiple Access | 17 |
| **OFDM** | Orthogonal Frequency Division Multiplexing | 48 |
| **PAPR** | Peak-to-Average-Power-Ratio | 18 |

| | | |
|---|---|---|
| **PER** | Packet Error Rate | 144 |
| **PHY** | Physical | 144 |
| **PIC** | Parallel Interference Cancellation | 78 |
| **PSK** | Phase Shift Keying | 17 |
| **QAM** | Quadrature Amplitude Modulation | 17 |
| **QPSK** | Quadrature Phase Shift Keying | 128 |
| **QoS** | Quality-of-Service | 144 |
| **RCPC** | Rate-Compatible Punctured Convolutional | 32 |
| **RCPT** | Rate-Compatible Punctured Turbo | 33 |
| **RRC** | Repetition Redundancy error Correction | 35 |
| **RS** | Reed-Solomon | 28 |
| **RTD** | Round-Trip Delay time | 24 |
| **RTS** | Request to Send | 74 |
| **RTT** | Round-Trip Time | 24 |
| **RV** | Random Variable | 106 |
| **SC-FDE** | Single Carrier with Frequency Division Equalizer | 143 |
| **SC-FDMA** | Single Carrier with Frequency Division Multiple Access | 17 |
| **SDMA** | Space Division Multiple Access | 11 |
| **SDU** | Service Data Unit | 40 |
| **SICTA/F1** | SICTA with First 1 feedback | 86 |
| **SICTA/FS** | SICTA with First Success | 86 |
| **SICTA** | Successive Interference Cancellation in a Tree Algorithm | 84 |
| **SIC** | Successive Interference Cancellation | 77 |
| **SIFS** | Short InterFrame Space | 136 |
| **SIMO** | Single Input Multiple Output | 11 |
| **SINR** | Signal-to-Interference-plus-Noise Ratio | 80 |
| **SISO** | Single Input Single Output | 93 |
| **SNR** | Signal-to-Noise Ratio | 143 |
| **SPR** | Single Packet Reception | 119 |

**SP** Shifted Packet . . . 131
**STC** Space-Time Codes . . . 11
**TDMA** Time Division Multiple Access . . . 143
**TDM** Time Division Multiplex . . . 17
**UC** Uncorrelated Channel . . . 131
**UMTS** Universal Mobile Telecommunication System . . . 95
**UWB** Ultra Wide Band . . . 38
**W-CDMA** Wideband-CDMA . . . 3
**WLAN** Wireless Local Area Network . . . 10
**WSN** Wireless Sensor Networks . . . 145
**WiMAX** Worldwide Interoperability for Microwave Access . . . 3

# List of Symbols

$\alpha_1$ .......... 52

The expected number of packet arrivals per slot.

$a_m$ .......... 52

The probability of arriving $m$ packets during a slot.

$A(z)$ .......... 54

The z-transform of $a_m$ which represents the moment-generating function of the packet arrival process.

$\alpha$ .......... 45

Inverse of the Signal-to-Noise Ratio (SNR).

$B_{k,q}^{(q',i)}$ .......... 103

The $q'$th feedback filter used to remove the interference from the $q'$th packet in the NDMA system.

$B_k^{(i)}$ .......... 44

The feedback coefficient in the DC Hybrid ARQ system.

$\beta(\chi^k)$ .......... 110

Represents the fact that at least one MT has a packet to transmit.

$\bar{\beta}$ .......... 112

The event of a MT being idle at the beginning of the epoch for $\chi^k \simeq \pi^{ndma}$.

$C_k^{(i)}$ .......... 46

Set of coefficients that characterize the feedforward filter.

$\chi$ .......... 106

Random variable representing the number of MTs with packets available to transmit in the first slot of an epoch.

$\chi^k$ .......... 109

The number of MTs with packets available to transmit in the first slot of the epoch $k$.

$\hat{\chi}^k$ .......... 110

Represents the conditional distribution of $\chi^k$ given $\beta(\chi^k)$.

$D_{min}$ . . . . . 114

The minimum average delay for a given load and a given number of MTs.

$\Delta_k^{(i)}$ . . . . . 45

Represent the noise component associated to decision errors.

$\delta(Q)$ . . . . . 106

The duration of the detection process in the NDMA system.

$dur_c^m(\chi)$ . . . . . 107

The $m$th moment for the duration expectation of an epoch with $c$ groups of detection slots and $\chi$ MTs with packets to transmit.

$E_b$ . . . . . 51

Represents the average bit energy associated to a given packet transmission.

$E[D]^{dc}$ . . . . . 58

Packet delay, defined as the time interval between the packet arrival and its removal from the queue (at the end of a slot).

$E[D_b](\chi^k)$ . . . . . 110

The expectation of the packet service time in the NDMA system.

$E[D']^{dc}$ . . . . . 58

Discrete packet delay, defined by the number of slots elapsed between the slot of the packet arrival and the instant it departs from the system.

$E[\Phi]$ . . . . . 58

The mean packet arrival deviation.

$E[\mathcal{S}]$ . . . . . 59

The average packet's service time for the DC Hybrid ARQ system.

$E[Z]$ . . . . . 53

Average number of transmissions for a packet until its successful reception or until being dropped.

$E[D]^{ndma}$ . . . . . 112

The average system delay for a packet in the NDMA system.

$E[D_b^2]$ . . . . . 112

Represents the second order moment for the packet service time in the NDMA system.

$E[\Delta^m](\chi)$ . . . . . 108

The $m$th moment of the expected duration of an epoch when $\chi$ MTs have packets to transmit.

$\eta_c$ . . . . . 106

Random variable representing the number of MTs that transmit a packet during the $c$th group of detection slots of an epoch.

$E\left[\Theta^m\right](\chi)$ ........ 108

The $m$th moment of the expected number of bytes transmitted during the epoch when $\chi$ MTs have packets to transmit.

$F_{k,q}^{(r,i)}$ ........ 103

The $r$th feedforward filter associated to the signal of the $r$th collision in the NDMA system.

$F_k^{(r,i)}$ ........ 44

The feedforward coefficient in the DC Hybrid ARQ system.

$G^{dc}$ ........ 60

The channel goodput in the DC Hybrid ARQ system.

$G_{fd-bound}^{ndma}$ ........ 101

The bound of the full-duplex throughput in the NDMA system.

$G_{hd-bound}^{ndma}$ ........ 101

The bound of the half-duplex throughput in the NDMA system.

$G_{max}^{ndma}$ ........ 113

The maximum saturated throughput in the NDMA system.

$G(\chi, p)^{ndma}$ ........ 108

The average normalized throughput for an epoch in the NDMA system.

$G_{sat}$ ........ 53

Saturation goodput in the DC Hybrid ARQ system.

$G_{sat}^{ndma}$ ........ 113

The saturation throughput in the NDMA system.

$H_{k,q}^{(r)}$ ........ 103

Represents the channel frequency response for the $k$th subcarrier, the $q$th user and the $r$th version of the collision in the NDMA system.

$H_k^{(r)}$ ........ 43

The overall channel frequency response for the $r$th transmission attempt in the DC Hybrid ARQ system.

$J$ ........ 97

Number of Mobile Terminals that send data to a Base Station.

$l$ ........ 43

Represents the number of transmission attempts in the DC Hybrid ARQ system.

$L_{data}$ ........ 97

Packet length.

$Limit\,(c)$ ........ 107

Represents the number of transmitting MTs that close an epoch during the $c$ group of detection slots.

$L_n^{I(i)}$ ........................................ 45

The LogLikelihood Ratios of the "in-phase bit" associated to $s_n$.

$L_n^{Q(i)}$ ........................................ 45

The LogLikelihood Ratios of the "quadrature bit" associated to $s_n$.

$M_C$ ........................................ 99

Maximum number of groups of detection slots during an epoch.

$M_n$ ........................................ 52

Represents the number of packet arrivals at the station during the $n$th slot in the DC Hybrid ARQ system, according to a stochastic process.

$M_R$ ........................................ 99

Maximum number of retransmissions in different epochs in the NDMA system.

$N_0$ ........................................ 51

Represents the one-sided power spectral density of the noise.

$N_{Block}$ ........................................ 97

Number of symbols in each FFT block.

$N_{FFT}$ ........................................ 97

Number of FFT blocks in a packet.

$N_k^{(r)}$ ........................................ 43

The channel noise associated to the $r$th transmission attempt.

$N_{PhyPreamble}$ ........................................ 97

Number of symbols representing a physical preamble overhead in the packet.

$\nu$ ........................................ 106

Random variable with the number of unintelligible groups of detection slots during an epoch.

$num_c^m(\chi)$ ........................................ 108

The $m$th moment of the expected number of bytes received during an epoch with $c$ groups of detection slots and $\chi$ MTs with packets to transmit.

$\Omega_{Q,k}$ ........................................ 105

The set with all error patterns of the detection process when $Q$ MTs transmit, that lead to $k$ packets received.

$\omega_{Q,q}$ ........................................ 105

On-off random variable that represents the outcome of the detection process for the packet from the $q$th MT when $Q$ MTs transmit.

$p_c$ ........................................ 97

Represents the MTs access probability for the $c$th transmission attempt in the NDMA system.

$P_{j,i,l}$ .................................................................... 53

The steady-state conditional probability of $X_k = i$ and $Y_k = j$, knowing that $Z_k = l$.

$p_{opt}{}^{\star}$ .................................................................... 114

The $p$ value that minimizes the packet delay in the conditions of (5.35).

$p_{QE}\left(\chi^k\right)$ .................................................................... 109

Represents the probability that the MAC queue of the MT is empty after a successful transmission in the NDMA system.

$p_{sat}{}^{\star}$ .................................................................... 113

The $p$ value that maximizes the saturated throughput in the NDMA system.

$p_{suc}(\chi)$ .................................................................... 109

The expected packet successful transmission probability during an epoch with an initial state where $\chi$ MTs had packets to transmit.

$PER_Q$ .................................................................... 105

Average Packet Error Rate for all users when $Q$ MTs are transmitting simultaneously.

$PER_{Q,q}$ .................................................................... 105

Packet Error Rate for the $q$th user when $Q$ MTs are transmitting simultaneously.

$\Phi$ .................................................................... 58

The discrete mean packet delay.

$\phi_l$ .................................................................... 110

The event of a MT having $l$ successive epochs with failed transmissions.

$\check{\pi}^{ndma}$ .................................................................... 112

Represents the conditional distribution of $\pi^{ndma}$ given $\bar{\beta}$.

$\hat{\pi}^{ndma}$ .................................................................... 112

Represents the conditional distribution of $\pi^{ndma}$ given $\beta\left(\pi^{ndma}\right)$.

$\pi\left(i\right)$ .................................................................... 54

The steady-state probability of the presented DC Hybrid ARQ system.

$\pi_i^{ndma}$ .................................................................... 111

The steady-state probability distribution of $\left\{\chi^k\right\}_{k=0}^{\infty}$.

$\pi^{ndma}$ .................................................................... 111

The system's equilibrium state.

$\Pi\left(z\right)$ .................................................................... 54

The z-transform of $\pi\left(i\right)$ which represents the moment-generating function of the steady-state probability in the DC Hybrid ARQ system.

$\psi\left(Q\right)$ .................................................................... 105

The random variable associated with the number of packets correctly received when

$Q$ MTs transmit during the detection process.

$Q$ . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . 99

Number of transmitting MTs in the NDMA system.

$q_l$ . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . 52

Characterizes the probability of successfully transmitting a packet with $l$ transmission attempts.

$Q_{max}$ . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . 98

The maximum number of concurrent MTs transmissions that the receiver can successfully decode.

$Q_x$ . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . 53

The probability of having $x-1$ failed transmissions.

$R$ . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . 52

Represents the maximum number of transmissions of a single packet allowed in the DC Hybrid ARQ system.

$R_{dat}$ . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . 108

Data rate.

$\rho^{(i)}$ . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . 45

The correlation coefficient.

$\hat{s}_n^{(i)}$ . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . 45

Represents the hard-decisions associated to $\tilde{s}_n^{(i)}$.

$\hat{S}_k^i$ . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . 45

The Discrete Fourier Transform of the block $\hat{s}_n^{(i)}$.

$\overline{S}_{k,q'}^{(i)}$ . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . 103

The Discrete Fourier Transform of the block $\overline{s}_{n,q'}^{(i)}$.

$S_{k,q}^{(r)}$ . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . 103

The Discrete Fourier Transform of the block of the $q$th user associated to the $r$th version of the collision $s_{n,q}^{(r)}$.

$\tilde{S}_{k,q}^{(i)}$ . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . 103

The frequency-domain samples at the output of the $q$ user for a given iteration $i$.

$S_k^{(r)}$ . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . 43

The Discrete Fourier Transform of $s_n^{(r)}$.

$S_n$ . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . 42

The Discrete Fourier Transform of $s_n$.

$s_n$ . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . 42

Data symbol from a given constellation.

$s_{n,q}$ . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . . 102

The time-domain block with data symbols associated to the $q$th user in a $Q$ user collision.

$\hat{s}_{n,q}^{(i)}$ ........ 103

The hard-decisions associated to $\tilde{s}_{n,q}^{(i)}$.

$\overline{s}_{n,q'}^{(i)}$ ........ 103

Represents the average symbol values conditioned to the Frequency Domain Equalization output in the NDMA system.

$s_{n,q}^{(r)}$ ........ 102

The time-domain block associated to the $q$th user and to the $r$th transmission attempt.

$\tilde{s}_{n,q}^{(i)}$ ........ 103

The Inverse Discrete Fourier Transform of $\tilde{S}_{k,q}^{(i)}$.

$s_n^{(r)}$ ........ 42

Data symbol from a given constellation associated to the $r$th transmission attempt.

$\overline{S}_k^{(i)}$ ........ 44

The Discrete Fourier Transform of $\overline{s}_n^{(i)}$.

$\overline{s}_n^{(i)}$ ........ 44

The average symbol values conditioned to the Frequency Domain Equalization output.

$\tilde{S}_k^{(i)}$ ........ 43

The frequency-domain samples at the output for a given iteration $i$.

$\tilde{s}_n^{(i)}$ ........ 45

The inverse Discrete Fourier Transform of $\tilde{S}_k^{(i)}$.

$\sigma_{eq}$ ........ 45

The variance coefficient of LogLikelihood Ratios values.

$T$ ........ 52

The number of slots scheduled to other stations in the TDMA super-frame.

$t_{ack}$ ........ 101

Duration of the ACK packet payload.

$t_{dat}$ ........ 101

Duration of the average data packet payload.

$t_h$ ........ 101

Duration of the MAC header.

$t_p$ ........ 101

Duration of the physical preamble.

$t_{sifs}$ ........ 101

Duration of the SIFS.

$t_{sy}$ ........ 101

Duration of the SYNC packet payload.

$\theta(Q)$ ........ 105

The expected number of packets jointly detected by the Base Station (BS) in the NDMA system.

$U_{j,l}(z)$ ........ 54

The z-transform of $P_{j,i,l}$.

$u(x)$ ........ 103

Unitary step function.

$\varrho$ ........ 110

The network's utilization rate which represents the probability that the MAC queue of the MT is not empty.

$\varsigma_i(\chi)$ ........ 109

Probability of having $i$ packets transmitted with success during an epoch, when $\chi$ MTs have packets to transmit.

$X_k$ ........ 53

The number of packets in the station's queue waiting for transmission (backlogged).

$Y_k$ ........ 53

Represents the scheduled slot index in the TDMA frame of slot $k$.

$Y_k^{(r)}$ ........ 43

The Discrete Fourier Transform of $y_n^{(r)}$.

$y_n^{(r)}$ ........ 43

Data symbol from the received signal associated to the $r$th transmission attempt.

$Z_k$ ........ 53

The number of transmission attempts of the head-of-line (HoL) packet at the end of slot $k$.

$\zeta_r$ ........ 47

Represents a shift in the frequency, for the $r$th transmission attempt, used with SP technique.

# CHAPTER 1

## Introduction

In the latest years wireless communications have become one of the most vibrant areas in the communications field, experiencing an exponential growth from the early 1990s [Lescuyer and Lucidarme, 2008]. Although it has been a topic of study since the 1960s, only over the last decades the industry was able to embrace the research theory and activities in the area. This is a direct consequence of two factors: first, research and development of digital communications systems is undergoing a revolution fuelled by rapid advances in technology. With the sophistication of signal processing and computation, advances on communication theory have an increasing potential to bridge the gap between practically feasible channel utilization and the fundamental information theoretical limit on channel capacity [Verdu, 1998]. Secondly, the successful implementation of low power signal processing algorithms and coding techniques allowed the proliferation of wireless cellular systems (e.g. the IS-95 Code Division Multiple Access (CDMA) standard). This provided a real and visible demonstration that good ideas from communication theory can have a significant impact in practice.

Two fundamental aspects need to be considered in wireless communications, since they are not as significant in wireline communication [Tse and Viswanath, 2005]. The first is the phenomenon of fading: the time variation of the received signal strength due to the small scale effect of multipath fading, as well as large scale effects such as path loss via distance attenuation and shadowing by obstacles. The second aspect is the interference, between users. The interference can be between transmitters communicating with a common receiver (e.g. uplink of a cellular system), between signals from a single transmitter to multiple receivers (e.g. downlink of a cellular system), or between transmitter-receiver pairs (e.g. interference between users in different cells). An efficient way of coping with these two phenomenons is central to the design of wireless communication systems. Although usually only Physical (PHY) layer solutions deal with these phenomenons, recent cross-layer efforts have been proposed which are the central theme of this thesis. Cross-layer design refers to protocol design done by actively exploiting the dependence between protocol layers to obtain performance gains, which nowadays has become increasingly relevant [Srivastava and Motani, 2005].

Traditionally the design of broadband wireless communication systems has focused on increasing the reliability of the air interface. In this context, fading and interference play a key role. More recently, the demand for higher bit rates has shifted recent developments towards increasing the spectral and power efficiency. Associated with this shift is a new point of view that fading can be viewed as an opportunity to be exploited, by implementing diversity techniques.

## 1.1 Motivation

Wireless communications is a field that has been around for over a hundred years, starting around 1897 with Marconi's successful demonstrations of wireless telegraphy [Marconi, 1897][Fahie, 1901, pages 316–340]. The focus of this thesis is on cellular networks, essentially because they are a topic of current interest. In addition, other wireless systems can be usually observed as special cases or generalizations of the features of cellular networks. A cellular network consists of a large number of wireless subscribers who have a Mobile Terminal (MT), that can be used in cars, in buildings, on the street, or almost anywhere. There is also a number of fixed Base Stations (BSs), arranged to provide coverage of the subscribers. Usually, the BSs are placed somewhat irregularly, depending on the location of places such as building tops or hill tops that have good communication coverage and that can be leased or bought. The area covered by a BS, i.e. the area from which incoming signals reach that BS, is called a cell. When an MT needs to transmit data, it is connected to the BS to which it appears to have the best path (often but not always the closest BS).

The wireless link from a BS to the MT is interchangeably called the downlink or the forward channel, and the link from the MT to a BS is called the uplink or a reverse channel. There are usually many MTs connected to a single BS, and thus, for the downlink channel, the BS must multiplex together the signals to the various connected MTs and then broadcast one waveform from which each MT can extract its own signal. For the uplink channel, each MT connected to a given BS transmits its own waveform, and the BS receives the sum of the waveforms from the various users plus noise. The BS must then separate the signals from each MT and forward these signals to its central.

With more than five billion worldwide mobile connections, there is no doubt that Second-Generation (2G) and Third-Generation (3G) cellular technologies are worldwide successes, adopted by most countries and network operators. Mobile communication technologies are often divided into generations, with First-Generation (1G) being the analogue mobile radio systems of the 1980s, 2G the first digital mobile systems, 3G the first mobile systems handling broadband data, and future Fourth-Generation (4G) networks (also known as International Mobile Telecommunications - Advanced (IMT-Advanced)[1] networks) which is the first cellular communication system optimized from the outset to support packet-switched data services, within which packetized voice communications are just one part. While the 2G family was based on Time Division Multiple Access (TDMA) and Frequency Division Multiple Access (FDMA) schemes, the 3G family uses CDMA, known as Wideband-CDMA (W-CDMA) owing to its 5 MHz carrier bandwidth [Sesia *et al.*, 2011]. 3rd Generation Partnership Project (3GPP) group, which is currently the dominant standards development group for mobile radio systems, has significantly evolved 3G technology since its first introduction. The first release of 3G networks, the Universal Mobile Telecommunication System (UMTS) standard, “Release 99” [3GPP, 2000] published in 1999, was mostly oriented towards dedicated channel allocation, and circuit-switched service support. Later, the standard evolved to an high-speed packet radio interface for downlink transmission (High-Speed Downlink Packet Access (HSDPA), on “Release 5” [3GPP, 2002]) and uplink transmission (High-Speed Uplink Packet Access (HSUPA), on “Release 6” [3GPP, 2004]) denoting a clear orientation towards IP-based services. These enhancements, known collectively as High-Speed Packet Access (HSPA) have been further improved in “Release 7” [3GPP, 2007] (known as HSPA+) with higher-order modulation and, for the first time in a cellular communication system, multi-stream Multiple Input Multiple Output (MIMO) operation. These enhancements of 3G technology have ensured backward compatibility with earlier releases, enabling network operators who have invested heavily in the W-CDMA technology of UMTS to generate new revenues from new features, while still providing service to their existing subscribers using legacy terminals.

The first pre-4G standard appeared on “Release 8” [3GPP, 2008] of the 3GPP specification series, the so-called Long Term Evolution (LTE) [Dahlman *et al.*, 2011]. LTE has adopted Orthogonal Frequency Division Multiplexing (OFDM) [Hanzo, 2003] and Single Carrier with Frequency Division Multiple Access (SC-FDMA) [Sesia *et al.*, 2011] (for the downlink and uplink respectively) technologies, which are dominating the latest evolutions of all mobile radio standards[2]. The LTE system was designed from the start with the goal of evolving the radio access technology under the assumption that all services would be

[1]IMT-Advanced are requirements issued by the International Telecommunication Union (ITU)-Radio Communication Sector of the ITU in 2008 for what is marketed as 4G mobile phone and Internet access service.

[2]Except in Worldwide Interoperability for Microwave Access (WiMAX) which still uses Orthogonal Frequency Division Multiple Access (OFDMA) for the uplink.

packet-switched, rather than following the circuit-switched model of earlier systems. LTE benefits also from being free to adopt radical new technology without the constraints of backward compatibility. Although LTE, often called 4G, already satisfies to a large extent the requirements defined for 4G networks, only LTE Advanced ("Release 10" [3GPP, 2011] of the 3GPP specification series) fully satisfies these requirements and even exceeds them in several aspects, being LTE labelled as "3.9G" [Dahlman *et al.*, 2011]. Where the first version of LTE exploited MIMO antenna techniques to deliver high data rates, the evolution of LTE towards LTE-Advanced extends such techniques further for both downlink and uplink communication, together with support for yet wider bandwidths. It also includes new features, such as support for relaying or enhancing inter-cell interference coordination [Boudreau *et al.*, 2009].

In order to have high bit rates we need to take out the most of the available bandwidth putting stress on spectral efficiency in all its aspects (modulation, error recovery, etc.). On the other hand, the power requirements increase with the bit rate, which means that it also needs high power efficiency. This is more critical when considering the uplink transmission in a cellular system, due to the limited autonomy of the batteries in the MTs.

The multipath propagation characteristics of the cellular wireless channels can be severely time-dispersive (i.e. very selective in the frequency), especially for broadband systems, introducing further difficulties. It is well-known that block transmission schemes, combined with frequency-domain processing are appropriate for these types of channels. Single Carrier with Frequency Division Equalizer (SC-FDE) (the modulation scheme) [Falconer *et al.*, 2002] and SC-FDMA [Tse and Viswanath, 2005] (the multi-access version) schemes are widely recognized as the best candidates for the uplink transmission of broadband wireless systems, since the transmitted signals have reduced envelope fluctuations, allowing efficient power amplification, and the signal processing requirements are concentrated at the receiver (i.e. the BS) [Falconer *et al.*, 2002]. More recent techniques can be employed to further improve the power efficiency, such as Iteractive Block-Decision Feedback Equalization (IB-DFE), which can be regarded as turbo equalization schemes implemented in the frequency-domain [Benvenuto *et al.*, 2010].

One of the most challenging topics is how to cope with lost packets in wireless random access scenarios where data transmission is unreliable. Packets are prone to get lost due to poor propagation conditions (e.g., deep fading, strong shadowing effects and/or strong interference levels) or collisions (associated with the medium access). The traditional approach is to discard lost packets and retransmit them, which leads to throughput degradation (that can be thought of as a degradation in the system's spectral efficiency) and increased delays, not to mention the additional power spent in the retransmissions. If the reason was poor propagation conditions, they are likely to persist and retransmission attempts are bound to fail. If it was a collision, further attempts might still have more than one user causing inefficiencies again.

However, the signal associated to a lost packet (regardless of the reason) has information concerning the packet (or packets) involved and should not be wasted [Sindhu, 1977; Costello *et al.*, 1998]. In fact, by combining several "lost packets" it is possible to end up receiving successfully a packet (or packets). This can be regarded as the use of adaptive time diversity techniques.

The consideration about the implementation complexity comes into play here. The simplest way of coping with lost packets is to employ Diversity-Combining (DC) techniques [Benelli, 1985; Yu and Giannakis, 2005; Dinis *et al.*, 2008]. In these techniques, the individual symbols from multiple identical copies of a packet are combined to create a single packet with more reliable constituent symbols. They are based on repetition codes with soft decision, which are not bounded by the performance of a base code. DC systems are simpler to implement, and allow efficient implementations with performances comparable to the more general, complex and powerful codes with puncturing when iterative detectors are used [Dinis *et al.*, 2008]. The performance gains at high Signal-to-Noise Ratio (SNR) conditions are similar to the ones with conventional Automatic Repeat reQuest (ARQ) schemes. However, since the individual Packet Error Rate (PER) decreases as the number of packet retransmissions grows, a significant performance boost is obtained in the presence of unfavourable propagation conditions (allowing better system power efficiency when compared to traditional retransmission approaches). TDMA is used in several current wireless network systems (e.g. 802.16 [Nuaymi, 2007], LTE [Lescuyer and Lucidarme, 2008], etc.) when hard Quality-of-Service (QoS) guarantees are needed. This turns out to be a match candidate to employ DC techniques since both share the same purpose: deliver hard QoS guarantees in a collision free channel of a shared link.

However, DC techniques are useless if the packets were lost due to a collision. A MultiPacket Reception (MPR) scheme is probably the best solution to cope with collisions [Sadjadpour *et al.*, 2010]. The most common MPR solution is employing Direct Sequence - Code Division Multiple Access (DS-CDMA) [Viterbi *et al.*, 1995] techniques when employing multiuser receivers [Verdu, 1998]. These approaches allow significant improvements in the spectral efficiency in high load conditions, i.e. when the number of simultaneous users is close to the maximum allowed. Nevertheless, when in the presence of low load conditions (i.e. when just a few packets are involved in a collision) there is a significant waste of resources since the spreading factor needs to be as high as the maximum number of packets expected to be involved in the collisions.

A promising MPR technique called Network-assisted Diversity Multiple Access (NDMA) was presented in [Tsatsanis *et al.*, 2000]. As with other MPR schemes proposed for CDMA or Space Division Multiple Access (SDMA), it takes advantage of the redundancy inherent to multiple collisions to separate the packets involved. However, contrarily to CDMA or SDMA, the additional redundancy in NDMA is dynamic (i.e. it depends on the number of packets involved in the collision), making it very efficient for both low and high load conditions.

## 1.2 Research Goals

In this thesis I propose two distinct schemes for the uplink transmission to cope with lost packets in broadband wireless random access scenarios. The use of SC-FDE schemes is considered, combined with either frequency-domain DC or NDMA techniques to cope with lost packets in an answer to reduce complexity. The major goals of these thesis are:

- Theoretical analysis of a DC Hybrid Automatic Repeat reQuest (Hybrid ARQ) solution employing SC-FDE for the uplink of wireless systems and proposition of an analytical performance model for aspects like goodput and delay.
- Analysis of an NDMA solution employing SC-FDE for the uplink of wireless systems, and proposition of a Medium Access Control (MAC) protocol able to control the number of concurrent users for limiting the complexity of the receiver.
- Development of an analytical model for the NDMA MAC protocol that considers throughput and delay, and allows system optimization.
- Comparison of the performance of the two proposed solutions, regarding the physical aspects as well as the overall system. In the end, it should be possible to identify the operation regions where each one has the best performance.

All in all, the work provided in this thesis has allowed the publication of ten conference papers, and the submission of five journal papers, two of which accepted for publication.

## 1.3 Thesis Outline

The forthcoming chapters present an overview of existing solutions, and a thorough view of the PHY and MAC layers for each of the solutions analysed in this thesis. In the end, their performance is compared focusing on throughput, delay and scalability. The focus and contributions of each chapter are given in the next paragraphs.

Chapter 2 presents different methods to cope with multiple users accessing a shared medium and transmission reliability issues. The chapter is divided into four big sections: in the first section, a short introduction is given about diversity concepts; in the second section, a short overview is presented for multiple access schemes on today's broadband wireless systems; in the third and fourth sections, the state of the art of Hybrid ARQ schemes is presented.

In chapter 3 I present a DC Hybrid ARQ cross-layer solution for a TDMA architecture on a SC-FDE scheme, which copes with lost packets due to poor propagation conditions. The chapter provides an analytical model for the throughput and delay, valid for generic traffic sources. A thorough study is provided about this system's performance, considering simulation results that validate the analytical model.

Chapter 4 overviews the state of the art on collision resolution techniques in random access scenarios. It starts by presenting an historical perspective, where separate proposals for MAC and PHY layers are presented, and ends presenting cross-layer solutions.

In chapter 5 I propose an NDMA cross-layer solution that copes with lost packets due to collisions in a traditional cellular wireless network. A new MAC protocol was designed considering a BS with MPR capability, that is able to separate up to $Q_{max}$ simultaneous packet receptions. This chapter provides an analytical model for the throughput and delay considering Poisson sources and different backoff algorithms. A thorough study is provided about this system's performance, considering simulation results that validate the analytical model.

The two proposed solutions are compared in chapter 6, on the PHY layer aspects as well as the MAC ones. The chapter begins with an overview of the PHY and MAC layers' parameters for each solution, and ends with a comparative performance analysis focusing on throughput, delay and scalability.

Finally, chapter 7 presents the global conclusions of this thesis, summarises its contributions and examines the scope for future work.

# CHAPTER 2
# Diversity Methods and Error Control Schemes

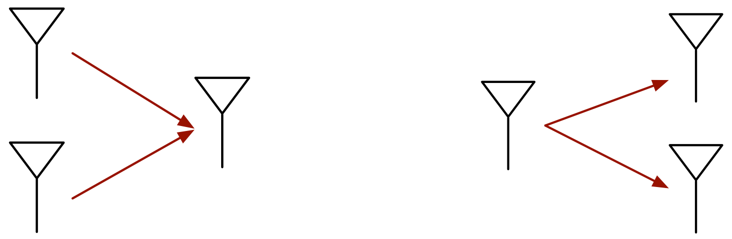
Telecommunication companies compete ferociously with each other and against extreme propagation conditions, to offer the best service conditions and data rates to its clients. Sharing medium access in a efficient way over multiple terminals has always been a challenging issue, since the early ages of telecommunication history. In addition, propagation conditions assumed another dimension in wireless environments, with disruptive phenomenons like fading or multipath, leading to the development of different techniques to allow reliable transmissions.

The purpose of this chapter is to present an overview of the different methods to cope with multiple users accessing a shared medium and about transmission reliability issues.

In the first section, a short introduction about the diversity concept is given, followed by a short overview over multiple access schemes on today's broadband wireless systems. The last two sections, embrace an extensive overview about the development of Hybrid Automatic Repeat reQuest (Hybrid ARQ) schemes and studies in order to frame the reader's view for later chapters.

## 2.1 Diversity Techniques

In the telecommunication's point of view, the term "diversity technique" is implicitly connected to the use of a diversity scheme. This usually refers to a method for improving the reliability of a message signal by using two or more communication channels with different characteristics. On today's technology, "diversity techniques" plays an important role in order to mitigate fading effects and channel interference as well as avoiding error bursts. This mitigation process relies on the assumption that individual channels experience different levels of fading and interference. It assumes that, multiple versions of the same signal



(a) Transmit diversity. (b) Receive diversity.

Figure 2.1: Spatial diversity examples.

may be transmitted and/or received and combined in the receiver side.

Much work has been done over this topic and diversity schemes may exploit different perspectives from simple retransmissions to multipath propagation. The diversity concept can be roughly divided in three major classes: *a*) spacial diversity; *b*) frequency diversity; and *c*) time diversity.

### 2.1.1 Spatial Diversity

A common approach, which recently has become one of the hot topics on researchers' agenda is spacial diversity (also usually called antenna diversity) [Tse and Viswanath, 2005]. Here the signal is transmitted over several different propagation paths. In the case of wireless transmission, it can be achieved by using multiple transmitter antennas (transmit diversity) as exemplified in Figure 2.1a and/or multiple receiving antennas (reception diversity) exemplified in Figure 2.1b.

If the antennas are placed sufficiently far apart, for example at different cellular base station sites or Wireless Local Area Network (WLAN) access points, the channel gains between different antenna pairs fluctuate more or less independently, and independent signal paths are created (it is normally assumed that in the case of Base Station (BS) antennas, a distance of $10\lambda$ is sufficient to ensure a low mutual fading correlation, where $\lambda$ represents the wavelength of a sinusoidal wave [Dahlman *et al.*, 2011]). The required antenna separation depends on the local scattering environment as well as on the carrier frequency. For a mobile terminal which is near the ground with many scatterers around, the channel decorrelates over shorter spatial distances, and typical antenna separation of half to one carrier wavelength is sufficient [Tse and Viswanath, 2005]; this is called micro-diversity. For base-stations on high towers, larger antenna separation of several to tens of wavelengths may be required [Tse and Viswanath, 2005]; this is called macro-diversity or site diversity.

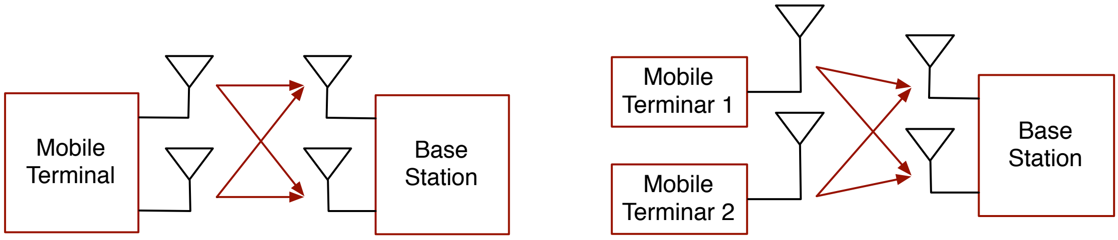
Channels with multiple transmit and multiple receive antennas provide even more potential. Transmit diversity is the case when there are multiple transmit antennas and one or more receive antenna (Multiple Input Single Output (MISO) or Multiple Input Multiple Output (MIMO) channel), which is common in the downlink of a cellular system since it is often cheaper to have multiple antennas at the BS than to have multiple antennas at every Mobile Terminal (MT). In these schemes, such as Space-Time Codes (STC), multiple antennas are positioned as far apart as possible, so that the transmitted signals of the different antennas experience independent fading in the maximum achievable diversity gain. As result, a mitigation on fading is achieved as well as a significant increase of system's capacity[1].

On the other hand, receive diversity is the case when there are one or more transmit antenna and multiple receive antennas (Single Input Multiple Output (SIMO) or MIMO channel), which is common in the uplink of a cellular system. These schemes, such as Layered Space-Time (LST) architecture, beamforming or Space Division Multiple Access (SDMA), constitutes a cost-effective approach to high-throughput wireless communications: in the case of MIMO solutions, they basically increase significantly both the system's capacity and spectral efficiency[2]. The LST architecture proposed by Foschini in [Foschini, 1996] is a framework with multiple antennas at both the transmitter and receiver, which achieves high spectral efficiency (see Figure 2.2a). By employing smart antenna arrays, beamforming mitigates the effects of interfering users roaming in the vicinity of the desired user, provided that their received signals are angularly separable. SDMA exploits the unique, user-specific "spacial signature", i.e. the Channel Impulse Response (CIR) of the individual users for differentiating amongst them and achieves a better spectral efficiency (see Figure 2.2b).

### 2.1.2 Frequency Diversity

In wideband channels, the transmitted signal arrives over multiple time periods and the multipaths can be resolved at the receiver. This phenomenon provides another form of diversity: frequency. The frequency response is no longer flat in this conditions, essentially due to the occurrence of fading and inter-symbol interference[3]. The main problem is then how to deal with the inter-symbol interference while at the same time exploiting the inherent frequency diversity in the channel. In general there are three approaches: single-carrier systems with equalization (e.g. Global System for Mobile communication (GSM)) (see section 2.2.4.2), spread spectrum methods like Direct Sequence - Code Division Multiple

[1] Only diversity gain is achieved.
[2] Both diversity and power gain are achieved.
[3] When the delayed replicas of previous symbols interfere with the current symbol.




(a) The uplink with multiple transmit and multiple receive antennas at each terminal.

(b) The uplink with single transmit antenna at each user and multiple receive antennas at the BS.

Figure 2.2: Illustration of MIMO solutions with spatial multiplexing (LST (a) and SDMA (b)).

Access (DS-CDMA) (e.g. IEEE 802.11b [IEEE, 2007]) (see section 2.2.2) and multi-carrier systems like Orthogonal Frequency Division Multiplexing (OFDM) where it is used a set of non-interfering orthogonal sub-carriers, each experiencing narrowband flat fading (e.g. IEEE 802.11a/g [IEEE, 2007]) (see section 2.2.4.1). An important conceptual point is that, while frequency diversity is something intrinsic in a wideband channel, the presence of inter-symbol interference is not, as it depends on the modulation technique used. These frequency diversity techniques are nowadays widely used on wireless broadband technologies [Tse and Viswanath, 2005].

### 2.1.3 Time Diversity

In a few words, time diversity consists on obtaining multiple versions of the same signal, as any other diversity scheme, but at different time instants. Transmissions over the channel may suffer from error bursts, since it is assumed time varying channel conditions. These errors may be caused due to a set of factors like fading in combination with a moving receiver, transmitter or obstacle or by co-channel interference from other transmitters. To mitigate this problem, the same data is transmitted multiple times and in some cases a redundant error-correcting code is added and the message is spread in time by means of bit-interleaving before it is transmitted.

Automatic Repeat reQuest (ARQ) and Hybrid-ARQ schemes can be considered as a subset of the time diversity class. As a result, an historical overview will be given in chapter 2.3. Throughout this thesis, the focus of development will rely almost exclusively on time diversity concepts.

All these diversity techniques are employed in conjunction with multiple access schemes. The most common ones are subsequently shortly overviewed.

## 2.2 Multiple Access Schemes

In the telecommunications field, a multiple access scheme is a channel access method that allows several terminals connecting to the same shared multi-point transmission medium to transmit/receive over it. Several different ways exist of multiplexing a communication channel or physical medium: starting with the most basic one, Time Division Multiple Access (TDMA), and ending on more complex architectures such as OFDM. An analogy to the problem of multiple access is a room (channel) in which people wish to talk to each other simultaneously. To avoid confusion, people could take turns speaking (time division), speak at different pitches or in different languages (frequency division with frequency or code multiplicity), or speak at different zones in the room (spacial division). In the next sub-sections, a short overview is given.

### 2.2.1 Time Division Multiple Access

Time Division Multiple Access (TDMA) is one of the most traditional multiple access methods. In general, it consists on allowing several users to share the same frequency channel by dividing the signal into different time slots, i.e. each user owns a time slot in a periodic frame. The users transmit consecutively, one after the other, each using its own time slot. In the end, multiple stations share the same transmission medium while using only a part of its channel capacity. TDMA schemes have a drawback: they require a high synchronization overhead in order to maintain the time slots synchronized. That is one of the reasons why in the case of the uplink (i.e. communication from a mobile user or terminal to a typical base station) some difficulties arise, particularly because the mobile user can move around and vary the time advance required to make its transmission match the gap in transmission from its peers. Another relevant topic is the procedure of matching each slot to its owner, usually called scheduling.

TDMA is a scheme widely used on today's communication standards, such as GSM or satellite systems. Part of the work in this thesis considers a TDMA access scheme, and proposes an extensive analytical performance model in chapter 3.

### 2.2.2 Code Division Multiple Access

Code Division Multiple Access (CDMA) is a channel access method used by various radio communication technologies, e.g. Universal Mobile Telecommunication System (UMTS). CDMA employs spread-spectrum technology and a special coding scheme to allow multiple users to be multiplexed over the same physical channel. In a CDMA radio system, each user (or group of users) is given a shared code. Many codes occupy the same channel, but only users associated with a particular code can communicate over it. All the communications of the other users, that have other codes, are perceived as noise and are rejected. Spread-spectrum techniques involve the transmission of a signal in a radio frequency bandwidth substantially larger than the information bandwidth, in order to achieve a particular operational advantage. In addition, a spread spectrum technique spreads the information bandwidth uniformly for the same transmitted power. Different types of spread-spectrum exist, where the two most important are: Direct Sequence Spread Spectrum (DSSS) and Frequency Hopping Spread Spectrum (FHSS), which consequently leads to the access methods DS-CDMA and Frequency Hoping - Code Division Multiple Access (FH-CDMA) respectively.

The code used to spread the information bandwidth can be quasi–orthogonal or orthogonal. By employing orthogonal codes, each user uses a code orthogonal to the others' codes to modulate their signal. Orthogonal codes, such as Walsh–Hadamard codes, have a cross-correlation equal to zero; in other words, they do not interfere with each other. However, the orthogonality between users can be lost due to multipath propagation effects and/or synchronization errors, among other effects. Another alternative is employing quasi-orthogonal codes, such as Maximum Length Sequence, Gold Sequences or Kasami Sequences. As the name suggests, all of these sequences are not exactly orthogonal. However, their cross-correlation levels are low (usually below a certain level to limit multiple access interference). These sequences are a binary sequence that appears random (pseudo-random sequences), but are statistically uncorrelated and can be reproduced in a deterministic manner by intended receivers. Conventionally, synchronous CDMA employs orthogonal codes, while asynchronous CDMA employs pseudo-random sequences.

The codes are generated locally at a much higher rate than the data to be transmitted. Then, the information data is combined via bitwise XNOR (negative exclusive OR) with the faster code. In general, Figure 2.3 exemplifies how this process works. The data signal with pulse duration of $T_s$ is XNOR'ed with the code signal with pulse duration of $T_c$. Therefore, the bandwidth of the data signal is $1/T_s$ and the bandwidth of the spread spectrum signal is $1/T_c$. Since $T_c$ is much smaller than $T_s$, the bandwidth of the spread spectrum signal is much larger than the bandwidth of the original signal. The ratio $T_s/T_c$ can be viewed as the spreading factor or processing gain and determines to a certain extent the upper limit of the total number of users supported simultaneously by a given link.

In general, it is very important to choose the codes correctly in a CDMA system, since the best performance will occur when there is good separation between the signal of the pretended user and the signals of the other users. The separation process of the signals involved in the reception is made using cross correlation. The receiver correlates the received

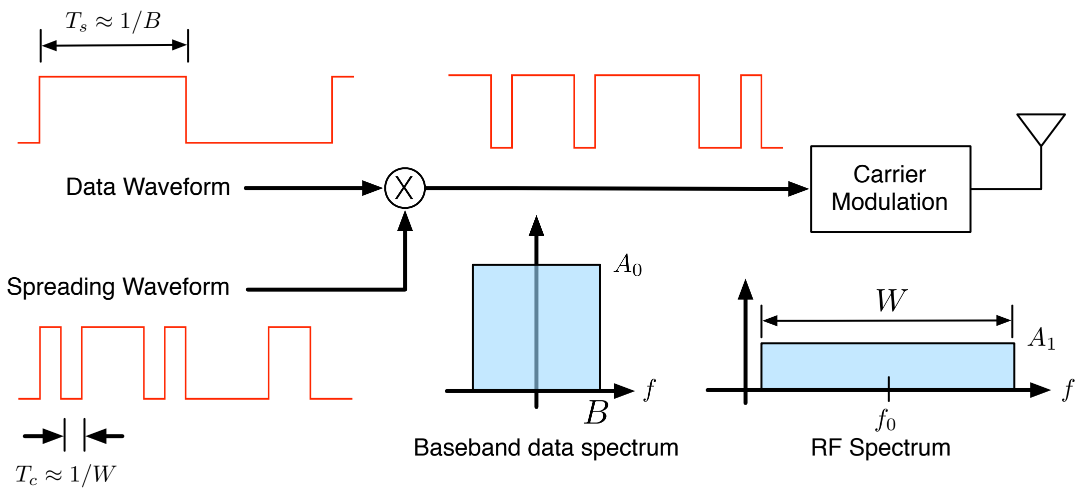



Figure 2.3: Transmitter side of a typically CDMA system.

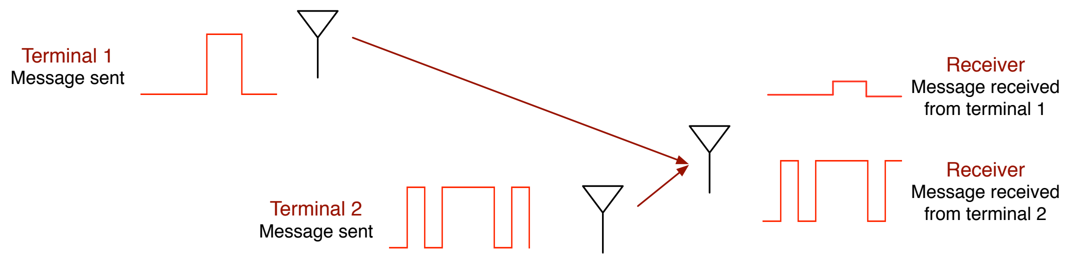



Figure 2.4: Illustration of the near-far problem.

signal with the locally generated code of the pretended user. If the correlation function is high, then the system can extract that signal, if not (value close to zero), the received signal represents transmissions from other users. Regarding synchronization, two approaches exists: a synchronous approach where orthogonal codes are used (less used); and an asynchronous approach with pseudo random codes (more used).

One of the biggest problem in CDMA systems is the near-far problem [Rappaport, 2002; Tse and Viswanath, 2005]. The near-far problem, represented in Figure 2.4, basically consists on a condition where a strong signal captures a receiver, making it impossible for the receiver to detect a weaker signal. Let us consider this example: a receiver and two transmitters, one close to the receiver, the other far away. If both transmitters transmit simultaneously and with equal powers, then the receiver will receive more power from the nearer transmitter. Since one transmission's signal is the other's noise, the Signal-to-Noise Ratio (SNR) for the farther transmitter is much lower. This makes the farther transmitter more difficult to understand, if not impossible. The SNR for the farther transmitter may be below detectability if the nearer transmitter transmits a signal that is orders of magnitude higher than the farther transmitter, and the farther transmitter may just as well not transmit. This effectively jams the communication channel.

To place this problem in more common terms, let us imagine one person talking to someone 5 meters away. If the two users are in a quiet, empty room then a conversation is quite easy to hold at normal voice levels. In a loud, crowded bar, it would be impossible to hear the same voice level, and the only solution is for both users to speak louder. As a consequence, this increases the overall noise level in the bar, and every other costumer has to talk louder too (this is equivalent to power control runaway). Eventually, everyone has to shout to make themselves heard by a person standing right beside them, and it is impossible to communicate with anyone more than a few of centimetres away. A possible solution is a adopting dynamic output power adjustment in the transmitters, i.e. the closer transmitters use less power so that the SNR for all transmitters at the receiver is roughly equivalent.

Another questions and topics may arise in a CDMA environment and can be observed in [Viterbi *et al.*, 1995] or [Lee and Miller, 1998].

### 2.2.3 Space Division Multiple Access

In the literature, the use of multiple receive antennas in the uplink and downlink is often designated as SDMA. It is possible to discriminate amongst the users by exploiting the fact that different users inflict different spatial signatures on the receive antenna array. SDMA exploits the unique, user-specific "spacial signature", i.e. the CIR of the individual users for differentiating amongst them. This allows the system to support multiple users within the same frequency band and/or time slot, provided that their CIRs are sufficiently different and are accurately measured. As one of the most promising techniques aiming at solving the capacity problem of wireless communication systems, SDMA enables multiple users to simultaneously share the same bandwidth in different geographical locations. More specifically, the exploitation of the spatial dimension, namely the so-called spatial signature, makes it possible to identify the individual users, even when they are in the same time/frequency/code domains, thus increasing the system's capacity. Further information about the SDMA techniques can be found at [Tse and Viswanath, 2005] and [Jiang and Hanzo, 2007].

### 2.2.4 Frequency Division Multiple Access

Frequency Division Multiple Access (FDMA) is a channel access method based on the Frequency Division Multiplex (FDM) scheme, which assumes different frequency bands to different data-streams. In the FDMA case, the data streams are allocated to different users or nodes. As a result, each user is individually allocated to one or several frequency bands (or channels). Typically, there is a single channel per carrier, though multi channels can exist[4].

In general, in an FDMA system, when a MT accesses the system, usually two carriers (channels) are assigned, one downlink (BS to MT) and one for the uplink (MT to BS). Separation of the downlink and uplink is necessary to allow the implementation of a duplexer, an arrangement of filters that isolates both channels, thus preventing a radio transceiver from jamming itself. FDMA scheme is particularly commonplace in satellite communication, where it coordinates access between multiple users. Beyond satellite communications, an example of FDMA system was the First-Generation (1G) cell-phone system. The main disadvantage in FDMA systems is related with cross-talks, that may cause interference among frequencies and disrupt the transmission. The complexity involved in the system is also an issue.

OFDM [Hanzo, 2003] and Single Carrier with Frequency Division Equalizer (SC-FDE) [Falconer *et al.*, 2002] are two known modulation techniques used in recent wireless access technologies. Both are modulation techniques identical to the multiplexing schemes, Orthogonal Frequency Division Multiple Access (OFDMA) [Yang, 2010] and Single Carrier with Frequency Division Multiple Access (SC-FDMA) [Myung and Goodman, 2008], but optimized to enhance the performance of a single user. Since this thesis focus on the SC-FDE approach, which is an alternative to OFDM approach, both schemes are going to be shortly overviewed and debated.

#### 2.2.4.1 Orthogonal Frequency Division Multiplexing

Orthogonal Frequency Division Multiplexing is the wireless version of the wired Discrete Multi-Tone (DMT) modulation, a FDM scheme used as a digital multi-carrier modulation method and a close brother to the multiple access scheme OFDMA. The concept is simple: a large number of closely spaced orthogonal sub-carriers are used to carry data. The data is divided into several parallel data streams or channels, one for each sub-carrier. Then, each sub-carrier is modulated with a conventional modulation scheme (like Quadrature Amplitude Modulation (QAM) or Phase Shift Keying (PSK)) at much lower symbol rate than the original data stream.

Recently, OFDM has evolved into a popular scheme for wideband wireless communications since it provides a big set of advantages [Hanzo, 2003]. The primary advantage

[4]Several sub-carriers can be combined into a single bit-stream, modulated onto a carrier and transmitted to one or more users using Time Division Multiplex (TDM).

of OFDM is its ability to handle severe channel conditions, e.g. frequency-selective fading due to multipath or narrowband interference, without complex time-domain equalization. Channel equalization is simplified essentially because it is performed on many slowly modulated narrowband signals rather than in one rapidly modulated wideband signal. The use of efficient implementation of Discrete Fourier Transform (DFT), named as Fast Fourier Transform (FFT), also helps. In addition, the low symbol rate makes the use of guard intervals between symbols affordable, making it possible to eliminate Inter-Symbol Interference (ISI) and time-spreading (on analogue TV it shows up as ghosting) to achieve diversity gain, i.e. a SNR improvement. Though these improvements, OFDM is not perfect. Its biggest drawback is the high Peak-to-Average-Power-Ratio (PAPR), which requires linear transmitter circuitry that can be expensive. Sensitivity to Doppler effects[5] and frequency synchronization can also be issues. More information about this technique can be obtained in [Cimini, 1985; Benedetto and Biglieri, 1999; Hanzo, 2003; Tse and Viswanath, 2005].

#### 2.2.4.2 Single Carrier with Frequency Division Equalizer

OFDM is a recognized multi-carrier solution to combat the effects of multipath conditions, primarily because of the favourable trade-off it offers between performance in severe multipath and signal processing complexity [Cimini, 1985]. But the single-carrier modulation method, when combined with frequency domain equalization (as in SC-FDE) is capable of delivering a performance similar to OFDM with essentially the same overall complexity, plus some advantages inherent to single-carrier architectures.

In a single-carrier system, as the name suggests, the signal is transmitted over a single carrier using some modulation (e.g. QAM or PSK) at a high rate[6]. A Frequency Domain Equalization (FDE) is simply the frequency domain analogue of what is done by a conventional linear time domain equalizer. In channels with severe delay spread, an FDE is computationally simpler than a conventional time domain equalizer. As pointed out in [Sari *et al.*, 1995], when a single-carrier system is combined with FFT and uses a cyclic prefix technique (also employed in OFDM systems), the same performance and low complexity as in an OFDM system is achieved assuming that coding is employed. In general, without coding SC-FDE presents better performance than OFDM, except when a flat fading channel is considered, where the performance of both systems is very similar [Gusmão *et al.*, 2003]. Note that when coding is used if the employed code is powerful, OFDM can achieve a slightly better performance than SC-FDE [Dinis *et al.*, 2003; Montezuma and Gusmao, 2001], but if

[5] Is the change in frequency of a wave for an observer moving relative to the source of the wave.

[6] In an OFDM system, each subcarrier also uses a modulation technique like QAM, PSK, etc. as referred in 2.2.4.1.

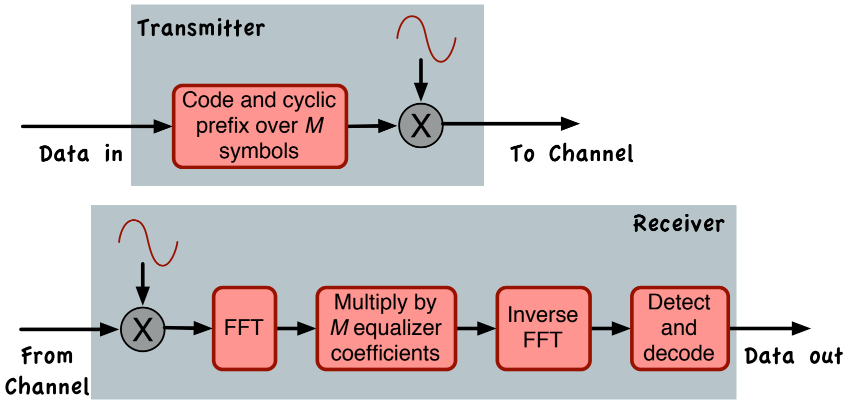



Figure 2.5: SC-FDE with linear equalization.

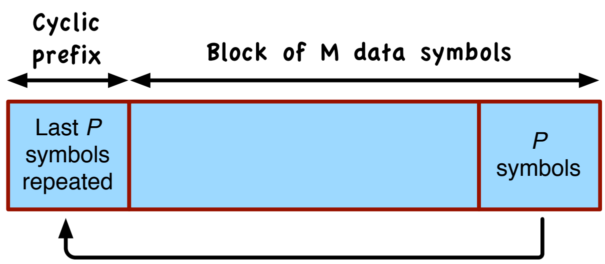



Figure 2.6: Block processing in frequency domain equalization.

not, OFDM performance can be much worse than SC-FDE performance. It is also pointed out, that both SC-FDE and OFDM systems can easily coexist, and extract advantages from such coexistence, due to the number of common signal processing functions that both share.

Figure 2.5 shows conventional linear equalization, using a transversal filter with $M$ tap coefficients, but with filtering done in the frequency domain. With this scheme, several advantages [Falconer *et al.*, 2002] are achieved: a reduced PAPR compared to OFDM is obtained, thereby allowing the use of less costly power amplifiers; the performance is similar to that of OFDM when coding is employed; the reduced complexity advantage of OFDM is also achieved; and coding is not essential for fighting frequency selectivity, as it is in non-adaptive OFDM. Figure 2.6 illustrates the cyclic prefix appending process. The cyclic prefix can be combined with a training sequence for equalizer adaptation [Falconer *et al.*, 2002]. Figure 2.7 [Falconer *et al.*, 2002] shows a comparison of the complexities of time domain and frequency domain processing (either SC-FDE and OFDM linear equalizers) as a function of the length of the channel impulse response, measured in symbol intervals.

When in the presence of frequency-selective radio channels, Decision Feedback Equalization (DFE) gives better performance than linear equalization. Typically, conventional DFE

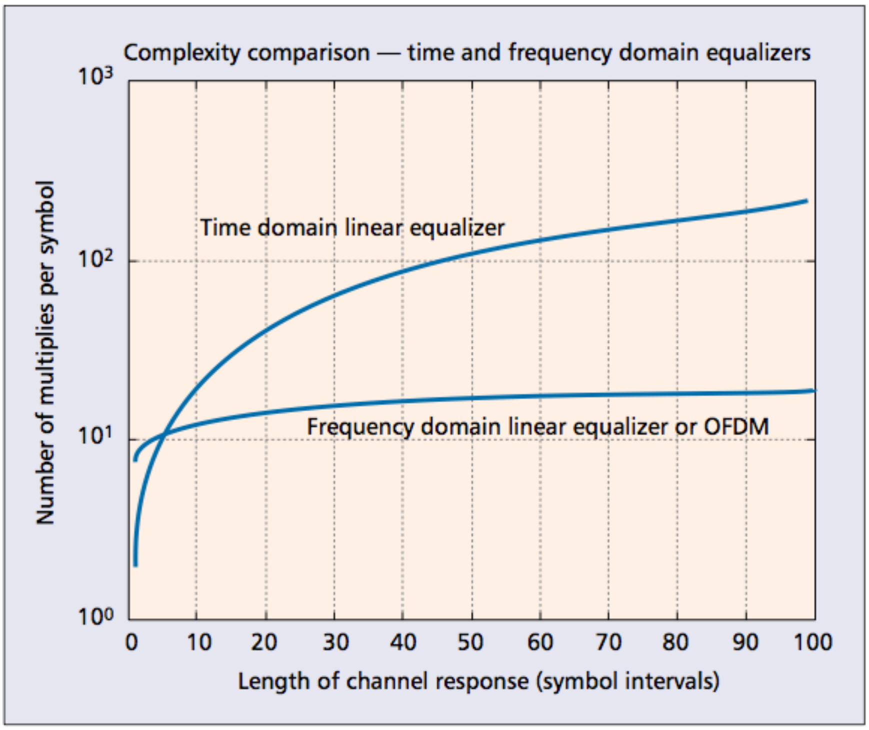



Figure 2.7: Complexity comparison of time and frequency domain linear equalizers [Falconer *et al.*, 2002].

equalizers perform symbol-by-symbol data symbol decisions, then filter and finally the result is fed back to remove their interference effect from subsequently detected symbols. Note that due to the delay inherently in the block FFT signal processing, this immediate filtered decision feedback cannot be done in a frequency domain DFE. To avoid the above mentioned delay problem, a hybrid time frequency domain DFE approach solution would be to use frequency domain filtering only for the forward filter part of the DFE and conventional transversal filtering for the feedback part. The latter is relatively simple since it performs multiplications only on data symbols, and it could be made short or long as required for performance adjustment. Figure 2.8 illustrates such a hybrid time frequency domain DFE topology. Note, as mentioned before, that the forward filter (the complex-valued $M$ forward equalizer coefficients $\{W_l\}$, which compensate for the frequency-selective channel's variations of amplitude and phase with frequency) is applied in frequency domain, when the feedback filter (the $B$ feedback taps $\{f_k\}$ which compensate ISI effects), are applied in time domain.

While attractive for its performance, this hybrid time frequency domain DFE topology still presents a significant complexity, especially due to the signal processing and the design of the feedback filter which is applied in time domain. Benvenuto and Tomasin [Benvenuto and Tomasin, 2002] proposed a non-linear Iteractive Block-Decision Feedback

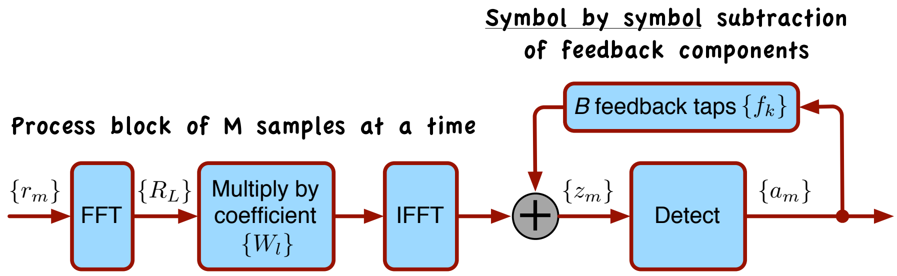



Figure 2.8: SC-FDE decision feedback equalizer.

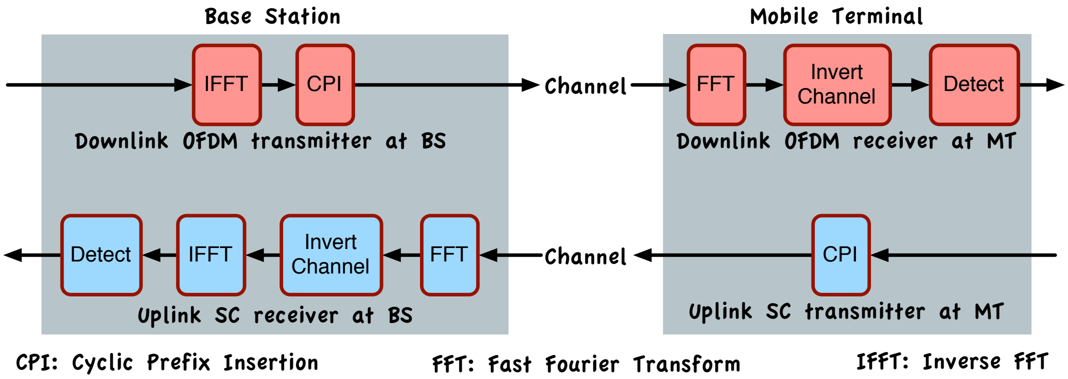



Figure 2.9: Coexistence of SC-FDE and OFDM: uplink and downlink asymmetry.

Equalization (IB-DFE) [Benvenuto *et al.*, 2010] over frequency selective fading channels, when both single branch transmitters and single branch receivers are employed. In this scheme both feedforward and feedback filters are implemented by DFTs, and this yields a significantly lower complexity than traditional DFEs. In addition, a lower complexity in the filter design is also obtained, since it does not require any matrix inversion. In [Dinis *et al.*, 2003, 2004a], Dinis et al. proposed an IB-DFE receiver with receiver diversity, where transmit/receiver spatial diversity techniques are jointly employed in a IB-DFE receiver. MIMO detection techniques were also added in IB-DFE [Dinis *et al.*, 2004b].

As referred earlier, both OFDM and SC-FDE schemes can coexist since the main difference is the placement of an Inverse Fast Fourier Transform (IFFT) operation: in OFDM it is placed at the transmitter to multiplex the data into parallel sub-carriers; in SC-FDE it is placed in the receiver to convert FDE signals back into time domains symbols. Figure 2.9 illustrates this arrangement proposed in [Gusmão *et al.*, 2000; Falconer *et al.*, 2002].

There are great advantages of adopting a dual mode system, where the BS uses an OFDM transmitter and an SC-FDE receiver, and the MT uses an SC-FDE transmitter and an OFDM receiver. One advantage is that most of the cumbersome signal processing is shifted to the BS, since it has two IFFTs and one FFT, while the MT has just one FFT. Another advantage is that the MT transmitter is SC-FDE, which results in a more efficient power consumption due to the reduced power backoff requirements of the SC-FDE system. As a result, this may reduce the cost of a MT's power amplifier.

Further information about SC-FDE schemes can be found in [Sari *et al.*, 1995; Falconer *et al.*, 2002; Gusmão *et al.*, 2003]. In this thesis, an SC-FDE scheme is implemented at the receiver for both studied approaches, in chapter 3 and chapter 5 respectively.

## 2.3 Hybrid ARQ architectures

With the purpose of coping with errors in data transmissions, different approaches have been proposed along the years. Error detection incorporated with ARQ schemes has been widely used for error-control in data communications systems until today. This method of error-control is simple and provides high system reliability. Since part of the work done in this thesis relies on Hybrid-ARQ architectures, an extensive overview will be given about this topic. The primary objective here is to frame the development of Hybrid ARQ studies into later chapters allowing a better understanding of this concept in the reader's perspective.

### 2.3.1 The Early Ages

Transmission errors have been a major concern in data communications since the early ages. These errors, mainly caused by channel noise, can be mitigated with the use of error-detecting or error-control schemes for data communications [Lin and Costello, 1983; Lin *et al.*, 1984; Costello *et al.*, 1998]: ARQ schemes and Forward Error Correction (FEC) schemes.

An ARQ error-control system consists in the incorporation of an high-rate error detection code, say a $(n, k)$ block code, with a certain retransmission protocol. When a message of $k$ useful data bits is ready for transmission, $n - k$ parity-check bits (which are formed based on the code used by the system) are appended to it to form what is usually called the codeword. This codeword is then transmitted and contaminated by the channel noise. At the receiver side, when the data is received, the receiver checks if the received word belongs to the code alphabet, which in the case of linear block codes is usually done by computing its syndrome[7]: if it is zero, the received data is a codeword in the code being used, if not the received data is corrupted. In the first case, the received data is assumed to be error free and is delivered, after removing the parity-check bits, to the user. In the second case, the presence of errors is detected, the receiver discards the erroneously received data, and requests a retransmission

[7] A syndrome is a binary word computed by the decoder and used in making the decision as to which codeword was transmitted.

of the same data. This process is repeat over and over, until the codeword is successfully received.

With this architecture, erroneous data is delivered to the user only if the receiver (also known as decoder) fails to detect the presence of errors. As a result, a proper error-detecting code ought to be used to attain a low probability value of undetected errors[Lin and Costello, 1983]. The main assets of ARQ schemes are the simplicity and the high system reliability extracted from it. As a drawback, the throughput is not constant and falls rapidly with increasing channel error rate.

An alternative is to employ an FEC scheme where an error-correcting code (block or convolutional) is used for fighting transmission errors. The initial procedure is the same as with the ARQ schemes: parity-check bits are added to each transmitted message to form a codeword. However, when the decoder detects the presence of errors in the transmitted data, now it attempts to locate and correct the errors. Regardless of the error correction has been successful or not, the decoded word is then delivered to the user.

In this architecture, erroneous data is delivered to the user if the decoder fails to detect the presence of errors or if it fails to determine the exact locations of the errors. As a result, FEC schemes almost behave in the opposite way to the ARQ scheme: the main advantage is the constant system's throughput, equal to the rate of the used code, i.e. $k/n$ of the available bandwidth; as a drawback, since the probability of a decoding error is much higher than the probability of an undetected error, it is much harder to achieve high system reliability with FEC schemes. An adoption of a long, powerful error-correcting code can mitigate this problem with the cost of making decoding harder to implement, and more expensive, not forgetting the larger delay associated.

These reasons made ARQ schemes the most common solution in data communication systems, whereas FEC schemes are still attractive in situations (e.g. data storage) where feedback channels are not available or retransmission is not suitable for some reason.

#### 2.3.1.1 Basic retransmission strategies

Several retransmission strategies could be taken in order to improve the performance of ARQ schemes. There are three basic types of ARQ schemes: stop-and-wait ARQ, go-back-N ARQ, and selective-repeat ARQ [Burton and Sullivan, 1972; Lin and Costello, 1983]. Stop-and-wait scheme is the most basic one of the three schemes. It represents the simplest ARQ procedure and was implemented in early error-control systems like the IBM Binary Synchronous Communication (BISYNC) [IBM, 1969; Cullen, 1969]. Figure 2.10 exemplifies how stop-and-wait strategy works. The number of functions of the transmitter and the receiver are reduced to

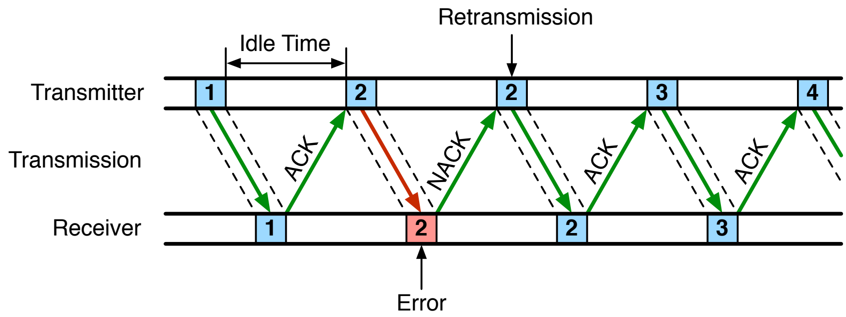



Figure 2.10: Stop-and-wait ARQ scheme.

the minimum: the transmitter sends a codeword to the receiver and waits for an acknowledgement; the receiver waits for the codeword, and when it arrives checks its integrity and transmits the acknowledgement with the result. A Positive Acknowledgement (ACK) from the receiver indicates that the transmitted codeword has been successfully received and it is a green light for the transmitter to send the next codeword in the queue. A Negative Acknowledgement (NACK) from the receiver indicates that in the transmitted codeword has been detected an error, triggering the resending of the codeword by the transmitter. This process of retransmission is repeated until the transmitter receives an ACK.

Despite its simplicity, this strategy is inherently inefficient because of the idle time spent waiting for an acknowledgement of each transmitted codeword. One possible solution[8] is to make the codeword length extremely long. With this, the idle time will become negligible when compared with the transmitting time. Albeit this making sense, it does not provide a viable solution since the probability of the codeword containing errors increases with its length. As a consequence, using long codewords will increase the number of retransmissions[9].

In the 70's, ARQ systems were proliferated to packet-switched and other data networks. Satellite communications with high data rates had become more and more important making the Round-Trip Delay time (RTD)[10] or RTT an issue. A viable alternative to replace the stop-and-wait procedure was needed and go-back-N ARQ systems were proposed. A basic go-back-N system is illustrated in Figure 2.11. The main difference between the go-back-N strategy and the stop-and-wait, is that the transmitter continuously transmits codewords during the previous idle period. Since the acknowledgement arrives at the end of the round-trip delay, the transmitter sends $N-1$ other codewords. Whenever the transmitter receives

[8]Only possible when there is no restrictions imposed by data formats or standards.
[9]Unless the codeword is divided into several parts.
[10]The length of time it takes for a signal to be sent plus the length of time it takes for an acknowledgement of that signal to be received - an internet user can determine the Round-Trip Time (RTT) by using the ping command.

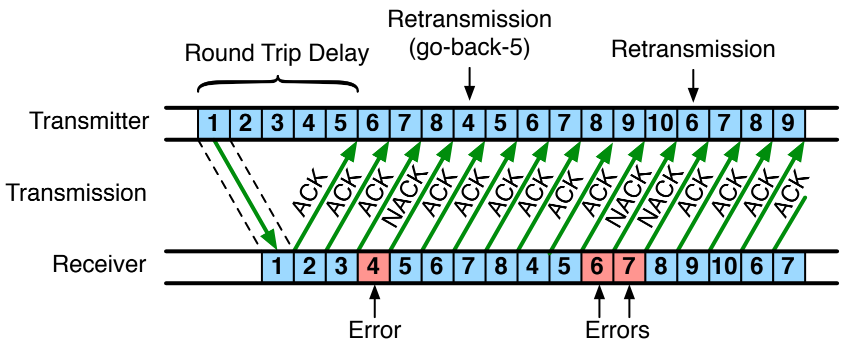



Figure 2.11: Go-back-N ARQ scheme.

a NACK indicating that a particular codeword, let us say codeword $i$, contains an error, it stops transmitting new codewords. It goes back to codeword $i$ (as the name suggests) and proceeds with the retransmission of the respective codeword and the $N-1$ succeeding codewords, which were transmitted during the round-trip delay time. At the receiver end, since the receiver detects the erroneously codeword $i$, it discards the respective codeword and also all the $N-1$ subsequently received codewords, whether they are error free or not. Retransmissions, as in stop-and-wait strategy, continue until the erroneously codeword is positively acknowledged. As soon as it happens the transmitter continues transmitting sequentially the remaining codewords.

This solution has still a main drawback: whenever a received codeword is detected in error, the receiver also rejects the next $N-1$ received codewords, even though many of them may be error free, triggering unnecessary retransmissions. This may represent a resource waste which can result in severe deterioration of throughput performance if a large RTD is involved. Let us consider the following example: a satellite channel with a RTD around 300ms; considering a codeword length of 1000 bits and a bit rate of 10Mb/s, the round-trip time covers around 3000 codeword's time. Therefore, when one codeword is not correctly received, 3000 codewords are rejected.

With the evolution of communications, data rates have grown very quickly. As a result, go-back-N scheme became quite ineffective, essentially due to the retransmission of many error free codewords following an erroneous codeword. The solution passes through the use of the selective-repeat ARQ scheme. The selective-repeat ARQ error-control scheme works in a similar way to go-back-N scheme, but adds a specific improvement: the transmitter now only resends those codewords that are negatively acknowledged (NACK) or that exceed a delay threshold (e.g. timeout threshold). Figure 2.12 illustrates how selective-repeat ARQ scheme works. It is possible to observe that after a retransmission, the transmitter continues

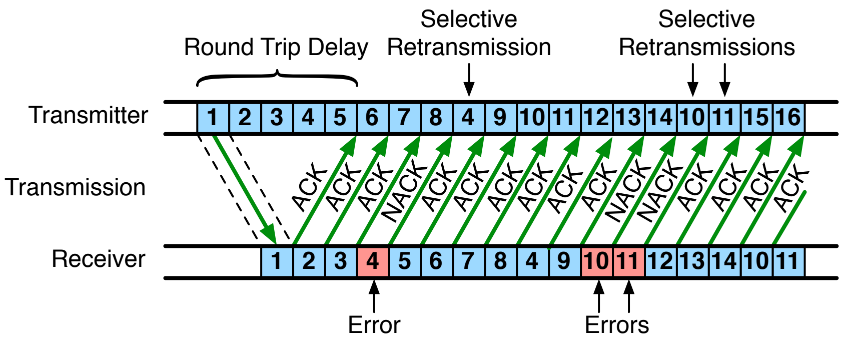



Figure 2.12: Selective-repeat ARQ scheme.

sending new codewords from the transmitter buffer, and not the $N-1$ previously transmitted codewords as in go-back-N scheme. In this scheme, a buffer must be provided at the receiver side with the purpose of storing all the error free codewords that have followed an erroneous codeword, since ordinarily, codewords must be delivered to the end user in correct order [Lin *et al.*, 1984]. After receiving successfully a retransmitted packet (a NACK'ed one), the error free buffered codewords are released and delivered to the end user in consecutive order until the next erroneous codeword is received. In the case of not enough buffer size supplied, the buffer may overflow and some codewords may be lost [Lin *et al.*, 1984].

Two approaches exist to cope with buffer overflow in a selective-repeat ARQ system with a finite receiver buffer: 1) providing a retransmission strategy so that buffer overflow can be prevented, e.g. [Miller and Lin, 1981]; 2) devising a mechanism for the transmitter to detect the occurrence of a buffer overflow so that the lost codewords can be properly retransmitted, e.g. [Yu and Lin, 1981]. Albeit the first approach is usually simpler, the second one offers better throughput performance [Lin *et al.*, 1984].

## 2.4 Hybrid-ARQ Schemes

In error-control systems, ARQ and FEC schemes are only just the tip of the iceberg[11]. Comparing both, we see that ARQ schemes are simpler and provide high system reliability; however their throughput falls off rapidly with increasing channel error rate. In FEC systems we have constant throughput, equal to the code rate used, regardless of the channel error rate; however erroneous codewords could be delivered to end user, destroying the system's reliability.

What if the drawbacks of the ARQ and FEC schemes can be overcome when the two

[11]They are usually seen as extreme opposite cases.

are properly combined? Such combination is usually referred as Hybrid Automatic Repeat reQuest (Hybrid ARQ) [Lin and Costello, 1983; Lin *et al.*, 1984] and basically consists of an FEC subsystem contained in an ARQ system. The purpose of embracing FEC as a function of an ARQ system is to reduce the number and the frequency of retransmissions by correcting some error patterns which may occur most frequently. Thus, the system throughput may be increased when some of the errors are corrected. Another advantage is that, unlike pure FEC systems, even when an uncommon error pattern is detected the receiver requests a retransmission rather than passing the erroneous decoded codeword to the end user. In general, a proper combination of ARQ and FEC schemes open the door to an higher reliability system than an FEC only and also an higher throughput system than an ARQ only, thereby combining the assets of each system.

Over the years, Hybrid ARQ schemes were classified and split into two categories: type-I and type-II schemes [Lin and Costello, 1983]. A simple and straightforward type-I Hybrid ARQ scheme uses a code which is designed for simultaneously detect and correct errors and which was firstly presented in [Wozencraft and Horstein, 1960, 1961]. Let us imagine that a received codeword is detected in error: the first step for the receiver is attempting to correct the error, which is possible if the number of errors and its burst length is within the designed error correcting code capability; if so, the codeword is corrected and delivered to the end user; however, if a error pattern is uncorrectable, it starts the second step, where the receiver rejects the codeword and asks for its retransmission. The retransmitted codeword is exactly the same as transmitted before. This process proceed and repeats itself until the codeword is successfully received/decoded.

Any ARQ scheme (and strategy) can embrace error correction capabilities. Nevertheless, since a code is used for both error correction and detection, as in a type-I Hybrid ARQ system, it inherently requires more parity-check bits than a code only used for error detection purpose in a pure[12] ARQ system. As a direct consequence, the overhead for each transmission increases. Therefore, when the channel error rate is low, a type-I Hybrid ARQ systems has implicitly lower throughput than its congenital pure ARQ system. The good news is, when the channel error rate is high, a type-I Hybrid ARQ system provides higher throughput than its corresponding pure ARQ system, in virtue of its error correction capability that has reduced the number of retransmissions.

Type-I Hybrid ARQ schemes are suited for communication systems when the presence of a fairly constant level of noise and interference is predicted on the channel. The fact of being possible to anticipate a certain constant level of noise and interference, allows

[12] ARQ system alone.

the design and choice of an error code capacity able to correct the vast majority of received codewords, thereby greatly reducing the number of retransmissions and enhancing the system performance. But for non-stationary channels the bit error rate tends to change substantially leading to difficult and mistaken choices on the error code parameters. Let us make some considerations: when the channel bit error rate is low, the transmission is ordinarily flawless, no error correction capability is needed and due to the extra parity-check bits for error correction included in each transmission, a waste of bandwidth is presented; but if instead that, the channel is very noisy, the designed error-correcting capability may not be enough to cope with the errors. As a result, the number of retransmissions increases and hence reduces the system throughput. Error-control schemes based on algebraic block codes (e.g., Reed-Solomon (RS) and BCH codes) and hard-decision decoding provide a natural form of type-I Hybrid ARQ scheme, as exemplified in [Wicker, 1992]. The Cellular Digital Packet Data (CDPD) wireless data protocols are an application that use these strategies [Costello *et al.*, 1998]. Along the years several studies of type-I Hybrid ARQ schemes using either block or convolutional codes have been proposed and analysed [Yamamoto and Itoh, 1980; Drukarev and Costello, 1983].

Taking into account the previous non-stationary bit error rate case, a sort of adaptive Hybrid ARQ scheme can be the right choice. When the channel is smooth and steady, the system should behave just like a pure ARQ (i.e. with parity-check bits just to detect the errors), while when the channel becomes noisy, extra parity-check bits should be added to the codeword. This is the main concept of type-II Hybrid ARQ schemes, a system where the throughput performance is always at least as good as in a pure ARQ scheme.

Mandelbaum was the first to introduce the concept of parity retransmission for error correction [Mandelbaum, 1974]. It was however Sindhu in [Sindhu, 1977] the first to discuss a scheme that made use of previous transmitted packets. Sindhu's idea was that such packets can be stored and later combined with additional copies of the packet, creating a more reliable single packet than any of its constituent packets. An illustration of Sindhu's idea is shown in Figure 2.13. And with this, we enter in the second category of Hybrid ARQ schemes: the type-II. In a type-II Hybrid ARQ scheme, a message in its first transmission is coded with parity-check bits for error detection only, as in a pure ARQ scheme, forming a codeword. If the receiver detects an error in the respective codeword, it saves the erroneous word and requests a retransmission. Now, the retransmission is not necessarily the original codeword, as is in type-I scheme, but can be a block of parity-check bits formed based on the original codewords and an error-correcting code. After this block of parity-check bits has been received, it is used to correct the errors presented in the previous stored codeword. If it does not succeed, a second retransmission is requested. The second retransmission may

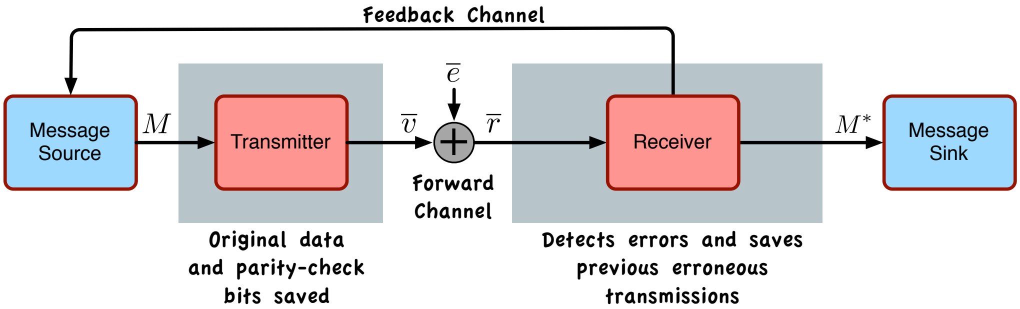



Figure 2.13: Sindhu's scheme for Hybrid ARQ architectures.

be either a repetition of the first and original codeword or another block of parity-check bits, depending on the retransmission strategy and the type of error-correcting code to be used.

There have been innumerable systems proposed that involve some form of packet combining since Sindhu's proposal in 1977. Chiti and Fantacci [Chiti and Fantacci, 2007] and also Caire and Tuninetti [Caire and Tuninetti, 2001] categorized the type-II Hybrid ARQ family in two branches: the hard-decision and the soft-output decision. Costello et al. took a wider and more clear perspective. In [Costello *et al.*, 1998] they defined two main categories to classify the type-II Hybrid ARQ family, commonly known and referred as packet combining systems: Code-Combining (CC) systems and Diversity-Combining (DC) systems.

### 2.4.1 Code Combining

The main idea behind Code-Combining (CC) systems is that the packets are concatenated to form noise-corrupted codewords from increasingly longer and of lower code rates. An early version of a code-combining system was the type-II Hybrid ARQ scheme invented by Lin and Yu [Lin and Ma, 1979; Lin and Yu, 1979, 1982]. In this scheme, two codes are used: one is a high rate code ($C_0$) which is designed for error detection only and the other is a half-rate invertible[13] code ($C_1$), which is designed for simultaneous error correction and error detection. In the first transmission, a codeword is transmitted with the message embedded. In the case of detected errors (by the highest rate code), the transmitter resends now the parity-check word (created with the help of the half rate invertible code). The receiver can recover the original codeword whether by successfully receiving the parity-check word and recovering the original message by its inversion afterwards, or also by using together the two words (the original and the parity-check) for error correction based on the half-rate code $C_1$. Therefore the retransmissions are alternate repetitions of the parity word and

[13] A code is said to be invertible if only knowing the parity-check bits of a codeword, the corresponding information bits can be uniquely determined by an inversion operation [Lin and Costello, 1983].

the original codeword. It should be pointed that the receiver needs to store alternately the original word and the parity block. The retransmission continues until the original message is recovered whether by inversion or by decoding. The work was analysed in a selective-repeat strategy with a receiver buffer of size $N$, where $N$ is the number of blocks that can be transmitted in a RTD period. The decoding complexity for the type-II hybrid ARQ scheme using alternate parity-data retransmission and a half-rate invertible code is only slightly greater than that of a corresponding type-I hybrid ARQ scheme with the same designed error-correcting capability. The extra circuits needed are an inversion circuit based on the half rate invertible code $C_1$, which is simply a linear sequential circuit and an error detection circuit based on the error-detecting code $C_0$ [Lin *et al.*, 1984].

The work presented by Lin and Yu in [Lin and Yu, 1982] is one of the most studied and was the starting point for many other schemes like [Miller, 1982; Comroe and Costello, 1984] with special attention to [Wang and Lin, 1983]. In this work, Wang and Lin considered again two codes, $C_0$ and $C_1$, but now $C_1$ is only used for error correction while the error detection for each transmission and retransmission is accomplished by $C_0$. He also presented an analysis for the throughput performance for any size of receiver buffer, with a rate 1/2 convolutional code using Viterbi decoding.

Inspired by these works, Chase [Chase, 1985] introduced a breakthrough scheme, and coined the term Code-Combining systems[14]. Briefly, CC represents a technique for combining the minimum number of repeated packets encoded with a code rate, let us say $R$, to obtain a lower rate, and thus more powerful, error-correcting code, capable of allowing communications when channel error rates are very high. Two critical features introduced by [Chase, 1985] should be pointed out: a maximum-likelihood decoding of $L$ noisy packets as a single low rate code of rate $R/L$; and a weighting of each packet by an estimate of its reliability, that can be viewed as a soft decision on a packet as opposed to the more conventional soft decisions used on a bit-by-bit basis. Figure 2.14 illustrates the sequence of mappings used in code combining systems. An information packet denoted by $I$, composed by $k$ bits, is mapped into an encoded packet with a code rate $R$. A parity-check sequence is treated as part of the information packet, so that the decoder can determine when to stop combining packets. Thus, the information packet is actually an error detection code. An estimate of the individual packet reliability is obtained by adding additional overhead bits. The name "Code-Combining" comes from the ability of the decoder by combining successive received codewords until the code rate is low enough to provide a successful decoding. Note that codewords can be combined in any order, so that even if several packets are lost, given

[14]The Chase work [Chase, 1985] is also often referred as "Chase Combining" in the wireless industry [Cheng, 2006].

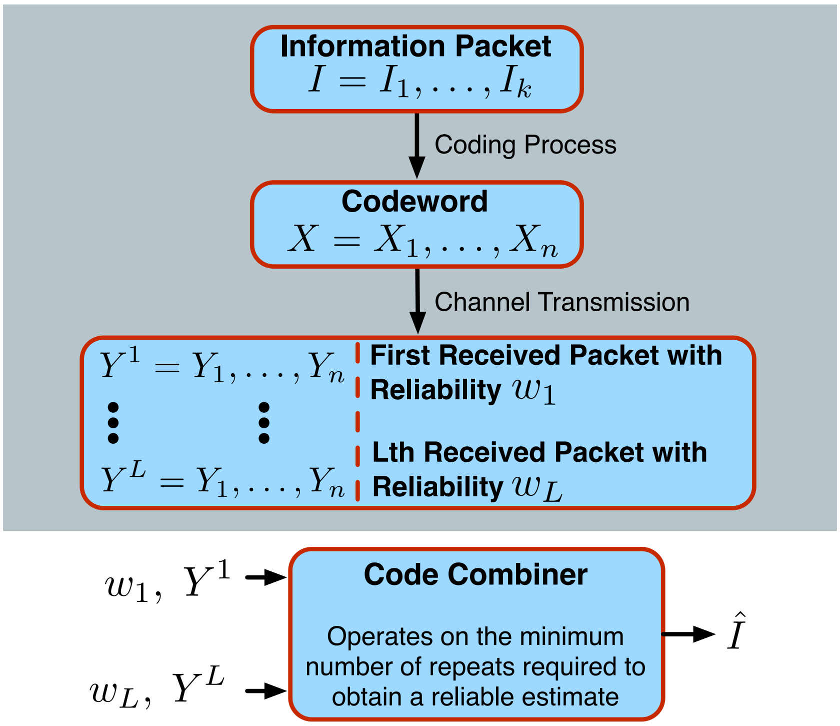



Figure 2.14: Mapping sequence of Code Combining systems.

a reliability estimation of 0, decoding is still achievable.

The CC technique treats the received packets $Y^1, Y^2, \ldots, Y^L$ as a code rate $R/L$. Each packet is assumed to have a reliability weight of $w_i$. As a basis, CC is designed to work in a very noisy environment, where conventional DC systems at that time [Sindhu, 1977], could easily break down. At that time, in DC systems (see section 2.4.2) a single error due to a massive fading effect, or equivalently, could cause an entire sum of $L$ received symbols to be incorrect, while with CC, a single error in a sum of $L$ symbols can typically be corrected by the more powerful rate $R/L$ code. CC represents an added dimension to diversity concepts which were at the time, limited to combining just individual symbols. It should be pointed out that [Chase, 1985] was one of the first works to propose error correction capabilities extracted from more than two packets. The schemes [Lin and Yu, 1982] and [Wang and Lin, 1983] allow just the combination of two packets (and not representing the same codeword, as explained before) and can be seen as a truncated CC system. However, their performance are always limited to the maximum error correction code capability chosen in the design.

Subsequent CC systems are exemplified by the works of Krishna and Morgera [Krishna and Morgera, 1987] or Morgera and Oduol [Morgera and Oduol, 1989]. Krishna and Morgera extended the work done in [Wang and Lin, 1983], by allowing combination of multiple retransmissions in a way that the receiver uses the same decoder for decoding the received information after every retransmission while the error correction capability of the code increases, thereby leading to an improved performance and minimum complexity for the overall system implementation. Thus, [Lin and Yu, 1982] can be considered as a special case of [Krishna and Morgera, 1987]. Morgera and Oduol [Morgera and Oduol, 1989] improved the performance of [Krishna and Morgera, 1987] by adding a soft-decision decoder.

Ideally, the code rate should be adjusted accordingly to the source and channel needs during the transmission of an information frame. Also, for practical purposes, switching between a set of encoders and decoders increases the complexity, making one encoder and one decoder, that can be modified without changing their basic structure, a step forward on that matter. As already pointed out, [Mandelbaum, 1974] introduced the concept of parity retransmission for error correction. In addition, [Mandelbaum, 1974] was the first to propose punctured codes for transmitting redundancy (the parity-check bits) in incremental steps by using RS codes. [Hagenauer, 1988] extended that notion and gave birth to a subclass usually named as Incremental Redundancy Automatic Repeat reQuest (IR-ARQ) scheme. In [Hagenauer, 1988] the concept of punctured convolutional codes is extended for the generation of a family of codes, known as Rate-Compatible Punctured Convolutional (RCPC), by adding a rate-compatibility restriction to the puncturing rule. The restriction implies that all the code bits of a high rate punctured code are used by the lower rate codes, i.e. the high rate codes are embedded into the lower rate codes of the family. If higher rate codes are not sufficiently powerful to decode channel errors, only supplemental bits which were previously punctured have to be transmitted in order to upgrade the code. An analytical and simulated analysis of the performance over an Additive White Gaussian Noise (AWGN), ideally interleaved Rayleigh and Rice fading channels as well as a modified soft-decision and Channel State Information (CSI) adaptive decoding process is presented in [Hagenauer, 1988].

Several other techniques were later developed, adding extra features and improvements to CC systems: Pursley and Sandberg [Pursley and Sandberg, 1991] proposed the use of RS codes in an incremental redundancy system for meteor-burst channels; Wicker provided a complete analysis and characterization of a punctured Maximum Distance Separable (MDS) type-II Hybrid ARQ scheme [Wicker and Bartz, 1994a]; in [Wicker and Bartz, 1994b] the approach was different, by adapting the highly efficient maximum likelihood decoding algorithm by R. R. Green [Posner, 1968], called "Green Machine", for use in type-I and type-II Hybrid ARQ schemes; Kallel [Kallel, 1990] focused on optimizing the work done by Chase in [Chase, 1985] and afterwards he evolved the work previously made in [Hagenauer, 1988]. Kallel [Kallel and Haccoun, 1990; Kallel and Leung, 1991; Kallel, 1992, 1994] developed several improvements on the IR-ARQ scheme subclass and ended up presenting the Complementary Punctured Convolutional (CPC) codes [Kallel, 1995]. The main drawback of IR-ARQ schemes is that additional incremental code bits sent for a packet received with errors (or a packet that is lost) are not in general self decodable. That is the decoder must rely on both, the initially transmitted packet as well as the additional incremental code bits for decoding. In situations where a packet can be lost or severely damaged, it is desirable to have a scheme where all additional information sent is self decodable. CPC codes adds that feature and the decoder does not have to rely on previously received transmissions for the same data packet for decoding, an approach that the authors called as a type-III Hybrid ARQ scheme.

Turbo codes [Berrou *et al.*, 1993; Berrou and Glavieux, 1996], were firstly used in an ARQ scheme in [Narayanan and Stuber, 1997], in which additional constituent encoders are added to obtain rates below that of a conventional turbo encoder. Zhang et al. [Zhang *et al.*, 1999] investigated the application of a type-II Hybrid ARQ scheme in a slotted Direct Sequence - Spread-Spectrum Multiple Access (DS-SSMA) packet radio system. Also in 1999, [Zhang and Kassam, 1999] proposed and analysed a Hybrid ARQ with selective combining scheme using RCPC codes for fading channels. In conjunction with the work previously made by [Hagenauer, 1988], it originated the Rate-Compatible Punctured Turbo (RCPT) codes, presented by [Rowitch and Milstein, 2000], to provide higher throughputs at the cost of higher code rates (e.g. 8/9). More recently, Low-Density Parity-Check (LDPC) codes have become an hot topic, whether combining with Hagenauer's work [Yazdani and Banihashemi, 2004] or extending IR-ARQ subclass family [Sesia *et al.*, 2004; Varnica *et al.*, 2005]. In [Visotsky *et al.*, 2005], the authors propose a Hybrid ARQ algorithm that exploits received packet reliability to improve system performance by averaging the magnitude of the log-likelihood ratios of the information bits as the packet reliability metric.

Many other works exists concerning the CC systems. A timeline table with the most important works done in the CC area is provided below in table 2.1.

### 2.4.2 Diversity Combining

Chung and Goldsmith [Chung and Goldsmith, 2001] have shown that the maximum spectral efficiency in ARQ schemes is possible to be obtained almost independently of the chosen degree of freedom. Sindhu's work [Sindhu, 1977] is considered the father and the creator of type-II Hybrid ARQ schemes [Costello *et al.*, 1998], as well as the pioneer of the second main category of Hybrid ARQ protocols family, the DC schemes. The conceptual idea behind the DC systems is that individual symbols from multiple, identical copies of a packet are combined in such way to create a single packet with a more reliable constituent symbols.

| Year | Author(s) | Contribution |
|---|---|---|
| 1982 | [Lin and Yu, 1982] | Pre-CC era; truncated CC system where the original codeword is combined with a parity codeword in an optimum way. |
| 1983 | [Wang and Lin, 1983] | Same scheme as [Lin and Yu, 1982] but with one code just for error-detecting and another to error-correcting. |
| 1985 | [Chase, 1985] | Proposed the CC scheme. |
| 1987 | [Krishna and Morgera, 1987] | Extended the work done in [Wang and Lin, 1983] by allowing combination of multiple retransmissions. |
| 1988 | [Hagenauer, 1988] | Proposed the RCPC codes. |
| 1989 | [Morgera and Oduol, 1989] | Extended the work of [Krishna and Morgera, 1987] by presenting a soft-decision receiver. |
| 1990 | [Kallel, 1990] | Optimized the performance of CC systems presented by [Chase, 1985]. |
| 1994 | [Wicker and Bartz, 1994b] | Adapted the "Green Machine" to type-I and type-II Hybrid ARQ schemes. |
| | [Wicker and Bartz, 1994a] | Provided an analysis and characterization of a punctured MDS type-II scheme. |
| 1995 | [Kallel, 1995] | Introduced the CPC codes for type-II Hybrid ARQ schemes. |
| 1997 | [Narayanan and Stuber, 1997] | Concept of Turbo Codes applied on Hybrid ARQ schemes. |
| 1999 | [Zhang *et al.*, 1999] | Combined a type-II Hybrid ARQ with a DS-SSMA packet radio system. |
| | [Zhang and Kassam, 1999] | Presented a selective combining scheme using RCPC codes for fading channels. |
| 2000 | [Rowitch and Milstein, 2000] | Proposed the RCPT codes. |
| 2004 | [Sesia *et al.*, 2004] | Combined LDPC codes with IR-ARQ schemes. |
| 2005 | [Visotsky *et al.*, 2005] | An algorithm is proposed for improving the performance using received packet reliability parameters. |

Table 2.1: Main contributions on CC schemes of type-II Hybrid ARQ.

These identical copies, are obtained by straightforward retransmissions.

In [Sindhu, 1977] erroneous copies of the same packet are XORed[15] to locate the errors in a combined copy. The correct packet can then be retrieved by an exhaustive search method. One of the advantages of this concept is that since only error detection is necessary, existing transmitters which only append frame check sequence can very well be used. Also, unlike a number of type-II schemes which employ soft-decisions [Chase, 1985; Hagenauer, 1988; Kallel, 1990; Narayanan and Stuber, 1997], hard-decisions were used here to maintain a low

[15] The logical operation exclusive disjunction [Jennings and Hartline, 2001], also called exclusive or, is a type of logical disjunction on two operands that results in a value of true if exactly one of the operands has a value of true. A simple way to state this is "one or the other but not both."

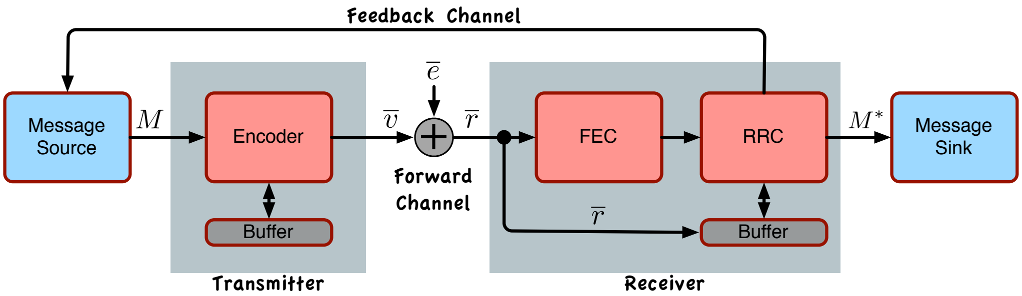



Figure 2.15: Sindhu's proposal for Hybrid ARQ architectures.

complexity approach at the receiver side.

Figure 2.15 exemplifies the structure adopted by Sindhu's work in 1977. Considering the transmitter side, a binary stream of data is packed into $k$ bit message blocks which are encoded into $n$ bit codewords from a linear code, i.e. $C(n, k)$. The codeword $v$ corresponding to a message block $M$ is sent over the forward channel and is also saved in a transmitter buffer. At the receiver side, the corresponding received $n$ bit block $\overline{r}_1$ is checked to see if it is FEC correctable into some codeword $\overline{v}$. If so, the received codeword is assumed to be recovered and decoded, so a positive ACK is sent to the transmitter. If not, $\overline{r}_1$ is stored in the buffer and a retransmission is requested. If this retransmission $\overline{r}_2$ is FEC correctable, that codeword is assumed to be $v$ and an ACK is sent, otherwise $\overline{r}_1$ and $\overline{r}_2$ are input to a Repetition Redundancy error Correction (RRC) procedure. The errors in $\overline{r}_1$ and $\overline{r}_2$ may overlap, causing this procedure to fail. In this event, another retransmission $\overline{r}_3$ is requested. If $\overline{r}_3$ is also not FEC correctable, RRC procedure is reapplied with $\overline{r}_1$, $\overline{r}_2$ and $\overline{r}_3$ as inputs. Notice that RRC procedure just combines two codewords at each time, despite having multiple identical codewords as input.

In 1979 Metzner [Metzner, 1979] improved the aforementioned work. Until then, the second transmission was just a retransmission of the first block, for further symbol by symbol combination. By maintaining simple encoding and decoding procedures Metzner was able to obtain significant performance improvements. This was achieved by using two approaches: one is to break up the $n$ bit codeword into small sub-blocks and use them as the data digits of a short rate 1/2 code, and since the code is short, likelihood ratio information can be used to make a combined hard-decision for the sub-block; these hard-decisions, now more reliable than with the first sending alone, are then used by the original $(n, k)$ code decoder; the second approach is to treat the first sending as the data digits of a systematic convolutional code of short constraint length. Note, in both approaches the codes are chosen so that all the information resided independently (i.e. should be extractable) in the second sending as well as the first sending, thereby introducing some constraints to the code selection. Also as in [Sindhu, 1977], just two codewords are combined at each time.

A few years later [Metzner and Chang, 1985], Metzner and Chang extended Metzner previous work by employing efficient selective-repeat ARQ strategies designed to cope with very noisy and variable conditions in the forward channel as well as in the return channel. Both of Metzner's works can be referred of sharing some principles presented in [Wang and Lin, 1983] which is classified as CC scheme, raising some doubts about their classification as DC architecture. However, both [Costello *et al.*, 1998] and [Chase, 1985] classified Metzner work in DC family since despite sharing some similarities, it is suggested a symbol by symbol diversity combining before the rate 1/2 decoding.

In 1985 Benelli's work in [Benelli, 1985], gave a steep boost on the development of DC schemes. According to Benelli, the method proposed by Sindhu offers transmission rates higher than those obtainable with convolutional ARQ system and can be used with acceptable performance even for low SNR conditions. Nevertheless, it has the disadvantage of requiring two FEC type correcting phases, enhancing with this the complexity of the system. What Benelli proposed was adding to Sindhu's idea, a reliability value associated to each demodulated symbol. Therefore, a reliability vector is associated to each word received, where the $i$th component of this vector denotes the reliability of the $i$th demodulated symbol. In addition, the reliability vector is updated every time a new repetition of the same word is received.

Figure 2.16 shows the general block diagram of the Benelli proposed scheme. At the transmitter each sequence of $k$ symbols coming from the source is codified by means of code $C$ of type $(n, k)$. This codeword is sent to the modulator and subsequently to the communication channel. The demodulator in the first place, carries out a hard estimate $r_{j,i}$, with $j$ representing the number of the transmission and $i$ the respective $i$th constituent symbol. The demodulator then associates a reliability symbol $w_{j,i}$ to each symbol $r_{j,i}$. The reliability value is tabulated accordingly to a threshold value and is subsequently summed with the reliability values of the $i$th value of previous transmissions, performing then a hard-decision for each symbol $v_j i$, thereby achieving a kind of weighted average decision of the values received on each symbol. Thus, the decoder first determines whether or not $v_j$ is a codeword. If it is, the codeword is accepted as the transmitted codeword. In the opposite case, the decoder analyses $r_j$. If $r_j$ is a codeword, then it is accepted as that, otherwise another retransmission is needed and requested.

This scheme requires a larger memory at the receiver when compared to conventional ARQ schemes, but contrarily to Sindhu's method, this scheme does not require any correction

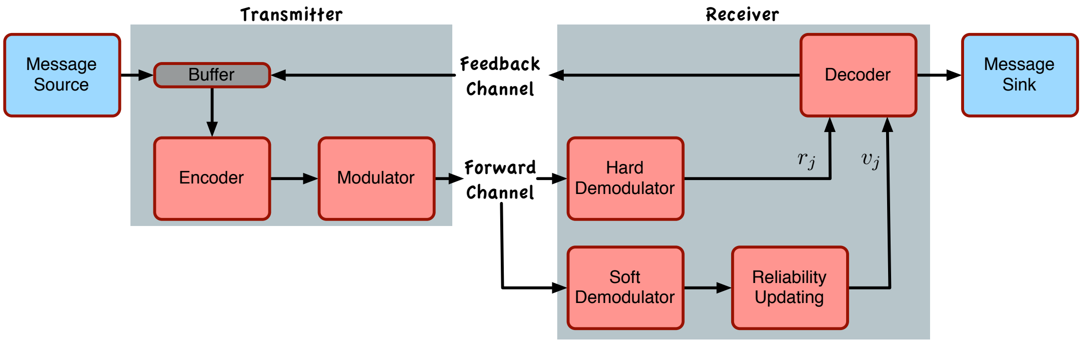



Figure 2.16: Block diagram of Benelli's proposal for Hybrid ARQ.

procedure. In fact, the correction of erroneous symbols in many cases takes place automatically through the updating of the reliability symbols. In this approach, the probability that the $i$th symbol of the vector $v_j$ is an error depends on the number $j$ of transmissions.

Errors in symbols with low reliability are in many cases much more probable than errors in symbols with high reliability and therefore, errors in symbols with low reliability can be easily erased in an automatic way through the reliability updating process. Errors in symbols with high reliability are more difficult to erase, due to their greater weight in the updating process. That is why, Chase [Chase, 1985], the creator of the CC schemes, highlight the fact in 1985 that these approaches (i.e. DC schemes) when in the presence of strong jamming effects are destiny to fail since combining severe constant noisy packets is a limited solution.

The basis for a soft-decision decoder with the reliability update procedure was introduced in [Benelli, 1985]. As a result, one year after, Benelli presented the first scheme adopting a soft-decision decoder in DC systems [Benelli, 1986]. The scheme is very similar, where the main change relies on the reliability value being quantised in $2n_L$ levels instead of just four levels using an input threshold value as in [Benelli, 1985]. A scheme adopting a soft decoder with the modulation operation integrated was proposed in [Benelli, 1987] in order to achieve a higher performance. In fact, that integration, if suitably executed, permits to increase the Euclidean distance[16] among the signals each time a new transmission of a codeword is performed. The error probability at the input of the channel decoder is then decreased.

Adachi and Ito [Adachi and Ito, 1986] made some improvements on [Benelli, 1985] by adopting a frequency demodulator for a digital frequency modulated signal where the receiver demodulator output associated with each symbol is weighted and combined every time a new retransmission is made. Thus, a considerable reduction in the average number of transmissions is achieved. Later on, in [Adachi *et al.*, 1989], the mentioned work was

[16]In mathematics, the Euclidean distance between points $p$ and $q$ is the length of the line segment connecting them.

featured with a finite number of retransmissions and is employed considering mobile radio channels with diversity.

Several works have been proposed for the DC approach: in [Wicker, 1991] an adaptive rate error-control system based on majority-logic decoding provides a significant improvement in error protection through the incorporation of DC techniques; Harvey and Wicker [Harvey and Wicker, 1994] presents several packet combining schemes for use in conjunction with the Viterbi decoder over stationary and time varying channels; Chakraborty et al. [Chakraborty *et al.*, 1998] presented a ready to implement algorithm, based on a DC scheme, and an approximate analytical description of it in a binary symmetric channel; the same authors in [Chakraborty *et al.*, 1999] expand their work studying the dependence of efficiency of type-II Hybrid ARQ schemes on packet size and propose a scheme for throughput optimization in time varying channels, by dynamically adjusting the packet size. More recently, Chiti and Fantacci [Chiti and Fantacci, 2007] introduced a modified Hybrid ARQ scheme based on the use of the turbo codes' error correction capabilities to deal with reliable end-to-end data communications in satellite networks, whose applications have to cope with accurate Quality-of-Service (QoS) requirements. According to his work, each packet is turbo coded, and packet retransmissions are lowered by keeping track of previous erroneous packet decoding results. Moreover, two kinds of these techniques are proposed and applied: the new input combined with a previous input (soft-input) or the new input combined with the output of previous erroneous frame decoding procedure (soft-output).

Dinis et al. [Dinis and Gusmão, 1998; Gusmão *et al.*, 1999; Dinis *et al.*, 2008, 2009a] worked in Hybrid ARQ schemes with DC approach. Soft packet combining techniques were proposed in [Dinis and Gusmão, 1998] which can be regarded as Hybrid ARQ schemes based on repetition codes with DC approach where the encoding/decoding complexity is very low. A study and analysis on the subject of adaptive error control schemes based on punctured RS codes, for an Asynchronous Transfer Mode (ATM) compatible mobile broadband system was presented in [Gusmão *et al.*, 1999]. A low complexity DC scheme employing SC-FDE techniques was presented in [Dinis *et al.*, 2008] for the uplink of wireless systems (more details about this approach are given in chapter 3). An Ultra Wide Band (UWB) system employing SC-FDE techniques in the presence of strong interference signals was presented in [Dinis *et al.*, 2009a], where DC systems are addressed.

A timeline table with the most important works done in the DC area is provided in 2.2. In general DC systems are suboptimal with respect to CC systems, but usually are simpler to implement, even considering soft-decision architectures. The usage of repetition codes with soft-decision can be regarded as a low-complexity version of type-II Hybrid ARQ family, since the decoder is much simpler.

| Year | Author(s) | Contribution |
|---|---|---|
| 1977 | [Sindhu, 1977] | Proposed a scheme where a erroneous packet is combined with its previous transmission to improve the decoded process. |
| 1985 | [Benelli, 1985] | Improved [Sindhu, 1977] by combining multiple identical copies of the same packet symbol-by-symbol. |
| 1986 | [Benelli, 1986] | Presented the first scheme adopting a soft-decision decoder in DC systems. |
| 1987 | [Benelli, 1987] | Added integration of the modulation operation in order to achieve a higher performance. |
| 1989 | [Adachi *et al.*, 1989] | Improvements are presented concerning mobile radio channels and frequency modulators for finite number of retransmissions. |
| 1991 | [Wicker, 1991] | An adaptive rate error control system with DC techniques is presented. |
| 1994 | [Harvey and Wicker, 1994] | Several packet combining schemes are introduced with the Viterbi decoder over stationary and time varying channels. |
| 1998 | [Chakraborty *et al.*, 1998] | The binary symmetric channel for DC schemes is studied and an algorithm is suggested. |
| 1999 | [Chakraborty *et al.*, 1999] | A study about the dependence of efficiency of type-II Hybrid ARQ schemes on packet size is presented. |
| 2008 | [Dinis *et al.*, 2008] | A low complexity DC scheme employing SC-FDE techniques for the uplink of wireless systems is presented. |

Table 2.2: Main contributions on DC schemes of type-II Hybrid ARQ.

In the last years, the main research and development focus in the Hybrid ARQ topic has been performance evaluation (either analytical or simulated). To emphasize the development done in this topic a short overview about it is given below.

### 2.4.3 Performance Studies

The current research in the Hybrid ARQ topic is mainly focused on performance evaluations. Generally, this evaluation consists on the use of analytical models or Monte Carlo simulations studies. Caire and Tuninetti [Caire and Tuninetti, 2001] applied the renewal-reward theorem [Wolff, 1989] to study the system performance in terms of throughput (total bit per second per hertz) and average delay for three idealized protocols: a coded version of Aloha, a repetition scheme with maximum ratio packet combining, and an incremental redundancy scheme with general coding. Cheng [Cheng, 2006] developed a unified performance metric and a detailed analysis for Hybrid ARQ schemes based on IR-ARQ and CC architectures. Huang et al. [Huang *et al.*, 2005] analysed the optimal scheduling policy for an Hybrid ARQ downlink, which maximizes the link's throughput (minimizes the number of retransmissions) for Poisson arrival processes. Boujemâa et al. [Boujemâa *et al.*, 2008] analysed the maximum throughput performance of Hybrid ARQ schemes with DC and CC using Direct Sequence - Carrier Sense Multiple Access (DS-CSMA) with a Rake receiver. Boujemâa also analysed [Boujemâa, 2009] the delay of cooperative truncated Hybrid ARQ with opportunistic relaying for Poisson sources, where relay nodes with better SNR may handle retransmissions. Choi et al. [Choi *et al.*, 2005] and Le Duc et al. [Le Duc *et al.*, 2009b], [Le Duc *et al.*, 2009a] considered the Hybrid ARQ performance for the group of fragments that compose an Internet Protocol (IP) packet. They proposed IP-Medium Access Control (MAC) cross-layer optimizations and a model for the IP packet error rate and transmission delay. Luo et al. [Luo *et al.*, 2005] analysed the Service Data Unit (SDU) delivery delay of selective-repeat ARQ as a function of the SDU size and the channel coding scheme. Badia et al. [Badia *et al.*, 2006], [Badia *et al.*, 2008], analysed the selective-repeat ARQ packet delay statistics using Markov channel models and assuming a constant RTT, which originates periodic packet retransmissions. Paper [Badia *et al.*, 2008] extends [Badia *et al.*, 2006] (uncoded systems), analysing the performance of selective-repeat ARQ with a truncated type II H-ARQ technique based on RS erasure codes.

Recent interest in Hybrid ARQ schemes comes from the quest for reliable and efficient transmission under fluctuating conditions in future wireless networks. Hybrid ARQ techniques are currently used in 3GPPs High-Speed Downlink Packet Access (HSDPA), High-Speed Uplink Packet Access (HSUPA), and Long Term Evolution (LTE) [Lescuyer and Lucidarme, 2008; Dahlman *et al.*, 2011].

# CHAPTER 3

## An Hybrid-ARQ SC-FDE System: an efficient way of coping with lost packets due to channel effects

As mentioned in the previous chapter, Hybrid Automatic Repeat reQuest (Hybrid ARQ) schemes have been proposed, where the basic idea behind these techniques is to retain the signal associated to an erroneous packet and to ask for additional redundancy. In typical Code-Combining (CC) schemes, the performance is limited by the basic code that was adopted, so if a more powerful basic code is required, a more complex receiver will be also needed. As an alternative, a recent effort was presented in [Dinis *et al.*, 2008] where a soft packet combining technique was proposed. In this proposal, packets associated to different transmission attempts are combined in a soft way, allowing improved performances [Dinis *et al.*, 2008]. As a result, these techniques can be regarded as a Diversity-Combining (DC) Hybrid ARQ scheme, since it can be viewed as repetition codes with soft decision. Moreover, the encoding/decoding complexity is low, specially when compared with CC approaches. In addition, this scheme considered the use of Single Carrier with Frequency Division Equalizer (SC-FDE) schemes, due to its weight in the wireless systems, since it is generally accepted as one of the best candidates for the uplink of future broadband wireless systems [Gusmão *et al.*, 2000; Falconer *et al.*, 2002]. The paper [Dinis *et al.*, 2008] was the starting point for the work presented in this thesis regarding Hybrid ARQ schemes.

This chapter presents a thorough study about the DC Hybrid ARQ technique employing SC-FDE that was presented in [Dinis *et al.*, 2008] for the uplink of wireless systems. A new Hybrid ARQ retransmission strategy for SC-FDE is proposed and analysed where the transmitter employs a shifted packet approach to improve the Hybrid ARQ performance. This strategy is particularly suited when the channel presents severely time-dispersive characteristics. Considering a Time Division Multiple Access (TDMA) scheme, it is described a new analytical model valid for a generic Hybrid ARQ TDMA scheme. In addition, it is proposed an accurate model concerning the queue-size and the packet delay analysis of a TDMA system with an Hybrid ARQ Type-II technique, for a generic packet arrival distribution. Numerical simulation results are presented in order to validate the system characterization. The analytical model is also validated using ns-2 [Information Sciences Institute, 2007] simulations, showing its accuracy. Results concerning goodput and packet delay using two traffic distributions (Poisson and Geometric) and taking into account the transmission technique simulations, are properly addressed. As expected, H-ARQ reduces significantly its packet delay compared to the conventional ARQ technique.

Although the analytical model could be applied to any transmission technique, in this chapter it is focused its application in SC-FDE. This model can be regarded as a tool for system analysis and for system configuration (e.g. to determine the minimum $E_b/N_0$ value that satisfies an average delay bound given a traffic model bound). The work presented in this chapter, was accepted for publication in IEEE Transaction on Communications [Pereira *et al.*, 2012] and was partially presented in [Pereira *et al.*, 2010b,d]. The model here introduced was used as a tool in subsequent works [Ganhão *et al.*, 2010b,a, 2011a].

## 3.1 Hybrid-ARQ System Design

A SC-FDE system is considered in this chapter. As with other Frequency Domain Equalization (FDE) systems, data is transmitted in fixed size blocks and the time-domain block associated to a given user (i.e. the corresponding packet) is $\{s_n; n = 0, 1, \ldots, N-1\}$, where $s_n$ is a data symbol selected from a given constellation. To each time-domain block $\{s_n; n = 0, 1, \ldots, N-1\}$ it is associated a frequency-domain block $\{S_k; k = 0, 1, \ldots, N-1\}$ which is the Decision Feedback Equalization (DFE) of $\{s_n; n = 0, 1, \ldots, N-1\}$. As with other block transmission techniques, a suitable cyclic prefix is added to each time-domain block.

When errors are detected in a given packet the receiver asks for its retransmission, but the signal associated to each transmission attempt is stored as shown in Figure 3.1. The packet associated to the $r$th attempt to transmit $\{s_n; n = 0, 1, \ldots, N-1\}$ is $\{s_n^{(r)}; n = 0, 1, \ldots, N-1\}$.

### 3.1.1 Receiver Characterization

Let us assume that $l$ versions of the packet are available (i.e. there were $l$ transmission attempts[1]) and the receiver combines them in an efficient way. The received signal associated

[1] $r = 1, 2, ..., l$

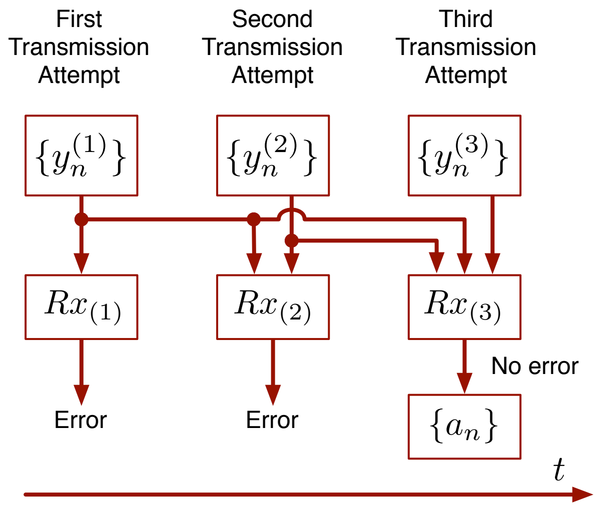



Figure 3.1: Receiver process.

to the $r$th transmission attempt is sampled and the cyclic prefix is removed, leading to the time-domain block $\{y_n^{(r)}; n = 0, 1, \ldots, N-1\}$. If the cyclic prefix is longer than the overall channel impulse response then the corresponding frequency-domain block is $\{Y_k^{(r)}; k = 0, 1, \ldots, N-1\}$, where

$$Y_k^{(r)} = S_k^{(r)} H_k^{(r)} + N_k^{(r)}, \quad r = 1, 2, \ldots, l, \tag{3.1}$$

with $N_k^{(r)}$ denoting the channel noise. $\{S_k^{(r)}; k = 0, 1, \ldots, N-1\}$ is the Discrete Fourier Transform (DFT) of $\{s_n^{(r)}; n = 0, 1, \ldots, N-1\}$ and $H_k^{(r)}$ is the overall channel frequency response for the $r$th transmission attempt.

The receiver, which is based on the Iteractive Block-Decision Feedback Equalization (IB-DFE) receiver proposed in [Benvenuto *et al.*, 2010], is depicted in Figures 3.2 and 3.3. An iterative frequency-domain receiver is assumed where, for a given iteration $i$, the frequency-domain samples at the output are given by

$$\tilde{S}_k^{(i)} = \sum_{r=1}^{l} F_k^{(r,i)} Y_k^{(r)} - B_k^{(i)} \overline{S}_k^{(i-1)}, \tag{3.2}$$

where $\{F_k^{(r,i)}; k = 0, 1, \ldots, N-1\}$ $(r = 1, 2, \ldots, l)$ and $\{B_k^{(i)}; k = 0, 1, \ldots, N-1\}$ can be regarded as the feedforward and the feedback coefficients, respectively. $\{\overline{S}_k^{(i-1)}; k = 0, 1, \ldots, N-1\}$ denotes the DFT of the average data estimates $\{\overline{s}_n^{(i-1)}; n = 0, 1, \ldots, N-1\}$, where $\overline{s}_n$ denotes the average symbol values conditioned to the FDE output. For Quadrature

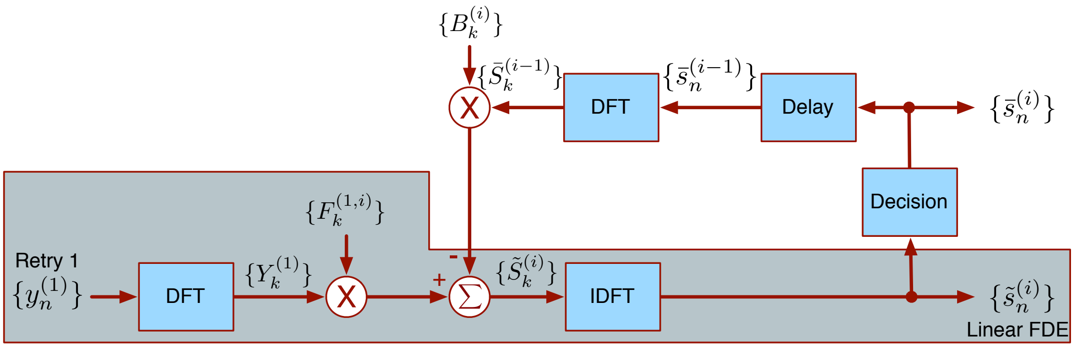



Figure 3.2: Structure of the iterative receiver for the first packet reception (the shadowed part corresponds to a linear FDE).

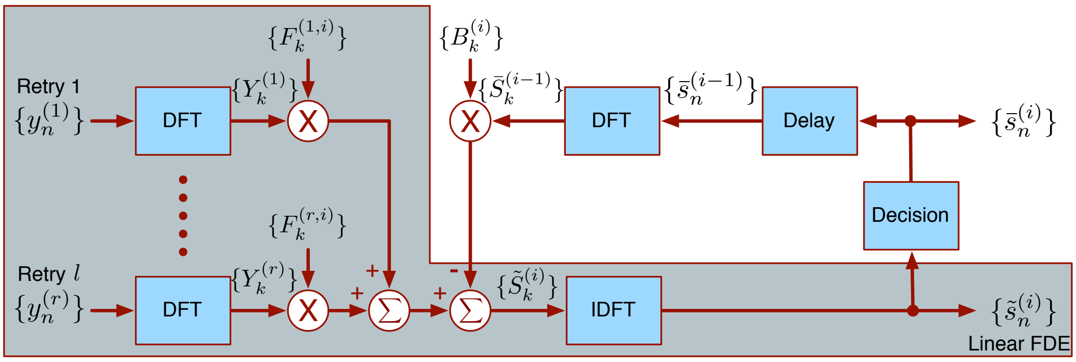



Figure 3.3: Structure of the iterative receiver for the $l$ packet reception (the shadowed part corresponds to a linear FDE).

Phase Shift Keying (QPSK) constellations it can be shown that (see [Gusmão *et al.*, 2007]) these average values are given by

$$\bar{s}_n^{(i)} = \tanh\left(\frac{L_n^{I(i)}}{2}\right) + j\tanh\left(\frac{L_n^{Q(i)}}{2}\right) \tag{3.3}$$

(without loss of generality, it is assumed that $|s_n|^2 = 2$, i.e. $s_n = \pm 1 \pm j$), with

$$L_n^{I(i)} = \frac{2}{\sigma_{eq}^2}\mathrm{Re}\{\tilde{s}_n^{(i)}\} \tag{3.4}$$

and

$$L_n^{Q(i)} = \frac{2}{\sigma_{eq}^2}\mathrm{Im}\{\tilde{s}_n^{(i)}\} \tag{3.5}$$

denoting the LogLikelihood Ratioss (LLRs) of the “in-phase bit” and the “quadrature bit”, associated to $s_n$, respectively, and $\{\tilde{s}_n^{(i)}; n = 0, 1, \ldots, N-1\}$ = Inverse DFT $\{\tilde{S}_k^{(i)}; k = 0, 1, \ldots, N-1\}$. The variance $\sigma_{eq}^2$ is given by

$$\sigma_{eq}^2 = \frac{1}{2} E[|s_n - \tilde{s}_n^{(i)}|^2] \approx \frac{1}{2N} \sum_{n=0}^{N-1} |\hat{s}_n^{(i)} - \tilde{s}_n^{(i)}|^2, \tag{3.6}$$

where $\hat{s}_n^{(i)} = \pm 1 \pm j$ are the hard-decisions associated to $\tilde{s}_n^{(i)}$.

The feedforward and feedback coefficients are selected to minimize the "signal-to-noise plus interference ratio", for a given iteration. It is assumed that $\hat{S}_k^{(i)} \approx \rho^{(i)} S_k + \Delta_k^{(i)}$ ($\{\hat{S}_k^{(i)}; k = 0, 1, \ldots, N-1\}$ denotes the frequency-domain block associated to the hard decisions, i.e. the DFT of the block $\{\hat{s}_n^{(i)}; n = 0, 1, \ldots, N-1\}$). The correlation coefficient $\rho^{(i)}$ is given by

$$\rho^{(i)} = \frac{1}{2N} \sum_{n=0}^{N-1} (|\mathrm{Re}\{\overline{s}_n^{(i)}\}| + |\mathrm{Im}\{\overline{s}_n^{(i)}\}|) \tag{3.7}$$

and $\Delta_k^{(i)}$ is a "noise component" associated to decision errors that is uncorrelated with the data samples and with $E[|\Delta_k^{(i)}|^2] = (1 - |\rho^{(i)}|^2) E[|S_k|^2]$ [Dinis *et al.*, 2003]. To simplify the analysis it is also assumed that $\overline{S}_k^{(i)} \approx \rho^{(i)} \hat{S}_k^{(i)}$ [Gusmão *et al.*, 2007].

The optimum feedforward coefficients for a given iteration, can be written as (see [Dinis *et al.*, 2008])

$$F_k^{(r,i)} = \frac{\mathcal{K}^{(i)} H_k^{(r)*}}{\alpha + (1 - (\rho^{(i-1)})^2) \sum_{r'=1}^{l} |H_k^{(r')}|^2}, r = 1, 2, \ldots, l,$$

where

$$\alpha = \frac{E[|N_k^{(r)}|^2]}{E[|S_k|^2]} \tag{3.8}$$

(i.e. $\alpha$ is the inverse of the Signal-to-Noise Ratio (SNR)), and $\mathcal{K}^{(i)}$ is selected so that $\frac{1}{N} \sum_{k=0}^{N-1} \sum_{r=1}^{l} F_k^{(r,i)} H_k^{(r)} = 1$.

The optimum feedback coefficients that minimize the signal-to-noise plus interference ratio are given by (see [Dinis *et al.*, 2008])

$$B_k^{(i)} = \sum_{r=1}^{l} F_k^{(r,i)} H_k^{(r)} - 1. \tag{3.9}$$

For the first iteration ($i = 1$) no information previously acquired is presented about $S_k$ and the correlation coefficient $\rho^{(0)}$ is zero. This means that $B_k^{(1)} = 0$ and

$$F_k^{(r,1)} = \frac{H_k^{(r)*}}{\alpha + \sum_{r'=1}^{l} |H_k^{(r')}|^2}, \quad r = 1, 2, \ldots, l, \tag{3.10}$$

corresponding to the optimum coefficients for a linear FDE [Gusmão *et al.*, 2003].

The feedforward coefficients, can be written as

$$F_k^{(r,i)} = H_k^{(r)*} C_k^{(i)}, r = 1, 2, \ldots, l \tag{3.11}$$

with

$$C_k^{(i)} = \frac{1/\mathcal{K}^{(i)}}{\alpha + (1 - (\rho^{(i-1)})^2) \sum_{r'=1}^{l} |H_k^{(r')}|^2}. \tag{3.12}$$

This means that the bank of feedforward filters can be replaced by a bank of matched filters, corresponding to an ideal Maximum Ratio Combining (MRC), followed by a single feedforward filter characterized by the set of coefficients $\{C_k^{(i)}; k = 0, 1, \ldots, N-1\}$.

This IB-DFE receiver can be implemented in two different ways, depending where the channel decoding output is employed. If it is employed within the feedback loop the IB-DFE can be regarded as a turbo equalizer implemented in the frequency domain and therefore it can be denoted as "Turbo IB-DFE" [Gusmão *et al.*, 2007; Dinis *et al.*, 2009b]. If the channel decoding output is not employed within the feedback loop, this receiver can be regarded as a low complexity turbo equalizer implemented in the frequency domain. Since this is not a true "turbo" scheme, hereafter this "Conventional IB-DFE" will be referred as not containing turbo equalization. Thus, the main difference between "Conventional IB-DFE" and "Turbo IB-DFE" is in the decision device: in the first case the decision device is a symbol-by-symbol soft-decision; for the Turbo IB-DFE a Soft-In, Soft-Out channel decoder is employed in the feedback loop. The Soft-In, Soft-Out block can be implemented as defined in [Vucetic and Yuan, 2002] and provides the LLRs of both the "information bits" and the "coded bits". The input of the Soft-In, Soft-Out block are LLRs of the "coded bits" at the multi-packet receiver, and the feed-forward coefficients are still obtained from (3.11).

For uncoded scenarios it only makes sense to employ conventional IB-DFE schemes. In coded scenarios we could still employ a "Conventional IB-DFE" and perform the channel decoding procedure after all the iterations of the IB-DFE. However, since the gains associated to the iterations are very low at low to moderate SNR values, it is preferable to involve the channel decoder in the feedback loop, i.e. to use the "Turbo IB-DFE".

### 3.1.2 Coping With Static Channel Conditions

In conventional Automatic Repeat reQuest (ARQ) systems when the transmitter receives a retransmission request (e.g. a Negative Acknowledgement (NACK) or a suitable timeout), it simply retransmits the erroneous packet. Ideally, it would be better to have different channel conditions not to repeat the failure, i.e. the channel conditions should be uncorrelated from retransmission to retransmission attempt. This condition is denoted as Uncorrelated Channel (UC). Typically, to have UC conditions a change in the working band for each retransmission attempt is needed (something not practical), and/or is required an interval between retransmissions large enough to allow significant channel variations due to Doppler effects (which might lead to significant delays in the ARQ case) and/or significant user movement (something very particular).

When these reasons do not happen the channel remains constant or almost constant during the different retransmission attempts, and this condition is denoted as Equal Channel (EC).

However, for the severely time-dispersive channels considered in this chapter, the transmitter can employ a strategy to reduce the apparent correlation between the equivalent channel for different retransmission attempts. The basic idea is that the correlation between the channel frequency response for the $k$th subcarrier[2] and the $k'$th subcarrier ($H_k$ and $H_{k'}$, respectively) can be very low if $k$ and $k'$ are not adjacent nor even very close. To take advantage of this, it could be assumed that the frequency domain block associated to the $r$th retransmission of a given packet, $\{S_k^{(r)}; k = 0, 1, \ldots, N-1\}$, is an interleaved version of $\{S_k; k = 0, 1, \ldots, N-1\}$. Since this is formally equivalent to assume that $\{H_k^{(r)}; k = 0, 1, \ldots, N-1\}$ is an interleaved version of $\{H_k; k = 0, 1, \ldots, N-1\}$, the interference correlations for each frequency can be very small. However, to avoid transmitting signals with very large envelope fluctuations, it is better to assume that $\{S_k^{(r)} = S_{k+\zeta_r}; k = 0, 1, \ldots, N-1\}$, i.e. it is a cyclic-shifted version of $\{S_k; k = 0, 1, \ldots, N-1\}$, with a shift $\zeta_r$, which means that the corresponding time-domain block is $\{S_n^{(r)} = S_n \exp(j2\pi\zeta_r n/N); n = 0, 1, \ldots, N-1\}$. Clearly, this is formally equivalent to have $S_k^{(r)} = S_k$ and $H_k^{(r)}$ a cyclic-shifted version of $H_k^{(1)}$, with a shift $-\zeta_r$ (a similar reasoning is behind the diversity techniques of [Gore *et al.*, 2002; Gidlund and Ahag, 2004; Bauch, 2006]) as shown in Figure 3.4. In general, the larger $\zeta_r$ the smaller the correlation between $H_k^{(r)}$ and $H_k^{(1)}$, provided that $\zeta_r < N/2$ (this is also effective against narrow-band interference)[3]. In this chapter it is assumed that the different $\zeta_r$ are the odd multiples of $N/2$, $N/4$, $N/8$, etc., i.e.

$$\begin{array}{llll} \zeta_2 = N/2, & \zeta_4 = 3N/4, & \zeta_6 = 3N/8, & \zeta_8 = 7N/8 \\ \zeta_3 = N/4, & \zeta_5 = N/8, & \zeta_7 = 5N/8, & \ldots \end{array}$$

An illustration of the concept is presented in Figure 3.5. Note that, if it was known *a priori*

[2]Since it is considered frequency-domain receivers, the “sub-carrier” concept is still valid for single-carrier schemes. Naturally, these sub-carriers are assumed to be in the in-band region.

[3]$\zeta_r > N/2$ is formally equivalent to have a negative value of zeta, i.e. a left shift instead of a right shift. Since the purpose of the shift is to reduce frequency correlations in the equivalent channel for each retransmission, this does not pose any problem.

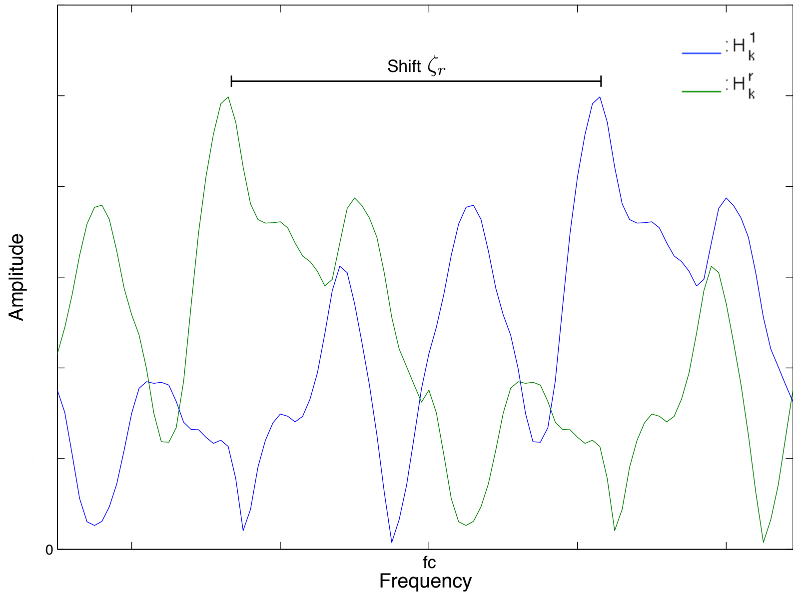



Figure 3.4: Example of a possible channel response for two different transmissions employing SP strategy.

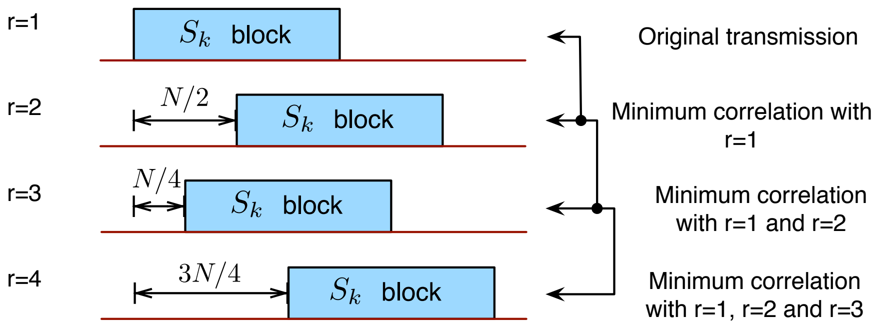



Figure 3.5: Concept illustration of the SP strategy.

that it would take for example three transmissions to successfully receive a given packet, the two ideal values for $\zeta_r$ (the ones that would achieve lower correlation with the original transmission) would be $N/3$ and $2N/3$ for $r = 2$ and 3 respectively. This retransmission strategy has the additional advantage of having the envelope fluctuations on the time-domain signal associated to $\{s_n^{(r)}; n = 0, 1, \ldots, N-1\}$ similar to the ones associated to $\{s_n; n = 0, 1, \ldots, N-1\}$ (in fact, the transmitted signal is still QPSK $r = 2, 3$ and 4), something not critical when it is employed Orthogonal Frequency Division Multiplexing (OFDM) signals as in [Gidlund and Ahag, 2004; Bauch, 2006]. This retransmission strategy will be denoted as SP.

### 3.1.3 Receiver Performance

A set of performance results concerning the proposed packet combining ARQ technique for SC-FDE schemes, obtained using Monte Carlo simulations is presented in this section. A table containing all the information regarding the simulations' setup used is illustrated in Table 3.1. Each time-domain block has $N = 256$ data symbols selected from a QPSK constellation under a Gray mapping rule, totalling 512 bits per data block. Therefore, each data packet has 64 bytes. A severely time-dispersive multipath channel considering the power delay profile type C for HIgh PERformance Local Area Network (HIPERLAN) with uncorrelated Rayleigh-distributed fading on different paths was assumed. Perfect synchronization and channel estimation conditions at the receiver are assumed. In terms of the characteristics of the channel it is considered the two conditions referred above and also the new retransmission strategy:

- UC, when in the presence of uncorrelated fading for different retransmission attempts.
- EC, when in the presence of the same channel for different retransmission attempts.
- SP, when in the presence of the same channel for different retransmission attempts (as with the EC case), but the transmitter performs cyclic shifts in the frequency-domain as described before. It can be seen as a EC channel condition where it is employed the SP strategy. This retransmission strategy will be denoted as SP.

Figures 3.6 and 3.7 show the average Packet Error Rate (PER) of the receiver when in the presence of $l$ transmission attempts. It is considered both the linear receiver (single iteration) and the iterative receiver (with 4 iterations), the UC and EC conditions as well as the SP retransmission strategy. For the sake of comparisons, it is also included the PER performance when not having the packet combining (in this case assuming only EC conditions). The results are expressed as function of $E_b/N_0$, where $N_0$ is the one-sided power spectral density of the noise and $E_b$ is the average bit energy associated to a given packet transmission. In both figures, the SP strategy outperforms the EC condition as expected, with higher gains for the linear receiver. The reason for this behaviour is that the linear receiver has a high sensitivity to deep in-band frequency notches, and the SP strategy reduces the probability of these notches when the signals associated to different retransmissions are combined; the iterative receiver can be regarded as a turbo FDE, which has significant robustness against these frequency notches. These figures show that these packet combining techniques allow significant PER reduction as it is increased the number of transmission attempts. This is not surprising, since the total transmitted power grows with the number

| | Figures | | | | | | | | | |
|---|---|---|---|---|---|---|---|---|---|---|
| | **3.6** | **3.7** | **3.8** | **3.9** | **3.10** | **3.11** | **3.12** | **3.13** | **3.14** | **3.15** |
| **Data Symbols** | 256 | | | | | | | | | |
| **Modulation** | Quadrature Phase Shift Keying | | | | | | | | | |
| **Block (Bits)** | 512 | | | | | | | | | |
| **Packet (Bytes)** | 64 | | | | | | | | | |
| **Coding** | No | | | | | | | | | |
| **Turbo equalization** | No | | | | | | | | | |
| **Iterations** | 1 | 4 | 1 and 4 | 1 and 4 | 4 | 4 | 4 | 4 | 4 | 4 |
| **Channel Conditions** | UC, EC and EC+SP | | | | EC+SP | | | | | |
| **No PC** | Yes | Yes | No | No | Yes | Yes | Yes | Yes | No | Yes |
| $1/\lambda$ | — | — | saturation | 20 | 3 loads | 3 loads | variable | 3 loads | variable | variable |
| **Type of Traffic** | — | — | Poisson | Poisson | Poisson | Poisson | Poisson | Geometric | Poisson | Poisson |
| **R** | 4 | 4 | 5 | 5 | 5 | 5 | 5 | 5 | 5 | 5 |
| **Number of Stations** | — | — | 8 | 8 | 8 | 8 | 8 | 8 | 8 | 8 |

Table 3.1: Simulations setups.

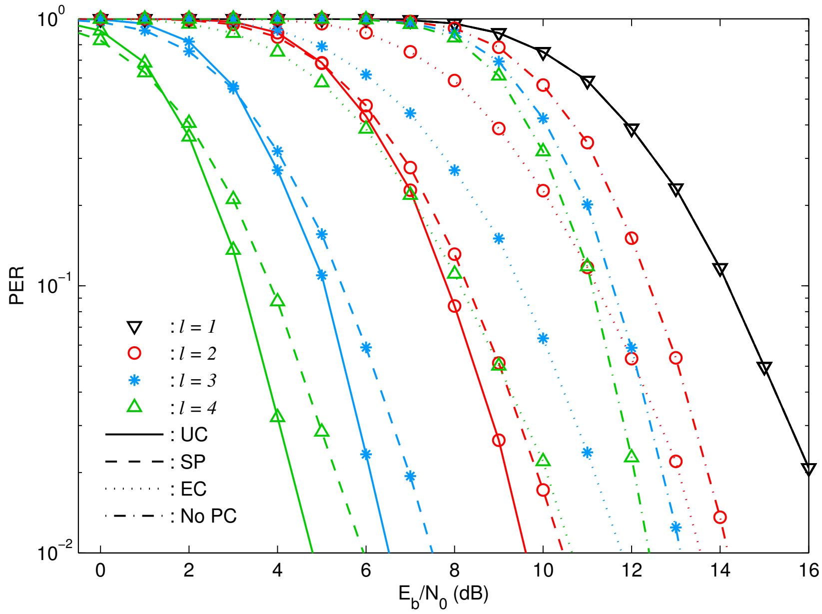



Figure 3.6: PER performance for the linear FDE receiver.

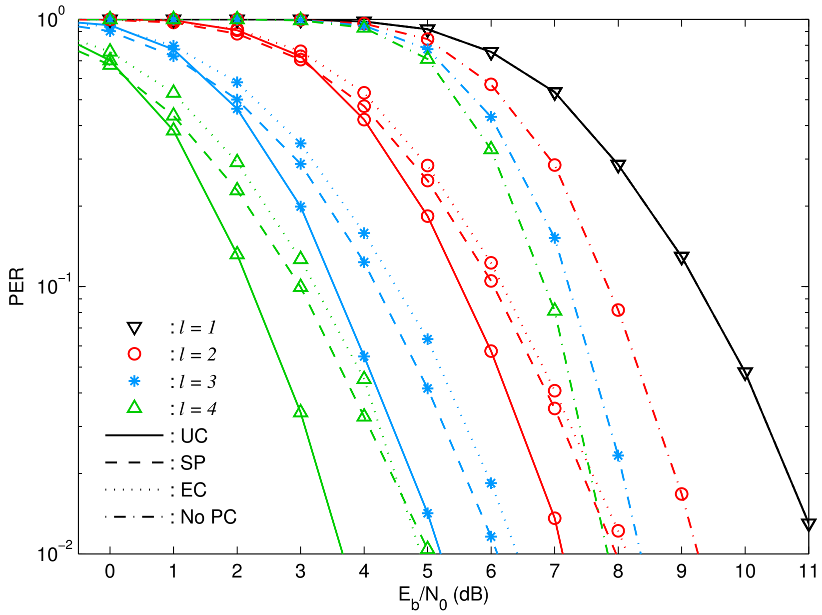



Figure 3.7: PER performance for an iterative FDE receiver with 4 iterations.

of transmission attempts and the receiver takes full advantage of all transmitted power. As expected, the iterative receiver has better performance than the linear receiver, with higher differences when the number of transmission attempts is lower: the differences for PER=$10^{-1}$ are about 5dB when $l = 1$, and 1dB when $l = 4$.

## 3.2 Hybrid-ARQ Performance on a TDMA System

In this section, it is presented the packet delay and goodput analysis for a generic TDMA Hybrid ARQ system and for a generic packet arrival distribution. An Hybrid ARQ system can be characterized by its probability of successfully transmitting a packet given by $q_l$, $0 < q_l < 1$, with $l$, $1 \leq l \leq R$, representing the number of transmission attempts of each packet and $R$ representing the maximum number of transmissions allowed for each packet. This probability can be described by analytical expressions for linear receivers and uncoded systems, but for iterative receivers or coded systems it is very hard to have an exact expression for $q_l$ (usually it is only possible to have analytical bounds [Kallel, 1990]). However, $q_l$ can still be obtained using simulations, as the ones presented in the section 3.1.3 for the detector proposed in this manuscript.

The proposed performance models can also be used for classical TDMA systems, where $q_l = q_1$ for all $l$ values.

### 3.2.1 System Characterization

It is considered a full-duplex communication channel on a TDMA basis. Time is divided into equal length slots. The beginning of a packet transmission across the channel coincides with the start of a slot.

It is assumed that each packet contains one block of data[4], so that a packet transmission time is equal to one slot. $M_n$ represents the number of packet arrivals at the station during the $n$th slot, according to a stochastic process. It is assumed that $\{M_n, n \geq 1\}$ is a sequence of i.i.d. random variables, and it is set $a_m = P(M_n = m)$, $m \geq 0$, and $\alpha_1 = E[M_n]$ is the expected number of packet arrivals per slot. The station is scheduled to transmit in one slot per TDMA frame. A TDMA frame contains $T+1$ slots, with $T \geq 1$, representing the group of slots scheduled to other stations[5]. Packets can only be transmitted in one of the T+1 frame slots, which is defined according to the TDMA's scheduling policy. The scheduled slot repeats itself with a periodicity equal to the TDMA frame duration. In the model, it is also assumed that acknowledgements are received before the beginning of the next slot. This section proposes a complete solution for the delay when just one scheduled slot per station is considered.

[4]The model can still be used to calculate the block level delay for variable packet length if the block arrival statistics are obtained by combining the packet length and the packet arrival statistics.

[5]Slot scheduling is outside the scope of this thesis. It is assumed that data slots are scheduled after reservation requests sent over a control channel.

### 3.2.2 Generating Function of the System Size

Let $X_k$ represent the number of packets in the station's queue waiting for transmission (backlogged), and $Z_k$ represent the number of transmission attempts of the Head-of-Line (HoL) packet at the end of slot $k$. Let $Y_k$ represent the scheduled slot index in the TDMA frame of slot $k$, with $k \geq 1$ and $1 \leq Y_k \leq T+1$. At the start of slot $k$ the station has $X_k$ packets in the queue. If a given station is scheduled to transmit in the first slot of the frame, then during slot $k$ the station tries to transmit the HoL packet for the $Z_k$th time when $Y_k = 1$, not transmitting when $Y_k \neq 1$. The random variables $X_k$, $Y_k$ and $Z_k$ define a tri-dimensional Markov Chain represented by $\{(X_k, Y_k, Z_k), k \geq 1\}$.
The system is stable if the average inter-packet arrival time (i.e. $1/\alpha_1$) is higher than the average packet service time[6], $(T+1)\,E[Z]$ (measured in slots per packet), i.e.

$$\alpha_1 (T+1)\, E[Z] < 1,$$

where $E[Z]$ represents the average number of transmissions for a packet until its successful reception or until being dropped. It is given by

$$E[Z] = \lim_{k\to\infty} E[Z_k] = \left[\sum_{l=1}^{R} l q_l Q_l\right] + R Q_{R+1}. \tag{3.13}$$

where $Q_x$ is the probability of having $x-1$ failed transmissions:

$$Q_x = \prod_{m=1}^{x-1} (1 - q_m)\,. \tag{3.14}$$

Under the stability condition, the conditional steady-state distribution $\{X_k\}$ given that $Y_k = j$ and $Z_k = l$ and the steady-state distribution of $\{X_k\}$ in the Cesàro sense exists. Denote the steady-state conditional probability

$$P_{j,i,l} = \lim_{k\to\infty} P\left(Y_k = j, X_k = i | Z_k = l\right) \tag{3.15}$$

and the corresponding steady-state probability

$$\pi(i) = \lim_{J\to\infty} \frac{1}{J} \sum_{k=1}^{J} \sum_{j=1}^{T+1} \sum_{l=1}^{R} P\left(Y_k = j, X_k = i | Z_k = l\right). \tag{3.16}$$

[6]Packets may not be correctly received after $R$ transmissions. When error recovery protocols are run on top of the MAC protocol, $\alpha_1$ must be at least less than the maximum average number of packets successfully transmitted, denoted by the saturation goodput $G_{sat}$, calculated in (3.43). The actual stability bound value depends on the efficiency of the error recovery protocol used.

Next, it is defined a set of $z$-transforms, $|z| \leq 1$.

$$U_{j,l}(z) = \sum_{i=0}^{\infty} P_{j,i,l} z^i, \qquad 1 \leq j \leq T+1,\ 1 \leq l \leq R \tag{3.17}$$

$$\Pi(z) = \sum_{i=0}^{\infty} \pi(i) z^i, \tag{3.18}$$

$$A(z) = \sum_{i=0}^{\infty} a_i z^i, \qquad A'(1) = \alpha_1,\ A''(1) = \alpha_2 \tag{3.19}$$

The steady-state probabilities referred previously satisfy the equilibrium equations represented in (3.20) considering $i \geq 0$. The steady-state conditional probability of $X_k$ (the number of packets backlogged in the stations' queue at the beginning of slot $k$) for the TDMA slot index next to the slot scheduled to the station (i.e. for $j = 2$ when the first slot is scheduled) and $Z_k$ are influenced by what happened during the previous slot: $Z_k$ is incremented after a failed transmission (3.20d); $Z_k$ is set to 1 and $X_k$ is decremented after a successful transmission (3.20c). Equation (3.20c) also handles the last retransmission of a packet ($Z_{k-1} = R$) or an idle slot ($X_{k-1} = 0$) during the previous slot. The steady state conditional probability of $X_k$ is only influenced by the arrival of new packets for the other TDMA slot index values.

$$P_{j,i,l} = \begin{cases} \sum_{m=0}^{i} a_m P_{T+1,i-m,l} & \text{if } j = 1 \quad (3.20a) \\ \sum_{m=0}^{i} a_m P_{j-1,i-m,l} & \text{if } T \geq 2,\ 3 \leq j \leq T+1 \quad (3.20b) \\ a_i P_{j-1,0,1} + \sum_{l=1}^{R} q_l \sum_{m=0}^{i} a_m P_{j-1,i+1-m,l} & \\ + (1-q_R) \sum_{m=0}^{i} a_m P_{j-1,i+1-m,R} & \text{if } l = 1, j = 2 \quad (3.20c) \\ (1-q_{l-1}) \sum_{m=0}^{i-1} a_m P_{j-1,i-m,l-1} & \text{if } 2 \leq l \leq R,\ j = 2 \quad (3.20d) \end{cases}$$

Noting that $P(Y_k = j) = (T+1)^{-1}$, for $1 \leq j \leq T+1$, it is obtained

$$\pi(i) = \frac{1}{T+1} \sum_{j=1}^{T+1} \sum_{l=1}^{R} P_{j,i,l}. \tag{3.21}$$

Considering (3.17) and (3.18), it is obtained

$$\Pi(z) = \frac{1}{T+1} \sum_{j=1}^{T+1} \sum_{l=1}^{R} U_{j,l}(z). \tag{3.22}$$

**Proposition 1.** $\Pi(z)$ *is given by the set of equations represented in (3.27).*

*Proof.* $\Pi(z)$ is calculated multiplying the $i$th equation defined in (3.20) by $z^i$ and summing over $i$. Considering the following algebraic manipulations

$$\sum_{i=0}^{\infty} \sum_{m=0}^{i} a_m P_{T+1,i-m,l} z^i = \left(\sum_{i=0}^{\infty} a_i z^i\right) \left(\sum_{i=0}^{\infty} P_{T+1,i,l} z^i\right), \tag{3.23}$$

$$\sum_{i=0}^{\infty} \sum_{m=0}^{i} a_m P_{j-1,i-m,l} z^i = \left(\sum_{i=0}^{\infty} a_i z^i\right) \left(\sum_{i=0}^{\infty} P_{j-1,i,l} z^i\right), \tag{3.24}$$

$$\begin{aligned}\sum_{i=0}^{\infty} \sum_{m=0}^{i} a_m P_{j-1,i+1-m,l} z^i &= \left(\sum_{i=0}^{\infty} a_i z^i\right) \left(\sum_{i=0}^{\infty} P_{j-1,i+1,l} z^i\right) = \left(\sum_{i=0}^{\infty} a_i z^i\right) .z^{-1} \left(\sum_{i=1}^{\infty} P_{j-1,i,l} z^i\right) \\ &= \left(\sum_{i=0}^{\infty} a_i z^i\right) .z^{-1} \left(\sum_{i=0}^{\infty} P_{j-1,i,l} z^i - P_{j-1,0,l}\right),\end{aligned} \tag{3.25}$$

and

$$\begin{aligned}\sum_{i=0}^{\infty} \sum_{m=0}^{i-1} a_m P_{j-1,i-m,l-1} z^i &= \sum_{i=0}^{\infty} \left(\sum_{m=0}^{i} a_i P_{j-1,i-m,l-1} z^i - a_i P_{j-1,0,l-1} z^i\right) \\ &= \left(\sum_{i=0}^{\infty} a_i z^i\right) \left(\sum_{i=0}^{\infty} P_{j-1,i,l-1} z^i\right) - \left(\sum_{i=0}^{\infty} a_i z^i P_{j-1,0,l-1} z^i\right)\end{aligned} \tag{3.26}$$

it is obtained the $z$-transforms represented in (3.27). □

$$U_{j,l}(z) = \begin{cases} A(z)\, U_{T+1,l}(z) & \text{if } j = 1 \quad (3.27a) \\ A(z)\, U_{j-1,l}(z) & \text{if } T \geq 2,\ 3 \leq j \leq T+1 \quad (3.27b) \\ \sum_{l=1}^{R} q_l A(z)\, z^{-1} \left(U_{j-1,l}(z) - P_{j-1,0,l}\right) + & \\ A(z)(1-q_R)\, z^{-1} \left(U_{j-1,R}(z) - P_{j-1,0,R}\right) + & \\ A(z)\, P_{j-1,0,1} & \text{if } l = 1, j = 2 \quad (3.27c) \\ (1-q_{l-1})\, A(z) \left(U_{j-1,l-1}(z) - P_{j-1,0,l-1}\right) & \text{if } 2 \leq l \leq R,\ j = 2 \quad (3.27d) \end{cases}$$

Substituting $U_{j,l}(z)$ in (3.22) by the expression (3.27) and after several simplifications, the equation (3.22) may be derived to the form

$$\Pi(z) = \frac{1}{T+1}\sum_{j=0}^{T} A(z)^j \sum_{l=1}^{R} U_{2,l}(z). \tag{3.28}$$

Using (3.27a) to (3.27d) iteratively, $U_{2,l}(z)$ can be expressed as follows

$$\begin{aligned}
U_{2,l}(z) &= (1-q_{l-1}) A(z) (U_{1,l-1} - P_{1,0,l-1}) \\
&= (1-q_{l-1}) A(z) \left(A(z)^T U_{2,l-1} - P_{1,0,l-1}\right) \\
&= (1-q_{l-1}) \Big(A(z)^{2(T+1)} (1-q_{l-2}) U_{2,l-2} - A(z)^{T+2} (1-q_{l-2}) P_{1,0,l-2} - \\
&\quad A(z) P_{1,0,l-1}\Big).
\end{aligned}$$

Due to the fact of $P_{1,0,l} = 0$ (an empty queue does not have packets previously transmitted) for $l \geq 2$ it is obtained

$$U_{2,l}(z) = \left[\prod_{m=1}^{l-1} (1-q_m) A(z)^{(l-2)(T+1)+1}\right] \left(A(z)^T U_{2,1}(z) - P_{1,0,1}\right). \tag{3.29}$$

Now, let us focus on $U_{2,1}(z)$ since $\Pi(z)$ still depends on it. $U_{2,1}(z)$ can be defined as

$$\begin{aligned}
U_{2,1}(z) =& \sum_{l=1}^{R} q_l A(z) z^{-1} (U_{1,l}(z) - P_{1,0,l}) + (1-q_R) A(z) z^{-1} (U_{1,R}(z) - P_{1,0,R}) + \\
& A(z) P_{1,0,1} \\
=& q_1 A(z) z^{-1} \left(A(z)^T U_{2,1}(z) - P_{1,0,1}\right) + \sum_{l=2}^{R} q_l A(z) z^{-1} \left(A(z)^T U_{2,l}(z) - P_{1,0,l}\right) + \\
& (1-q_R) A(z) z^{-1} \left(A(z)^T U_{2,R} - P_{1,0,R}\right) + A(z) P_{1,0,1}.
\end{aligned}$$

To solve it, it is considered again the fact of $P_{1,0,l} = 0$ for $l \geq 2$ and after using (3.27a)-(3.27d) iteratively it is obtained

$$U_{2,1}(z) = \frac{f_0(z)}{g_0(z)}, \tag{3.30}$$

where

$$\begin{aligned}
f_0(z) =& A(z) P_{1,0,1} \left[1 - z^{-1}\left(q_1 + (1-q_R) Q_R A(z)^{(R-1)(T+1)} + \sum_{l=2}^{R} q_l Q_l A(z)^{(l-1)(T+1)}\right)\right], \\
g_0(z) =& 1 - z^{-1}\left(q_1 A(z)^{T+1} + (1-q_R) Q_R A(z)^{R(T+1)} + \sum_{l=2}^{R} q_l Q_l A(z)^{l(T+1)}\right),
\end{aligned}$$

and $Q_x$ is the probability of having $x-1$ failed transmissions, as defined in (3.14).

By the normalization condition, $\sum_{l=1}^{R} U_{2,l}(1) = 1$, it is obtained

$$P_{1,0,1} = \left[\left(1 + \sum_{l=2}^{R} Q_l\right)\frac{D_z}{g_1} - \sum_{l=2}^{R} Q_l\right]^{-1}, \tag{3.31}$$

where $U_{2,1}(1)$ is determined applying the L'Hôpital's rule and is equal to

$$U_{2,1}(1) = \frac{f_1}{g_1}, \tag{3.32}$$

where

$$\begin{aligned} f_1 =& P_{1,0,1} D_z, \\ g_1 =& 1 - (T+1)\,\alpha_1 \left[(1-q_R)\,Q_R R + \sum_{l=1}^{R} q_l Q_l l\right] \end{aligned}$$

and

$$D_z = 1 + \alpha_1 \left(1 - (1-q_R)\,Q_R\left[(R-1)(T+1)+1\right] - \sum_{l=1}^{R} q_l Q_l \left[(l-1)(T+1)+1\right]\right).$$

From (3.28) and (3.29) it is possible to define the generating function

$$\Pi(z) = \frac{1}{T+1}\left(\sum_{j=0}^{T} A(z)^j\right)\left[U_{2,1}(z) + \sum_{l=2}^{R} W_l(z)\right], \tag{3.33}$$

where

$$W_l(z) = \left(Q_l A(z)^{(l-2)(T+1)+1}\right)\left(A(z)^T U_{2,1}(z) - P_{1,0,1}\right).$$

### 3.2.3 Mean System Size

The mean number of packets in the station's queue can be computed by taking the derivative of $\Pi(z)$ and letting $z \to 1$.From (3.33) it is obtained (3.34).

$$\begin{aligned} \Pi'(1) =& \alpha_1 \left[U_{2,1}(1) + \sum_{l=2}^{R} Q_l \left(U_{2,1}(1) - P_{1,0,1}\right)\right] + \left[U'_{2,1}(1) + \right. \\ & \left. \sum_{l=2}^{R} Q_l \left[(l-1)(T+1)\,\alpha_1 U_{2,1}(1) + U'_{2,1}(1) - \left[(l-2)(T+1)+1\right]\alpha_1 P_{1,0,1}\right]\right]. \end{aligned} \tag{3.34}$$

To accomplish $U'_{2,1}(1)$ it is computed the derivative of (3.30) and apply twice the L'Hôpital's rule. Thus, it is possible to obtain

$$U'_{2,1}(1) = \frac{f_2 g_1 - f_1 g_2}{2g_1^2}, \tag{3.35}$$

where $f_2$ and $g_2$ are respectively defined by (3.38) and (3.39).

$$\begin{aligned} f_2 =& P_{1,0,1}\Bigg[\alpha_2(1-q_1) - (1-q_R)Q_R[(R-1)(T+1)+1]\Big[(R-1)(T+1)\alpha_1^2+\alpha_2\Big] + \\ & 2\alpha_1 - \sum_{l=2}^{R} q_l Q_l[(l-1)(T+1)+1]\Big[(l-1)(T+1)\alpha_1^2+\alpha_2\Big]\Bigg] \end{aligned} \tag{3.38}$$

$$\begin{aligned} g_2 =& -q_1(T+1)\Big(T\alpha_1^2+\alpha_2\Big) - (1-q_R)Q_R R(T+1)\Big([R(T+1)-1]\alpha_1^2+\alpha_2\Big) - \\ & \sum_{l=2}^{R} q_l Q_l l(T+1)\Big[(l(T+1)-1)\alpha_1^2-\alpha_2\Big] \end{aligned} \tag{3.39}$$

### 3.2.4 Delay Analysis

The packet delay $E[D]^{dc}$ is defined as the time interval between the packet arrival and its removal from the queue (at the end of a slot). Due to the continuous time packet arrival distribution nature, $E[D]^{dc}$ is composed by an initial vacation time (until the beginning of a slot) and by a discrete packet delay time (until the packet departs from the system). The discrete packet delay, denoted by $E[D']^{dc}$, is defined by the number of slots elapsed between the slot of the packet arrival and the instant it departs from the system. Using the Little's formula and assuming a First Come First Served (FCFS) service discipline, the discrete mean packet delay is given by $E[D']^{dc} = E[X]/\alpha_1$, where $E[X] = \Pi'(1)$ and is calculated using (3.34). The vacation time mean value depends on the distribution of the packet arrival deviation from the slot boundary, denoted by $\Phi$. The deviation from the slot boundary of packet $k \in [1,\infty[$ that arrives at time $t_k$ is given by $\phi_k = \lceil t_k \rceil - t_k$. The mean packet arrival deviation is defined in a Cesàro sense by the following equation.

$$E[\Phi] = \lim_{k\to\infty} \frac{1}{k}\sum_{i=1}^{k}\phi_i$$

Therefore, the packet delay is given by

$$E[D]^{dc} = \frac{E[X]}{\alpha_1} + E[\Phi]. \tag{3.40}$$

This equation can be used with any packet arrival distribution, as long as the packet arrival process, $A(z)$, and the packet arrival deviation from the slot boundary, are defined. If the packet arrival deviation from the slot boundary is unknown, a uniform distribution approximation can be considered, with $E[\Phi] \approx 1/2$. This approach is more generic than the traditional approach for delay analysis, which considers a Poisson process packet arrival and the use of the theory of embedded Discrete Time Markov Chain (DTMC), specially the Pollaczek-Khintchine formula. The next section validates the proposed model, assuming arrivals to be governed by the following two streams.

#### 3.2.4.1 Poisson Arrival Process

Considering arrivals to be defined by a Poisson Process given by

$$P\left(M_n = m\right) = \frac{\lambda^m}{m!} e^{-\lambda}, \qquad m \geq 0, n \geq 1,$$

it is obtained

$$A\left(z\right) = e^{-\lambda(1-z)}, \qquad A'\left(1\right) = \alpha_1 = \lambda, \qquad A''\left(1\right) = \alpha_2 = \lambda^2. \tag{3.41}$$

For a Poisson packet arrival process the deviation from the slot boundary is $E[\Phi] = 1/2$.

#### 3.2.4.2 Geometric Arrival Stream

Considering a geometric stream of independently arrival events, with probability $p$ of occurring a single packet arrival and probability $(1 - p)$ of no occurring, $0 < p < 1$, the moment generating function of it can be described [Saeki and Rubin, 1982] as

$$A\left(z\right) = 1 - p + pz, \qquad A'\left(1\right) = \alpha_1 = p, \qquad A''\left(1\right) = \alpha_2 = 0. \tag{3.42}$$

### 3.2.5 Goodput Analysis

The average packet's service time counts the number of slots used to transmit a packet or to cancel its transmission after $R$ retries, and is equal to

$$E[\mathcal{S}] = (T+1)\left[\sum_{i=1}^{R} \left[iq_iQ_i\right] + RQ_{R+1}\right].$$

Let $G_{sat}^{dc}$ be the channel's saturation goodput. It is limited by the average packet's service time and by the probability of successful transmission, and is given by

$$G_{sat}^{dc} = \frac{1 - Q_{R+1}}{E[\mathcal{S}]}. \tag{3.43}$$

The channel goodput, $G^{dc}$, follows $(1 - Q_{R+1})\,\alpha_1$ as long as the channel is not saturated.

$$G^{dc} = \begin{cases} G_{sat}^{dc}, & \alpha_1 \geq \frac{1}{E[\mathcal{S}]} \\ (1 - Q_{R+1})\,\alpha_1, & \alpha_1 < \frac{1}{E[\mathcal{S}]} \end{cases}. \tag{3.44}$$

## 3.3 Performance Results

A set of performance results concerning the proposed Hybrid ARQ for a TDMA uplink channel is presented in this section. The system-level performance is analyzed using the results obtained in Sec. 3.1.3 and the analytical model of Sec. 3.2. It is considered, as referred earlier, one slot per station and 8 stations ($T = 7$) generating traffic following a Poisson process or a Geometric stream with $\alpha_1 = \lambda$ packets/slot. In the Geometric stream case, it was considered a uniform packet arrival deviation from the slot boundary in $[0, 1]$, defining $E[\Phi] = 1/2$.

It is implemented the described retransmission strategies (two channel conditions and the shifted packet strategy) in the ns-2 simulator [Information Sciences Institute, 2007]. It is assumed that $q_l$ for packets is a mean-ergodic process, and the PER variation with $l$ is modelled using the PER values with 4 iterations measured in the first set of experiments, and represented in Figure 3.7. The $q_l$ as a function of $E_b/N_0$ is introduced as a matrix parameter for the simulations. In the analysis, Hybrid ARQ with $R = 5$ (it attempts up to 5 transmissions per packet) and the conventional ARQ scheme (equivalent to packet combining with $q_l = q_1$, $2 \leq l \leq R$) are compared.

Figures 3.8 and 3.9, show the main performance differences under Poisson traffic load between each retransmission strategy considered and the number of iterations, for goodput and delay respectively. It can be observed, as shown in Figures 3.6 and 3.7, that the linear FDE (1 iteration on the receiver) presents worse performance (lower goodput and higher delays) than the iterative receiver (4 iterations considered). This is more visible for the EC condition. UC condition outperforms any other transmission strategy. This is expected and already anticipated when the UC was introduced. However, it is quite unrealistic to consider such an uncorrelation between retransmissions. The traditional channel condition adopted is the EC, and the SP strategy presents equal or better results whatever set of $E_b/N_0$ weighted.

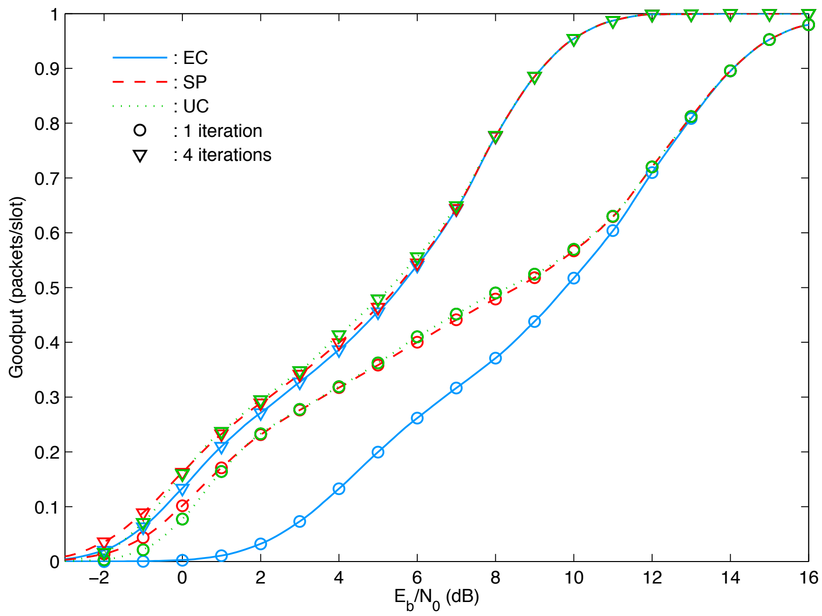



Figure 3.8: Goodput as function of $E_b/N_0$ with $R = 5$ and saturated load.

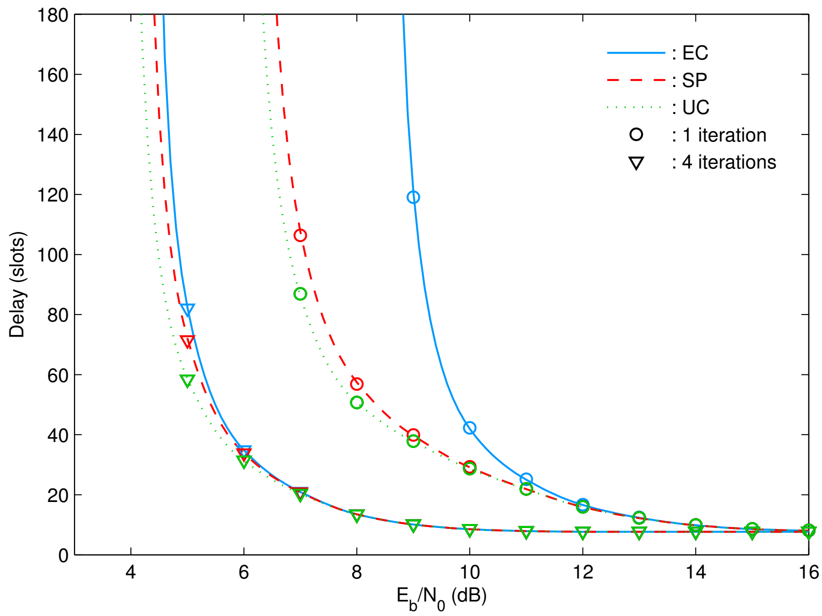



Figure 3.9: Packet delay with $E_b/N_0$ for Poisson traffic, $R = 5$ and $1/\lambda$=20 slots/packet.

Therefore, from this point on it is considered the SP strategy.

Figure 3.10 depicts the simulation and the analytical model of goodput in function of $E_b/N_0$ for different traffic load scenarios ($1/\lambda$=$\{$Saturated, 10, 20$\}$ slots/packet). The analytical results were computed using (3.44). The figure shows that the measured goodput follows the analytical values, thus validating it. It also shows huge saturation goodput gains

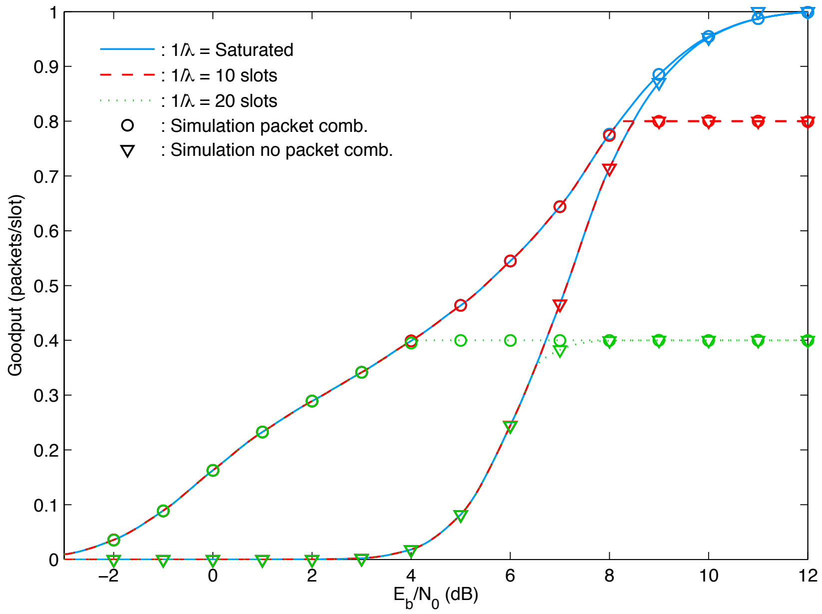



Figure 3.10: Goodput as function of $E_b/N_0$ with $R = 5$.

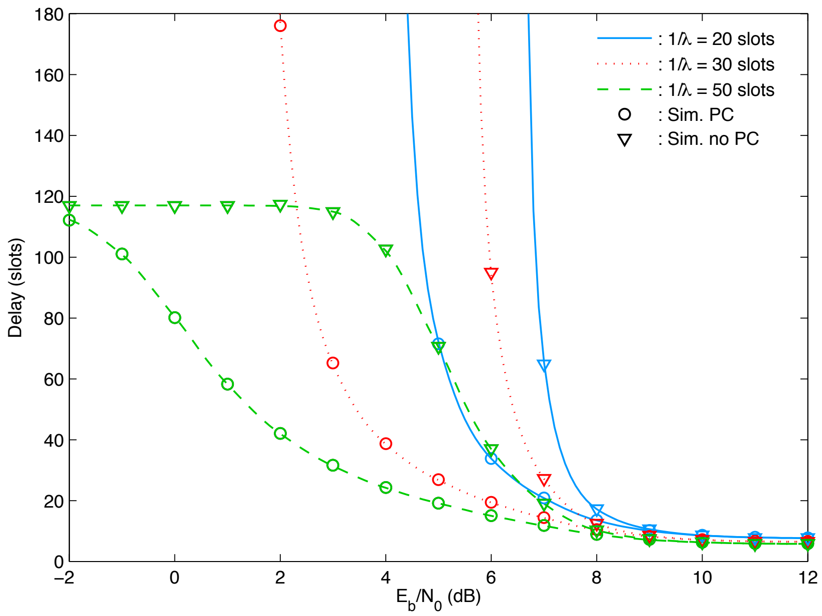



Figure 3.11: Packet delay with $E_b/N_0$ for Poisson traffic and $R = 5$.

achieved with packet combining for $E_b/N_0 < 9$dB, where $q_1$ is low but a higher $q_l$ with $l > 1$ allows the SP retransmission strategy to be successful. It can be shown that it is always possible to achieve a higher or equal goodput with SP strategy over conventional ARQ schemes on the same conditions.

Figure 3.11 depicts the simulation and the analytical model for packet delay in function

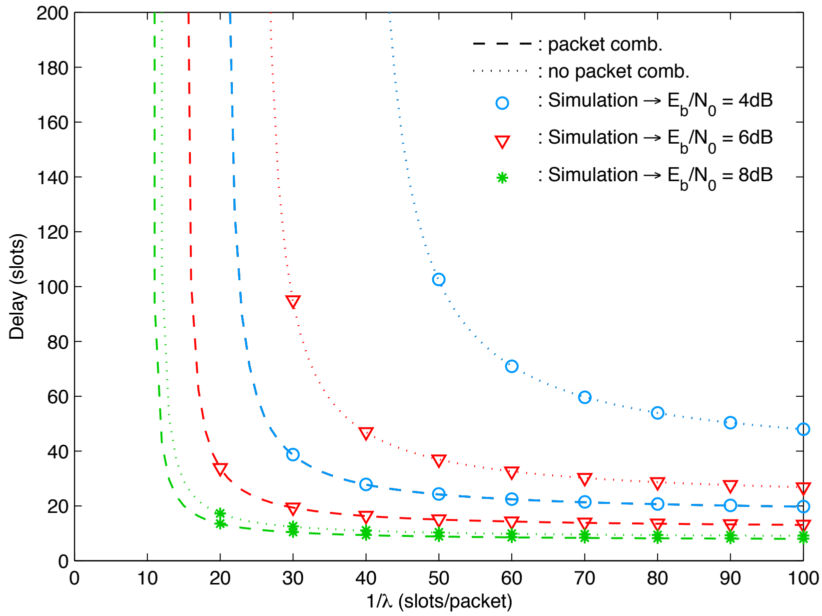



Figure 3.12: Packet delay with $1/\lambda$ for Poisson traffic and $R = 5$.

of $E_b/N_0$ for $1/\lambda$ equal to 20, 30 and 50 slots/packet (considering Poisson process), both for the SP strategy and conventional ARQ scheme. The analytical results were computed using (3.40) and (3.41). It shows that the measured delay follows precisely the analytical model values, thus validating it. It also shows that the SP strategy outperforms systematically the conventional ARQ, producing lower or equal delays, and having finite delays for lower $E_b/N_0$ values. Notice that packet combining allows a finite delay even for an initial very high PER ($q_1 \approx 0$ for 4dB). The figure also shows that the packet delay is bounded to a maximum value. The maximum delay occurs when packets are retransmitted $R$ times, for a high PER (associated to a low $E_b/N_0$), and varies with the packet load.

Figure 3.12 presents the simulation and the analytical model for packet delay in function of $1/\lambda$ for different $E_b/N_0$ with and without packet combining strategy. It shows that the packet delay for the SP strategy is less sensible to the load than for conventional ARQ. SP strategy's delay grows slowly when the load increases ($1/\lambda$ decrease) except when the load approaches the saturation limit (defined by the saturation goodput in Figure 3.10). Conventional ARQ scheme delay grows faster outside the saturation limit ($E_b/N_0 = 6$dB), unless the success probability $q_1$ is high ($E_b/N_0 = 8$dB).

Figure 3.13 presents the simulation and analytical model values for packet delay in function of $E_b/N_0$ for $1/\lambda$ equal to 10, 20 and 50 slots/packet considering a geometric arrival stream and $R = 5$. Both, SP strategy and conventional ARQ schemes, are represented. The analytical results were computed using (3.40), (3.42) and as it can be seen, the measured

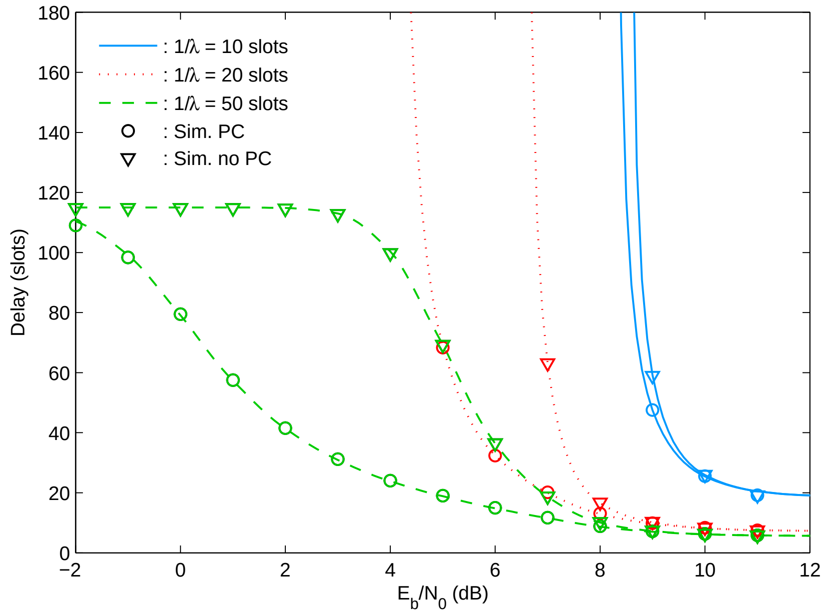



Figure 3.13: Packet delay with $1/\lambda$ for Geometric traffic and $R = 5$.

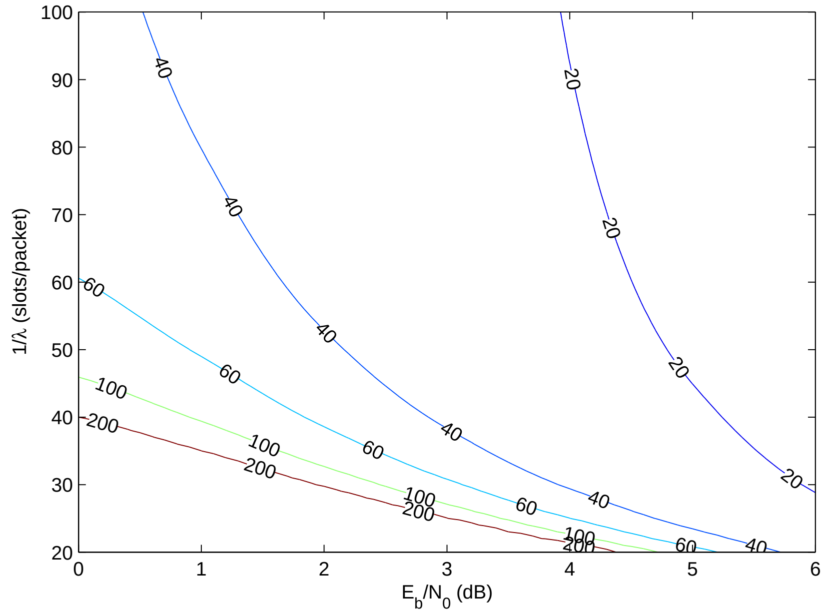



Figure 3.14: Packet delay for $1/\lambda$ and $E_b/N_0$ with packet combining for Poisson traffic and $R = 5$.

delay follows precisely the analytical model values, thus validating it for non-Poisson traffic. The obtained results were similarly close to the results using Poisson process. SP strategy outperforms systematically conventional ARQ, producing lower or equal delay values, independently of the arrival stream selected.

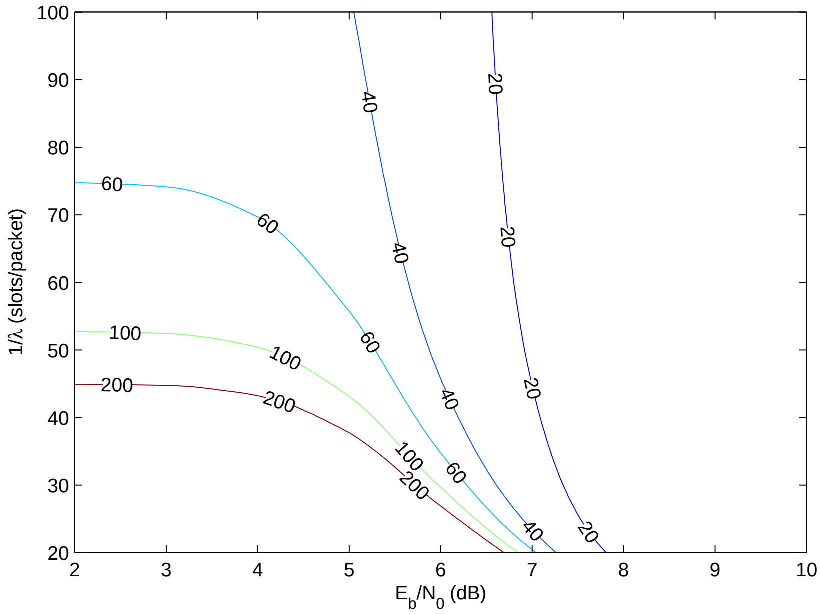



Figure 3.15: Packet delay for $1/\lambda$ and $E_b/N_0$ without packet combining for Poisson traffic and $R = 5$.

Figure 3.14 and Figure 3.15 present the results of the analytical model for packet delay in function of $E_b/N_0$ and $1/\lambda$ for Poisson arrival stream, respectively for the SP and conventional ARQ schemes. With SP strategy, a much larger useful range of $E_b/N_0$ than in conventional ARQ is achieved. Figure 3.10 showed that conventional ARQ does not allow traffic to flow for $E_b/N_0 < 3$dB, resulting on the delay bounds represented in Figure 3.15 associated to $R$ unsuccessful retransmissions of the packets.

SP, in the same conditions, allows a higher traffic flow (as seen in Figure 3.10) and a feasible packet delay reduction, proven in Figure 3.14. Therefore, it is possible to offer a more robust system in scenarios where $E_b/N_0$ is critical. With the analysis performed on aforementioned figures, a correct and effective system configuration for pre-determined delay requirements is possible. It is viable to compute the minimum needed $E_b/N_0$ value within an average delay bound requirement for a given generated traffic value. Let's consider a generating traffic value (Poisson process) of $1/\lambda = 40$ slots/packet and a delay bound value of 40 slots. Looking at Figure 3.14, with packet combining technique, a minimum $E_b/N_0 =$ 3dB is needed to fulfil aforementioned requirements. Without packet combining technique, Figure 3.15, a minimum value of $E_b/N_0 = 6$dB is necessary. Therefore, the analytical model presented allows an effective power control when throughput and delay bounds are defined.

## 3.4 Conclusions

This chapter carries out exact goodput, packet delay and queue-size analysis for a single slot per frame packet-switched TDMA applying a packet combining ARQ scheme in a full-duplex system. It was proposed a generic analytical model that applies to any packet arrival distribution, which obtains the generating function of the steady-state system size. This model also applies to any transmission technique employing a TDMA hybrid ARQ scheme. The packet delay and goodput are calculated using the model output. Although the proposed analysis is valid for any transmission technique with packet combining, provided that the PER values for each transmission attempt are known (obtained either analytically or by simulation), it is considered an SC-FDE scenario. The simulation results show its correctness and accuracy for two distinctive packet arrival streams: Poisson and Geometric. Moreover, effective power control is achievable by defining a minimum $E_b/N_0$ bound that satisfies throughput and delay requirements.

The analysis performed in this chapter assumed a collision free environment which is not always possible to provide. To handle with collisions, collision resolution schemes need to be adopted.

# CHAPTER 4

# Collision Resolution Techniques

A Perspective of fixed allocation resources (i.e. Time Division Multiple Access (TDMA) scheme) was presented in the previous chapter. Whereas fixed allocation schemes are generally good for constant bit rate traffic with limited buffering or strict delay constraints, they fail when arrivals are more bursty. In this case, random access schemes are generally preferable since they allow low mean delay, provided that the overall traffic load is limited. However, they also open the door for a collision environment.

Traditionally, the design of Medium Access Control (MAC) protocols has been separated from the design of the Physical (PHY) layer. Researchers on the MAC field usually consider a perfect channel abstraction for the PHY layer, which behaves as a black box that only has errors due to collisions: when only one user transmits, the packet arrives at the receiving end error free, but when simultaneous transmissions occur, packets are lost due to collisions. Random access protocols were until recently based on such an idealized model and, as a consequence, random access protocols were viewed as collision resolution or collision avoidance techniques.

However, a closer look on the collision model can lead us to observe that it comprehends both optimist and pessimistic assumptions. It is optimistic because it ignores channel effects such as fading or noise in the reception, but since it does not accommodate the possibility that packets may be successfully decoded in the occurrence of a collision it is also pessimistic. As a result, it is possible to conclude that with the recent advent of multiuser communications at the PHY layer, the collision model became obsolete and no longer represents all the characteristics of PHY layer. The need to go beyond the collision model is crucial as wireless communications evolve.

In 1985 Gallager [Gallager, 1985] pointed out as fundamental challenge the choice of a proper model that interfaces the PHY layer and network layers. In addition, a major point of Gallager's work is that, in his own words, "a better set of models and approaches are needed for multiaccess communication than collision resolution or information theory alone." More recently, Ephremides and Hajek [Ephremides and Hajek, 1998] emphasized the previous question, noting that information-theoretic techniques were not yet of widespread use in the domain of networking with random user activity until that moment.

This chapter provides an overview on collision resolution techniques in random access scenarios. A brief historical perspective is given, starting on MAC and PHY layer separate proposals and ending in more cross-layer/joint detection fashion schemes, as an answer to [Gallager, 1985] and [Ephremides and Hajek, 1998] demands.

## 4.1 MAC-Layer Solutions

### 4.1.1 ALOHA and its Evolutions

The simplified idea in ALOHA random access schemes is that each terminal may transmit as soon as it has a packet in the buffer. ALOHA random access schemes stem from Abramson's landmark work [Abramson, 1970], pure ALOHA[1]. The design consisted in a multiaccess scheme that allows different terminals in different islands access a central unit through a shared wireless channel: whenever a terminal has data, it transmits; the sender finds out whether transmission was successfully or experienced a collision by listening to the broadcast from a destination station; if the latter case occurred, the sender retransmits after some random time. In the presence of light traffic loads with bursty data sources, it is unlikely that two or more terminals transmit at the same time. Hence delay is much lower than in a fixed allocation scheme. However, in the presence of higher loads, collisions do occur and since after each collision the probability to occur a new collision is higher, limited performance is obtained. In the latter case, multiple collisions of the same packet can occur (and consequently multiple backoff periods) leading to a relatively low maximum throughput and excessive delay, even under moderate traffic load. The analysis of ALOHA by Abramson [Abramson, 1970] concludes that the maximum throughput achievable is 18.4% of the total capacity of the system.

A few years later, a variation of ALOHA was presented by Roberts [Roberts, 1975], the slotted ALOHA. The bigger change is the fact that the time is slotted into equal size periods of time and a packet can only be transmitted at the beginning of one slot. Thus, it can reduce the collision duration. The *modus operandi* is similar: whenever a terminal has data, it transmits in the next slot; if no collision occurs, the terminal is able to transmit again in the next slot; but when in the presence of a collision, the terminal transmits in the

[1]The designated name came from the research program name itself, called "The ALOHA system."

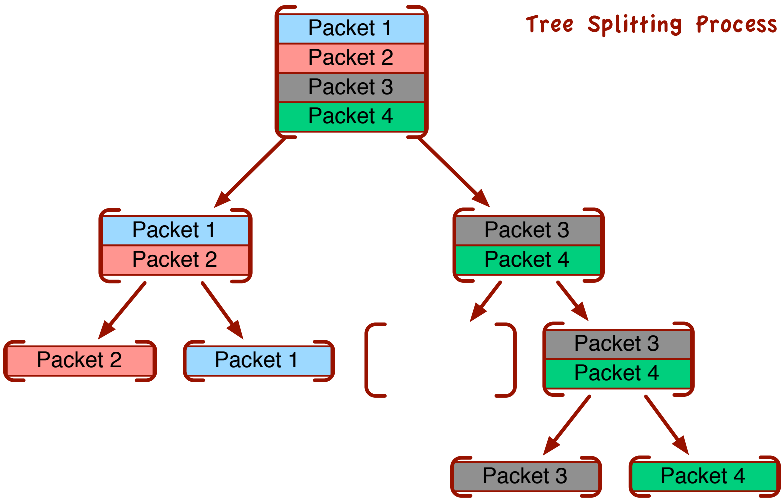
next slots with a probability $p$ unaware of other terminals. Note that some synchronization control is needed in the slots. The slotted ALOHA algorithm allows a duplication of the maximum throughput when compared to pure ALOHA, fixing the limit at 0.368 packets/slot (i.e. 36.8% of the total capacity of the system).

Over the years, many improved solutions and studies employing ALOHA schemes were proposed, e.g. [Tsybakov and Mikhailov, 1979], [Anantharam, 1991] regarding stability issues for finite populations or [Habbab *et al.*, 1989] concerning the capture model. More recently Abramson [Abramson, 1994] introduced a spread-spectrum channel using an ALOHA packet separation mechanism that opened the door to a variety of techniques used in conventional ALOHA channels, which can significantly increase both the throughput and the efficiency of the channel. Although better tailored to handle variable bit rate traffic than common multiple access schemes as TDMA or Frequency Division Multiple Access (FDMA), ALOHA schemes suffer from substantial throughput penalties and wasted resources under heavy loads. An alternative way is needed for the MAC layer to coordinate the retransmissions of collided packets such that secondary collisions can be mitigated.

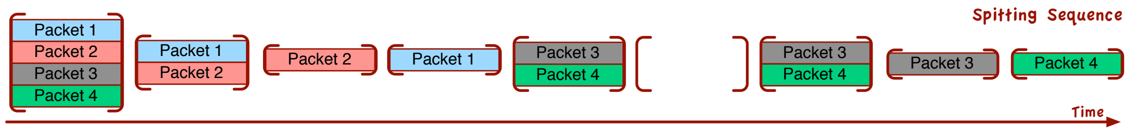
### 4.1.2 Tree Algorithms

In 1979 a breakthrough development was introduced by Capetanakis [Capetanakis, 1979b,a], the tree algorithm. The main idea consists on splitting the transmissions of collided users, thus obtaining a random access protocol that offers a higher throughput than ALOHA traditional approach. The tree algorithm assumes ternary feedback: 0 for an empty slot; 1 for a successful transmission of a packet; and $e$ for a collision. The feedback is given to the terminals immediately after the packet transmission is complete. When a collision occurs and the feedback message $e$ is made available, the terminals that had transmitted are split into two subgroups, in contrast the terminals not involved switch to backoff mode. It should be pointed out, that in a ternary feedback system, the nodes involved in a collision cannot know the number of terminals involved in the collision or their identities [Bersakas and Gallager, 1992; Dimic *et al.*, 2004].

An illustration of the binary tree using a tree algorithm can be seen in Figure 4.1. As the name suggests, the schematic representation is similar to a tree. It starts on the top and develops towards the bottom. The root or top node of the tree is the initial collision group. Each node in the tree represents an available slot, which can be an empty slot, a successful transmission or a collision. The collision nodes are split into left and right branches or subgroups. Splitting can be statistical or previously defined. In the statistical case, each terminal randomly chooses to fit in the first or the second subgroup accordingly




(a) Illustration of a tree schematic.




(b) Packet transmission sequence.

Figure 4.1: Tree splitting algorithm behaviour.

with a probability value of 1/2. On the other hand, the split can be computed using the node's binary ID in the previously defined case (a proper explanation of this process will be given below). Usually the left branch represents the first subgroup, which corresponds to the transmission after the collision. The right branch represents the second subgroup, which only transmits after completed the collision resolution process of the first subgroup. Picking up the example of Figure 4.1, the packets of terminal 1 (packet 1, the blue one) and terminal 2 (packet 2, the red one) are retransmitted in the first subgroup. As a result, they collide again and a new process of splitting is needed. They split and two successful retransmissions are obtained. After finishing the retransmission process of the first subgroup's terminals, the terminals of the second subgroup enter in action. Terminal 3 (packet 3, the grey one) and 4 (packet 4, the green one) retransmit and collide. Then, due to its statistical behaviour, both backoff and retransmit in the second subgroup, resulting in another collision. After the splitting process, both successfully retransmit. It can be seen in the representation of the packet sequence, Figure 4.1b, that the original collision with four packets involved resulted in a collision resolution process involving 8 slots.

The previous example represents the random addressing scheme where the splitting process is performed statistically. This addressing scheme is capable of dealing with the infinite user case scenario. However, in a ternary feedback scheme and a finite user system, each terminal can setup the collision resolution tree using a binary addressing scheme (e.g. the node's binary ID) and thus determining when to transmit its packet. For example, the tree resolution scheme of Figure 4.1 is obtained if the node IDs are 100 (terminal 1), 110 (terminal 2), 001 (terminal 3) and 000 (terminal 4). After an initial collision, the terminals 1 and 2 go to the left branch and both transmit immediately because their first bit is 1; terminals 3 and 4 go to the right branch because their first bit is 0 and wait for the collision resolution of the first subgroup to be resolved. Since the terminals 1 and 2 transmitted due to their first bit equal to 1, they collide again. The second bit of terminal 1 is 0, so it goes to the right branch and remains silent in the next slot; meanwhile, the second bit in terminal 2 is 1 so it transmits in the next slot. As a result, terminal 2 was alone in the first subgroup and its successful transmission was observed (due to the feedback $0/1/e$) by terminals 1, 3 and 4. Now terminal 1 transmits and since it transmits alone, its transmission is successful. Terminals 3 and 4 observe the feedback and conclude by hearing the feedback, that the left branch has split once and two successful transmissions have occurred. Hence nothing remains in the left branch and they can now proceed. Both of them immediately transmit (since both were assigned previously to the right branch) and collide. Then, both terminals look to their second bit and since both bits are 0, they go to the right group and an empty slot occurs. On the next slot they collide again. Looking at their third bit, they now go into separate branches, and the collision is resolved after the next two slots.

In terms of stable throughput the tree splitting algorithm (as described above) is able to achieve 0.43 packets/slot [Capetanakis, 1979b], representing a boost when compared with the previous standard of 0.37 packets/slot for slotted ALOHA (mentioned in section 4.1.1). However, the ALOHA algorithm, when applied to the Poisson source model (i.e., an infinite number of sources that collectively generate traffic satisfying the Poisson statistic) has throughput stability[2] issues [Bersakas and Gallager, 1992]. Under similar source and channel conditions, the tree protocol is stable, has higher maximum average throughput, and better delay properties [Capetanakis, 1979a]. In addition, instead of a fixed binary tree, a traffic dependent dynamic tree can be employed using the dynamic tree algorithm to split the collision group into $j$ subgroups, where $j$ is chosen in order to minimize the expected length of the collision resolution period based on the previous collision resolution periods [Capetanakis, 1979b]. As shown in [Capetanakis, 1979b], an interesting feature of the dynamic tree algorithm is that it embodies TDMA as a special case when the splitting factor $j$

[2]A multiple access system with sources satisfying the Poisson source model is defined to be stable if the average delay is finite [Capetanakis, 1979a].

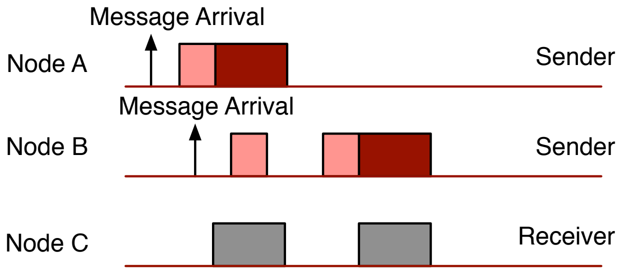
is fixed and equal to the finite number of the nodes. In this way, the dynamic tree algorithm offers a graceful transformation from slotted ALOHA at low loads to TDMA at high loads[3]. As a result, if the probability that a user has a packet to transmit is greater than a specific threshold, then the TDMA and the dynamic tree algorithm performance are the same, but if on the other hand the probability is lower, then the tree algorithm is more efficient than TDMA. Despite all these advantages, the dynamic tree algorithm requires on-the-fly estimation of the average arrival rate for each node, which is problematic for bursty sources, and adaptation of the retransmission probabilities, implying in a certain manner a relatively complex control mechanism.

Along the years, many improvements to the tree algorithms have been proposed. But among them, two probably arose with simple but effective ideas. The first [Massey, 1981] is applied when an idle slot follows a collision, which implicitly means in a binary tree that terminals whose packets collided are assigned to the second branch, so if all the terminals from the second branch transmit after the idle slot, a collision will definitely happen (which was what happen in the second group of Figure 4.1). To bypass this problem, the second branch should immediately split into two subgroups after the idle slot and only enable the first of the newly created subgroups to transmit. With this improvement, a maximum throughput of 0.46 packets/slot is achieved. Another approach to ameliorate the performance is the First Come First Served (FCFS) algorithm [Gallager, 1978; Tsybakov and Mikhailov, 1980; Gallager, 1985]. The idea behind the FCFS algorithm as the name suggest, is to split the colliding packets according to their arrival time, i.e. packets arriving earlier than a predetermined instance go to the first branch, and packets arriving later go to the second branch. Continuously doing that, leads to successive transmission of packets in the order of their arrival. This improvement brings the maximum throughput of the tree algorithm up to 0.4871 packets/slot. A little improvement in the algorithm was made by Mosely and Humblet [Mosely and Humblet, 1985], achieving 0.4878 packets/slot of throughput, the best known[4] to the author's knowledge, variation of the tree algorithm. This improvement is obtained by choosing the optimal number of subgroups (or branches) after a collision, since two subgroups are not optimal when more than two packets are involved in a collision.

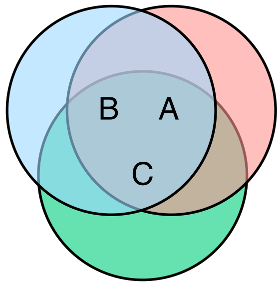
By that time, Mikhailov and Tsybakov [Mikhailov and Tsybakov, 1981] had already characterized the upper bound for the maximum throughput of random access protocols, under the aforementioned ternary feedback assumption (e.g. ALOHA or tree algorithms), by a limit of 0.587 packets/slot. To improve this value, collision multiplicity feedback needs

[3]Note that for random arrivals TDMA is not optimal at high loads.

[4]Another improvement of $3.6\times10^{-7}$ has been made by Vvedenskaya and Pinsker [Vvedenskaya and Pinsker, 1983], but this gain is so small that it has just theoretical interest.




(a) Example of a non-persisting CSMA protocol.




(b) Scheme of three MTs communicating.

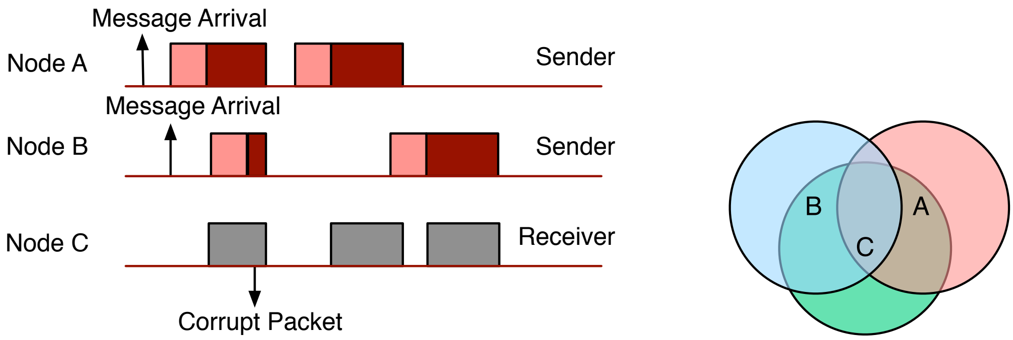
Figure 4.2: Example of a non-persisting CSMA protocol in a three node wireless communication without hidden node problem.

to be employed, wherein the number of collided packets is made available to the transmitting terminals. Although it opens the doors to an optimization of the retransmission probability, the delay remains poor at higher loads since collisions are still wasteful.

### 4.1.3 Carrier Sensing

In 1975 Kleinrock and Tobagi introduced to the world the Carrier Sense Multiple Access (CSMA) protocols for radio channels [Kleinrock and Tobagi, 1975; Tobagi and Kleinrock, 1975]. In few words, CSMA consists in listening (i.e. "carrier sense"[5]) to the channel before transmitting, escaping avoidable collisions. Two protocols were proposed [Kleinrock and Tobagi, 1975]: the non-persisting and the $p$-persisting. The idea in the non-persisting protocol is to limit the interference among packets by always rescheduling a packet which finds the channel busy upon arrival. In other words, a terminal senses the channel and operates as follows: if the channel is sensed idle, it transmits the packet; if the channel is sensed busy, the terminal waits a random period of time before performing a new channel sensing process. This process is illustrated on Figure 4.2. In the case of the $p$-persistent protocol, the scheme consists on including an additional parameter $p$ representing the probability that a ready packet persists. More precisely, after a terminal sensing the channel the protocol consists on the following: if the channel is sensed idle, the terminal transmits with probability $p$ or it delays, for the worst case propagation delay of one packet, with probability $(1 - p)$; if at this new point of time, the channel is still detected idle, the same process is repeated; if the channel is sensed busy, it waits until the channel becomes idle and then operates as aforementioned.

[5]It tries to detect the presence of an encoded signal from another station before attempting to transmit.




(a) Example of a CSMA/CD protocol.

(b) Scheme of three MTs communicating.

Figure 4.3: Example of a CSMA/CD protocol in a three node wireless communication with hidden node problem.

In spite of the good possible performance of CSMA protocols when employed to radio channels, they still suffer from chronicle issues such as hidden node or exposed node problems [Tobagi and Kleinrock, 1975]. Improvements such as the CSMA/Collision Avoidance (CSMA/CA), Request to Send (RTS)/Clear to Send (CTS) packet exchanges or the Distributed Coordination Function (DCF) in 802.11 protocol [IEEE, 2007; Bianchi, 2000; Oliveira *et al.*, 2009] exist but not resolve the problem in an optimal way. Collision detection methods can also be used, as in CSMA/Collision Detection (CSMA/CD) (illustrated on Figure 4.3), where terminals sense the common bus for a carrier before transmitting, and then listen for collisions during transmission. If a collision is detected, the transmission is immediately aborted, so if propagation delay is small relatively to packet duration, CSMA/CD diminish the impact of collisions. This distinguishes CSMA/CD from ALOHA-type protocols, which assume that feedback is made available only after packet transmission is complete.

The main drawback of these approaches is that these techniques (e.g. CSMA/CD or CSMA/CA) are difficult to implement reliably in channels with fading, shadowing effects (which enhance hidden node problems) and relatively large propagation delays, which are usually common on wireless environments. Moreover "listen while you talk" features are not feasible to the magnitude of the signal attenuation. Carrier sensing may not even be reliable due to unpredictability of the wireless networks [Rappaport *et al.*, 1996].

### 4.1.4 Reservation Schemes

Reservation based schemes like Reservation-Based TDMA try to bridge the gap between random access and fixed allocation protocols. These protocols usually employ a short reservation temporal phase followed by a longer contention free payload phase. The reservation phase is usually performed in a random access scheme, albeit polling is also viable for small terminal populations. The contention free phase consists in a scheduled fixed allocation scenario. To be effective, reservation based schemes require non-bursty traffic since they do not eliminate the problem of multiple access under varying traffic, they just shift it to the reservation phase. Note that the overall performance of this type of protocols is implicitly connected to the chosen random access protocol for the reservation phase.

## 4.2 PHY-Layer Solutions

As mentioned in the introduction of this chapter, the collision model no longer represents all the characteristics of PHY layer. It represents a simplistic PHY layer that leaves to the MAC layer the difficult task of separating users via scheduling. Signal processing techniques on the PHY layer have evolved and the problem of packet collisions can already be solved. For instance, whenever in a collision of multiple packets, one of them is received with much higher power than others in the colliding group, it can be successfully decoded by using signal capture mechanisms [Zorzi and Rao, 1994; Hajek *et al.*, 1997]. This opens the door for exploiting the capture effect to improve the performance of random access schemes.

The purpose of this section is to present a short historical overview about common PHY layer solutions that separate colliding packets at the signalling level, without the involvement of the MAC layer, although these solutions are not the main topic of this thesis.

### 4.2.1 Multiple Packet Reception

The performance of today's wireless systems is clearly more limited by interference than by any other single effect. Interference is distinguished from noise in that it is caused by other human designed devices, usually designed to use the same network. Whereas conventional noise can be overcome by increasing transmit power, overall interference is increased by this simple approach, since neighbouring devices would have to deal with even more interference than before. As a result, increasing transmit power is not a solution. The minimum required transmission power should be adopted so that the interference caused to other devices is also minimized.

To understand how signal processing can be used to increase the capacity of cellular systems, let us take into consideration a centralized system (e.g. a cellular network). It should be pointed first that the downlink and uplink scenarios have very different characteristics. In the downlink, each receiver only needs to decode a single desired signal, among adjacent intra-cell signals. In addition, it is easier to implement a scheduled and/or synchronized access. On the other hand, in the uplink the base station receiver must detect the signal of all users while suppressing other cell interference from many other sources. It is possible to conclude that the uplink scenario will need to employ a much more sophisticated signal processing detector than in the downlink scenario. This is fortunate, since the mobile units will be highly power limited and hence have limited processing power. Uplink receivers at the base station may employ multiuser receivers that are capable of robustly detecting all desired and interfering users in its cell, even in the presence of non-predictable interference from other cells. Note that some recent technologies such as multiple antenna techniques (e.g. Multiple Input Multiple Output (MIMO)) do not change this fundamental reality in cellular networks. In fact, multiple antenna systems will increase the total interference imposed on neighbour cells [Blum *et al.*, 2002; Paulraj *et al.*, 2004; Jiang and Hanzo, 2007] by increasing the data rate per user and using many transmit antennas. To cope with this increase of interference, recent inter-cell interference cancellation techniques (where cooperative Base Stations (BSs) exist) have been studied for future wireless standards [Boudreau *et al.*, 2009].

The idea of simultaneously receiving multiple users' interference is not particularly new. Most current wireless communications systems already have to cope with some degree of multiple access interference. There is no fundamental reason that collided transmissions cannot be handled and recovered by means such as coding and signal processing. The advent of multiaccess techniques employing schemes such as Code Division Multiple Access (CDMA), or more recently multiuser receivers, led to a new model conception. In 1988, Ghez et al. [Ghez *et al.*, 1988] made a fundamental change in the collision model that thereafter has been the foundation of the majority of several PHY protocols. In a few words what they propose was the following: when there are simultaneous transmissions, the reception can be described by conditional probabilities, instead of single success/failure status described in collision model. They proposed the MultiPacket Reception (MPR) model (represented by a matrix), where $C_{ij}$ is the conditional probability that, given $i$ users transmitting, $j$ out of $i$ transmissions are successful. Note that $C_{ij}$ models only $j$ successful out of $i$ transmissions but not which $j$ transmissions are successful. Therefore this model is inadequate for some practical spatial diversity systems in which one set of $j$ users may be quite different from the other. Further improvements to cope spatial diversity systems were later presented by Naware et al. [Tong *et al.*, 2004; Naware *et al.*, 2005]. Beyond describing the MPR model, Ghez et al. [Ghez *et al.*, 1988] also presented a study focusing on ALOHA under MPR model.

In the 70s Van Etten [van Etten, 1976] and Cover [Cover, 1972] foreshadowed some evolution in the multiuser receivers. One of the most active persons in the area was Verdu with extended and remarkable scientific contributions [Verdu, 1986, 1998; Ghez *et al.*, 1988]. The optimum multiuser receiver for CDMA was discovered and advanced in the mid 80s [Verdu, 1986]. Although Verdu's maximum likelihood receiver was able to optimally decode multiple users in parallel with dramatic gains, it is extremely complex to implement and computationally exhaustive [Andrews, 2005]. This led the research industry for a staggering effort by achieving suboptimal multiuser receivers with "supportable" complexity [Moshavi, 1996; Verdu, 1998].

A controversial paper from QUALCOMM [Vembu and Viterbi, 1996] identified at an early stage many of the fundamental problems with academic research on MPR. Despite the fact that this article made a number of mistakes, properly documented in [Verdu, 1997], it probably helped thereafter spur more intensive research on interference cancellation. More extensive information about the work done by Ghez et al. [Ghez *et al.*, 1988] as well as its improvements can be found in [Tong *et al.*, 2004].

### 4.2.2 Interference Cancellation: Conceptual Schemes

Along the years, many different conceptions and forms of interference cancellation have been implemented. Interference cancellation should be interpreted as the class of techniques that demodulate and/or detect desired information, and then use this information along with channel estimates to cancel received interference from the received signal [Andrews, 2005]. In other words, for the system to be classified as interference cancellation, signal processing techniques are used after detection to reduce the influence of the interference on future decisions. This conceptual idea is also justified from a theoretical point of view as well, with Cover [Cover, 1972] being the first to suggest such principle. A simple Successive Interference Cancellation (SIC) implementation with suboptimal coding was shown by Viterbi [Viterbi, 1990] in 1990, assuming accurate channel estimation and a large spreading factor.

A well-known and common successful example of interference cancellation principle, is the Decision Feedback Equalization (DFE) already presented in 2.2.4.2, which is used to cancel Inter-Symbol Interference (ISI) in frequency selective channels [Andrews, 2005]. The DFE is known to work well in practice and achieve far better performance than linear equalizers, which suffer from noise enhancement [Andrews, 2005]. Two different classes of receivers can be employed with the purpose of interference cancellation: linear and non-linear receivers. Linear techniques (e.g. Zero Forcing or Minimum Mean Square Error designs) are attractive from a complexity point of view but not very robust. They tend to amplify noise when inverting the spatial matrix channel. Non-linear interference cancellation techniques are hardest to analyse but they tend to present a better performance. Receivers equalization is outside the scope of this thesis, and more information about it can be found in [Verdu, 1998].

Interference cancellation for multi-user systems has commonly been split into two categories: the SIC and the Parallel Interference Cancellation (PIC). Over other types of multiuser receivers, both techniques have advantages, such as integrated error correction coding, demonstrated for SIC in [Viterbi, 1990] and in PIC in [Giallorenzi and Wilson, 1996]. A succinct description of the two schemes is given bellow. In a SIC [Viterbi, 1990; Patel and Holtzman, 1994] scheme, just one user is detected per stage: first, it detects the highest-power user's signal, since it is the least contaminated by multiaccess interference; then, it is detected the next strongest and so on. The received signal for that user is regenerated between the estimation of each user's transmitted data and a replica of the channel is applied to it. The regenerated signal is subtracted afterwards from the composite received signal allowing subsequent user's detection to experience less interference. The approach taken in PIC schemes is slightly different [Varanasi and Aazhang, 1990; Sawahashi *et al.*, 2002]. Instead of detecting the users in a successive sequence, they are detected in a parallel fashion: first all users are detected simultaneously (first stage); then, this first very crude estimate can be used to cancel some interference from the composite received signal; finally, a new parallel detection can be run again (second stage). This process can be repeated over several stages[6]. Some approaches have been made by some authors so as to achieve a balance between the two schemes, where groups of users are detected in parallel, and the subtraction of the interference from each group is performed in a successive manner [Varanasi, 1995].

There are a set of trade-off aspects between the SIC and PIC approaches. A high-level comparison of some of these aspects are illustrated in Table 4.1 [Andrews, 2005]. Since a vast number of different schemes exists, for each of these techniques the values for complexity order and latency should be interpreted as general trends. PIC has usually a lower latency (since the number of iterative stages are usually lower than the number of users) but a higher level of complexity (total number of detection steps) than its rival, since all the users must be detected in parallel $P$ times. SIC, has complexity and latency proportionally to the number of users, so its adoption may be critical when lower delay values are fundamental. Some conclusions taken in [Patel and Holtzman, 1994], refers that if all users are received with equal power, PIC performs better, whereas if users have unequal received power levels, SIC is definably preferable. In the end, both PIC [Divsalar *et al.*, 1998] and SIC [Agrawal *et al.*, 2005] appear to be readily able to help solving the well-known near-far[7] problem without

[6] Note the fact that SIC schemes can also have stages if an iterative receiver is considered.

[7] The near-far problem usually refers to an issue in the dynamic range of one or more stages of a receiver in which Analog-to-Digital Converter (ADC) resolution limits the range of signals. In the presence of a strong

| Technique | Complexity Order (detection steps) | Latency (stages) |
|---|---|---|
| **Successive IC** | $K$ | K |
| **Parallel IC** | $PK$ | P |
| **Turbo** | $PK$ to $2^K$ | 2P |

Table 4.1: Different interference cancellation techniques with $K$ number of users and $P$ stages.

changing the existing standards.

With the development of turbo codes in the early 1990s [Berrou *et al.*, 1993; Berrou and Glavieux, 1996], a new class of interference cancellation schemes has arisen. Usually mentioned as iterative interference cancellation (also called turbo MPR) schemes, they appeared firstly in the works of Moher [Moher, 1998], Reed et al. [Reed *et al.*, 1998], Alexander et al. [Alexander *et al.*, 1999] and Wang and Poor [Wang and Poor, 1999], where all presented a near single user performance with various CDMA configurations and receivers strategies. There has been a large amount of research on this topic since these early pioneering work [Andrews, 2005, and references therein].

Note that the entire concept of interference cancellation relies on the premise that the received signal can be reliably estimated, which requires accurate description of what was transmitted and what the channel did to that transmission. As a result, inaccurate channel estimation[8] is a problem for all approaches, SIC, PIC and iterative. Some work has been made in this topic proposing different methods of addressing this problem: iterative channel estimation [Kopbayashi *et al.*, 2001]; or power control algorithms where channel estimation error is taking into account [Andrews and Meng, 2003]. More details about this topic (which is out of scope of the current thesis), and other subjects such as power control, multipath channels, complexity and implementation issues, among others, can be consulted in [Andrews, 2005].

## 4.3 PHY-MAC Joint Resolution

As mentioned in the introduction of this chapter, both firstly Gallager [Gallager, 1985] and after Ephremides and Hajek [Ephremides and Hajek, 1998] foreshadowed as fundamental challenge the need to integrate the benefits from information-theoretic techniques by choosing a proper model that interfaces the PHY layer and network layers. In an effort to achieve this performance improvement there has been an increasing interest in cross-layer protocols.

signal, the receiver must reduce its gain to prevent ADC saturation, which causes the weaker signal to fall into the noise of the ADC.

[8]In fact, channel estimation is a critical factor in innumerable wireless communication systems.

Generally speaking, cross-layer design refers to protocol design done by actively exploiting interactions between various layers of the network stack so as to obtain performance gains. However, as debated in [Kawadia and Kumar, 2005] and [Srivastava and Motani, 2005] such cross-layer design can run at cross purposes with sound and longer-term architectural principles and lead to various negative consequences: it is generally agreed that a good architectural design can spark proliferation and longevity and cross-layer design can compromise that; it can also, if one is not careful, produce unintended cross-layer interactions leading to undesirable consequences on overall system performance; the luxury of designing a protocol in isolation is lost and the effect of any single design choice on the whole system needs to be considered.

Nevertheless, three main reasons motivate designers to violate the layered architectural in a wireless network [Srivastava and Motani, 2005]:

- wireless links create unique problems for protocol design that cannot be handled well in the framework of layered architecture [Shakkottai *et al.*, 2003];
- wireless networks open the doors for opportunistic communication that cannot be properly exploited in a strictly layered design (e.g. the time varying link quality allows opportunistic usage of the channel by dynamically changing transmission parameters according to the variations);
- the wireless medium offers some new modalities of communication and cooperation (e.g. the advent of MPR, cooperative interference cancellation, etc.).

All in all, cross-layer design represents an open challenge that can bring many advantages specifically when wireless networks may be on the cusp of massive proliferation, but it needs to be considered in a holistic way.

**The First Step: Capture Effect**

The advent of wireless networks, sparked the interest in cross-layer designs by the need to provide a greater level of adaptivity to variations of wireless channels, specifically joint and collaborative designs between PHY and MAC layers. The first hybrid schemes to be proposed, studied the perfect capture model on existing MAC protocols [Metzner, 1976; Abramson, 1977; Habbab *et al.*, 1989] in which the packet with the strongest received power is captured. A more accurate Signal-to-Interference-plus-Noise Ratio (SINR) based capture model, in which a single packet is captured if the SINR is above a given threshold, was studied in [Arnbak and van Blitterswijk, 1987; Lau and Leung, 1992]. The performance of the FCFS algorithm in channels with capture was analysed in [Sidi and Cidon, 1985]. Subsequent works [van der Plas and Linnartz, 1990; Zorzi, 1998; Luo and Ephremides, 2002] concerning the capture model extended it by studying the effects of power levels, channel effects and retransmission control respectively. Zorzi and Rao [Zorzi and Rao, 1994; Nguyen *et al.*, 2006; Zorzi and Rao, 2006] analysed an ALOHA system with Rayleigh fading and capture. The stability condition for a finite-population slotted ALOHA protocol in a CDMA network is analysed in [Sant and Sharma, 2000]. Recent work about capture focused mostly on studying its effects on IEEE 802.11 networks [Hadzi-Velkov and Spasenovski, 2002a,b, 2003; Nyandoro *et al.*, 2007; Garetto *et al.*, 2008].

### The Advent of MultiPacket Reception Solutions

Most of the aforementioned work is focused on systems in which at most a single packet can be captured at any given time. Research regarding MPR systems, i.e. where more than a single packet can be captured, was triggered by Ghez et al. [Ghez *et al.*, 1988, 1989] in which the channel was modelled such that the number of successfully received packets is a random variable which depends on the number of simultaneous transmissions. In their work, the effect of this MPR model on the ALOHA protocol with adaptive transmission probabilities was studied. MacKenzie and Wicker [MacKenzie and Wicker, 2003] revisited the model of Ghez et al. [Ghez *et al.*, 1988] and presented a game-theoretic approach for a MPR slotted ALOHA. In 2005 [Naware *et al.*, 2005], a more generic MPR model was presented. The MPR model of Ghez et al. [Ghez *et al.*, 1988] assumes a symmetrical model where users are indistinguishable as mentioned earlier. Naware et al. proposed a general asymmetric MPR model and studied the stability and delay of slotted ALOHA based random access systems.

Zhao and Tong [Zhao and Tong, 2000] proposed an algorithm that is capable of separating colliding packets when exploiting channel diversities and known symbols embedded in data packets without assuming the knowledge of propagation channels and signal waveforms. Finally, a study involving the SINR capture model in which more than a single packet can be captured is addressed in [Hajek *et al.*, 1997].

When MPR systems started to be considered, Tong et al. [Tong *et al.*, 2001] gave an overview about the interaction between the PHY and MAC layers, explicitly considering the impact of MPR technique on the design of future protocols. Such overview was later extended and improved [Tong *et al.*, 2004] by considering random access issues.

### Joint Resolution Techniques

Different approaches have been proposed where MAC behaviour takes into consideration the MPR challenges. A complete set of studies on several collision resolution algorithms for networks with MPR capability following the model proposed in [Ghez *et al.*, 1988] can be consulted in [Likhanov *et al.*, 1993]. Celik et al. [Celik *et al.*, 2008, 2010] performed an evaluation of simple generic MAC protocols when the network is capable of MPR. They also proposed an alternative backoff algorithm that decreases the transmission probability after a success and increases it after a failure, improving the algorithm's fairness for heterogeneous distances to the receiver.

Optimal centralized scheduling algorithms in TDMA networks with multiple channels and MPR capability were studied in [Chlamtac and Farago, 1994]. These algorithms provide optimal throughput for saturated loads. Collision avoidance can be achieved for variable load conditions by applying a channel slot reservation mechanism which can be controlled either by the sender, e.g. [Eisenberg *et al.*, 2007], or by the receivers, e.g. [Zhao and Tong, 2003]. In the sender's case, Eisenberg et al. [Eisenberg *et al.*, 2007] assign a pair of RTS / CTS bits to each MT, and the BS controls who transmits in the next data slot, generating significant overhead for a large number of MTs. On the other hand, Zhao and Tong [Zhao and Tong, 2003] proposed the service room protocol designed explicitly for general MPR channels where the receiver controls the slot reservation mechanism. The protocol has the objective to accommodate groups of users with different delay requirements avoiding unnecessary empty slots for light traffic and excessive collisions for heavy traffic. The BS selects the set of transmitters for each slot, based on the information (channel history and the Quality-of-Service (QoS) constraints) about their previous slot usage. The main drawback of this scheme is its computational cost. A much simpler protocol was proposed with similar performance in [Zhao and Tong, 2004]. In [Gau and Chen, 2006] Gau et al. proposed a multicast polling scheme based on channel state predictions. A centralized implementation was considered relying on the unique access point to decide which nodes could transmit packets in a time slot.

Another alternative is to let collisions happen, and resolve them by employing ALOHA algorithms, Tree Algorithms or CSMA schemes. Adireddy and Tong [Adireddy and Tong, 2005] considered the effect of having knowledge of fading at the transmitters on the design of ALOHA. It is shown that significant gains can be obtained by allowing the transmission probability to be a function of the channel state (as opposed to conventional power control). Luo and Ephremides [Luo and Ephremides, 2006a] considered a standard MPR channel and characterized the relations among the throughput region of random multiple access, the capacity region of multiple access without code synchronization and feedback, and also the stability region of finite terminal ALOHA protocol. In [Gau, 2006] Gau presents an analytical approach to derive the exact value of saturation throughput of Slotted ALOHA in an interference dominating wireless ad-hoc network. Two years later, the non-saturated case was addressed in [Gau, 2008]. Delay and throughput is analysed for a finite-user slotted ALOHA with MPR, when error free conditions and random traffic are assumed. More recently, Lotfinezhad et al. [Lotfinezhad *et al.*, 2008] derived the optimal retransmission probabilities for slotted ALOHA in wireless sensor networks with MPR.

As an alternative to the more common ALOHA solutions, Gau and Chen [Gau and Chen, 2008] studied the throughput and delay of the tree algorithm for MPR and packet queues with capacity for one and two packets. Gau in [Gau, 2011] recently proposed a tree/stack splitting with remainder algorithm for distributed MAC in a MPR wireless network. Regarding CSMA solutions, Gau [Gau, 2009] in 2009 proposed probability models for performance analysis of the slotted non-persistent CSMA protocol in wireless networks with MPR.

#### Playing With Channel State Information

Another approach of joint resolution, is the use of channel state information by MAC layer [Knopp and Humblet, 1995; Tse and Hanly, 1998; Viswanath *et al.*, 2002; Qin and Berry, 2003; Adireddy and Tong, 2005; Huang *et al.*, 2008]. Channel fluctuations at the PHY layer provide valuable information. For example, if a user is in a deep fade and its transmission has little chance of being decoded successfully, then it would be better if the user does not transmit and waits for a better channel state. Exploiting channel state information induces multiuser diversity, and the performance improves with the increase of the number of users [Tse and Hanly, 1998; Viswanath *et al.*, 2002]. Qin and Berry [Qin and Berry, 2003] proposed the use of channel state information to vary the transmission probability in ALOHA. They analysed the system under the collision model and demonstrated the effect of multiuser diversity on the throughput.

#### 802.11-like Solutions

More recently, there has been extensive contributions by Zhang et al. [Huang *et al.*, 2008; Zhang *et al.*, 2009; Zhang, 2010; Zhang *et al.*, 2010] focusing MPR in wireless local networks. A Channel State Information (CSI) based random access protocol was proposed by Zhang et al. [Huang *et al.*, 2008] that takes advantage of a CDMA MPR system in IEEE 802.11-like wireless local area networks. A study on how such MPR capability can enhance the capacity of future wireless networks is presented in [Zhang *et al.*, 2009]. In addition, a joint MAC-PHY layer protocol for an IEEE 802.11-like wireless network that incorporates advanced signal processing techniques to implement MPR is also addressed. Since the wireless channel is often not properly explored in MPR wireless local area networks, a novel multi-round contention random access protocol that addresses this issue is presented in [Zhang, 2010].

Both saturation and non-saturation throughput under mean bounded values of delay and jitter are evaluated in [Zhang *et al.*, 2010] when IEEE 802.11-like wireless networks are considered.

#### Revisiting Challenges

Several other works regarding MPR capabilities were presented, where different MAC challenges are addressed. Luo and Ephremides [Luo and Ephremides, 2006b] extended their previous work of [Luo and Ephremides, 2002] and showed that in MPR networks, the optimal throughput can be achieved when employing a single power level system. Garcia-Luna-Aceves et al. studied the effect of various MPR models on the network capacity [Garcia-Luna-Aceves *et al.*, 2007], energy efficiency [Wang and Garcia-Luna-Aceves, 2008], and also provided a joint routing and scheduling approach [Wang *et al.*, 2008b]. Finally, Guo et al. [Guo *et al.*, 2009] analysed the capacity region of a MPR scheme CDMA-like, wherein each node can decode at most $k$ simultaneous transmissions within its receiving range.

Among the above proposals, two schemes with joint PHY-MAC resolutions stood out. One of them, the Successive Interference Cancellation in a Tree Algorithm (SICTA) created a new benchmark for random access maximum stable throughput. The other, the Network-assisted Diversity Multiple Access (NDMA) scheme, lead to the creation of a new class of protocols. A closer look on the aforementioned proposals is given below.

### 4.3.1 SICTA - Successive Interference Cancellation in a Tree Algorithm

When the traffic is discontinuous and bursty, dynamic allocations based on random access schemes such as ALOHA, CSMA or tree algorithms already mentioned in section 4.1, usually provide interesting delay-throughput trade-off characteristics, since average load traffic is low. With the advent of interference cancellation techniques as well as MPR schemes, previous protocols that had been proposed for the classical single packet receivers, have been also applied to the MPR scenarios. Taking into consideration the fact that tree algorithm approach falls into the set of limited collision algorithms which guarantees an upper bound transmission delay, Yu and Giannakis [Yu and Giannakis, 2005, 2007] developed a protocol named SICTA. SICTA relies on SIC to take advantage of collided packets in a conventional tree algorithm. With this cross-layer design approach, SICTA spread SIC benefits into MAC layer.

In traditional tree algorithms, collided packets are discarded with no attempt to extract useful packet information. In contrast, SICTA retains them for future reuse. In few words, it exploits the structure of a conventional tree algorithm and employs SIC to resolve collisions.

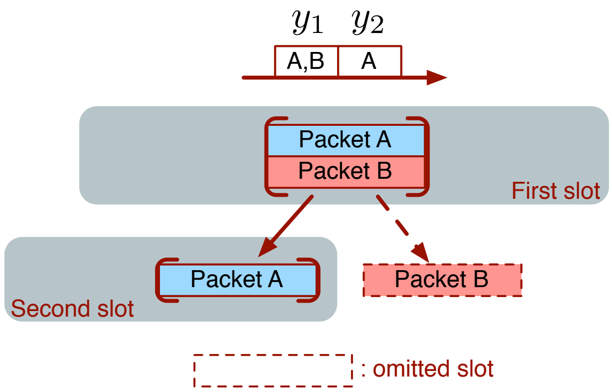



Figure 4.4: Example of the tree concept with SICTA protocol.

In order to illustrate its behaviour, consider the example of Figure 4.4. Let $\mathbf{y}_t$ denote the received signal vector at the end of slot $t$. From the received signal at the second slot, $\mathbf{y}_2$, the receiver decodes the packet A and subsequently, the interference from packet A to packet B in the first slot is cancelled to obtain $\tilde{\mathbf{y}}_1 = \mathbf{y}_1 - \mathbf{y}_A$. Based on $\tilde{\mathbf{y}}_1$, the packet B can be recovered as well. Notice that, in a quick comparison to traditional tree algorithms previously exemplified in Figure 4.1, only two slots are sufficient to decode the packets rather than three. With this approach, SICTA can reduce the number of transmission slots, which clearly offers the potential to improve throughput. As a result, a maximum stable throughput of 0.693 is achievable, which exceeds the bound 0.587 [Mikhailov and Tsybakov, 1981] previously defined as the maximum possible throughput of random access protocols under ternary feedback, since collided packets are not discarded.

In spite of establishing a new benchmark concerning the maximum throughput for random access protocols under ternary feedback when errors are not prevailing, SICTA performance may be hold up by the deadlock problem arising due to error propagation and/or channel fading. Let us considered the example of Figure 4.5. Two packets collide at the first slot and the received signal vector is $\mathbf{y}_1 = \mathbf{x}_A + \mathbf{x}_B + \mathbf{n}$, where $\mathbf{x}_A$ and $\mathbf{x}_B$ denote packets A and B, and $\mathbf{n}$ denotes the noise vector. At the end of the second slot, packet A is decoded and then, cancelled to obtain $\tilde{\mathbf{y}}_1 = \mathbf{x}_B + \mathbf{n}_A + \mathbf{n}$. If $\mathbf{n}_A + \mathbf{n}$ is sufficiently large to preclude the receiver from recovering the packet B, then $\tilde{\mathbf{y}}_1$ is incorrectly assumed to be a new collision of packets. Consequently, the BS requires the remaining fake multiple users to split. After the remaining packets have been decoded, the receiver cannot discern the remaining noise terms from a real collision and for this reason, it forces non-existing users to infinitely split process, until an external criterion terminates it. Note that besides noise, also channel fading effects may lead to deadlock scenarios.

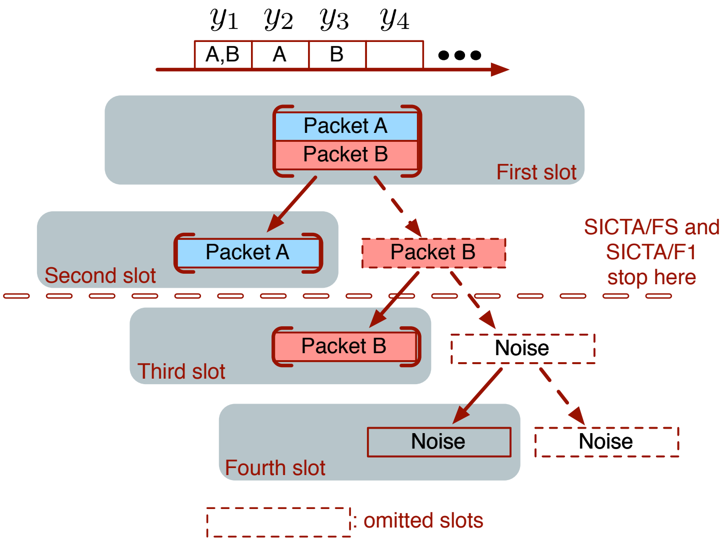



Figure 4.5: Example of the deadlock problem.

To handle the deadlock problem, Wang et al. [Wang *et al.*, 2007] proposed the SICTA with First Success (SICTA/FS) in which combines the SICTA protocol with the first success concept, already applied in a different context by [Garces and Garcia-Luna-Aceves, 1997]. The SICTA/FS protocol behaves as SICTA until it arrives at its first success. At that instant, a single packet is decoded and the SIC is employed to extract as many extra packets as possible from previous collisions. However, even though packets may remain unresolved, the SICTA/FS terminates the collision resolution process and starts a new one. The SICTA/FS is in fact a truncated version of the SICTA as exemplified in Figure 4.5. Therefore, although SICTA/FS cannot achieve a maximum stable throughput as high as the SICTA, it can provide a value around 0.6, which is close to the SICTA value.

Further examinations of the SICTA/FS still found another issue concerning deadlock stages [Wang *et al.*, 2008a]. If a single idle slot is incorrectly interpreted as a collision a new deadlock problem may arise. To completely avoid the deadlock, an evolution of SICTA/FS was presented, the SICTA with First 1 feedback (SICTA/F1) [Wang *et al.*, 2008a]: it makes use of binary feedback only (collision represented by $e$ and no-collision represented by 1); and in addition to SICTA/FS it truncates the collision resolution not only upon a success but also after an idle. As in SICTA/FS, SICTA/F1 relies on SIC to separate packets from the reserved collided packets. SICTA/FS can afford the maximum stable throughput in Additive White Gaussian Noise (AWGN) channels at the price of a large packet loss. To enable SICTA/FS and SICTA/F1 gain in a system where minimal packet loss is allowed, a Binary Exponential Backoff (EB) (BEB) algorithm is employed and studied in [Wang *et al.*, 2008a]. Analysis and simulations reveal that BEB-SICTA/FS can achieve 0.6 of maximum stable throughput without packet loss, and BEB-SICTA/F1 achieve a maximum of 0.55.

However, it should be mentioned that SICTA requires the receiver to store soft information of the received signal of all previously undecodable messages. In addition this implies that decoding successively many packets that have collided over time can lead to long delays. Another important aspect that needs to be emphasized, is the feedback message size. Most protocols use a set of 2 bit feedback messages: idle (0), success (1) and collision ($e$) messages. Instead, SICTA's set of feedback messages[9] consists of (0), ($e$) and the number of packets that were finally resolved in the previous time slot, so if this number is large it requires allocation of more bits for feedback signalling. An alternative to SICTA's architecture was proposed in [Yim *et al.*, 2009], where a simple multiple access paradigm that uses CSI is employed at the transmitter to control the each node's transmit power. MPR is achieved for a small number of users by carefully setting two power levels so as to using SIC at the receiver, and thus achieving a theoretical stable throughput higher than SICTA, of 0.793 packets/slot.

### 4.3.2 Network-assisted Diversity Multiple Access Protocols

Random access schemes rely on retransmissions to resolve current collisions, and most of the effort made in the random access literature has been essentially on retransmission schemes that minimize future collisions. In those schemes, the throughput penalty incurred by collisions cannot be eliminated unless some way is devised to extract useful information from that collided packets, since the collided packets are typically discarded when a collision occurs. Tsatsanis et al. [Tsatsanis *et al.*, 2000] proposed a novel approach to the collision resolution problem to recover packets from multiple collided packets and thus boost throughput, so-called Network-assisted Diversity Multiple Access (NDMA). The NDMA class of protocols essentially consists in not discarding collided packets and relying on proper retransmissions and signal separation principles to resolve collisions. In particular, if $Q$ users collide in a given time slot, they repeat their transmission for a total of $Q$ times so that $Q$ copies of the collided packets are received. Then, the receiver has to resolve a $Q \times Q$ source mixing problem and separate each individual user. All in all, it requires only $Q$ slots to transmit $Q$ colliding packets, so NDMA protocols could not introduce throughput penalty in the presence of collisions at least in theory.

This technique exploits diversity combining ideas to separate the collided packets, but explores MAC layer functionality to create diversity at the physical layer. In the end, the

[9]Except SICTA/F1 which uses only binary feedback: collision ($e$) and no-collision (1).

result can be viewed as a “spreading on demand”, wherein demand depends on the collision multiplicity determined by the BS.

In NDMA protocols the feedback is $0/1/e$ as in ALOHA-like protocols: 0 clears all terminals for transmission; 1 enables those that transmitted in the previous slot and disables all others; $e$ signals error in packet recovery. In general, when the BS detects a collision it sets feedback to 1, which induces another collision of the same packets in the following slot. After sufficient number of retransmissions, the BS sets feedback to 0 if all packets were properly decoded or $e$ if an error occurred in the decoding process. The set of slots that comprises the first transmission and subsequent retransmissions is denoted as a collision resolution epoch.

In the end of a collision resolution epoch, when the receiver has the $T$ transmissions of the $Q$ colliding packet, the signal received can be described as

$$\mathbf{Y} = \mathbf{H} \times \mathbf{S} + \mathbf{W}. \tag{4.1}$$

$\mathbf{Y}$ is the received data matrix and can be written as

$$\mathbf{Y} = \begin{bmatrix} Y_1^{(1)} & Y_2^{(1)} & \cdots & Y_N^{(1)} \\ Y_1^{(2)} & Y_2^{(2)} & \cdots & Y_N^{(2)} \\ \vdots & \vdots & \ddots & \vdots \\ Y_1^{(T)} & Y_2^{(T)} & \cdots & Y_N^{(T)} \end{bmatrix}, \tag{4.2}$$

where $N$ denotes packet length in symbols and $T$ the number of total transmissions. $\mathbf{H}$ is the mixing matrix denoting the complex gain of the symbols due to channel effects of the $q$th packet during the $t$th transmission and can be written as

$$\mathbf{H} = \begin{bmatrix} H_1^{(1)} & H_2^{(1)} & \cdots & H_Q^{(1)} \\ H_1^{(2)} & H_2^{(2)} & \cdots & H_Q^{(2)} \\ \vdots & \vdots & \ddots & \vdots \\ H_1^{(T)} & H_2^{(T)} & \cdots & H_Q^{(T)} \end{bmatrix}, \tag{4.3}$$

where $Q$ is the number of collided packets (which is a random variable). $\mathbf{S}$ is the collided packet signal matrix whose $q$th row is the packet of the $q$th terminal and can be described as

$$\mathbf{S} = \begin{bmatrix} S_1^{(1)} & S_2^{(1)} & \cdots & S_N^{(1)} \\ S_1^{(2)} & S_2^{(2)} & \cdots & S_N^{(2)} \\ \vdots & \vdots & \ddots & \vdots \\ S_1^{(Q)} & S_2^{(Q)} & \cdots & S_N^{(Q)} \end{bmatrix}. \tag{4.4}$$

$\mathbf{W}$ is the white Gaussian noise matrix and can be written as

$$\mathbf{W} = \begin{bmatrix} W_1^{(1)} & W_2^{(1)} & \cdots & W_N^{(1)} \\ W_1^{(2)} & W_2^{(2)} & \cdots & W_N^{(2)} \\ \vdots & \vdots & \ddots & \vdots \\ W_1^{(T)} & W_2^{(T)} & \cdots & W_N^{(T)} \end{bmatrix}. \tag{4.5}$$

The model in (4.1) is a classical signal separation problem where $\mathbf{S}$ needs to be recovered from $\mathbf{Y}$ and the number of collided packets $Q$ has to be detected. The decoding steps consists in: 1) detecting $Q$; 2) determining the appropriate $T$; and 3) estimating $H$. $H$ needs to be a full column rank[10] [Tsatsanis *et al.*, 2000], as a result, $T \geq Q$ transmissions are needed to ensure that.

#### 4.3.2.1 A Training Based Collision Resolution

In order for the BS to discriminate the users, an address field is required in the packet containing a unique ID sequence for each MT. This terminal identifying sequence is know to the BS and can be embedded in the header of each packet. A simple collision detection mechanism is possible if orthogonal IDs are used and the BS employs matched filters corresponding to these IDs [Tsatsanis *et al.*, 2000]. Hence, the joint terminal detection problem can be decoupled into independent single terminal detection problems. $K$ is set to the total number of detected users.

Note the fact that NDMA is only suitable if the channel between every MT and the BS is frequency flat and constant over each packet slot but different from slot to slot [Tsatsanis *et al.*, 2000]. An evolution of [Tsatsanis *et al.*, 2000] was later proposed by Zhang and Tsatsanis [Zhang and Tsatsanis, 2002] where NDMA was extended to multi-path time dispersive channels. In a frequency selective channel, orthogonality is maintained by using complex exponential ID sequences and a cyclic prefix or guard time [Zhang and Tsatsanis, 2002; Dimic *et al.*, 2004]. Since [Zhang and Tsatsanis, 2002] considers a time-domain receiver implementation, its complexity can be very high for severely time-dispersive channels. Moreover, due to the linear nature of the receivers of [Tsatsanis *et al.*, 2000; Zhang and Tsatsanis, 2002], the residual interference levels can be high and/or significant noise enhancement can be obtained. To address these issues, promising frequency-domain multipacket detection schemes that allow efficient packet separation in the presence of successive collisions were proposed in [Dinis *et al.*, 2007, 2009b]. These receivers are suitable to severely time dispersive channels

[10] The column rank of a matrix S is the maximum number of linearly independent column vectors of S. S matrix that has a rank as large as possible is said to have full rank; otherwise, the matrix is rank deficient.

and do not require different channels for different retransmissions.

**Evolutions on NDMA protocols**

Several improvements have been proposed since the first appearance of the NDMA protocol, with a training based collision resolution. In finite Signal-to-Noise Ratio (SNR) environments, two kinds of errors may result in packet corruption: when the BS incorrectly determines the collision multiplicity, i.e. the number of packets involved in the collision, and thus schedule an insufficient number of retransmissions resulting in failure to recover some or all the packets; or when in the presence of noise. An effort to cope with the effects of a finite SNR environment by employing a Automatic Repeat reQuest (ARQ) control is presented in [Zhang and Tsatsanis, 2000].

One of the most recently active group in this topic has been Samano-Robles et al. [Samano-Robles *et al.*, 2006, 2007, 2008a,b,c, 2009]. Unfavourable joint detection conditions at finite SNR can degrade the throughput performance. To mitigate this problem, Samano-Robles et al. [Samano-Robles *et al.*, 2006] proposed in 2006 a constant $p$-persistent access mode for NDMA protocol. This $p$-persistent mechanism is optimized to reduce the average number of colliding users so as to optimize the throughput at the expense of a small delay degradation. In [Samano-Robles *et al.*, 2007] it is presented a new resource allocation mechanism which is designed to improve the multiuser detection in NDMA protocols. The mechanism consists on adjusting the probability of false alarm of each user controlling the average number of active time-slots per user. By adjusting this PHY layer parameter, it is possible to regulate the typical delay degradation of a fixed resource allocation scheme, ensuring different levels of QoS. An infinite user population model with finite traffic load for NDMA protocols was presented in [Samano-Robles *et al.*, 2008b], where the effects of packet decoding errors as well as of imperfect detection are considered.

A new multiaccess protocol that combines the concept of splitting tree algorithms, MPR and NDMA is suggested in [Samano-Robles *et al.*, 2008c]. The proposed algorithm calculates the optimum set of users allowed to retransmit at each one of the following time-slots and two sub-optimal solutions were addressed:

(1) an enhanced version of NDMA (NDMA protocol combined with some extra MPR signal processing techniques) is presented, where after each received transmission the system attempts to decode the packets using MPR techniques;

(2) a fair splitting algorithm with equal probability is used by the users that were not successfully decoded in the current time slot, thus reducing the number of colliding users in the following time slot.

Since detecting the collision multiplicity in NDMA protocols is critical, two improved schemes of the previous training based solution of [Tsatsanis *et al.*, 2000] are proposed in [Samano-Robles *et al.*, 2008a]. The first solution consists of a set of cooperative nodes that independently collect information about the collision multiplicity and relay it afterwards to the BS. This solution is indicated for scenarios with low traffic loads and deep and long term fades. In comparison with conventional NDMA systems, in which the collision multiplicity estimation is only based upon the first received collision, the second solution consists in a sequential detection approach in which the estimate of the collision multiplicity and the detection threshold are updated upon the reception of each received retransmission. Deep and short term fades with high traffic loads are particularly suitable for this case. More recently [Samano-Robles *et al.*, 2009], it was presented a detailed study on stability, throughput and delay properties of asymmetrical non-blind NDMA protocols under an imperfect detection assumption.

The orthogonality assumption for the ID sequences is responsible for a major drawback, since the length of the user ID sequence is required to grow linearly in the number of the MTs and not logarithmically as usual [Zhang *et al.*, 2002]. Therefore, for a large user population (active plus inactive MTs), the introduced overhead could be substantial.

#### 4.3.2.2 Blind NDMA: a Blind Signal Separation Method

To overcome the difficulties mentioned in the above paragraph, Zhang et al. [Zhang *et al.*, 2002] explored the applicability of blind signal separation methods (i.e. methods that do not require a known ID sequence) to the collision resolution problem. The solution proposed is to employ exponential phase modulation at the packet level, in such a way that a structured mixing matrix is constructed through the retransmissions. The signature of each colliding MT's packet has a Vandermonde form and can be blindly estimated through rotational invariance techniques [Zhang *et al.*, 2002]. This proposed method is less computationally demanding since it is proportional to the number of colliding packets, in contrast to the method of [Tsatsanis *et al.*, 2000] that it is proportional to the total number of users in the system. Collision multiplicity is estimated at the BS using rank detection [Zhang *et al.*, 2002] and requires $T = K + 1$ slots to be executed.

#### Other Blind Solutions

Two additional blind solutions for NDMA schemes can be found in the literature. One, is the Independent Component Analysis - NDMA (ICA-NDMA) [Ozgul and Delic, 2006], a blind collision multiplicity detection technique that accomplishes collision resolution through independent component analysis (ICA, which coined the name) and that does not need any phase control like in Blind NDMA (BNDMA). However, the detection algorithm needs at least $(K + 2)$ slots when $K$ packets are involved in the collision, which results in a greater packet delay compared to BNDMA. The second solution is an improvement of ICA-NDMA protocol by adding cooperative transmission [Yao *et al.*, 2010]. In the ICA - Cooperative NDMA (ICA-CoopNDMA), when $K$ nodes collide, a cooperative transmission mechanism is triggered to collect different superposition[11] of the $K$ nodes' packets in the following slots.

#### Evolutions Regarding Blind NDMA protocol

Two distinct works have been proposed upon BNDMA protocol by Dimic et al. [Dimic and Sidiropoulos, 2000; Dimic *et al.*, 2003]. In [Dimic and Sidiropoulos, 2000] Dimic and Sidiropoulos proposed a multicode multicarrier scheme built upon BNDMA protocol called Multicode BNDMA, which allows users to transmit multiple packets per slot where each one is modulated by a random digital carrier. The stability region of BNDMA was formally analysed in [Dimic *et al.*, 2003]. A simpler and more general steady-state analysis was also presented, which embraces the Multicode BNDMA scheme.

#### 4.3.2.3 Different x-NDMA Evolutions

Since the emerge of Tsatsanis et al. [Tsatsanis *et al.*, 2000] work, many other improvements and studies have been presented concerning NDMA aspects or variations. The Frequency Hopping - NDMA (FH-NDMA) protocol was introduced in [Dimic and Sidiropoulos, 2003] proposing a wireless network setup that provides scalability and robustness to node failures. Dimic and Sidiropoulos proposed a semi-ad-hoc scenario where nodes are dynamically organized into clusters and where each of them operates under NDMA protocol with a pseudo random frequency hopping sequence on a per packet slot basis. All nodes within a cluster are synchronized and use the same frequency hopping sequence.

Madueño e Vidal were the first to address the applicability of the NDMA protocol in the pure ad-hoc case[12] [Madueno and Vidal, 2005a]. Guidelines for this design were presented as well as an investigation of the benefits of retransmission combining for broadcasting. As a result, it was proposed the Feedback Free - NDMA (FF-NDMA) protocol which does not require feedback from the receiver. The solution went through forcing every node to transmit

[11] The superposition principle, states that, for all linear systems, the net response at a given place and time caused by two or more stimuli is the sum of the responses which would have been caused by each stimulus individually.

[12] Dimic and Sidiropoulos in [Dimic and Sidiropoulos, 2003] considered clusters with a lead node that acts as a BS.

the same packet over a fixed number of $R$ slots, and thus allowing resolving collisions of up to $R$ packets without feedback. Different values of $R$ in different setups were also studied. A cross-layer analysis of the FF-NDMA for both Single Input Single Output (SISO) and MIMO configurations with orthogonal space time block codes is given in [Madueno and Vidal, 2005b]. In the SISO case, collision resolution is performed by repeating collided packets whereas in MIMO case both time and spatial diversity are exploited.

Finally, an hybrid solution that uses NDMA principals with Incremental Redundancy Automatic Repeat reQuest (IR-ARQ) protocols (see section 2.4.1) was proposed in [Nam *et al.*, 2007]. It tailors the IR-ARQ protocol to jointly combat the effects of user collisions, multipath fading and channel noise through the time diversity, with repetitions.

# CHAPTER 5

# A Network Diversity Multiple Access protocol: an efficient way of coping with lost packets due to collisions

As mentioned in previous chapters, Diversity-Combining (DC) schemes are not suitable to cope with packets lost due to collisions. A viable solution for this case consists in employing MultiPacket Reception (MPR) mechanisms. These mechanisms are proposed for scenarios where Mobile Terminals (MTs) contend to access the medium with parallel and simultaneous transmissions.

Most MPR approaches adopted for contemporary wireless access technology employ Code Division Multiple Access (CDMA) (e.g. Universal Mobile Telecommunication System (UMTS), etc.) and take advantage of the redundancy inherent to spreading, to separate the packets involved in the collisions. This means that the system needs to be designed for the worst-case scenario (i.e., the spreading factor needs to be as high as the maximum number of packets expected to be involved in the collisions). Therefore, there is a significant waste of resources when just a few packets are involved in a collision (which is the most common case), leading to significant performance degradation.

A different approach is adopted in Network-assisted Diversity Multiple Access (NDMA) [Tsatsanis *et al.*, 2000; Zhang and Tsatsanis, 2002] where the redundancy is handled over time: if $Q$ MTs are involved in a collision they retransmit their packets $Q-1$ times. The packet rate in this system is difficult to calculate because it depends on the value $Q$ of colliding MTs. For a certain $Q$, the packet rate would be $1/Q$th of the total available bandwidth (which might be roughly comparable to a CDMA system with a spreading factor of $Q$, in terms of efficiency).

A promising NDMA scheme was proposed in [Dinis *et al.*, 2007] for Single Carrier with Frequency Division Equalizer (SC-FDE) systems which allows an efficient packet separation. The receiver can be regarded as an iterative multi-packet detector with interference cancellation, contrarily to [Tsatsanis *et al.*, 2000] and [Zhang and Tsatsanis, 2002], where linear receivers are considered. Later on, in [Dinis *et al.*, 2009b] it is added to the previous scheme a turbo multi-packet receiver where, as in turbo equalizers [Tuchler and Hagenauer, 2000, 2001; Gusmão *et al.*, 2007; Benvenuto *et al.*, 2010], the channel decoder outputs are used in the feedback loop so as to improve equalization performance, as well as inter-packet interference cancellation. The receiver has relatively low complexity in severely time-dispersive channels, since it allows Fast Fourier Transform (FFT) based implementations [Dinis *et al.*, 2009b]. To be effective, this technique requires uncorrelated channels for different retransmissions. Since this is not practical in many systems, it is used the Shifted Packet (SP) technique already presented in section 3.1.2 for retransmissions where the frequency-domain block to be transmitted has different cyclic shifts for different retransmissions. By employing this technique, the receiver does not require uncorrelated channels for different retransmission attempts and (as with DC schemes) reduces the Packet Error Rate (PER) since it combines the signals associated to all $Q$ transmissions.

Although several improvements have been proposed in the NDMA scheme (see section 4.3.2), few are related with practical implementation issues. Considering a real implementation of the NDMA scheme and imagining that $Q$ packets are received at the same time, if $Q$ is too high, difficulties in the signal separation could arise (complexity and estimation issues), making the receiver infeasible. Ideally, in a practical implementation a maximum value for $Q$, $Q_{max}$, should exists so as to limit aspects like the complexity of the receiver and the overhead of the estimation. As a result, a Medium Access Control (MAC) protocol should exist that limits the $Q$ value, i.e. the number of packets involved in a collision, to a predefined maximum value of $Q_{max}$. Additionally, by controlling the number of MTs competing in a given channel and the number of retransmissions involved in each collision, the MAC protocol can properly coordinate the access in order to optimize the system performance. Note that, although the number of concurrent MT's transmissions should always be as close as possible to the maximum allowed ($Q_{max}$) to achieve higher system performance (lowering the PER), it should not exceed this value to avoid making the collision unrecoverable. This problem is particularly challenging if the number of MTs allowed to compete, $J$, is much higher than $Q_{max}$.

In this chapter it is considered a Base Station (BS) with MPR capability based on the NDMA scheme proposed in [Dinis *et al.*, 2007, 2009b] that is able to separate simultaneous packet receptions. The MAC layer is designed to control the number of accessing MTs, $Q$, and all the mechanics of the retransmissions, which are an essential routine in the NDMA scheme. In order to control the MT access in the medium, it is proposed a $p$-persistent Slotted ALOHA MAC protocol designed to cope with a number of MTs, $J$, higher than $Q_{max}$. It is considered a generic backoff algorithm and it is presented a suitable analytical model for the system's behaviour for saturated load (when all nodes always have packets to transmit) and for homogeneous unsaturated load. The model takes the physical layer characteristics into account, namely the variation of the PER and packet transmission time with the number of packets involved in a collision, as well as the number of packet retransmissions due to detection errors. Both half-duplex (slotted non-periodically) and full-duplex (slotted periodically) scenarios are characterized.

The work presented in this chapter was accepted for publication, firstly considering saturated scenarios [Pereira *et al.*, 2009a,b], and secondly considering unsaturated scenarios [Pereira *et al.*, 2010a].

## 5.1 System Overview

In this chapter it is considered the uplink transmission in structured wireless systems employing SC-FDE schemes where a set of $J$ MTs send data to a BS[1]. The BS runs the multi-packet detection algorithm in real time since it is normally denoted as a resource rich device. MTs have a half-duplex radio and employ a $p$-persistent Slotted ALOHA algorithm to send data packets using the time slots defined by the BS (for the sake of simplicity, it is assumed that the packets associated to each user have the same length, $L_{data}$). The interaction between MTs and the BS has two phases: an association phase using a dedicated control channel before the data transmission phase using a shared data channel. The BS uses the downlink channel to broadcast the MTs access probability ($p_c$ for the $c$th transmission attempt) and, possibly, to force packet retransmissions or block the transmission of new packets in the next slot. It is assumed that different data packets arrive simultaneously and perfect power control and time advance mechanisms exist, able to compensate a different attenuation and propagation times. It is assumed perfect channel estimation, user detection, and synchronization at the receiver side. Data packets are composed of $N_{FFT}$ FFT blocks and have a physical preamble overhead of $N_{PhyPreamble}$ symbols. Each FFT block carries $N_{Block}$ symbols. The physical preamble is used to estimate the channel, synchronize the reception and detect the users involved in a given collision.

[1]The system design also allows the definition of an ad-hoc operation mode, with MT to MT communication.

```
while (1) do
    c ← 1;
    send SYNC with parameter p_c packet;
    Wait end of slot;
    if no SignalReceived then
        continue;
    else
        repeat
            // detect the number of packets involved in the received
               signal
            Q ← ReceiverProcess(SignalReceived);
            if Q ≤ Q_max then
                send ACK asking (Q − 1) retransmissions;
                wait end of (Q − 1) retransmissions/slots;
                break;
            else if (c + 1) < Mc then
                send ACK with parameter p_{c+1} to exclude some transmitters;
                c ← c + 1;
                Wait end of slot;
            end
        until c > Mc;
    end
end
```

**Algorithm 1:** Algorithm at the Base Station.

### 5.1.1 MAC Protocol

The uplink slots are organized as a sequence of epochs, where one or more groups of detection slots form an epoch. An illustration of the algorithm employed in the BS is depicted in Algorithm 1. The BS broadcasts a SYNC (Synchronization) control packet through the downlink channel marking the beginning of each epoch, allowing any MT with data packets to contend in the next slot (the first slot of the first group of detections slots) with equal access probability $p_1$ (i.e. assuring fairness). No new MT is allowed to contend until the end of the epoch (in the second and further groups of slots).

If the BS does not receive any signal throughout the first slot of the group of detection slots $c$ (i.e. no MT is transmitting), it broadcasts a new SYNC packet, starting a new epoch. If at least one MT transmits, a signal is received by the BS and it is assumed that the BS detects the number of MTs' messages involved in the signal received. A first decision is whether this number is above the maximum bound allowed ($Q_{max}$), or not (meaning that the BS can handle them on the fly).

If the first situation occurs (more than $Q_{max}$ MTs transmit) the BS detects that it is a collision but it cannot detect the signal. This is designated as an unintelligible collision, where the group of detection slots has a single slot. The BS continues with this epoch but some MTs must leave the process. To achieve this the BS broadcasts a Positive Acknowledgement (ACK) packet defining a new value for the access probability $p_{c+1}$ for the next group of detection slots. It might happen that the following group still contains an unintelligible collision and the process must be repeated again, or the number of colliding MTs is not above $Q_{max}$ any more.

If the number of transmitting MTs, $Q$, is equal to or below $Q_{max}$, the BS broadcasts an ACK packet at the end of the slot with the value of $Q-1$ informing the MTs to retransmit the data packets, making possible the detection of the data packets. The data packets are retransmitted continuously and the last control packet sent by the BS is the SYNC one marking the end of this epoch and announcing the beginning of a new one. An epoch length is limited to a maximum number of groups of detection slots, denoted as $M_C$.

Two backoff algorithms are considered for the BS when the number of transmissions exceeds $Q_{max}$:

- Constant Backoff (CB): $p_c = p$;
- Exponential Backoff (EB): $p_c = p^c$.

The SYNC packets carry the value of $p_c$, a list of detected senders, and a bit mask informing whether each of the packets was well received or not in the previous epoch. The ACK packet sent in the situation of unintelligible collision carries the new value for $p_c$. The ACK packets sent on the other situations carry a list of detected senders and the number of transmissions needed until the end of the epoch. The confirmation for a data packet correctly received is transmitted in the SYNC packet. Failed data packets are retransmitted in the next epochs up to a maximum number of times, denoted by $M_R$, being discarded after that.

Two approaches are considered in this thesis: an half-duplex approach, where a non-periodical slotted scheme is employed, and a full-duplex approach, with a periodical slotted scheme. Regarding the half-duplex approach, Figure 5.1 illustrates four possible situations that may occur during the MAC operation, with 3 MTs and $Q_{max} = 2$. It also contains the instants when new packets are ready to be sent. The first represented epoch was successful with 2 active senders. The second epoch was empty as no-one transmitted. The third epoch starts with a unintelligible collision and in its second slot one MT gave up and two proceeded. Both succeeded to transmit. The last epoch of the figure shows a successful transmission of a single MT.

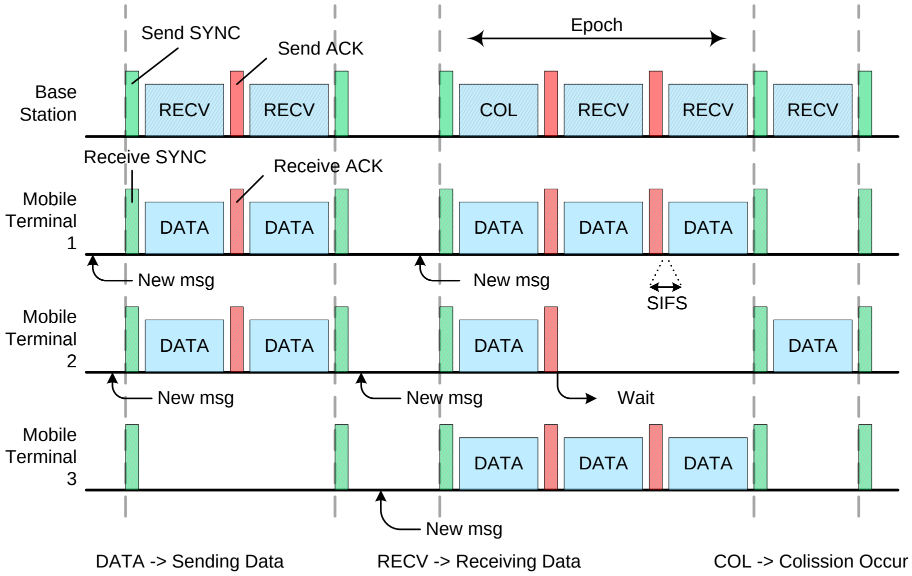



Figure 5.1: MAC protocol example.

Data, SYNC and ACK packets from different senders are spaced by a Short InterFrame Space (SIFS) time, to support the half-duplex commutation between sending and receiving mode. No spacing is used between retransmitted packets during the multi-packet detection phase.

Due to the half-duplex approach and the physical and MAC header length, it is not possible to achieve full channel utilization (100% throughput). For a given $Q_{max}$ and scheduled traffic, the throughput is maximized if $Q_{max}$ MTs are transmitting in each slot, since in this case there are no wasted slots (it is able to detect the $Q_{max}$ data packets involved in each collision), the number of required ACK and SIFS packets is minimized (one ACK and four SIFS for the $Q_{max}$ data packets) and the PER is minimized (in terms of the energy associated to each transmission attempt and/or the peak energy for each MT). The bound of the half-duplex throughput is given by

$$G_{hd-bound}^{ndma} \leq \frac{Q_{max} t_{dat}}{(t_p + t_{sy} + t_{sifs}) + (t_p + t_{ack} + t_{sifs}) + (Q_{max} (t_p + t_h + t_{dat}) + 2t_{sifs})}, \quad (5.1)$$

where, $t_{sifs}$ is the SIFS duration, $t_p$ and $t_h$ are the physical preamble and MAC data header duration, and $t_{sy}$ and $t_{ack}$ are the SYNC and ACK packet payload duration, respectively. Finally, $t_{dat}$ is the average data packet payload duration.

Concerning the full-duplex approach, it is considered a periodical slotted scheme with instantaneous ACK and SYNC packets, i.e. the duration of ACK and SYNC packets is assumed negligible when compared to the data duration. Therefore, the model for the full-duplex approach provides a maximum bound for the system performance, where it is possible to achieve almost full channel utilization. The maximum bound for the full-duplex throughput is given by

$$G_{fd-bound}^{ndma} \leq \frac{Q_{max} t_{dat}}{Q_{max} \left(t_p + t_h + t_{dat}\right)}. \tag{5.2}$$

### 5.1.2 Receiver Characterization

In a practical implementation, the MPR capability of the receiver is limited to a maximum value for $Q$, $Q_{max}$. The MAC protocol guarantees that a collision situation of $Q \leq Q_{max}$ packets is eventually reached. The receiver should then be able to separate the $Q$ packets while performing the Frequency Domain Equalization (FDE) procedure. For this purpose it is considered the iterative FDE receiver proposed in [Dinis *et al.*, 2009b]. The receiver does not require uncorrelated channels for different retransmission attempts and (as with DC schemes) reduces the PER since it combines the signals associated to all $Q$ transmissions. However, it is needed to know the number of users involved in a collision ($Q$) and which ones are they. This information can be obtained by transmitting suitable training blocks (quasi orthogonal for different users) before the data blocks [Souto *et al.*, 2010]. It should be pointed out that this can be done using synchronization and channel estimation blocks, with only a marginal overhead increase [Souto *et al.*, 2010]. The receiver has $Q$ versions of the signals associated to the $Q$ packets and jointly detects all packets involved. It is considered an iterative receiver that jointly performs the equalization and multi-packet detection procedures, where each iteration consists of $Q$ detection stages, one for each frame. Although the gains of the iterative procedure are very high for large Signal-to-Noise Ratio (SNR), it only has marginal gains for low SNR, which is the typical working region when it is employed channel coding. For this reason it is considered a turbo version of the receiver, the Turbo Iteractive Block-Decision Feedback Equalization (IB-DFE) proposed in [Dinis *et al.*, 2009b], by employing techniques presented in [Gusmão *et al.*, 2007] as explained in the final two paragraphs of section 3.1.1. This turbo approach can also be applied to DC schemes for coded scenarios and its gains can be observed in the chapter 6.

Figure 5.2 illustrates the frequency-domain multi-packet reception receiver able to separate up to $Q_{max}$ packets involved in a collision using the NDMA scheme [Tsatsanis *et al.*, 2000; Dinis *et al.*, 2009b]. It is an iterative receiver that jointly performs the equalization and multi-packet detection procedures, where each iteration consists of $Q$ detection stages, one for each packet. The detection scheme used in the block “Detect Packet i” is identical to the

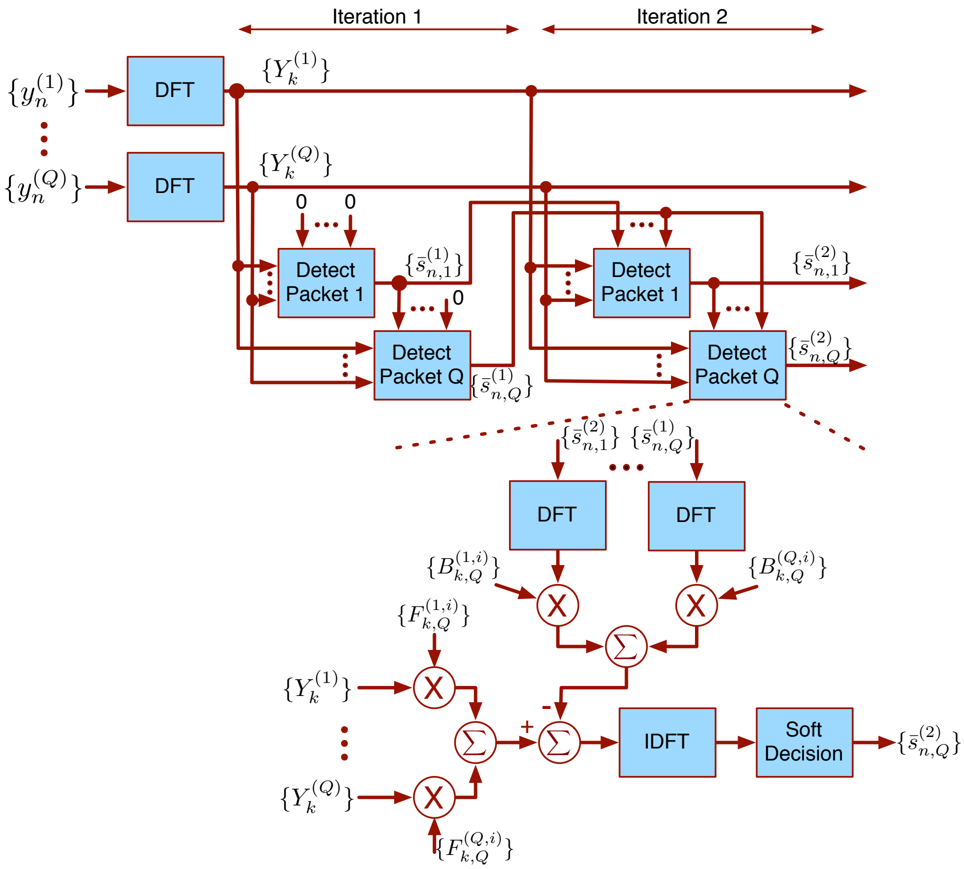



Figure 5.2: Network Diversity Multiple Access receiver

DC system presented in Figure 3.3 except for the feedback part. To achieve the separation of $Q$ packets involved in a collision the algorithm needs $Q$ versions of each packet, which means that $Q$ successive transmissions/collisions are required. An SC-FDE scheme is employed where the time-domain block associated to the $q$th user is $\{s_{n,q}; n = 0, 1, \ldots, N-1\}$, with $N$ denoting the FFT size. Whenever there is a collision, it is necessary to retransmit the packets involved. The packet associated to $r$th attempt to transmit $\{s_{n,q}; n = 0, 1, \ldots, N-1\}$ is $\{s_{n,q}^{(r)}; n = 0, 1, \ldots, N-1\}$. As with other SC-FDE schemes, a suitable cyclic prefix is added to each time-domain block [Dinis *et al.*, 2009b].

Let us assume that we have $Q$ versions of a collision of $Q$ packets. The frequency-domain received signal associated to the $r$th version of the collision is $\{Y_k^{(r)}; k = 0, 1, ..., N-1\}$, with

$$Y_k^{(r)} = \sum_{q=1}^{Q} S_{k,q}^{(r)} H_{k,q}^{(r)} + N_k^{(r)}, \tag{5.3}$$

where $H_{k,q}^{(r)}$ denotes the channel frequency response for the $k$th subcarrier, the $q$th user and the $r$th version of the collision and $N_k^{(r)}$ denotes the channel noise (as in 3.1). $\{S_{k,q}^{(r)}; k = 0, 1, ..., N-1\}$ is the Discrete Fourier Transform (DFT) of the block of the $q$th user associated to the $r$th version of the collision $\{s_{n,q}^{(r)}; n = 0, 1, ..., N-1\}$.

The residual interference from the other packets is removed when detecting a given packet. For the detection of the $q$th packet and the $i$th iteration it is used $Q$ frequency-domain feedforward filters and $Q$ frequency-domain feedback filters. The $r$th feedforward filter is associated to the signal of the $r$th collision and is characterized by the coefficients $F_{k,q}^{(r,i)}$, $k = 0, 1, ..., N-1$; the $q'$th feedback filter is used to remove the interference from the $q'$th packet and is characterized by the coefficients $B_{k,q}^{(q',i)}$, $k = 0, 1, ..., N-1$.

The $k$th frequency-domain sample associated to the $q$th packet is

$$\tilde{S}_{k,q}^{(i)} = \sum_{r=1}^{Q} F_{k,q}^{(r,i)} Y_k^{(r)} - \sum_{q'=1}^{Q} B_{k,q}^{(q',i)} \overline{S}_{k,q'}^{(i-u(q'-q))} \tag{5.4}$$

where $u(x)$ is the unitary step function, i.e., $u(x) = 0$ for $x < 0$ and 1 for $x \geq 0$. The block $\{\overline{S}_{k,q'}^{(i)}; k = 0, 1, \ldots, N-1\}$ is the DFT of the block $\{\overline{s}_{n,q'}^{(i)}; n = 0, 1, \ldots, N-1\}$, where $\overline{s}_{n,q'}^{(i)}$ is given as analogous to equation 3.3 and denotes the average symbol values conditioned to the FDE output [Silva and Dinis, 2006]. If it is assumed that $s_{n,q} = \pm 1 \pm j$ then it can be shown [Gusmão *et al.*, 2007] that these average values are given by equations (3.3) to (3.5) denoting $\{\tilde{s}_{n,q}^{(i)}; n = 0, 1, \ldots, N-1\}$ = Inverse Discrete Fourier Transform (IDFT) $\{\tilde{S}_{k,q}^{(i)}; k = 0, 1, \ldots, N-1\}$. The variance ${\sigma_{eq}^{(i)}}^2$ is given as analogous to equation (3.6), where $\hat{s}_{n,q}^{(i)} = \pm 1 \pm j$ are the hard-decisions associated to $\tilde{s}_{n,q}^{(i)}$.

After some straightforward but lengthy manipulations the optimum feedforward coefficients can be given by [Dinis *et al.*, 2009b]

$$F_{k,q}^{(r,i)} = \frac{\breve{F}_{k,q}^{(r,i)}}{\gamma_q^{(i)}}, \tag{5.5}$$

with

$$\gamma_q^{(i)} = \frac{1}{N} \sum_{k=0}^{N-1} \sum_{r=1}^{Q} \breve{F}_{k,q}^{(r,i)} H_{k,q}^{(r)} \tag{5.6}$$

and $\breve{F}_{k,q}^{(r,i)}$ is obtained from the set of $Q$ equations:

$$\sum_{q' \neq q} (1 - |\rho_{q'}^{(i)}|^2) H_{k,q'}^{(r)*} \sum_{r'=1}^{Q} \breve{F}_{k,q}^{(r',i)} H_{k,q'}^{(r')} + \alpha \breve{F}_{k,q}^{(r,i)} + (1 - |\rho_q^{(i)}|^2) H_{k,q}^{(r)*} \sum_{r'=1}^{Q} \breve{F}_{k,q}^{(r',i)} \breve{F}_{k,q}^{(r',i)} = H_{k,q}^{(r)*}, \quad r = 1, 2, \ldots, Q, \tag{5.7}$$

where $\alpha$ is obtained by (3.8) (i.e., $\alpha$ is the inverse of the SNR ) and the correlation coefficient $\rho_q^{(i)}$ is defined as $\rho_q^{(i)} = E[\hat{s}_{n,q}^{(i)}, s_{n,q}^{(i)*}]/E[|s_{n,q}^{(i)}|^2]$. The feedback coefficients are then given by [Dinis *et al.*, 2009b]

$$B_{k,q}^{(q',i)} = \sum_{r=1}^{Q} F_{k,q}^{(r)} H_{k,q'}^{(r,i)} - \delta_{q,q'} \tag{5.8}$$

($\delta_{q,q'} = 1$ if $q = q'$ and 0 otherwise). The $Q$ feedback coefficients are used to remove interference between packets (as well as residual inter-symbol interference for the packet that is being detected). The feedforward coefficients are selected to minimize the overall noise plus the residual interference due to the fact that it is not possible to have exact data estimates in the feedback loop. For high SNR and without any information about the data (i.e., when $\rho_q = 0$), the system of equations (5.5) gives the $F_{k,q}^{(r,i)}$ coefficients required to cancel the other packets when it detects the $q$th packet.

However, it should be pointed out that the correlation between channels associated to different retransmissions should be low. If not, the system of equations (5.5) might not have a solution or it can be ill conditioned. To sort this out, different transmission strategies could be employed as mentioned in 3.1.2.

By employing the turbo version of [Dinis *et al.*, 2009b] it is taken advantage of the channel decoder output within the feedback loop instead of the uncoded "soft decisions" as explained in the final two paragraphs of section 3.1.1.

## 5.2 Performance Analysis and Optimization

This section evaluates the system throughput and delay for homogeneous Poisson traffic, on a wireless network with $J$ MTs. Its main objective is to obtain an analytical model for the behaviour of the MAC protocol on a multi-packet reception system, considering different approaches for backoff algorithms as a function of the PER associated to different MTs so as to allow its optimization. In a real system this means that it should be able to estimate the PER fluctuations under given transmission conditions [Conti *et al.*, 2007, 2009] (fortunately, the PER fluctuations are usually relatively slow for the SC-FDE schemes considered in this paper, provided that we have rich multipath propagation).

To achieve this objective, in the first place the expected epoch duration is calculated as well as the expected number of bytes transmitted per MT and per epoch, to compute the average throughput. After this, it is modelled the MAC behaviour employing Markov Chain theory to get the system's steady state. Finally, it is characterized the packet service time and identified the throughput-optimal and delay-optimal $p_c$ transmission probabilities for the cases when the system is saturated or unsaturated, respectively.

### 5.2.1 Detection process

Due to the serial structure of the detector [Dinis *et al.*, 2009b] (see section 5.1.2), the PER ($PER_{Q,q}$) is influenced by the number of sending MTs ($Q$) and the order of the MT in the detection process, the $q$th MT. Note that in this model, it is considered that MTs are always handled in the same order (i.e., the $q$th MT is always the $q$th to be detected). It is also assumed that no errors occur in the downlink broadcast transmissions.

Let $\psi(Q)$ denote the random variable associated with the number of packets correctly received when $Q$ MTs transmit during the detection process. The probability of having $k$ packets successfully received ($P\{\psi(Q) = k\}$) can be calculated identifying the set with all error patterns that lead to $k$ packets received, denoted by $\Omega_{Q,k}$. This set is composed by $\binom{Q}{k}$ members (all the combinations of $Q$ packets with $k$ successes and $Q-k$ errors), each identifying an error pattern. The outcome of the detection process for the packet from the $q$th MT is an on-off random variable, $\omega_{Q,q}$, which can be 1 (success) or 0 (failure). Let $\omega_Q = \{\omega_{Q,q}; q = 1, \ldots, Q\}$ denote an error pattern, defined by the outcome of all $Q$ packets involved in a collision. The set $\Omega_{Q,k}$ contains all error patterns $\omega_Q$ that satisfy the condition $\sum_{q=1}^{Q} \omega_{Q,q} = k$. Therefore, $P\{\psi(Q) = k\}$ can be calculated summing the probability of all error patterns within $\Omega_{Q,k}$,

$$P\{\psi(Q) = k\} = \sum_{\omega_Q \in \Omega_{Q,k}} \prod_{q=1}^{Q} (1 - PER_{Q,q})^{\omega_{Q,q}} (PER_{Q,q})^{\left(1-\omega_{Q,q}\right)}. \tag{5.9}$$

When $Q$ MTs transmit, the expected number of packets jointly detected by the BS is

$$\theta(Q) = E[\psi(Q)] = \sum_{k=1}^{Q} kP\{\psi(Q) = k\}. \tag{5.10}$$

The equation above can be simplified considering the MT's average packet error rate for all $q$, denoted by $PER_Q$, valid for a low variation of $PER_{Q,q}$ with $q$. Since errors become independent of the detection order, $\psi(Q)$ has a binomial distribution. In result, the calculation of $\theta(Q)$ is simplified to

$$\theta(Q) = \begin{cases} \sum_{k=1}^{Q} bi(Q, k, 1 - PER_Q)\, k, & 1 \le Q \le Q_{max} \\ 0, & Q > Q_{max}, Q = 0 \end{cases}, \tag{5.11}$$

where $bi(Q, k, p) = \binom{Q}{k} p^k (1-p)^{(Q-k)}$ is the binomial probability mass function. On the other hand, the duration of the detection process in the half-duplex mode is

$$\delta\left(Q\right)=\begin{cases}2t_p+t_h+t_{dat}+2t_{sifs}+t_{sy}, & Q\leq 1,\ Q>Q_{max}\\ 2t_p+Q(t_p+t_h+t_{dat})+4t_{sifs}+t_{sy}+t_{ack}, & 1<Q\leq Q_{max}\end{cases}. \tag{5.12}$$

If $Q$ is zero or above $Q_{max}$ (the maximum number of detectable transmissions) the detection process fails and lasts a single slot. Otherwise, it lasts $Q$ slots. A group of detection slots always includes an initial contention slot defined by a SYNC or an ACK packets (both packets have the same duration, $t_p+t_{sy}=t_p+t_{ack}$), followed by a data packet time (it is assumed a constant duration equal to $t_p+t_h+t_{dat}$) and it has to change at least twice between receiving and transmitting modes ($2t_{sifs}$). A multi-packet detection phase includes an additional ACK packet, $Q-1$ packet retransmissions and two more changes between receiving and transmitting modes.

In the case of full-duplex, the duration of the detection process is defined by

$$\delta\left(Q\right)=\begin{cases}t_p+t_h+t_{dat}, & Q\leq 1,\ Q>Q_{max}\\ Q(t_p+t_h+t_{dat}), & 1<Q\leq Q_{max}\end{cases} \tag{5.13}$$

This chapter presents an analytical model valid for both approaches but only presents results of the half-duplex approach. Further details (e.g. example illustration or performance results) about the full-duplex approach are presented in chapter 6.

### 5.2.2 Epoch Analysis

An epoch may have multiple unintelligible groups of detection slots (with a single slot dimension) before the last group of detection slots. Let $\nu$ denote a Random Variable (RV) with the number of unintelligible groups of detection slots (hereafter named simply as unintelligible slots). RV $\nu$ can be defined based on another RV $\eta_c$, which denotes the number of MTs that transmit a packet during the $c$th group of detection slots of an epoch. The $\nu$'s probability mass function can be defined by $P\{\nu=c\}=P\{\eta_{c+1}\leq Q_{max}\mid \eta_c>Q_{max}\}$ for $0<c<M_C$, or by $P\{\nu=M_C\}=P\{\eta_{M_C-1}>Q_{max}\}$. Finally, $P\{\nu=0\}=P\{\eta_1\leq Q_{max}\}$.

In this subsection it is shown that both RVs, $\nu$ and $\eta_c$, can be described as stochastic processes that depend solely on the number of MTs with packets available to transmit in the first slot of an epoch, denoted by RV $\chi$, and by the value of $p_c$, the MTs' transmission probability during the $c$th group of detection slots.

RV $\eta_c$ is defined when the epoch lasts $c$ (for $0<c\leq M_C$) or more groups of detection slots (i.e. $\nu\geq c-1$). RV $\chi$ and $p_1$ influence the number of packets transmitted in the first

group of detection slots of an epoch. MTs that persist contending during an epoch, transmit independently with probability $p_c$. Therefore $\eta_c$ conditional probability mass function can be defined recursively using

$$P\{\eta_c = Q\ |\nu \geq c-1\}(\chi) = \tag{5.14}$$
$$= \begin{cases} \sum_{i=Q}^{J} P\{\chi = i\} bi(i, Q, p_1), & c = 1 \\ \displaystyle\sum_{i=Q_{max}+1}^{J} \frac{P\{\eta_{c-1} = i \mid \nu \geq c-2\}(\chi)}{P\{\nu \geq c-1 \mid \nu \geq c-2\}(\chi)} bi(i, Q, p_c), & 1 < c \leq M_C \end{cases},$$

where the expression for $1 < c \leq M_C$ is derived from $P\{\eta_c = Q \mid \nu \geq c-1\}(\chi) = P\{\eta_c = Q \mid \nu \geq c-2\}(\chi) / P\{\nu \geq c-1 \mid \nu \geq c-2\}(\chi)$. $P\{\nu \geq c-1 \mid \nu \geq c-2\}(\chi)$ denotes the probability of having an unintelligible slot at the $(c-1)$th slot of an epoch, and can be calculated as the probability of having more than $Q_{max}$ MTs transmitting a packet at that slot, i.e.

$$P\{\nu \geq c-1 \mid \nu \geq c-2\}(\chi) = \sum_{Q=Q_{max}+1}^{J} P\{\eta_{c-1} = Q \mid \nu \geq c-2\}(\chi). \tag{5.15}$$

Equation (5.14) applies only when $P\{\nu \geq c-1 \mid \nu \geq c-2\} > 0$ for $c > 1$. From (5.12) and (5.14), the $m$th moment for the duration expectation of an epoch with $c$ groups of detection slots is

$$dur_c^m(\chi) = \begin{cases} E\left[((c-1)\,\delta\,(Q_{max}+1) + \delta\,(\eta_c))^m \mid \nu = c-1\right], & 1 \leq c < M_C \\ E\left[((c-1)\,\delta\,(Q_{max}+1) + \delta\,(\eta_c))^m \mid \nu = c-1 \vee \nu = c\right], & c = M_C \end{cases}$$
$$= \sum_{Q=0}^{Limit(c)} (c\delta\,(Q_{max}+1) + \delta\,(Q))^m\, P\{\eta_c = Q \mid \nu \geq c-1\}(\chi), \tag{5.16}$$

with $\delta\,(Q_{max}+1)$ denoting the duration of a group with a unintelligible slot. $Limit\,(c)$ takes account of the number of transmitting MTs ($Q$) that close an epoch: an epoch with $c$ groups of detection slots has $\eta_c = Q \leq Q_{max}$ if $c < M_C$; an epoch with $M_C$ group of detection slots has $\eta_{M_C-1} = Q > Q_{max}$ and it does not depend on $\eta_{M_C}$, because it is the last possible group of detecting slots.

$Limit\,(c)$ is given by

$$Limit(c) = \begin{cases} Q_{max}, & 1 \leq c < M_C \\ J, & c = M_C \end{cases}. \tag{5.17}$$

Similarly, the $m$th moment of the expected number of bytes received during an epoch with $c$ groups of detection slots is

$$num_c^m(\chi) = \sum_{Q=0}^{Limit(c)} \left(L_{data}\theta\left(Q\right)\right)^m P\{\eta_c = Q \mid \nu \geq c-1\}\left(\chi\right), \tag{5.18}$$

where $\theta\left(Q\right)$, defined by (5.11), accounts for the successful detection rate.

The probability of occurring at least $c-1$ unintelligible slots during an epoch is calculated as follows

$$P\{\nu \geq c-1\}(\chi) = \begin{cases} \prod_{i=2}^{c} P\{\nu \geq i-1 \mid \nu \geq i-2\}(\chi), & 1 < c \leq M_C \\ 1, & c = 1. \end{cases} \tag{5.19}$$

From (5.16), (5.18) and (5.19), the $m$th moment of the expected duration of an epoch is given by

$$E\left[\Delta^m\right](\chi) = \sum_{c=1}^{M_C} dur_c^m(\chi) P\{\nu \geq c-1\}(\chi), \tag{5.20}$$

and the $m$th moment of the expected number of bytes transmitted during the epoch

$$E\left[\Theta^m\right](\chi) = \sum_{c=1}^{M_C} num_c^m(\chi) P\{\nu \geq c-1\}(\chi). \tag{5.21}$$

Finally, the average normalized throughput (also known as the channel utilization) for an epoch yields

$$G(\chi, p)^{ndma} = \frac{E\left[\Theta^1\right](\chi)}{E\left[\Delta^1\right](\chi) R_{dat}}, \tag{5.22}$$

with $R_{dat}$ denoting the data rate. Equation (5.22) can be used for both unsaturated and saturated scenarios. In saturation, all MTs always have a packet to transmit (i.e. $P\{\chi = J\} = 1, P\{\chi \neq J\} = 0$).

Given $\chi$ and using (5.9), (5.14) and (5.19), the probability of having $i$ packets transmitted with success during an epoch, denoted by $\varsigma_i\left(\chi\right)$ where $i = 0, 1, \ldots, J$, is determined taking account of the success probability and the number of packets transmitted at each group of detection slots ($\eta_c$ for the group after $c$ unintelligible slots), i.e.,

$$\varsigma_i\left(\chi\right) = \sum_{c=1}^{M_C} \sum_{Q=\max(i,1)}^{Q_{max}} P\left\{\eta_c = Q \mid \upsilon \geq c-1\right\}\left(\chi\right) P\left\{\upsilon \geq c-1\right\}\left(\chi\right) P\left\{\psi\left(Q\right) = i\right\}. \tag{5.23}$$

When the average PER is considered ($PER_Q$ for $Q$ transmissions), the equation above is written as

$$\varsigma_i(\chi) = \sum_{c=1}^{M_C} \sum_{Q=\max(i,1)}^{Q_{max}} P\{\eta_c = Q \mid \nu \geq c-1\}(\chi) P\{\nu \geq c-1\}(\chi) bi(Q, i, 1 - PER_Q). \quad (5.24)$$

Using (5.14) and (5.24), the expected packet successful transmission probability during an epoch with an initial state defined by the RV $\chi$ can be given by the ratio of the expected number of successful transmissions to the expected number of transmitting MTs, i.e.,

$$p_{suc}(\chi) = \frac{\sum_{i=1}^{Q_{max}} i\varsigma_i(\chi)}{\sum_{i=0}^{J} iP\{\chi = i\}}. \quad (5.25)$$

### 5.2.3 Network's Steady State

To derive the system's steady state for a given load, on a network with $J$ MTs it is considered the $(J+1)$-states Discrete Time Markov Chain (DTMC) defined by $\chi^k$, with $k \in [0, \infty[$. The random variable $\chi^k$ denotes the number of MTs with packets available to transmit in the first slot of the epoch $k$. The system behaviour can be defined by the transition probability between $\chi^k$ and $\chi^{k+1}$. From the perspective of the system, during an epoch the number of MTs with packets to transmit changes due to the departure of successful transmitted packets and the arrival of new packets. It is irrelevant if the packet arrivals occur during the epoch or if all packets arrive just before the beginning of the next epoch, because the system dynamics during an epoch is not influenced by the arrival of new packets. It is assumed that the packet generation follows a Poisson process, with an average rate of $\lambda$ packets per second for each MT. In consequence, the probability of appearing a new packet on a MT is $1 - e^{-\lambda E[\Delta^1](\chi^k)} \approx \lambda E[\Delta^1](\chi^k)$, given by [Bloch *et al.*, 1998], where $E[\Delta^1](\chi^k)$ is the expected epoch duration. The probability that an active MT becomes inactive depends on both $p_{suc}\left(\chi^k\right)$ and $p_{QE}\left(\chi^k\right)$, where $p_{QE}\left(\chi^k\right)$ represents the probability that the MAC queue of the MT is empty after a successful transmission. If $q$ MTs have packets to transmit at the beginning of epoch $k$, then there will be $i$ MTs with packet to transmit if $q - a$ MTs stop transmitting ($a$ MTs continue transmitting in epoch $k+1$) and $i - a$ of the idle MTs start transmitting in epoch $k+1$, for $a \in [\max(0, i+q-J), \min(i,q)]$. Therefore, the state transition probability is defined by:

$$\begin{aligned} P\{\chi^{k+1} = i \mid \chi^k = q\} = & \quad (5.26) \\ = \sum_{a=\max(0,i+q-J)}^{\min(i,q)} & bi\left(q, q-a, p_{suc}\left(\chi^k\right) p_{QE}\left(\chi^k\right)\right) bi\left(J-q, i-a, \lambda E\left[\Delta^1\right](\chi^k)\right), \end{aligned}$$

where min and max are respectively the minimum and maximum values of the parameters. $p_{QE}\left(\chi^k\right)$ depends on the network's utilization rate, defined as $\varrho = \lambda E\left[D_b\right](\chi^k)$, where $E\left[D_b\right](\chi^k)$ is the expectation of the packet service time. $\varrho$ can be interpreted as the probability that the MAC queue of the MT is not empty: for a M/G/1 queue with a time-invariant service time, Takács [Takács, 1982, pp. 66–76] shows that $p_{QE}\left(\chi^k\right) = 1 - \varrho$ for $\varrho < 1$ and that $p_{QE}\left(\chi^k\right) = 0$ for $\varrho \geq 1$.

Let $\phi_l$ as function of $\chi^k$ be the event of a MT having $l$ successive epochs with failed transmissions. The packet service time depends on the number of epochs used to transmit a packet, and their durations. An approximate value for $E\left[D_b\right](\chi^k)$ can be calculated using eq. (5.27), (5.20) and (5.25), where is assumed that different epochs have i.i.d. (independent and identically distributed) duration distributions, resulting in constant $E\left[\Delta^1\right](\chi^k)$ values for all epochs during the transmission of the packet. However, to improve its precision, the fact that at least one MT has a packet to transmit (denoted by $\beta(\chi^k)$) was considered. The expectation of the packet service time is given by

$$\begin{aligned} E\left[D_b\right](\chi^k) &= \sum_{l=0}^{M_R} E\left[D_b \mid \phi_l\right](\chi^k) P\left\{\phi_l(\chi^k)\right\} \\ &\approx \sum_{l=0}^{M_R} (l+1)\, E\left[\Delta^1\right](\hat{\chi}^k)\left(1 - p_{suc}(\hat{\chi}^k)\right)^l p_{suc}(\hat{\chi}^k) \\ &= \frac{E\left[\Delta^1\right](\hat{\chi}^k)}{p_{suc}(\hat{\chi}^k)}\left(1 - \left(1 + (2+M_R)\, p_{suc}(\hat{\chi}^k) + \left(p_{suc}(\hat{\chi}^k)\right)^2\right)\left(1 - p_{suc}(\hat{\chi}^k)\right)^{M_R+1}\right), \end{aligned} \tag{5.27}$$

where $M_R$ is the maximum number of retries and $\hat{\chi}^k$ refers to the conditional distribution of $\chi^k$ given $\beta(\chi^k)$,

$$\begin{aligned} P\left\{\hat{\chi}^k = i\right\} = P\left\{\chi^k = i \mid \beta(\chi^k)\right\} &= \frac{P\left\{\beta(\chi^k) \mid \chi^k = i\right\} P\{\chi^k = i\}}{\sum_{n=0}^{J} P\left\{\beta(\chi^k) \mid \chi^k = n\right\} P\{\chi^k = n\}} \\ &= \frac{\left(\frac{i}{J}\right) P\{\chi^k = i\}}{\sum_{n=0}^{J} \left(\frac{n}{J}\right) P\{\chi^k = n\}}. \end{aligned} \tag{5.28}$$

Defining the steady-state probability distribution of $\left\{\chi^k\right\}_{k=0}^{\infty}$ as follows

$$\pi_i^{ndma} = \lim_{k\to\infty} P\{\chi^k = i\},\ \forall i \in \{0, 1, \ldots, J\} \tag{5.29}$$

consider the following theorem.

**Theorem 1.** *The DTMC* $\left\{\chi^k\right\}_{k=0}^{\infty}$ *has a steady-state probability distribution that is independent of the initial state* $\chi^0$*, i.e.,* $\pi_i^{ndma}$ *exists,* $\forall i \in \{0, 1, \ldots, J\}$*.*

*Proof.* Let $T(s,r) = P\left\{\chi^{k+1} = r | \chi^k = s\right\}$ and $SS$ be the state space of the DTMC $\left\{\chi^k\right\}_{k=0}^{\infty}$.

1. Equation (5.26) shows that $T(0,r) > 0$ and that $T(r,0) > 0$, $\forall r \in SS$. Therefore the DTMC is irreducible because state 0 leads to every state in $SS$ and each state in $SS$ leads to state 0.

2. Since $T(0,0) > 0$, the period of state 0 is one. Since the DTMC is irreducible, the DTMC is aperiodic.

3. Since $|SS| < \infty$, there exists a positive-recurrent state in $SS$. Since the DTMC is irreducible, every state in $SS$ is positive-recurrent.

4. From 1), 2) and 3), the steady-state probability distribution for the DTMC exists [Bloch *et al.*, 1998].

$\square$

The system's equilibrium state, $\pi^{ndma}$, is characterized by the set of probabilities given by $\left\{\pi_i^{ndma}, i = \{0, 1, \ldots, J\}\right\}$. It can be determined by applying an interactive numerical method [Bloch *et al.*, 1998] to solve the DTMC defined by (5.26), using (5.20), (5.25) and (5.27). The system throughput follows directly, through (5.22).

### 5.2.4 Delay Analysis

For Poisson sources, the individual transmission delay on a MT can be modelled by a M/G/1 queue with vacations [Bersakas and Gallager, 1992]. A vacation time occurs when an idle MT gets a new packet while an epoch is active. The MT has to wait until the end of that epoch before starting the packet transmission. On this subsection the $\chi^k$ parameter is omitted when its distribution is equal to the steady state ($\chi^k \simeq \pi^{ndma}$).

Assuming that $\chi^k$ is mutually uncorrelated for different $k$ values after reaching the steady state, the expected variance (second order centred moment) of the packet service time is the sum of variances of all individual epochs that compose the packet service time. Therefore, the expected second order moment for the packet service time conditioned that transmission failed in the previous $l$ epochs depends on the moments of the epoch duration (defined by (5.20)) and is given by

$$E\left[D_b^2 \mid \phi_l\right] = E\left[(D_b - E\left[D_b \mid \phi_l\right])^2 \mid \phi_l\right] + E\left[D_b \mid \phi_l\right]^2$$

$$\approx \sum_{j=0}^{l}(E\left[\Delta^2\right] - \left(E\left[\Delta^1\right]\right)^2) + \left(\sum_{j=0}^{l} E\left[\Delta^1\right]\right)^2$$
$$= (l+1)\, E\left[\Delta^2\right] + l\,(l+1)\left(E\left[\Delta^1\right]\right)^2. \tag{5.30}$$

Notice that the independence and mutually uncorrelation hypothesis loose precision when the network load grows and approaches saturation. Near saturation, a detection failure in a contention slot induces a higher collision probability in future slots (this correlation grows with the relative load), originating longer epoch durations than the model predicts. The second order moment for the packet service time is simply

$$E\left[D_b^2\right] = \sum_{l=0}^{M_R} E\left[D_b^2 \mid \phi_l\right] \times P\left\{\phi_l\right\}$$
$$\approx \sum_{l=0}^{M_R}\left((l+1)\, E\left[\Delta^2\right] + l\,(l+1)\, E\left[\Delta^1\right]^2\right) \times \left(1 - p_{suc}\left(\hat{\pi}^{ndma}\right)\right)^l p_{suc}\left(\hat{\pi}^{ndma}\right), \tag{5.31}$$

where $\hat{\pi}^{ndma}$ refers to the conditional distribution of $\pi^{ndma}$ given $\beta\left(\pi^{ndma}\right)$, calculated using (5.28), and the moments of the epoch's delay and the success probability are defined in (5.20) and (5.25).

Let $\bar{\beta}$ be the event of a MT being idle at the beginning of the epoch. The vacation time depends on the conditional distribution of $\pi^{ndma}$ given $\bar{\beta}$, denoted by $\breve{\pi}^{ndma}$, which can be calculated using Bayes' theorem:

$$\breve{\pi}_i^{ndma} = P\left\{\pi^{ndma} = i \mid \bar{\beta}\right\} = \frac{P\left\{\bar{\beta} \mid \pi^{ndma} = i\right\}\pi_i^{ndma}}{\sum_{n=0}^{J} P\left\{\bar{\beta} \mid \pi^{ndma} = n\right\}\pi_n^{ndma}} = \frac{\left(1 - \frac{i}{J}\right)\pi_i^{ndma}}{\sum_{n=0}^{J}\left(1 - \frac{n}{J}\right)\pi_n^{ndma}}. \tag{5.32}$$

From the M/G/1 queue with vacation [Bersakas and Gallager, 1992], the average system delay for a packet can be expressed as

$$E\left[D\right]^{ndma} = E\left[D_b\right] + \frac{\lambda E\left[D_b^2\right]}{2(1-\lambda E\left[D_b\right])} + \frac{E\left[\Delta^2\right]\left(\breve{\pi}^{ndma}\right)}{2E\left[\Delta^1\right]\left(\breve{\pi}^{ndma}\right)}. \tag{5.33}$$

The effective packet transmission delay, measuring the time from entering the MT's MAC queue until being received by the network layer at the BS, is equal to $E\left[D\right]^{ndma} - t_{sifs} - t_p - t_{sy}$ because it does not include the time of the last SYNC transmission, acknowledging the data packet reception to the MT.

The packet delay follows directly from the system's steady state, using (5.33). Notice that (5.33) fails to capture the saturated system behaviour because it assumes an infinite MAC queue. Real systems have a finite MAC queue and when in the presence of saturated load, some packets are dropped while others are sent with a finite delay. Furthermore, for $p$ (the parameter of the backoff algorithms considered for the access probability) values in the saturation region but near the saturation border, the system has periods of time where the queues are not full. As a result, in these conditions the delay value is much lower than the presented by equation 5.33 for the saturation border.

### 5.2.5 Optimization of the transmission probability

Two backoff algorithms were proposed in section 5.1.1 for $p_c$, which control the MT's transmission probability and are function of a parameter $p$. The value of this parameter influences the system's throughput and the packet delay. The optimal value when $J \leq Q_{max}$ is $p = 1$, which corresponds to the classical NDMA protocol. Otherwise, the value can be calculated using the models presented above. This section analyses the $p$ value that maximizes throughput in saturated conditions for a known number of MTs ($J$), and the $p$ value that minimizes delay, considering a known number of MTs ($J$) injecting Poisson traffic with a known rate $\lambda$.

The saturation throughput can be computed through $G_{sat}^{ndma} = G^{ndma}$ given that $P\{\chi^k = J\} = 1, P\{\chi^k \neq J\} = 0, \forall k$, using (5.22). ${p_{sat}}^{\star}$ is defined as the $p$ value that maximizes the saturated throughput, and $G_{max}^{ndma} = G_{sat}^{ndma}\left({p_{sat}}^{\star}\right)$ as the maximum saturated throughput, i.e.,

$$ {p_{sat}}^{\star} \triangleq \arg\max_{p} G_{sat}^{ndma}. \tag{5.34} $$

$G_{sat}^{ndma}$ is generally differentiable because it is calculated from (5.22), which is the quotient of two differentiable functions (considering a differentiable backoff function) with a non-zero denominator. The value of ${p_{sat}}^{\star}$ can be obtained by solving numerically $\frac{\partial}{\partial p} G_{sat}^{ndma} = 0$. It has at least one minimum value in $[0, 1]$, since $G_{sat}^{ndma}$ is zero if $J > Q_{max}$, $p = 0$ and $p = 1$, and it is positive for $p \in ]0, 1[$. Thus, in saturated conditions, ${p_{sat}}^{\star}$ is the optimal value for the backoff algorithm's parameter ($p$) considered for the transmission probability for saturated conditions, and depends only on $J$, $Q_{max}$ and the PER values. The simulations presented in the next section indicate that $G_{max}^{ndma}$ is the system's maximum throughput.

For unsaturated conditions, the system can be stable with a given load $\lambda$ and a transmission data rate $R_{dat}$ for multiple values of $p$. The steady-state probability distribution ($\pi^{ndma}$) depends on the number of MTs ($J$), the average load per MT ($\lambda$) and on $Q_{max}$ and $p$ values. In the following analysis, it is assumed that MTs may retransmit unlimited times the failed packets until they are successfully received at the BS ($M_R \to \infty$). In these conditions, multiple solutions exist for (5.35), as long as $\lambda J$ is bellow $R_{dat}G_{max}^{ndma}$. However, for a given $p$ value the system equilibrium state, $\{\pi^{ndma}(p)\}$, is unique and can be computed as

$$R_{dat}G^{ndma}(\{\pi^{ndma}(p)\}, p) = \lambda J. \tag{5.35}$$

${p_{opt}}^{\star}$ is defined as the $p$ value that minimizes the packet delay in the conditions of (5.35), and $D_{min} = E[D]^{ndma}\left({p_{opt}}^{\star}\right)$ as the minimum average delay for a given load and a given number of MTs:

$${p_{opt}}^{\star} \triangleq \arg\min_{p} E[D]^{ndma}. \tag{5.36}$$

The delay, $E[D]^{ndma}$, is defined for the values of $p$ where $R_{dat}G_{sat}^{ndma}\left(p\right) > \lambda J$. These values can be represented by a set of disjoint intervals, delimited by the values of $p$ that satisfy the equation $R_{dat}G_{sat}^{ndma}\left(p\right) = \lambda J$. Since the delay is a differentiable function (5.33) of $p$ within these intervals, ${p_{opt}}^{\star}$ can be calculated solving numerically $\frac{\partial}{\partial p}E[D, \pi^{ndma}(p)]^{ndma} = 0$ for each of the continuous differentiable intervals. It has at least one minimum value in $[0, 1]$, since $E[D]^{ndma}$ is infinite for $p = 0$ and $p = 1$ (and for the intervals' border values, when they exist) and it is finite when the system is not saturated, if $J > Q_{max}$ and $\lambda J < R_{dat}G_{max}^{ndma}$. The optimal value for $p$ with homogeneous Poisson sources is ${p_{opt}}^{\star}$ and depends only on $J$, $Q_{max}$, $\lambda$, $R_{dat}$ and the PER values.

## 5.3 Performance Results

A set of performance results concerning the proposed MAC protocol in the half-duplex approach for multi-packet detection in SC-FDE systems is presented in this section. A table containing all the information regarding the simulations' setup used in this chapter is illustrated in Table 5.1 and Table 5.2. Each FFT block has $N$=512 data symbols, plus an appropriate cyclic prefix. The useful part of each FFT block has duration 4 ms and the cyclic prefix has duration 1.12 ms. The data symbols are selected from a QPSK (Quadrature Phase-Shift Keying) constellation under a Gray mapping rule and the channel encoder is the well-known [Viterbi *et al.*, 1989] rate-1/2 64-state convolutional code with generators $1 + D^2 + D^3 + D^5 + D^6$ and $1 + D + D^2 + D^3 + D^6$. The data packets corresponds to $N_{FFT} = 16$ FFT blocks, each one with $N_{Block} = 64$ Bytes. The physical preamble associated to each data packet has length $N_{PhyPreamble} = 24$ Bytes.

It is considered the turbo multi-packet detection scheme described in [Dinis *et al.*, 2009b] (i.e., the channel decoder outputs are employed in the feedback loop) and, unless otherwise stated, the channel remains fixed for each packet retransmission. To allow efficient packet

| | Figures | | | | |
|---|---|---|---|---|---|
| | **5.3** | **5.4** | **5.5** | **5.6** | **5.7** |
| **Data Symbols** | 512 | | | | |
| **Modulation** | Quadrature Phase Shift Keying | | | | |
| **Block (Bits)** | 512 | | | | |
| **Packet (Bytes)** | 1024 | | | | |
| **Coding** | 64-state convolutional code with rate 1/2 | | | | |
| **Turbo equalization** | Yes | | | | |
| **Iterations** | 4 | | | | |
| **Channel Conditions** | UC and EC+SP | EC+SP | EC+SP | EC+SP | EC+SP |
| **No PC** | — | — | — | — | — |
| $1/\lambda$ **(ms/packet)** | — | saturation, 90 and 150 | | | saturation |
| **Type of Traffic** | — | Poisson | Poisson | Poisson | Poisson |
| $E_b/N_0$ **(dB)** | variable | lossless | lossless | 1 and lossless | variable |
| $p$ **values** | — | variable | variable | variable | optimized |
| **Backoff algorithm** | — | CB | EB | CB and EB | CB and EB |
| **R or** $Q_{max}$ | 4 | 1, 2 and 4 | 1, 2 and 4 | 4 | 2 and 4 |
| **Number of Stations** | — | 8 | 8 | 8 | 8 |

Table 5.1: Simulations setups from Figure 5.3 to Figure 5.7.

| | Figures | | | | | |
|---|---|---|---|---|---|---|
| | **5.8** | **5.9** | **5.10** | **5.11** | **5.12** | **5.13** |
| **Data Symbols** | 512 | | | | | |
| **Modulation** | Quadrature Phase Shift Keying | | | | | |
| **Block (Bits)** | 512 | | | | | |
| **Packet (Bytes)** | 1024 | | | | | |
| **Coding** | 64-state convolutional code with rate 1/2 | | | | | |
| **Turbo equalization** | Yes | | | | | |
| **Iterations** | 4 | | | | | |
| **Channel Conditions** | EC+SP | | | | | |
| **No PC** | — | — | — | — | — | — |
| $1/\lambda$ **(ms/packet)** | 150 | 150 | saturation | 150 | variable | 100 and 150 |
| **Type of Traffic** | Poisson | Poisson | Poisson | Poisson | Poisson | Poisson |
| $E_b/N_0$ **(dB)** | variable | lossless | 1 and lossless | 1 | lossless | lossless |
| $p$ **values** | optimized | variable | variable | optimized | optimized | optimized |
| **Backoff algorithm** | CB and EB | CB and EB | CB and EB | CB and EB | CB and EB | CB and EB |
| **R or** $Q_{max}$ | 2 and 4 | 4 | 2, 4 and $J$ | 2 and 4 | 2 and 4 | 4 |
| **Number of Stations** | 8 | 8 | variable | variable | 8 | 8 |

Table 5.2: Simulations setups from Figure 5.8 to Figure 5.13.

| Delay (s) | Power (dB) |
|---|---|
| 0.0E+0 | -3.30 |
| 1.0E-5 | -3.60 |
| 2.0E-5 | -3.90 |
| 3.0E-5 | -4.20 |
| 5.0E-5 | 0.00 |
| 8.0E-5 | -0.90 |
| 1.1E-4 | -1.70 |
| 1.4E-4 | -2.60 |
| 1.8E-4 | -1.50 |
| 2.3E-4 | -3.00 |
| 2.8E-4 | -4.40 |
| 3.3E-4 | -5.90 |
| 4.0E-4 | -5.30 |
| 4.9E-4 | -7.90 |
| 6.0E-4 | -9.40 |
| 7.3E-4 | -13.20 |
| 8.8E-4 | -16.30 |
| 1.05E-3 | -21.20 |

Table 5.3: Adopted power delay profile.

separation, it is considered the cyclic-shifted versions of the FFT blocks for each retransmission attempt which formally is equivalent to have cyclic-shifted versions of the channel in each retransmission attempt [Dinis *et al.*, 2007]. The performance results show that the behaviour of the receiver is similar for different channels, provided that strong time-dispersion effects and rich multipath propagation are presented. As an example, the considered channel is characterized by the power delay profile of table 5.3, with uncorrelated Rayleigh fading on the different paths. It is assumed perfect "average power control" (i.e., the signals associated to all users have the same average power at the receiver (the BS)), as well as perfect synchronization and channel estimation conditions. It is considered Poisson sources and it is assumed that the receiver knows the number of MTs that attempted to transmit in a given slot ($Q$), when $Q > Q_{max}$, as well as the MTs that transmitted each packet, when $Q \leq Q_{max}$.

Figure 5.3 shows the average PER, averaged over all users, after four iterations, when $Q \in \{1, 2, 3, 4\}$ ($Q \leq Q_{max}$). A fixed channel is considered for all retransmission attempts, combined with the adoption of the SP technique. For the sake of comparisons it is also included in Figure 5.3 the PER values with Uncorrelated Channel (UC) for the different retransmission attempts and the PER value of the Hybrid Automatic Repeat reQuest (Hybrid ARQ) solution when up to four retransmissions are combined (see section 3).

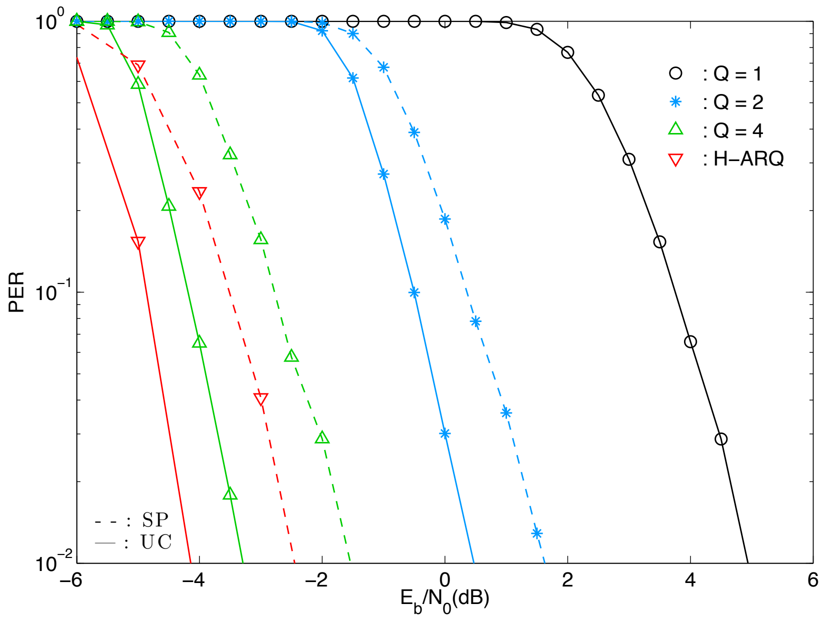



Figure 5.3: PER after 4 iterations, for $Q = 1$ (without collision), 2, 4 and for the Hybrid ARQ scheme where four transmissions of the same packet are combined.

The results are expressed as function of $E_b/N_0$ ratio[2]. Clearly this technique is able to cope with a large number of collisions, with improved performances as it is increased the number of packets involved in the collisions (and, consequently, the number of retransmissions), even with the SP technique (with the same channel for each retransmission). For this reason, it is considered the SP technique hereafter. It can also be observed that for $Q = 4$, the PER values are similar to the ones obtained in the Hybrid ARQ solution, despite the interference present due to multiple simultaneous transmissions.

To evaluate the MAC performance, it is added the assumption that $PER_{Q,q}$ for data packets is a mean-ergodic process. Therefore, the detection process can be modelled on a discrete event simulator by a matrix with the average packet error rates $PER_{Q,q}$, for the $qth$ packet when $Q$ packets ($Q \leq Q_{max}$) are jointly detected. The multi-packet detection physical layer and the MAC protocol was implemented in the ns-2 simulator [Information Sciences Institute, 2007]. The following simulation parameters were used below: Data Rate $R_{dat} = 1Mbps$; $t_{dat} = 990/R_{dat}$ s; $t_{ack} = t_{sy} = 64/R_{dat}$ s; $t_h = 34/R_{dat}$ s; $t_p = 24/R_{dat}$ s; $t_{sifs} = 10\mu$s; $M_C = 21$ and $M_R = 100$. The $M_R$ value assures that all packets are transmitted with success when the system is not saturated. $PER_{Q,q}$ values were taken from the same set of physical layer simulations (as SP results of Figure 5.3). The simulation

[2]As mentioned earlier $N_0$ is the one-sided power spectral density of the noise and $E_b$ is the average bit energy associated to a given packet transmission.

scenario considers $J$ MTs uniformly placed on a circle centred in the BS with radius 3m, within the perfect power control range. Each MT runs a Poisson packet generator and has a MAC queue with a capacity for 50 packets. An average rate of $\lambda = 200$ packets/s/MT was used to saturate the network. Unsaturated experiments applied average packet rates varying from $\lambda = \frac{1000}{250} = 4$ packets/s/MT to $\lambda = \frac{1000}{50} = 20$ packets/s/MT. MTs generate traffic in the simulation interval $[9, 10810]$s, but measurements were taken in the interval $[10, 10810]$s to exclude the initial transient MTs' states. The results shown below are the averages of five independent simulations, with different random seeds. The MPR scenario is compared to the normal Single Packet Reception (SPR) scenario when $Q_{max}$=1.

Figures 5.4, and 5.5 show analytical and simulation throughput values, as a function of $p$ (for the analytical results it is considered the network steady state and (5.22)). These figures show that the measured throughput follows the analytical values, validating them. Figure 5.4 shows the impact of different values of $Q_{max}$ when $E_b/N_0$ =7dB (i.e., $PER_{Q,p} \approx 0$, corresponding to an almost lossless scenario), $J = 8$ MTs and CB. Figure 5.5 presents the same information for EB. $1/\lambda$ was set to 5ms/packet/MT for saturated experiments and it was set to 90ms and 250ms respectively for $Q_{max} = 4$ and for $Q_{max} = 2$. Both figures show the potential throughput gains that can be achieved by employing multi-packet detection with different backoff algorithms. For higher $p$ values, the EB presents better throughput gains than the CB. Figure 5.6 shows the impact of $E_b/N_0$ for the same $1/\lambda$ values used in Figure 5.4 and Figure 5.5. The system has a maximum stable throughput value for a lossless scenario ($E_b/N_0 = 7$dB) of 0.835 and 0.821 for CB and EB respectively, with $p$ values of 0.325 and 0.304, respectively. For a scenario with $E_b/N_0 = 1$dB, the maximum stable throughput value is 0.777 and 0.747 for CB and EB, achieved with $p$ values of 0.4 and 0.36, respectively. As expected, the throughput degradation of the MAC protocol (relatively to the maximum bound) increases for lower $E_b/N_0$.

For unsaturated scenarios an interval of $p$ values exists where the throughput is equal to the forced load. This interval is delimited by the $p$ values where $\lambda J$ intersects the saturation throughput. Although EB outperforms CB for higher $p$ values, as referred previously, the maximum stable throughput gain is achieved with CB. Conventional SPR ($Q_{max} = 1$) behaves in the same manner as MPR, but supports lower values of $\lambda$.

Figure 5.7 shows the system's maximum saturation throughput ($G_{max}^{ndma} = G_{sat}^{ndma}(p_{sat}{}^{\star})$) for a set of $E_b/N_0$ values with $Q_{max} = 4$ and $Q_{max} = 2$, and different backoff algorithms. $p_{sat}{}^{\star}$ was calculated using the bisection method. As expected, the saturation throughput degrades (relatively to the maximum bound) for lower $E_b/N_0$. Despite this degradation, a sustainable throughput is achieved even in the presence of unfavourable propagation conditions (i.e. low $E_b/N_0$ values). It is also possible to observe that in terms of throughput the CB algorithm

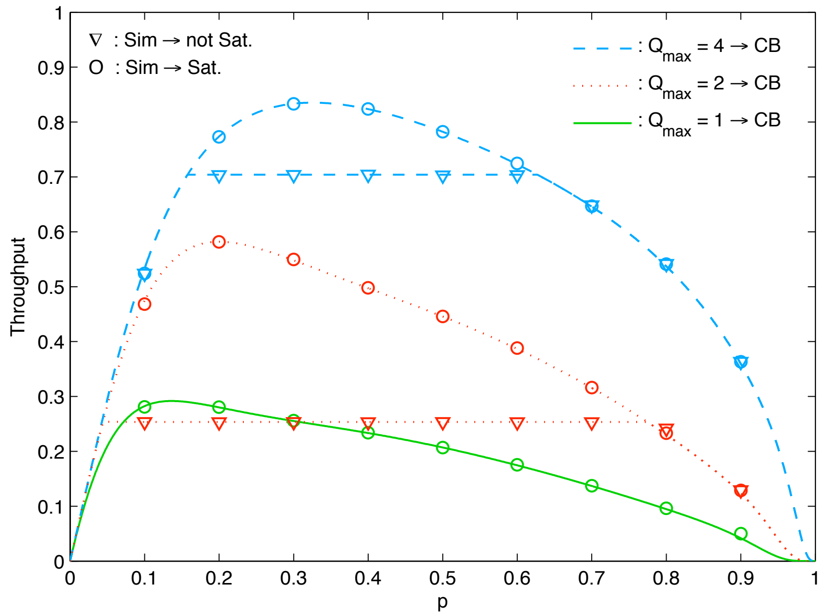



Figure 5.4: Throughput with $p$ and $Q_{max}$ for $J$ = 8 MTs, CB, $E_b/N_0$ =7dB and $1/\lambda \in \{5(sat), 90, 250\}$ ms/packet.

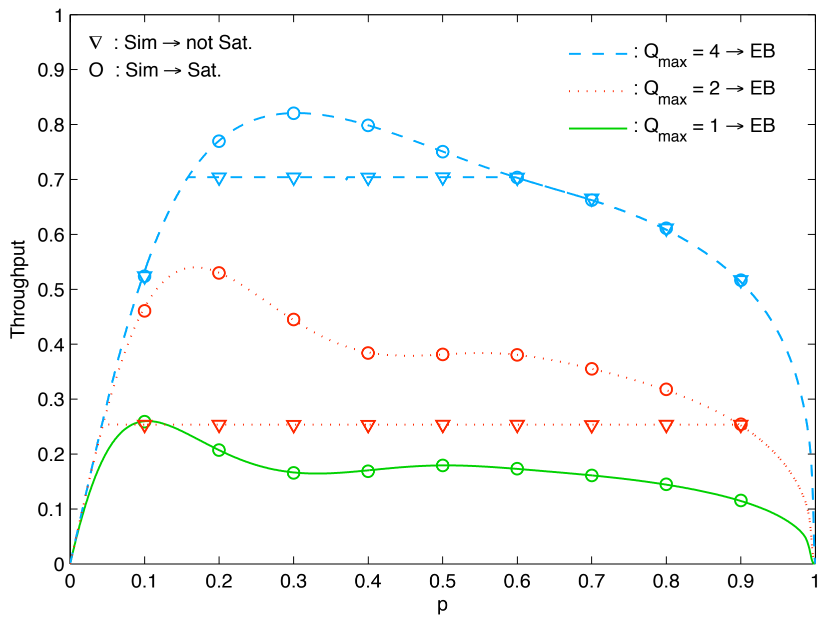



Figure 5.5: Throughput with $p$ and $Q_{max}$ for $J$ = 8 MTs, EB, $E_b/N_0$ =7dB and $1/\lambda \in \{5(sat), 90, 250\}$ ms/packet.

constantly outperforms the EB approach. Nevertheless, the difference between the two algorithms is not significant.

Figure 5.8 shows the analytical and simulation delay values with $p$ for $J$ = 8 MTs and

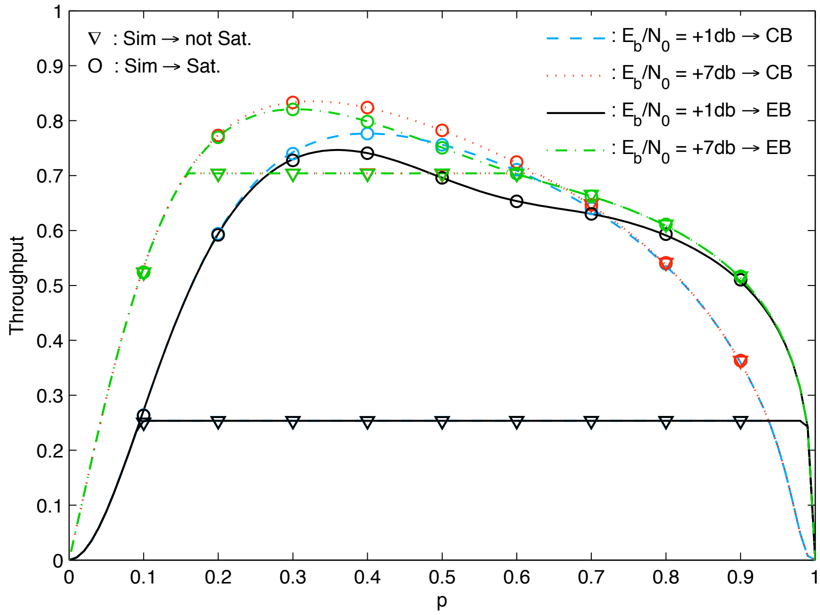



Figure 5.6: Throughput with $p$ and $E_b/N_0$ for $J = 8$ MTs, CB or EB, $Q_{max}$ =4 and $1/\lambda \in \{5(sat), 90, 250\}$ ms/packet.

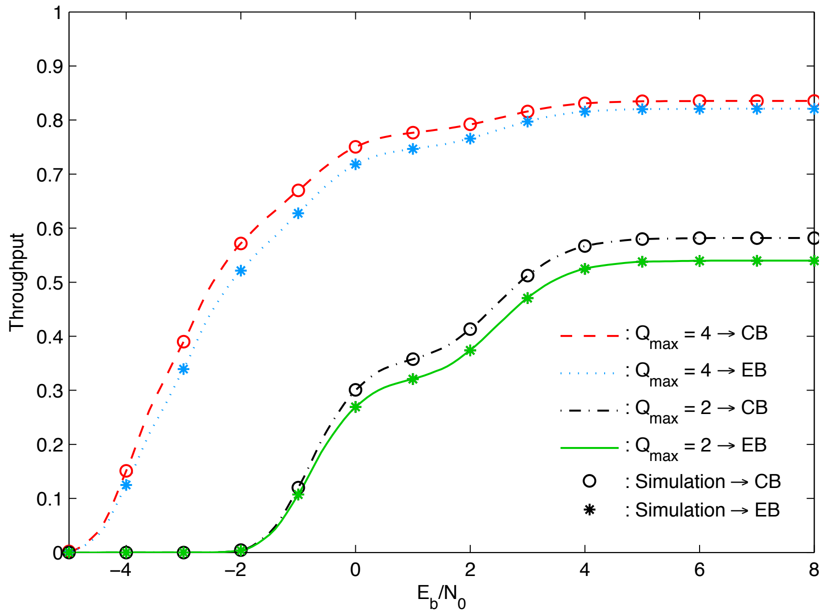



Figure 5.7: Throughput with CB, EB, $Q_{max}$ and $E_b/N_0$ for $J = 8$ MTs and saturation load.

$1/\lambda = 150$ ms/packet/MT. The analytical results were obtained using the model (5.33). Once again, the analytical model is validated by the simulations, except for $p = 0.9$ where the model does not apply because the system is already saturated (in this case, the saturation limit was reached for $p = 0.871$). The finite queue size and the proximity to the saturation

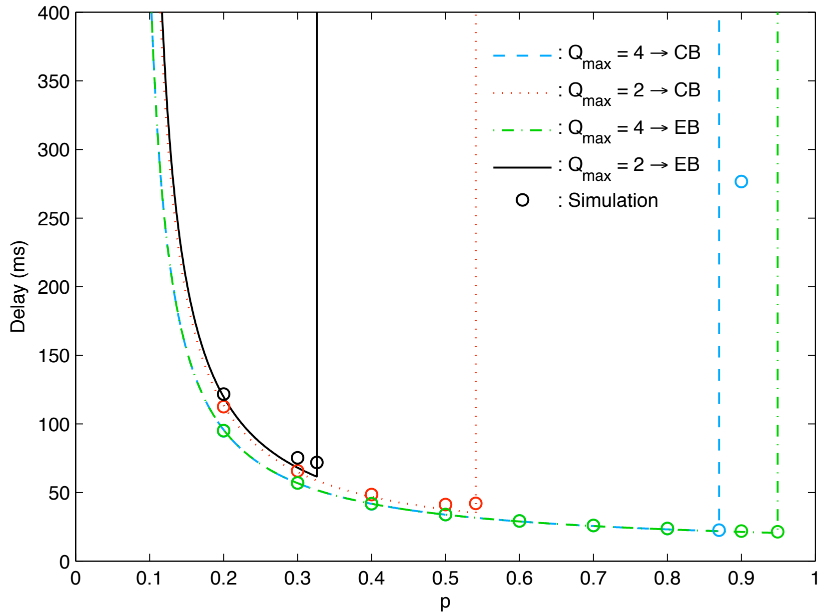



Figure 5.8: Delay with $p$ and $Q_{max}$ for $J = 8$ MTs, CB or EB, $E_b/N_0 = 7$dB and $1/\lambda = 150$ ms/packet.

bound explain the low delay value measured for $p = 0.9$. Both figures show that the delay decreases with $p$ until it reaches its minimum value at $p_{opt}{}^{\star}$. The minimum delay occurs for a $p$ value about 0.001 before the upper saturation limit. Packet delay is reduced for higher $p$ values because the first packet transmission is anticipated, albeit the higher collision probability. Figure 5.8 shows the delay variation with different $Q_{max}$ values and backoff algorithms for the lossless scenario. $Q_{max}$ reduces the system's maximum throughput, and therefore the range of stable $p$ values[3]. However, it does not reduce significantly the average delay value. The backoff algorithm plays an important role on delay values, since it creates a dependency with forced load. For EB, the range of stable $p$ values increases for low loads, because the upper saturation limit occurs for higher $p$ values. However, when the forced load starts to grow, the range of stable $p$ values decrease more abruptly than when it is used CB. Figure 5.9 shows that the delay increases for lower $E_b/N_0$, especially with low $p$ values. The error resilient properties of the detection mechanism for $Q > 2$ make the system less sensible to errors for higher $p$ values, where the average number of MTs transmitting packets per slot increases. These results show that with the correct dimensioning of $Q_{max}$, backoff algorithm and $p$, the system can provide an average delay for low $E_b/N_0$ scenarios comparable to the equivalent lossless scenario.

[3]In this case, the multi-packet detection scheme behaves as an adaptive coding technique where the "equivalent" code rate decreases as it is increased the number of transmission attempts. However, in practical implementations there is a maximum value of $Q_{max}$ due to complexity issues, as already pointed out.

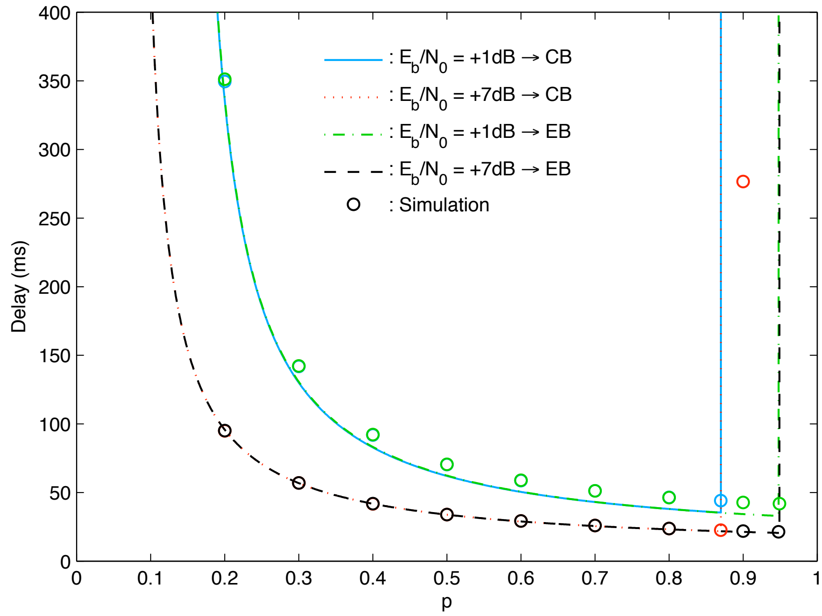



Figure 5.9: Delay with $p$ and $E_b/N_0$ for $J = 8$ MTs, CB or EB, $Q_{max} = 4$ and $1/\lambda = 150$ ms/packet.

Figure 5.10 shows the maximum saturation throughput ($G_{max}^{ndma}$) for $E_b/N_0 = 1$dB. It shows the maximum throughput gain achievable with the increase of $Q_{max}$, for a given $J$. The throughput increment decreases with $Q_{max}$, showing that there is a trade-off between the multi-packet detection complexity and the additional throughput gain when $Q_{max}$ is increased. It is also possible to observe that CB outperforms EB. Figure 5.11 depicts the minimum delay ($D_{min} = E\left[D\right]^{ndma}(p_{opt}{}^{\star})$) for a lossless scenario ($E_b/N_0 = 7$dB) and $\lambda J = \frac{1000}{150}$ packets/s. $p_{opt}{}^{\star}$ was calculated using the bisection method on the intervals of $p$ values where $R_{dat}G_{sat}^{ndma}(p) > \lambda J$. EB outperforms CB with low and constant load, regardless of $Q_{max}$, although the saturation limit is achieved more quickly as $J$ grows. The delay increases with $J$ due to the higher contention in the medium access. However, it is possible to contain the delay growth increasing $Q_{max}$ and the corresponding complexity. In the limit, if $Q_{max} = J$ the delay only depends on $J$ due to the $PER_Q$ variation [Dinis *et al.*, 2009b]. This figure also shows that the model follows the simulation results, except for EB, $Q_{max} = 2$ and $J = 100$ where the system is saturated and the model does not apply.

Figure 5.12 shows the minimum delay ($D_{min}$) on a lossless scenario (PER=0). Clearly, when the load is high (or $1/\lambda$ is low) CB outperforms EB regardless of the value of $Q_{max}$, but for low load the EB outperforms CB. The analytical model follows the simulation results, with a visible deviation only when the load is high, and consequently approaching the saturation limit. This results mainly from the fact that the i.i.d. and mutually uncorrelation

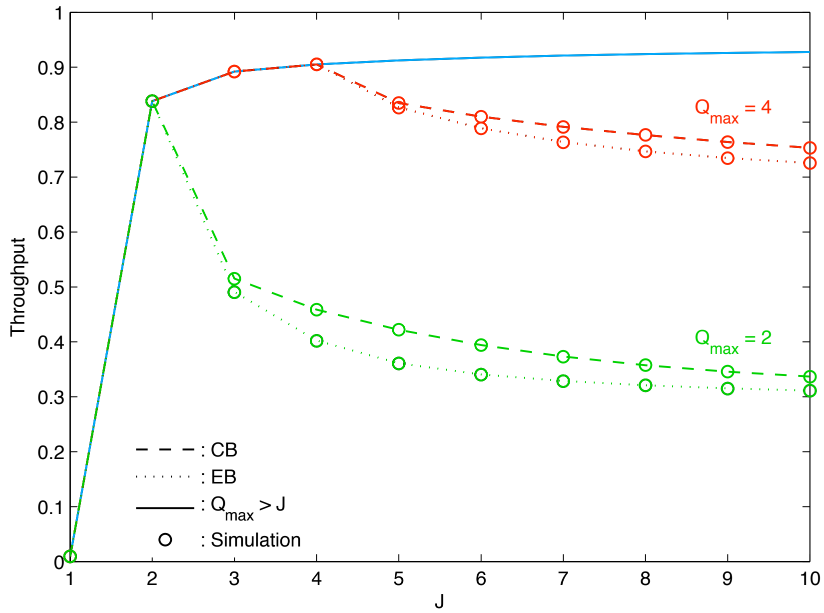



Figure 5.10: $G_{max}^{ndma}$ with $J$, CB or EB and $Q_{max}$ for $E_b/N_0 = 1$dB and saturation load.

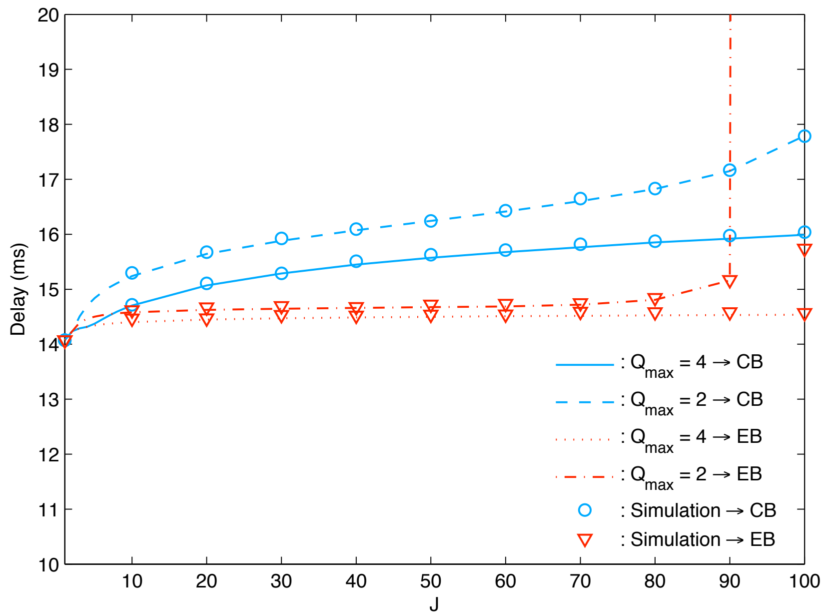



Figure 5.11: $D_{min}$ with $Q_{max}$, CB or EB and $J$, $E_b/N_0 = 7$dB and $1/(\lambda J) = 150$ ms/packet.

approximations in (5.27) and (5.30) loose accuracy near saturation. A deviation around 20% occurs for $Q_{max} = 2$ and high loads ($1/\lambda < 150$ ms/packet/MT), corresponding to $\lambda E[D]^{ndma}$ values above 75% of saturation load.

Figure 5.13 depicts the minimum delay ($D_{min}$) when it is considered $Q_{max} = 4$. When the load is low ($1/\lambda = 150$ ms/packet/MT in the figure), the analytical model follows the

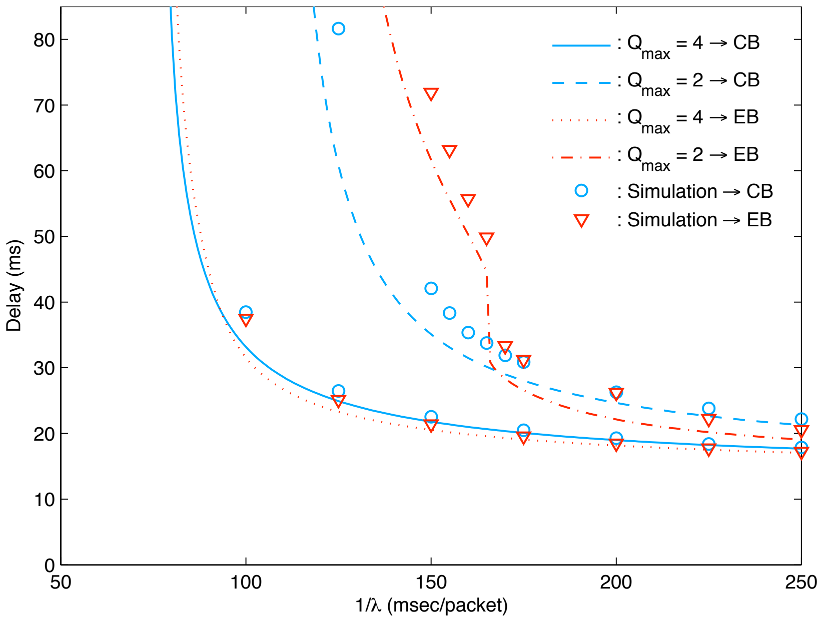



Figure 5.12: $D_{min}$ with $Q_{max}$, CB or EB and $1/\lambda$ for lossless scenario and $J = 8$ MTs.

simulation results with almost no deviation, for $E_b/N_0 > 2$ dB. However, for lower $E_b/N_0$, a deviation starts to appear, since it is approached the saturation limit. The system saturates for $\lambda \geq \frac{R_{dat} G_{max}^{ndma}}{J}$, which decreases when $E_b/N_0$ decreases. A maximum deviation of 15ms is obtained when $E_b/N_0 > 0$ dB. For $E_b/N_0 \leq 0$ dB (an unrealistic scenario), the model follows the simulation results, but with a visible deviation. For a higher load (e.g. $1/\lambda = 100$ ms/packet/MT), a constant deviation is observed (it was also present in Figure 5.12), for all range of $E_b/N_0$, due to the fact that the saturation limit is close. The system is saturated for $\frac{J}{R_{dat} G_{max}^{ndma}} = 76$ ms/packet/MT for the lossless scenario, and for higher values of $\frac{J}{R_{dat} G_{max}^{ndma}}$ for lower $E_b/N_0$ scenarios.

## 5.4 Conclusions

In this chapter it is considered the uplink of broadband wireless systems employing SC-FDE schemes and it is presented a MAC protocol for a frequency-domain multi-packet receiver. The simulation results follow precisely the analytical model results, except for the packet delay when the system approaches saturation, where a small deviation is visible due to the approximations used in the analytical model. The proposed analytical model shows that the system allows $J$ MTs (above the multi-packet detection capability) to efficiently transmit uniform Poisson traffic with a bounded average delay as long as the BS defines $p = p_{opt}{}^{\star}$. This can be calculated using local information ($R_{dat}$, $J$, $Q_{max}$, an estimation of the load

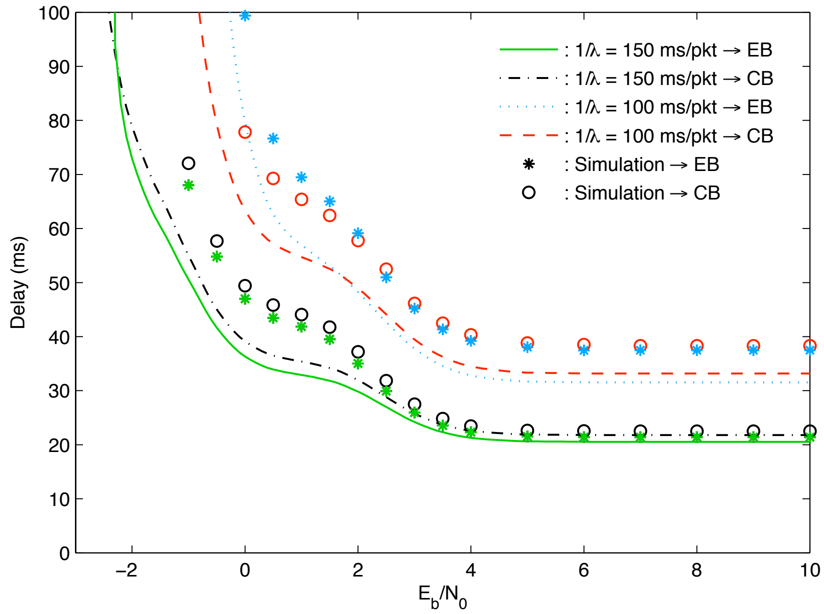



Figure 5.13: $D_{min}$ with $E_b/N_0$, CB or EB and $1/\lambda$ for $Q_{max} = 4$ and $J = 8$ MTs.

and the measured MT PER values). Besides, it shows how the saturation throughput can be maximized using the same information.

The proposed model can also be used for configuring the system. With a bounded $Q_{max}$ value the maximum number of MTs associated to a BS, or the maximum traffic packet rate in order to guarantee an objective maximum average packet delay can be computed.

# CHAPTER 6

## Performance comparison

Different approaches to cope with lost packets were addressed in the previous chapters of this thesis. One simple way of coping with lost packets is to avoid collisions using a Time Division Multiple Access (TDMA) approach and to employ the Diversity-Combining (DC) Hybrid Automatic Repeat reQuest (Hybrid ARQ) approach presented in the chapter 3. On the other hand, the Network-assisted Diversity Multiple Access (NDMA) approach proposed in chapter 5 takes advantage of the redundancy inherent to multiple collisions to separate the packets involved. Both approaches consider the use of Single Carrier with Frequency Division Equalizer (SC-FDE) schemes in the uplink transmission and they employ either frequency-domain DC or NDMA techniques to cope with lost packets in an answer to reduce complexity.

The physical level and system level performance of both systems is analysed in this chapter. This chapter begins with an overview of the Physical (PHY) and Medium Access Control (MAC) layers considered in the comparison for each case, and ends with a performance analysis focusing on throughput, delay and scalability. The work illustrated in this chapter was the subject of a conference publication [Pereira *et al.*, 2010c]and was submitted for journal publications.

## 6.1 Scenario Characterization

The uplink transmission in structured wireless systems employing SC-FDE schemes is considered, where a set of Mobile Terminals (MTs) send data to a Base Station (BS). MTs are low resource battery operated devices whereas the BS is a high resource device, which runs the DC or MultiPacket Reception (MPR)/NDMA algorithm in real-time. In the DC Hybrid ARQ approach, MTs send data packets using the time slots of a TDMA frame defined by the BS, as described in chapter 3. On the other hand, the NDMA approach consists on a

MPR receiver at the BS, able to cope with parallel transmissions as described in chapter 5. It also employs a $p$-persistent slotted ALOHA algorithm, where the BS controls the number of MTs competing for each time slot.

For the sake of comparison, some adjustments had to be made:

- Since NDMA approach includes turbo equalization, it is considered the turbo/coded version for the DC Hybrid ARQ approach, where channel coding is employed (referred in the end of section 3.1.1).

- DC Hybrid ARQ approach uses a full-duplex scheme. Therefore, it is also considered the full-duplex (slotted periodically) version of the NDMA system (mentioned in section 5.1.1).

- For the sake of simplicity, packets associated to each user from both approaches have the same duration/size.

Data packets are composed of $N_{FFT}$ Fast Fourier Transform (FFT) blocks and the physical preamble overhead is assumed to be negligible. Each FFT block carries $N_{Block}$ symbols. In both approaches, it is assumed perfect channel estimation and synchronization at the receiver side, i.e. at the BS. In addition, perfect power control and time advance mechanisms exist, able to compensate a different attenuation and propagation times.

Regarding the NDMA approach, MTs associate to a BS using a dedicated control channel before transmitting data. The BS uses the downlink channel to broadcast the MTs access probability and, possibly, to force packet retransmissions or block the transmission of new packets in the next slot. It is assumed perfect user detection and that different data packets arrive simultaneously at the receiver (i.e. BS).

## 6.2 Packet Detection Performance Results

A set of performance results concerning the two proposed schemes, DC Hybrid ARQ technique and NDMA for SC-FDE schemes is presented in this section. All the information regarding the simulations' setup used in this chapter are illustrated in Table 6.1 and 6.2. Each time-domain block has $N = 256$ data symbols selected from a Quadrature Phase Shift Keying (QPSK) constellation under a Gray mapping rule. It is considered a rate-1/2 64-state convolutional code with generators $1+D^2+D^3+D^5+D^6$ and $1+D+D^2+D^3+D^6$. The data packets correspond to one FFT block, with 32 bytes. A severely time-dispersive multipath channel with uncorrelated Rayleigh-distributed fading on different paths was assumed.

| | Figures | | | | |
|---|---|---|---|---|---|
| | **6.1** | **6.2** | **6.3** | **6.4** | **6.5** |
| **Data Symbols** | 256 | | | | |
| **Modulation** | Quadrature Phase Shift Keying | | | | |
| **Block (Bits)** | 256 | | | | |
| **Packet (Bytes)** | 32 | | | | |
| **Coding** | 64-state convolutional code with rate 1/2 | | | | |
| **Turbo equalization** | Yes | | | | |
| **Iterations** | 4 | | | | |
| **Channel conditions** | UC and EC | UC and EC+SP | UC and EC | UC and EC+SP | UC and EC+SP |
| **No PC** | No | | | | |
| $1/\lambda$ | — | — | — | — | — |
| **Type of Traffic** | — | — | — | — | — |
| $E_b/N_0$ **(dB)** | variable | | | | |
| $p$ **values (for NDMA)** | — | — | — | — | — |
| **Backoff algorithm (for NDMA)** | — | — | — | — | — |
| **R or** $Q_{max}$ | 4 | | | | |
| **Number of Stations** | — | — | — | — | — |

Table 6.1: Simulations setups from Figure 6.1 to Figure 6.5.

| | Figures | | | | |
|---|---|---|---|---|---|
| | **6.6** | **6.9** | **6.10** | **6.11** | **6.12** |
| **Data Symbols** | 256 | | | | |
| **Modulation** | Quadrature Phase Shift Keying | | | | |
| **Block (Bits)** | 256 | | | | |
| **Packet (Bytes)** | 32 | | | | |
| **Coding** | 64-state convolutional code with rate 1/2 | | | | |
| **Turbo equalization** | Yes | | | | |
| **Iterations** | 4 | | | | |
| **Channel conditions** | UC and EC+SP | EC+SP | | | |
| **No PC** | No | | | | |
| $1/\lambda$ | — | saturation | saturation | 2.5 | 2.5 |
| **Type of Traffic** | — | Poisson | Poisson | Poisson | Poisson |
| $E_b/N_0$ **(dB)** | variable | variable | -4, 0 and 4 | variable | -4, 0 and 4 |
| $p$ **values (for NDMA)** | — | optimized | optimized | optimized | optimized |
| **Backoff algorithm (for NDMA)** | — | CB | CB | CB | CB |
| **R or** $Q_{max}$ | 4 | | | | |
| **Number of Stations** | — | 20 and 50 | variable | 20 and 50 | variable |

Table 6.2: Simulations setups from Figure 6.6 to Figure 6.12.

In terms of the characteristics of the channel, three conditions and transmission strategies referred in the previous chapters were considered:

- Uncorrelated Channel (UC), when in the presence of uncorrelated fading for different retransmission attempts.
- Equal Channel (EC), when in the presence of the same channel for different retransmission attempts.
- Shifted Packet (SP), when in the presence of the same channel for different retransmission attempts (as with the EC case), but the transmitter performs cyclic shifts in the frequency-domain as described before.

It should be pointed that in the case of NDMA receiver, the correlation between channels associated to different retransmissions should be low, if not, the system of equations (5.7) might not have solution or it can be ill conditioned. As a result, the EC transmission strategy is not feasible.

Figure 6.1 to Figure 6.6 show the average Bit Error Rate (BER) and Packet Error Rate (PER) when in the presence of different number of transmission attempts ($l$ or $Q$ for the DC Hybrid ARQ or NDMA approach, respectively), different transmission strategies and considering four iterations. In Figure 6.5 and Figure 6.6 only UC and SP transmissions strategies were considered, whereas in Figure 6.1 to Figure 6.4 UC, EC and SP are presented. The results are expressed as function of $E_b/N_0$. In both cases, the SP strategy outperforms the EC strategy. UC strategy is, as expected, the strategy that presents better results, although it is an unrealistic scenario as explained before (see section 3.1.2).

It is possible to see that the DC Hybrid ARQ approach presents a lower BER/PER than the NDMA approach. The exception is for $Q$ or $l = 1$ (no collisions) where they are the same. This behaviour is justified by the lower residual interference on packet detection for the diversity combining, since it has just one MT transmitting. All the aforementioned figures also show a significant PER reduction with the increments of $l$ or $Q$, as expected since the total transmitted power grows and these approaches take full advantage of all the transmitted power.

These results point to higher rates of successful transmissions as the number of retransmissions grows, which is specially useful when in the presence of disruptive propagation conditions that lead to very low $E_b/N_0$ values. However, one “price” to pay is delay because retransmissions do take time.

The trade-off between these two aspects can be stated clearer: on one side, the DC Hybrid ARQ approach presents a better PER than the NDMA one; on the other side, the

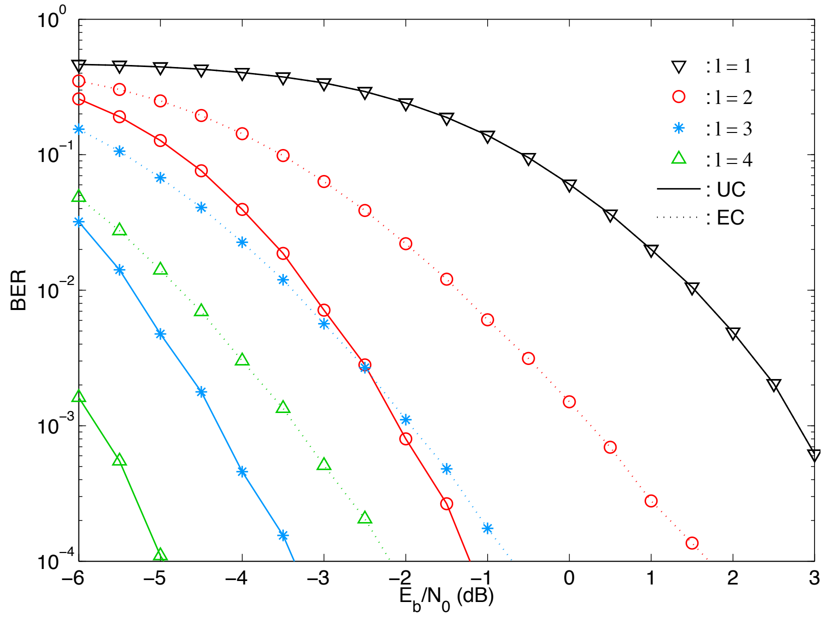



Figure 6.1: BER performance of the DC-HARQ receiver with 4 iterations when in the presence of UC and EC channel conditions.

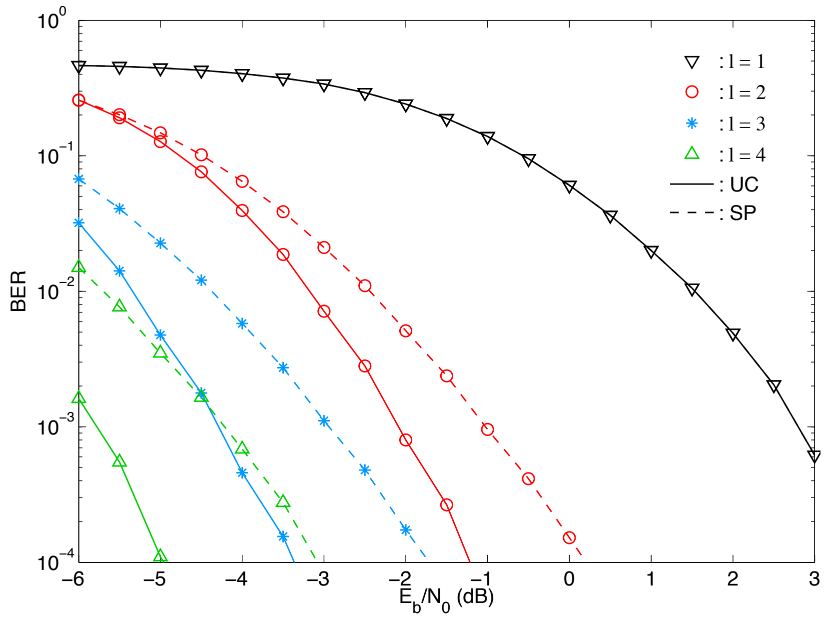



Figure 6.2: BER performance of the DC-HARQ receiver with 4 iterations when in the presence of UC channel condition and employing SP strategy.

durations of the retransmissions are different for the two systems (the DC Hybrid ARQ approach uses TDMA whereas NDMA retransmits packets sequentially). This leads us to a key question: in which scenarios is one approach faster than the other? Clearly the answer

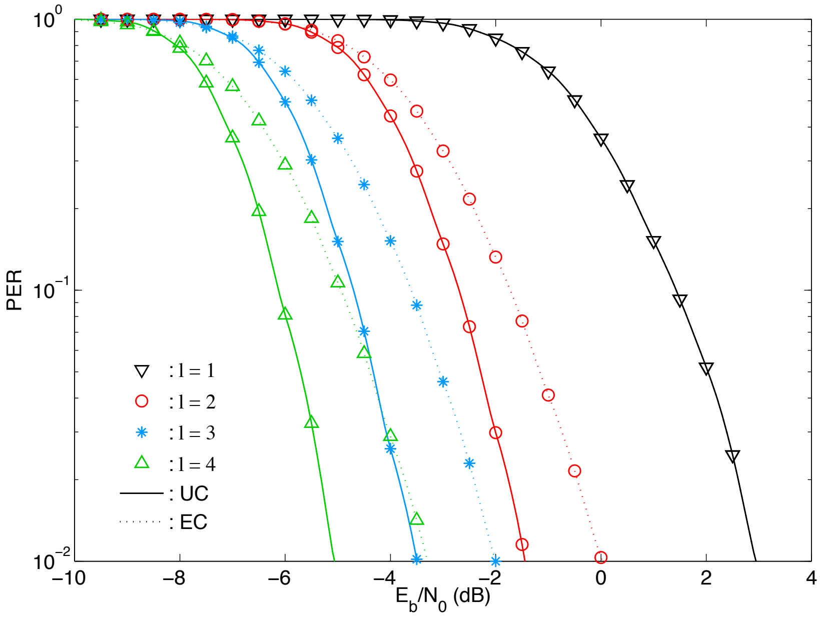



Figure 6.3: PER performance of the DC-HARQ receiver under the same conditions of Figure 6.1.

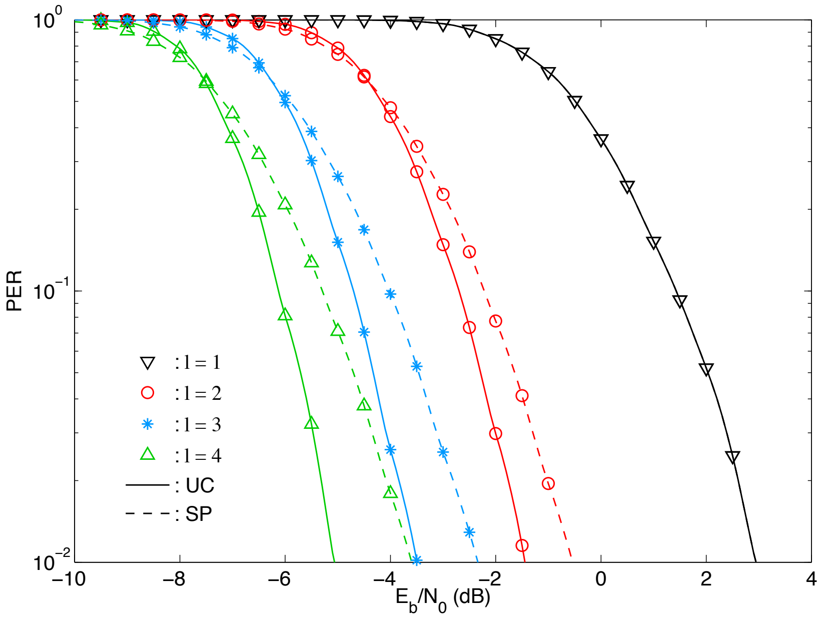



Figure 6.4: PER performance of the DC-HARQ receiver under the same conditions of Figure 6.2.

also depends on the type of MAC layer defined for each approach, which is the subject of the following section.

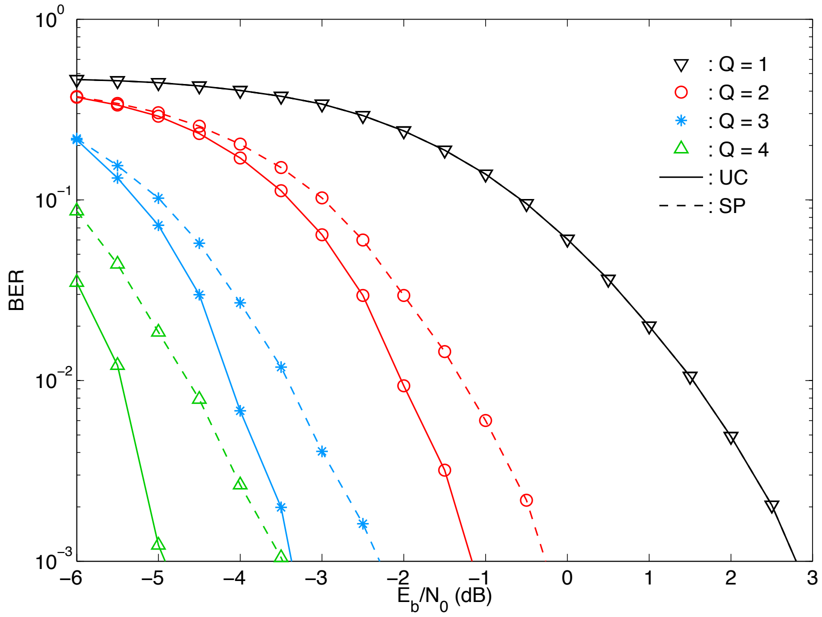



Figure 6.5: BER performance of the NDMA receiver with 4 iterations.

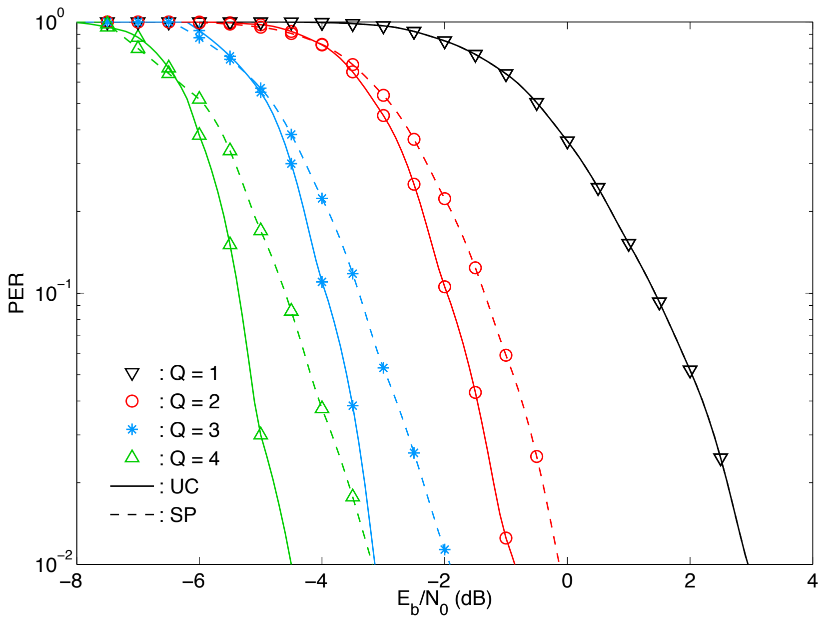



Figure 6.6: PER performance of the NDMA receiver with 4 iterations.

## 6.3 Medium Access Protocol and System Behaviour

### 6.3.1 Diversity Combining Hybrid ARQ

The DC Hybrid ARQ uses a static TDMA system. So, there is a frame that is repeated continuously and that is structured in slots. It is considered that users are assigned statically to one slot of the frame and each slot has the same length of a data block (or packet). There is a one-to-one correspondence between packets and slots. It is also assumed that a downlink channel is synchronized with the uplink one and acknowledgements of the packets arrive to the transmitter before the following slot in the next uplink frame. This usage can be compared to the usage of a dedicated channel and is quite common when hard Quality-of-Service (QoS) guarantees are needed.

In this chapter, the number of slots per TDMA frame is relevant for two reasons: it influences the delay and it has effects on the scalability of the system. To make a fair comparison with NDMA it will be assumed different frame sizes. If N users are considered, a N slot TDMA frame is assumed.

Figure 6.7 shows an example of the detection process. Each detection can result in a successful or unsuccessful reception of the data packets involved. The unsuccessful outcome could be caused by effects such as interference, channel noise, fading etc., denoted hereafter as "channel effects". Let's consider that a group of MTs are trying to transmit packets to the BS. Since each MT has a slot uniquely assigned, the successful reception of the data packets is exclusively bounded by the channel effects (i.e. there are no collisions). Aggressive channel effects, like fading or shadowing, are considered, thus leading to scenarios with low values for $E_b/N_0$.

Figure 6.7 shows some transmitting situations: user 1 transmits the first packet successfully but has to transmit twice the second packet; user 2 transmits three times the first packet; etc. Two observations can be stated: even with constant and aggressive channel effects, it is possible to achieve a successful transmission since as the number of retries grows so does the probability of successfully detecting the message at the receiver side (as illustrated in section 6.2); the time to transmit a packet successfully is not constant.

### 6.3.2 Network-assisted Diversity Multiple Access

The MAC protocol for the NDMA receiver is more active than in the previous case. As explained in chapter 5, the first objective is to avoid having more than $Q_{max}$ MTs transmitting packets simultaneously. A full-duplex approach is considered in this chapter (see section 5.1.1). The MAC protocol divides the time into sequences of epochs and behaves in a very

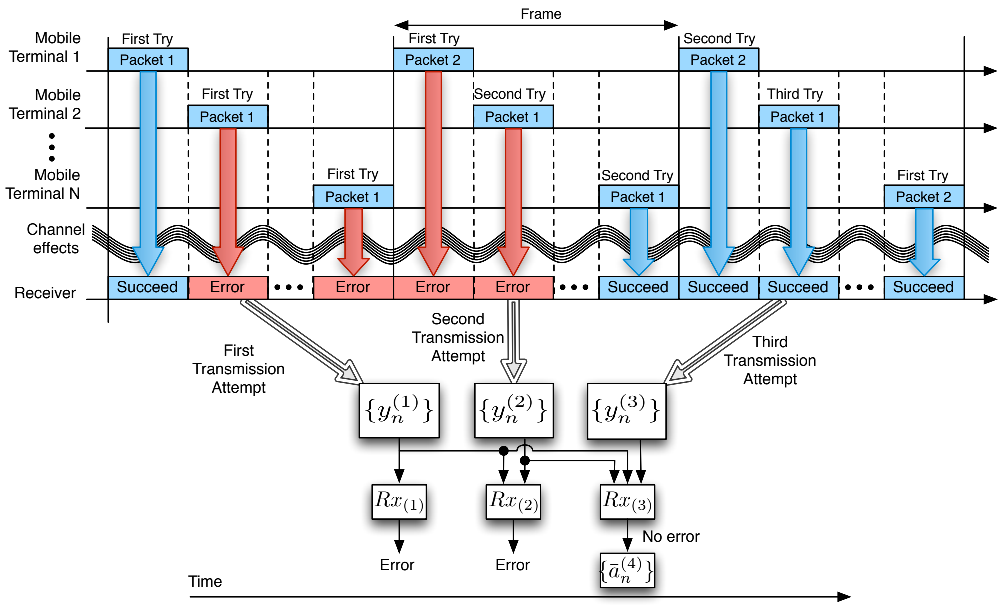



Figure 6.7: Diversity Combining Hybrid ARQ behaviour.

similar manner than in half-duplex approach described in section 5.1.1. The BS broadcasts a SYNC (Synchronization) control frame through the downlink channel (Short InterFrame Space (SIFS) periods are no more required) marking the beginning of each epoch. This frame contains the value of the access probability for this epoch, which is equal for every MT. The starting moments of an epoch are the moments for the MTs with data to access the medium with the probability of that epoch. After an epoch has started, no new MTs are allowed to start transmission. When an epoch starts three situations can happen:

i) No-one, or only a single MT transmits. The epoch finishes and only transmission errors can prevent a successful transmission (this is reported in the downlink channel).

ii) A number between two and $Q_{max}$ transmits. This epoch will last a number of slots identical to the number of transmitters (again reported in the downlink channel).

iii) More than $Q_{max}$ transmits. A "real collision" is detected and a new (lower) value for the probability is issued to the MTs via the downlink channel. All the participating MTs use this value and try to access the following slot.

One of these three situations can happen again.

Note that the system performance is optimized when the number of slots in situation iii) is minimized. Different backoff algorithms could be applied, but in this section it is considered a constant transmission probability at the beginning of each epoch. The optimal

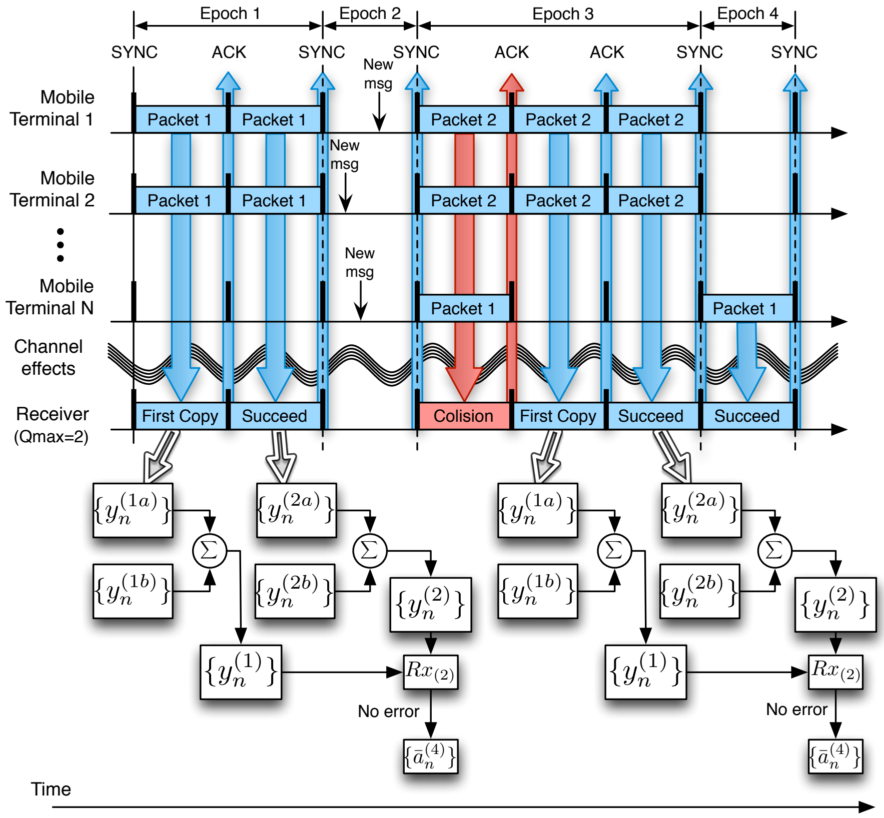



Figure 6.8: MAC behaviour for a Network-assisted Diversity Multiple Access protocol.

value for this transmission probability can be calculated based on the total number of MTs in the cell and their average load assuming Poisson sources (see chapter 5).

After receiving $Q-1$ retransmission data frames from the $Q$ MTs involved in a given collision, the detection process is executed. At the end, a data frame is correctly received if it is correctly detected by the BS. Failed data frames are retransmitted in following epochs, until a maximum number of retries, before being dropped.

Figure 6.8 illustrates the MAC behaviour of a BS with $Q_{max}$ equal to two. The first epoch illustrates situation ii) and the receiver succeeds to detect both packets. In the second epoch the situation i) happened and there was no transmission. In epoch 3 the situation iii) happened once. The new value for the transmitting probability was enough to have less than or equal to $Q_{max}$ transmitting MTs. The last epoch of the figure illustrates situation i) with one successful transmission. The figure also illustrates the moments when new messages are ready to be transmitted.

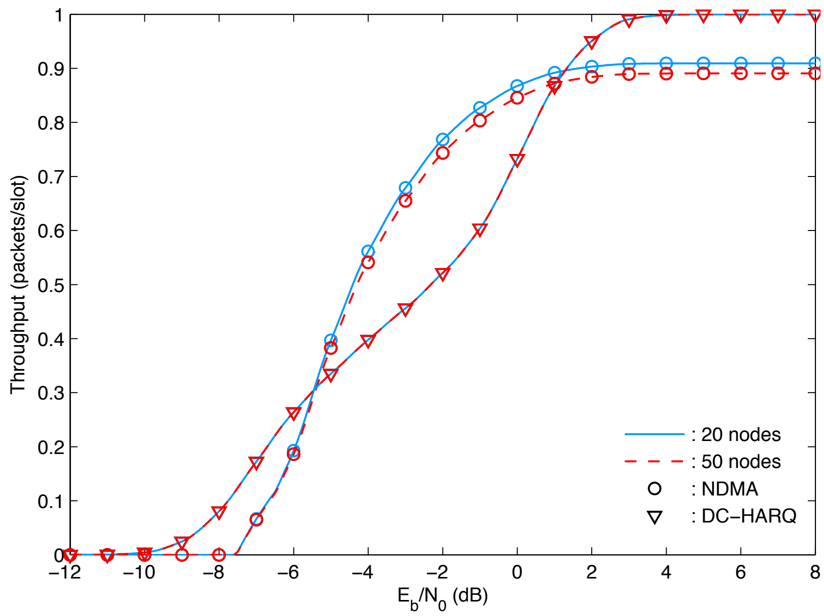



Figure 6.9: Saturation throughput over $E_b/N_0$.

## 6.4 DC Hybrid ARQ vs NDMA

From the previous descriptions, it is clear that NDMA epochs can be shorter than TDMA frames and might compensate in terms of delay the fact that the PER is higher for NDMA (forcing more retransmissions). In this section, the system level scalability and throughput-delay performances are evaluated for different $E_b/N_0$ conditions, with the help of the analytical tools presented in chapter 3 and chapter 5. Given the complexity of the NDMA system it was decided to have $Q_{max} = 4$ with an optimal MT transmission probability calculated using the model presented in chapter 5. For the DC Hybrid ARQ the complexity is not so serious but for the sake of comparisons it was also decided to have a maximum of four transmissions per packet. Having more (five) would create slightly better PER conditions but it could also deteriorate the delay. The number of slots per frame, $N$, has equally an important influence on the delay. In this case, the realized experiments considered the minimum value for $N$ that allows $Q$ MTs to communicate, i.e. $N = Q$.

Figures 6.9 and 6.10 depict the saturation throughput. Figure 6.9 shows its variation with $E_b/N_0$ when two different values for the number of MTs (or $N$) are considered. Figure 6.10 shows its variation with the number of MTs (or $N$), when three values of $E_b/N_0$ are considered.

Figure 6.9 shows that throughput gains are achievable with DC Hybrid ARQ for very low $E_b/N_0$ conditions, for $E_b/N_0 < -5.5$dB. This is a direct consequence of the lower residual

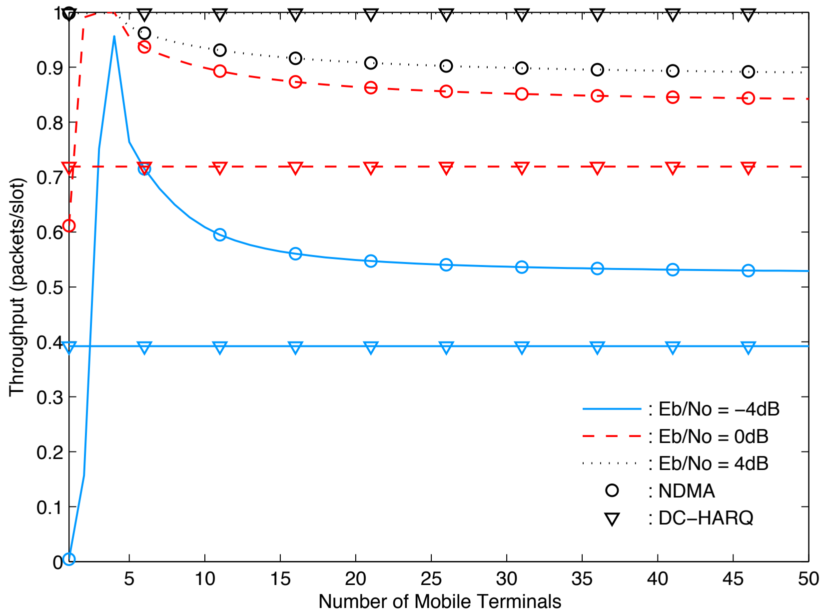



Figure 6.10: Saturation throughput over number of mobile terminals.

interference between the MTs and the features of the packet combination, translated into a lower PER. The relative gain for higher values of $E_b/N_0$ is not surprising since there are almost no transmission errors and the TDMA scheme prevents waste with unintelligible collisions. However, between $E_b/N_0 \approx -5.5$dB and $E_b/N_0 \approx 1$dB, the NDMA system outperforms the DC Hybrid ARQ approach. In spite of the higher PER, the slot mechanism of NDMA ends up being more efficient than the fixed TDMA structure. It should be pointed out that these $E_b/N_0$ values represents the energy for a single packet. When a packet is retransmitted $Q$ times and all those transmissions are combined, the energy associated to the reception of that packet is respectively $Q$ times superior than the represented by $E_b$. Figure 6.10 shows that the throughput is quite stable with the number of MTs (or $N$). The exception happens for the NDMA case when $Q$ is lower and near $Q_{max}$, where it exhibits its maximum throughput. The reason is simple: for these cases the transmission probability is one (see section 5.1.1) given that the number of users is very limited and the system behaves almost as an error-free TDMA. Apart from this aspect, the figure shows that both approaches are scalable, i.e. can be used for a higher number of MTs.

Figures 6.11 and 6.12 show the results of a similar setting of experiments but now measuring delay. The first refers to different ratios of $E_b/N_0$ for two different numbers of MTs (or $N$), and in the second the number of MTs (or $N$) was varied for three values of $E_b/N_0$. In both figures it is considered a uniform load based on a Poisson generator with $\lambda = 0.4$ packets/slot. Figure 6.11 depicts that within the stability bounds of the NDMA approach,

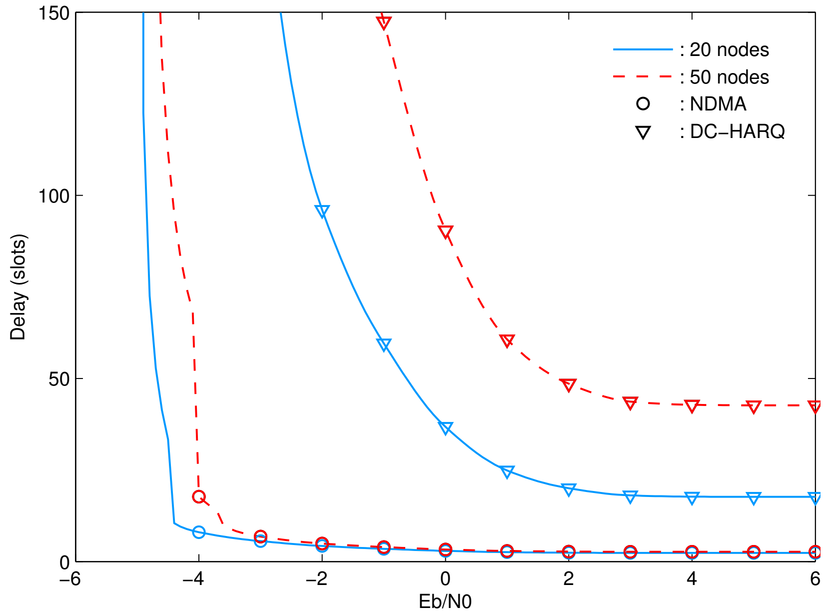



Figure 6.11: Delay over $E_b/N_0$.

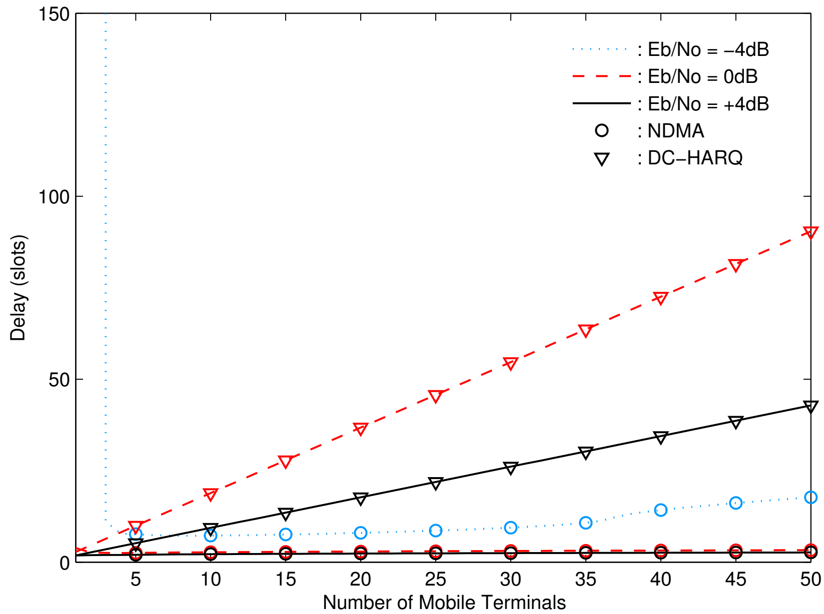



Figure 6.12: Delay over number of nodes.

this approach exhibits lower delays than the DC Hybrid ARQ ones. Figure 6.12 shows that the delay varies linearly with the number of MTs. It is also shown that the slope is higher for the DC Hybrid ARQ approach, and that the NDMA approach has always lower values. This stable behaviour of NDMA happens in what it is known as the “stability region” of NDMA [Dinis *et al.*, 2009b], and it features around 0.89 packets/slot with 50 nodes as it is shown on Figure 6.9. With these performance results, both NDMA[1] protocol and DC Hybrid ARQ techniques are viable solutions for future wireless communication systems.

## 6.5 Conclusions

This chapter reviewed two approaches of coping with corrupted packets in a broadband wireless system, employing SC-FDE schemes. These approaches are based on time diversity which is a new paradigm on PHY-MAC cross-layer design. NDMA approach relies on a random access protocol that with a jointly designed MAC-PHY layer provides increased throughput (around 0.89 packet/slot) and low delay characteristics over a wide range of $E_b/N_0$ values and number of MTs. DC Hybrid ARQ is more focused when strict QoS requirements (e.g. jitter guarantees or high throughput) enter into consideration. On both systems, complexity issues are relegated to the receiver side, which is the ideal case for the uplink direction on cellular radio systems and makes these approaches suitable for future wireless technologies.

This chapter also highlighted a curious aspect of complementarity between these two techniques: for certain values of $E_b/N_0$ a shared channel can be more efficient than a dedicated one and a radio resource control entity could take advantage of it easily. Considering all layers it is curious to see that for certain propagation and load conditions the rigidity of the TDMA structure associated with DC makes the NDMA approach more interesting. These conditions might be exploited at radio resource control.

[1]NDMA protocol is particularly suited for a random access traffic class.

# CHAPTER 7

## Conclusions and Discussion

Wireless communications have been a major topic of research over the years, where reliability issues have always played an important role. Regarding reliability, two different ways of losing packets exist: when in the presence of poor propagation conditions (e.g. deep fading, strong shadowing effects and/or strong interference levels) or when collisions happen (associated with the medium access).

This thesis proposes two distinct cross-layer time diversity approaches to cope with lost packets in a traditional broadband wireless cellular system, where several Mobile Terminals (MTs) exist for a single Base Station (BS). Both proposes are tailored to the uplink of a wireless communication system, where a Single Carrier with Frequency Division Equalizer (SC-FDE) scheme is employed.

Chapter 2 presents a general overview of different diversity methods and error control schemes. A Diversity-Combining (DC) Hybrid Automatic Repeat reQuest (Hybrid ARQ) system employed in a Time Division Multiple Access (TDMA) architecture is presented in chapter 3: an efficient way of coping with lost packets due to poor propagation conditions. By applying the DC proposal in a full-duplex TDMA system with a single slot per frame, chapter 3 carries out exact throughput, packet delay and queue-size analysis. The proposed analytical model applies to any packet arrival distribution and to any transmission technique employing a TDMA Hybrid ARQ scheme. Several simulation results are presented showing the correctness and accuracy of the proposed model for two distinctive packet arrival streams: Poisson and Geometric. Furthermore, effective power control is achievable with this model by defining minimum Signal-to-Noise Ratio (SNR) bounds that satisfy throughput and delay requirements. Nevertheless, the analysis performed in chapter 3 assumes a collision free environment, which is not always possible to provide.

Regarding random access channels, chapter 4 provides a general overview of conventional collision resolution schemes while chapter 5 presents a MultiPacket Reception (MPR) solution for it. I propose a $p$-persistent Network-assisted Diversity Multiple Access (NDMA) cross-layer protocol: an efficient way of coping with lost packet due to collisions. This proposal provides a Medium Access Control (MAC) protocol designed for a NDMA multi-packet receiver, where both half and full-duplex systems are employed for saturated and unsaturated loads. Once again, an analytical model is presented and the throughput and delay performance is analysed showing that the simulation results follow the analytical model results. The proposed cross-layer solution allows $J$ MTs competing and efficiently transmitting uniform Poisson traffic with a bounded average delay. By employing the proposed analytical model, the BS can optimize the system to limit the average delay or maximize the throughput. To achieve this, the BS needs local information, like the estimation of the load or the measured Packet Error Rate (PER) values for each MT. The proposed model can also be used to configure the system. Given $Q_{max}$, the bounded maximum number of MTs allowed to transmit in one slot, the model can be used to compute the maximum number of MTs associated to a BS, or the maximum traffic packet rate that guarantees an objective maximum average packet delay.

Chapter 6 reviews the two previously proposed approaches and presents a comparison between them. The successful implementation of a theoretical concept or a technique requires an understanding of how it interacts with the wireless system as a whole. As a result, in this chapter the system point of view is analysed: results concerning only the Physical (PHY) layer and the joint designed PHY-MAC layers are presented. From these results I can conclude that DC Hybrid ARQ receiver outperforms systematically the NDMA receiver, essentially due to its lower residual interference. However, despite that conclusion, NDMA approach relies on a jointly designed PHY-MAC random access protocol that provides increased throughput (around 0.89 packet/slot) and low delay characteristics over a wide range of $E_b/N_0$ values and number of MTs. DC Hybrid ARQ is more focused when strict Quality-of-Service (QoS) requirements (e.g. jitter guarantees or high throughput) enters into consideration. On the other hand, the $p$-persistent NDMA protocol is able to achieve higher throughput and lower delay values when in the presence of a specific range of $E_b/N_0$. In both systems the complexity issues are relegated to the receiver side, which is the ideal case for the uplink direction on wireless cellular radio systems and makes these approaches suitable for future wireless technologies.

Chapter 6 also highlights a curious aspect of complementarity between the two techniques analysed in this thesis: for certain values of $E_b/N_0$ a shared channel using NDMA can be more efficient than a dedicated one using Hybrid ARQ TDMA and a radio resource control entity could take advantage of it easily.

## 7.1 Contributions and Future Work

Several contributions for the cross-layer design of systems that cope with lost packets have been proposed in this thesis. Contributions regarding low complex receivers, analytical models, MAC protocols and how they influence the global system performance, have been presented. The two proposed analytical models are generic and flexible enough to allow future adaptation to conceptually similar schemes. The major contributions are associated to in the work developed in chapters 3, 5 and 6 and led to several publications on conferences and journals. A summary list of these publications is given:

- A first publication regarding a turbo iterative multi-packet detector with interference cancellation, where linear receivers are considered was proposed in [Dinis *et al.*, 2009b].
- The half-duplex $p$-persistent NDMA protocol for saturated loads was firstly proposed in [Pereira *et al.*, 2009a].
- A study concerning the backoff algorithms for the half-duplex NDMA protocol is presented in [Pereira *et al.*, 2009b].
- Firstly in [Pereira *et al.*, 2010d] with a short preview, and subsequently in [Pereira *et al.*, 2010b], it is proposed the DC Hybrid ARQ analytical model. An extensive study was given in [Pereira *et al.*, 2012].
- The half-duplex $p$-persistent NDMA protocol for unsaturated loads is presented in [Pereira *et al.*, 2010a].
- The comparison between the two approaches to cope with lost packets proposed in this thesis, is presented in [Pereira *et al.*, 2010c].
- An extensive study about the DC Hybrid ARQ proposal is given in [Pereira *et al.*, 2012].
- A set of interesting publications outside the scope of this thesis, where the analytical models proposed in this thesis were applied or extended can be consulted in [Ganhão *et al.*, 2010b,a, 2011c,a,b].

The out of scope work includes studies concerning energy efficiency [Ganhão *et al.*, 2010a,b, 2011a,c]. Considering the DC Hybrid ARQ proposed scheme, they analyse the energy per useful packet concerning satellite applications or High Altitude Platform (HAP) [Ganhão *et al.*, 2010b], as well as Wireless Sensor Networks (WSN) [Ganhão *et al.*, 2010a].

Both approaches need to consider minimum transmission power based on the goodput and delay constraints. Regarding the $p$-persistent NDMA proposal, it is assumed that perfect power control is used, where the average reception power is equal for all the stations. By employing multiple power levels [Ganhão *et al.*, 2011c], and studying the energy per useful packet, some of the stations may not retransmit its packets all the times.

Another approach mentioned in chapter 6, is the proper combination of the two proposed solutions. Following this perspective, an Hybrid ARQ NDMA access mechanism has been proposed [Ganhão *et al.*, 2011a,b]. In this approach, the number of retransmissions, $Q$, is no longer implicitly connected with the number of terminals involved in the collision. The PER can be reduced when more than $Q$ copies of the packets are available. As result, the access mechanism forces MTs with reception errors during a collision resolution epoch to transmit more than $Q$ times. Although still in an early stage it was already awarded as runner-up best paper award in a major international conference [Ganhão *et al.*, 2011a] proving its potential application in future wireless technologies.

**Future Work**

The work presented in this thesis can be seen as a basis for future ideas. Considering the analytical model of the DC Hybrid ARQ proposed scheme, a possible evolution may be related to the number of assigned slots to each MT by the BS. The proposed model is only capable of handling the static assignment of one slot per MT in each frame. A possible evolution would be the development of an analytical model able to cope with the scheduling of two or more slots per MT. In addition to the mathematical contribution per se, this new analytical model would open the door for possible studies concerning approaches to reach delay bounds (i.e. QoS restrictions) when an adaptive slot allocation by the BS is employed.

Regarding the proposed NDMA scheme, a possible future work is related with the backoff algorithm. Instead of choosing a fixed or exponential backoff, the transmission probability could be regulated by an adaptive solution. This adaptive backoff, processed by the BS and broadcast in the downlink, could be able to handle variable traffic with a high rate of success (i.e. minimizing the delay as well as maximizing the throughput).

Evolutions on the recently proposed Hybrid ARQ NDMA protocol [Ganhão *et al.*, 2011a] can also be linked for interesting future work. Since in this system, epochs duration (and consequently the number of retransmissions) are no longer implicitly connected with the number of collided packets, different levels of QoS can be defined for each user, i.e. different number of extra retransmissions. Furthermore, with the proliferation of Multiple Input Multiple Output (MIMO) systems, multiple simultaneous transmissions of the same packet by a single MT could increase the number of collided packets as well as the system's performance.

# Bibliography


General Information: Binary Synchronous Communication. Publication GA27-3004, IBM Corporation, WhitePlains, NY, 1969. (Cited on page 23.)

3GPP. Technical Specifications and Technical Reports for a UTRAN-based 3GPP system, Release 99, TR 21.101, March 2000. (Cited on page 3.)

3GPP. Technical Specifications and Technical Reports for a UTRAN-based 3GPP system, Release 5, TR 21.101, June 2002. URL http://www.3gpp.org. (Cited on page 3.)

3GPP. Technical Specifications and Technical Reports for a UTRAN-based 3GPP system, Release 6, TR 21.101, December 2004. URL http://www.3gpp.org. (Cited on page 3.)

3GPP. Technical Specifications and Technical Reports for a UTRAN-based 3GPP system, Release 7, TR 21.101, September 2007. URL http://www.3gpp.org. (Cited on page 3.)

3GPP. Technical Specifications and Technical Reports for a UTRAN-based 3GPP system, Release 8, TR 21.101, December 2008. URL http://www.3gpp.org. (Cited on page 3.)

3GPP. Technical Specifications and Technical Reports for a UTRAN-based 3GPP system, Release 10, TR 21.101, March 2011. URL http://www.3gpp.org. (Cited on page 4.)

N. Abramson. The Throughput of Packet Broadcasting Channels. *IEEE Transactions on Communications*, 25(1):117–128, 1977. doi: 10.1109/TCOM.1977.1093713. (Cited on page 80.)

N. Abramson. Multiple access in wireless digital networks. *Proceedings of the IEEE*, 82(9): 1360–1370, 1994. doi: 10.1109/5.317082. (Cited on page 69.)

Norman Abramson. THE ALOHA SYSTEM: another alternative for computer communications. In *Proceedings of the November 17-19, 1970, fall joint computer conference*, AFIPS '70 (Fall), pages 281–285, New York, NY, USA, 1970. ACM. (Cited on page 68.)

F. Adachi and S. Ito. Efficient ARQ with time diversity reception technique - a time diversity ARQ. *Electronics Letters*, 22(23):1257–1258, 1986. doi: 10.1049/el:19860861. (Cited on page 37.)

F. Adachi, S. Ito, and K. Ohno. Performance analysis of a time diversity ARQ in land mobile radio. *IEEE Transactions on Communications*, 37(2):177–183, 1989. doi: 10.1109/26.20086. (Cited on pages 37 and 39.)

S. Adireddy and L. Tong. Exploiting decentralized channel state information for random access. *IEEE Transactions on Information Theory*, 51(2):537–561, 2005. doi: 10.1109/TIT.2004.840878. (Cited on pages 82 and 83.)

A. Agrawal, J. G. Andrews, J. M. Cioffi, and Teresa Meng. Iterative power control for imperfect successive interference cancellation. *IEEE Transactions on Wireless Communications*, 4(3):878–884, 2005. doi: 10.1109/TWC.2005.846996. (Cited on page 78.)

P. D. Alexander, M. C. Reed, J. A. Asenstorfer, and C. B. Schlegel. Iterative multiuser interference reduction: turbo CDMA. *IEEE Transactions on Communications*, 47(7):1008–1014, 1999. doi: 10.1109/26.774851. (Cited on page 79.)

V. Anantharam. The stability region of the finite-user slotted ALOHA protocol. *IEEE Transactions on Information Theory*, 37(3):535–540, 1991. (Cited on page 69.)

J. G. Andrews. Interference cancellation for cellular systems: a contemporary overview. *IEEE Wireless Communications Magazine*, 12(2):19–29, 2005. doi: 10.1109/MWC.2005.1421925. (Cited on pages 77, 78, and 79.)

J. G. Andrews and T. H. Meng. Optimum power control for successive interference cancellation with imperfect channel estimation. *IEEE Transactions on Wireless Communications*, 2(2):375–383, 2003. doi: 10.1109/TWC.2003.809123. (Cited on page 79.)

J. Arnbak and W. van Blitterswijk. Capacity of Slotted ALOHA in Rayleigh-Fading Channels. *IEEE Journal on Selected Areas in Communications*, 5(2):261–269, 1987. doi: 10.1109/JSAC.1987.1146521. (Cited on page 80.)

L. Badia, M. Rossi, and M. Zorzi. SR ARQ packet delay statistics on markov channels in the presence of variable arrival rate. *IEEE Transactions on Wireless Communications*, 5(7):1639–1644, July 2006. ISSN 1536-1276. (Cited on page 40.)

L. Badia, M. Levorato, and M. Zorzi. Markov analysis of selective repeat type II hybrid ARQ using block codes. *IEEE Transactions on Communications*, 56(9):1434–1441, September 2008. ISSN 0090-6778. (Cited on page 40.)

G. Bauch. Differential modulation and cyclic delay diversity in orthogonal frequency-division multiplex. *IEEE Transactions on Communications*, 54(5):798–801, 2006. doi: 10.1109/TCOMM.2006.874005. (Cited on pages 47 and 48.)

S. Benedetto and E. Biglieri. *Principles of digital transmission: with wireless applications*. Kluwer academic publishers, 1999. (Cited on page 18.)

G. Benelli. An ARQ Scheme with Memory and Soft Error Detectors. *IEEE Transactions on Communications*, 33(3):285–288, Mar. 1985. ISSN 0090-6778. (Cited on pages 5, 36, 37, and 39.)

G. Benelli. Efficient ARQ technique using soft demodulation. *Electronics Letters*, 22(4):205–206, 1986. doi: 10.1049/el:19860143. (Cited on pages 37 and 39.)

G. Benelli. An ARQ Scheme with Memory and Integrated Modulation. *IEEE Transactions on Communications*, 35(7):689–697, 1987. doi: 10.1109/TCOM.1987.1096851. (Cited on pages 37 and 39.)

N. Benvenuto and S. Tomasin. Block iterative DFE for single carrier modulation. *IEEE Electronics Letters*, 38(19):1144–1145, 2002. ISSN 0013-5194. (Cited on page 20.)

N. Benvenuto, R. Dinis, D. Falconer, and S. Tomasin. Single Carrier Modulation With Nonlinear Frequency Domain Equalization: An Idea Whose Time Has Come - Again. *Proceedings of the IEEE*, 98(1):69 –96, Jan. 2010. ISSN 0018-9219. (Cited on pages 4, 21, 43, and 96.)

C. Berrou and A. Glavieux. Near optimum error correcting coding and decoding: turbo-codes. *IEEE Transactions on Communications*, 44(10):1261–1271, 1996. doi: 10.1109/26.539767. (Cited on pages 33 and 79.)

C. Berrou, A. Glavieux, and P. Thitimajshima. Near Shannon limit error-correcting coding and decoding: Turbo-codes. 1. In *Proc. IEEE Int Communications ICC 93. Geneva. Technical Program, Conf. Record Conf*, volume 2, pages 1064–1070, 1993. doi: 10.1109/ICC.1993.397441. (Cited on pages 33 and 79.)

D. Bersakas and R. Gallager. *Data Networks*. Prentice-Hall, 2nd edition, 1992. (Cited on pages 69, 71, 111, and 112.)

G. Bianchi. Performance analysis of the IEEE 802.11 distributed coordination function. *IEEE Journal on Selected Areas in Communications*, 18(3):535–547, 2000. doi: 10.1109/49.840210. (Cited on page 74.)

G. Bloch, S. Greiner, H. de Meer, and S. Trivedi. *Queueing networks and Markov chains: Modeling and performance evaluation with computer science applications*. John Wiley & Sons, Inc., New York, 1998. (Cited on pages 109 and 111.)

R. S. Blum, J. H. Winters, and N. R. Sollenberger. On the capacity of cellular systems with MIMO. *IEEE Communications Letters*, 6(6):242–244, 2002. doi: 10.1109/LCOMM.2002.1010868. (Cited on page 76.)

G. Boudreau, J. Panicker, Ning Guo, Rui Chang, Neng Wang, and S. Vrzic. Interference coordination and cancellation for 4G networks. *IEEE Communications Magazine*, 47(4):74–81, 2009. doi: 10.1109/MCOM.2009.4907410. (Cited on pages 4 and 76.)

H. Boujemâa. Delay Analysis of Cooperative Truncated HARQ With Opportunistic Relaying. *IEEE Transactions on Vehicular Technology*, 58(9):4795–4804, Nov. 2009. ISSN 0018-9545. (Cited on page 40.)

H. Boujemâa, M.W. El Bahri, and M. Siala. Throughput performance of Hybrid ARQ schemes with packet and code combining for DS-CDMA downlink systems using orthogonal spreading sequences. *European Transactions on Telecommunications*, 19(2):173–179, Feb. 2008. (Cited on page 40.)

H. O. Burton and D. D. Sullivan. Errors and error control. *Proceedings of the IEEE*, 60(11):1293–1301, 1972. doi: 10.1109/PROC.1972.8897. (Cited on page 23.)

G. Caire and D. Tuninetti. The throughput of Hybrid-ARQ protocols for the Gaussian collision channel. *IEEE Transactions on Information Theory*, 47(5):1971–1988, Jul 2001. ISSN 0018-9448. (Cited on pages 29 and 39.)

J. Capetanakis. Generalized TDMA: The Multi-Accessing Tree Protocol. *IEEE Transactions on Communications*, 27(10):1476–1484, 1979a. (Cited on pages 69 and 71.)

J. Capetanakis. Tree algorithms for packet broadcast channels. *IEEE Transactions on Information Theory*, 25(5):505–515, 1979b. (Cited on pages 69 and 71.)

G. D. Celik, G. Zussman, W. F. Khan, and E. Modiano. MAC for Networks with Multipacket Reception Capability and Spatially Distributed Nodes. In *Proc. INFOCOM 2008. The 27th Conf. Computer Communications. IEEE*, pages 1436–1444, 2008. doi: 10.1109/INFOCOM.2008.202. (Cited on page 82.)

G. D. Celik, G. Zussman, W. F. Khan, and E. Modiano. MAC for Networks with Multipacket Reception Capability and Spatially Distributed Nodes. *IEEE Transactions on Mobile Computing*, 9(2):226–240, 2010. doi: 10.1109/TMC.2009.135. (Cited on page 82.)

S. S. Chakraborty, E. Yli-Juuti, and M. Liinaharja. An ARQ scheme with packet combining. *IEEE Communications Letters*, 2(7):200–202, 1998. doi: 10.1109/4234.703915. (Cited on pages 38 and 39.)

S. S. Chakraborty, M. Liinabarja, and E. Yli-Juuti. An adaptive ARQ scheme with packet combining for time varying channels. *IEEE Communications Letters*, 3(2):52–54, 1999. doi: 10.1109/4234.749360. (Cited on pages 38 and 39.)

D. Chase. Code Combining–A Maximum-Likelihood Decoding Approach for Combining an Arbitrary Number of Noisy Packets. *IEEE Transactions on Communications*, 33(5):385–393, May 1985. ISSN 0090-6778. (Cited on pages 30, 31, 32, 34, 36, and 37.)

Jung-Fu Cheng. Coding performance of hybrid ARQ schemes. *IEEE Transactions on Communications*, 54(6):1017, June 2006. ISSN 0090-6778. (Cited on pages 30 and 39.)

F. Chiti and R. Fantacci. A Soft Combining Hybrid-ARQ Technique Applied to Throughput Maximization Within 3G Satellite IP Networks. *IEEE Transactions on Vehicular Technology*, 56(2):594–604, 2007. doi: 10.1109/TVT.2007.891483. (Cited on pages 29 and 38.)

I. Chlamtac and A. Farago. An optimal channel access protocol with multiple reception capacity. *IEEE Transactions on Computers*, 43(4):480–484, 1994. doi: 10.1109/12.278486. (Cited on page 82.)

Yongkyu Choi, Sunghyun Choi, and Seokhyun Yoon. MSDU-based ARQ scheme for IP-level performance maximization. In *IEEE Global Telecommunications Conference, 2005. GLOBECOM'05*, volume 5, Dec. 2005. (Cited on page 40.)

Seong Taek Chung and A. J. Goldsmith. Degrees of freedom in adaptive modulation: a unified view. *IEEE Transactions on Communications*, 49(9):1561–1571, 2001. doi: 10.1109/26.950343. (Cited on page 33.)

Jr. Cimini, L. Analysis and Simulation of a Digital Mobile Channel Using Orthogonal Frequency Division Multiplexing. *IEEE Transactions on Communications*, 33(7):665–675, 1985. (Cited on page 18.)

R. Comroe and Jr. Costello, D. ARQ Schemes for Data Transmission in Mobile Radio Systems. *IEEE Journal on Selected Areas in Communications*, 2(4):472–481, 1984. (Cited on page 30.)

A. Conti, M. Z. Win, and M. Chiani. Slow Adaptive M -QAM With Diversity in Fast Fading and Shadowing. *IEEE Transactions on Communications*, 55(5):895–905, 2007. doi: 10.1109/TCOMM.2007.892453. (Cited on page 104.)

A. Conti, D. Panchenko, S. Sidenko, and V. Tralli. Log-Concavity Property of the Error Probability With Application to Local Bounds for Wireless Communications. *IEEE Transactions on Information Theory*, 55(6):2766–2775, 2009. doi: 10.1109/TIT.2009.2018273. (Cited on page 104.)

Jr. Costello, D.J., J. Hagenauer, H. Imai, and S.B. Wicker. Applications of error-control coding. *Information Theory, IEEE Transactions on*, 44(6):2531–2560, Oct 1998. ISSN 0018-9448. (Cited on pages 5, 22, 28, 29, 33, and 36.)

T. Cover. Broadcast channels. *IEEE Transactions on Information Theory*, 18(1):2–14, 1972. doi: 10.1109/TIT.1972.1054727. (Cited on pages 76 and 77.)

J. Cullen. Binary Synchronous Communications. *IEEE Transactions on Communication Technology*, 17(6):654–658, 1969. doi: 10.1109/TCOM.1969.1090148. (Cited on page 23.)

E. Dahlman, S. Parkvall, and J. Skold. *4G LTE/LTE-Advanced for mobile broadband*. Academic Press, 2011. (Cited on pages 3, 4, 10, and 40.)

G. Dimic and N. D. Sidiropoulos. Multicode multicarrier random access. In *Proc. Conf Signals, Systems and Computers Record of the Thirty-Fourth Asilomar Conf*, volume 2, pages 1230–1234, 2000. (Cited on page 92.)

G. Dimic and N. D. Sidiropoulos. Frequency-hopped network diversity multiple access for semi-ad-hoc wireless networks. In *Proc. IEEE Int. Conf. Acoustics, Speech, and Signal Processing (ICASSP '03)*, volume 4, 2003. doi: 10.1109/ICASSP.2003.1202594. (Cited on page 92.)

G. Dimic, N. D. Sidiropoulos, and L. Tassiulas. Wireless networks with retransmission diversity access mechanisms: stable throughput and delay properties. *IEEE Transactions on Signal Processing*, 51(8):2019–2030, 2003. (Cited on page 92.)

G. Dimic, N. D. Sidiropoulos, and Ruifeng Zhang. Medium access control - physical cross-layer design. *IEEE Signal Processing Magazine*, 21(5):40–50, 2004. doi: 10.1109/MSP.2004.1328087. (Cited on pages 69 and 89.)

R. Dinis and A. Gusmão. Transmission techniques for increased power efficiency in OFDM-based wireless communication systems. In *Proc. IEEE Radio and Wireless Conf. RAWCON 98*, pages 15–18, 1998. doi: 10.1109/RAWCON.1998.709125. (Cited on page 38.)

R. Dinis, A. Gusmão, and N. Esteves. On broadband block transmission over strongly frequency-selective fading channels. In *Proc. Wireless 2003*, pages 261–269, Calgary, Canada, July 2003. (Cited on pages 18, 21, and 45.)

R. Dinis, A. Gusmão, and N. Esteves. Iterative block-DFE techniques for single-carrier-based broadband communications with transmit/receive space diversity. In *Proc. st Int Wireless Communication Systems Symp*, pages 347–351, 2004a. doi: 10.1109/ISWCS.2004.1407267. (Cited on page 21.)

R. Dinis, R. Kalbasi, D. Falconer, and A. H. Banihashemi. Iterative layered space-time receivers for single-carrier transmission over severe time-dispersive channels. *IEEE Communications Letters*, 8(9):579–581, 2004b. doi: 10.1109/LCOMM.2004.835339. (Cited on page 21.)

R. Dinis, P. Carvalho, L. Bernardo, R. Oliveira, M. Serrazina, and P. Pinto. Frequency-Domain Multipacket Detection: A High Throughput Technique for SC-FDE Systems. In *Proc. IEEE Global Telecommunications Conference GLOBECOM '07*, pages 4619–4624, 2007. (Cited on pages 89, 96, and 117.)

R. Dinis, P. Carvalho, and J. Martins. Soft Combining ARQ Techniques for Wireless Systems Employing SC-FDE Schemes. In *Proc. 17th International Conference on Computer Communications and Networks (ICCCN)*, pages 1–5, 2008. (Cited on pages 5, 38, 39, 41, and 45.)

R. Dinis, P. Carvalho, and M. Martins. SC-FDE with soft packet combining ARQ techniques for interference-limited UWB systems. In *Proc. 3rd Int. Conf. Signal Processing and Communication Systems ICSPCS 2009*, pages 1–6, 2009a. doi: 10.1109/ICSPCS.2009.5306412. (Cited on page 38.)

R. Dinis, P. Montezuma, L. Bernardo, R. Oliveira, M. Pereira, and P. Pinto. Frequency-domain multipacket detection: a high throughput technique for SC-FDE systems. *IEEE Transactions on Wireless Communications*, 8(7):3798–3807, 2009b. ISSN 1536-1276. (Cited on pages 46, 89, 96, 101, 102, 103, 104, 105, 114, 123, 140, and 145.)

D. Divsalar, M. K. Simon, and D. Raphaeli. Improved parallel interference cancellation for CDMA. *IEEE Transactions on Communications*, 46(2):258–268, 1998. doi: 10.1109/26.659484. (Cited on page 78.)

A. Drukarev and Jr. Costello, D. Hybrid ARQ error control using sequential decoding. *IEEE Transactions on Information Theory*, 29(4):521–535, 1983. doi: 10.1109/TIT.1983.1056711. (Cited on page 28.)

Yiftach Eisenberg, Keith Conner, Mathew Sherman, Joshua Niedzwiecki, and Reggie Brothers. MUD Enabled Media Access Control for High Capacity, Low-Latency Spread Spectrum Communications. In *Proc. IEEE Military Communications Conf. MILCOM 2007*, pages 1–7, 2007. doi: 10.1109/MILCOM.2007.4455193. (Cited on page 82.)

A. Ephremides and B. Hajek. Information theory and communication networks: an unconsummated union. *IEEE Transactions on Information Theory*, 44(6):2416–2434, 1998. doi: 10.1109/18.720543. (Cited on pages 68 and 79.)

J.J. Fahie. *A history of wireless telegraphy: including some bare-wire proposals for subaqueous telegraphs*. Blackwood, 1901. (Cited on page 2.)

D. Falconer, S.L. Ariyavisitakul, A. Benyamin-Seeyar, and B. Eidson. Frequency domain equalization for single-carrier broadband wireless systems. *IEEE Communications Magazine*, 40(4):58–66, Apr. 2002. ISSN 0163-6804. (Cited on pages xix, 4, 17, 19, 20, 21, 22, and 41.)

G.J. Foschini. Layered space-time architecture for wireless communication in a fading environment when using multi-element antennas. *Bell labs technical journal*, 1(2):41–59, 1996. (Cited on page 11.)

R. Gallager. A perspective on multiaccess channels. *IEEE Transactions on Information Theory*, 31(2):124–142, 1985. doi: 10.1109/TIT.1985.1057022. (Cited on pages 67, 68, 72, and 79.)

R.G. Gallager. Conflict resolution in random access broadcast networks. In *Proc. AFOSR Workshop Communications Theory and Applications*, pages 74–76, Sept. 1978. (Cited on page 72.)

F. Ganhão, M. Pereira, L. Bernardo, R. Dinis, R. Oliveira, and P. Pinto. Performance of Hybrid ARQ for Network Diversity Multiple Access Schemes. In *Proc. 20th Int Computer Communications and Networks (ICCCN) Conf*, pages 1–6, 2011a. doi: 10.1109/ICCCN.2011.6006031. (Cited on pages 42, 145, and 146.)

Francisco Ganhão, Miguel Pereira, Luís Bernardo, Rui Dinis, Rodolfo Oliveira, and Paulo Pinto. Energy per useful packet optimization on a TDMA WSN channel. In *19th IEEE International Conference on Computer Communications and Networks (ICCCN'10)*, Zurich, Switzerland, August 2-5 2010a. doi: 10.1109/ICCCN.2010.5560155. (Cited on pages 42 and 145.)

Francisco Ganhão, Miguel Pereira, Luís Bernardo, Rui Dinis, Nuno Souto, João Silva, Rodolfo Oliveira, and Paulo Pinto. Energy per useful packet optimization on a TDMA HAP channel. In *72nd IEEE Vehicular Technology Conference (VTC'10 Fall)*, Ottawa, Canada, September 6-9 2010b. doi: 10.1109/VETECF.2010.5594540. (Cited on pages 42 and 145.)

Francisco Ganhão, Miguel Pereira, Luis Bernardo, Rui Dinis, Rodolfo Oliveira, and Paulo Pinto. Performance of Hybrid ARQ for NDMA Access Schemes with Uniform Average Power. *Journal of Communications*, 6(9):691–699, Nov. 2011b. doi: 10.4304/jcm.6.9.691-699. Special Issue on Advances in Communications and Networking. (Cited on pages 145 and 146.)

Francisco Ganhão, Miguel Pereira, Luis Bernardo, Rui Dinis, Rodolfo Oliveira, Paulo Pinto, Mário Macedo, and Paulo Pereira. Energy Efficient NDMA Multi-packet Detection with Multiple Power Levels. In Luis Camarinha-Matos, editor, *Technological Innovation for Sustainability*, volume 349 of *IFIP Advances in Information and Communication Technology*, pages 581–588. Springer Boston, 2011c. doi: 10.1007/978-3-642-19170-1_64. (Cited on pages 145 and 146.)

R. Garces and I. J. Garcia-Luna-Aceves. Collision avoidance and resolution multiple access: first-success protocols. In *Proc. 'Towards the Knowledge Millennium' Communications ICC 97 Montreal 1997 IEEE Int. Conf*, volume 2, pages 699–703, 1997. doi: 10.1109/ICC.1997.609940. (Cited on page 86.)

JJ Garcia-Luna-Aceves, H.R. Sadjadpour, and Z. Wang. Challenges: towards truly scalable ad hoc networks. In *Proceedings of the 13th annual ACM international conference on Mobile computing and networking*, pages 207–214. ACM, 2007. (Cited on page 84.)

M. Garetto, T. Salonidis, and E. W. Knightly. Modeling Per-Flow Throughput and Capturing Starvation in CSMA Multi-Hop Wireless Networks. *IEEE/ACM Transactions on Networking*, 16(4):864–877, 2008. doi: 10.1109/TNET.2007.902687. (Cited on page 81.)

R. Gau. Tree/Stack Splitting with Remainder for Distributed Wireless Medium Access

Control with Multipacket Reception. *IEEE Transactions on Wireless Communications*, (99):1–15, 2011. doi: 10.1109/TWC.2011.083111.110284. Early Access. (Cited on page 83.)

Rung-Hung Gau. Performance analysis of slotted Aloha in interference-dominating wireless ad-hoc networks. *IEEE Communications Letters*, 10(5):402–404, 2006. doi: 10.1109/LCOMM.2006.1633337. (Cited on page 82.)

Rung-Hung Gau. Performance analysis of Finite-User slotted aloha in wireless networks with multiple packet reception and random traffic. *IEEE Communications Letters*, 12(2): 140–142, 2008. doi: 10.1109/LCOMM.2008.071654. (Cited on page 83.)

Rung-Hung Gau. Modeling the slotted nonpersistent CSMA protocol for wireless access networks with multiple packet reception. *IEEE Communications Letters*, 13(10):797–799, 2009. doi: 10.1109/LCOMM.2009.091306. (Cited on page 83.)

Rung-Hung Gau and Kuan-Mei Chen. Predictive multicast polling for wireless networks with multipacket reception and queuing. *IEEE Transactions on Mobile Computing*, 5(6): 725–737, 2006. doi: 10.1109/TMC.2006.84. (Cited on page 82.)

Rung-Hung Gau and Kuan-Mei Chen. Probability Models for the Splitting Algorithm in Wireless Access Networks with Multipacket Reception and Finite Nodes. *IEEE Transactions on Mobile Computing*, 7(12):1519–1535, 2008. doi: 10.1109/TMC.2008.83. (Cited on page 83.)

S. Ghez, S. Verdu, and S. C. Schwartz. Stability properties of slotted Aloha with multipacket reception capability. *IEEE Transactions on Automatic Control*, 33(7):640–649, 1988. (Cited on pages 76, 77, 81, and 82.)

S. Ghez, S. Verdu, and S. C. Schwartz. Optimal decentralized control in the random access multipacket channel. *IEEE Transactions on Automatic Control*, 34(11):1153–1163, 1989. doi: 10.1109/9.40760. (Cited on page 81.)

T. R. Giallorenzi and S. G. Wilson. Suboptimum multiuser receivers for convolutionally coded asynchronous DS-CDMA systems. *IEEE Transactions on Communications*, 44(9): 1183–1196, 1996. doi: 10.1109/26.536924. (Cited on page 78.)

M. Gidlund and P. Ahag. Enhanced HARQ scheme based on rearrangement of signal constellations and frequency diversity for OFDM systems. In *Proc. VTC 2004-Spring Vehicular Technology Conf. 2004 IEEE 59th*, volume 1, pages 500–504, 2004. doi: 10.1109/VETECS.2004.1388003. (Cited on pages 47 and 48.)

D. Gore, S. Sandhu, and A. Paulraj. Delay diversity codes for frequency selective channels. In *Proc. IEEE Int. Conf. Communications ICC 2002*, volume 3, pages 1949–1953, 2002. doi: 10.1109/ICC.2002.997188. (Cited on page 47.)

Ming-Fei Guo, Xinbing Wang, and Min-You Wu. On the capacity of k-MPR wireless networks. *IEEE Transactions on Wireless Communications*, 8(7):3878–3886, 2009. doi: 10.1109/TWC.2009.090265. (Cited on page 84.)

A. Gusmão, R. Dinis, and N. Esteves. Adaptive HARQ schemes using punctured RS codes for ATM-compatible broadband wireless communications. In *Vehicular Technology Conference, 1999. VTC 1999 - Fall. IEEE VTS 50th*, volume 5, pages 2530–2535 vol.5, 1999. (Cited on page 38.)

A. Gusmão, R. Dinis, J. Conceicao, and N. Esteves. Comparison of two modulation choices for broadband wireless communications. In *Proc. IEEE 51st VTC 2000-Spring Tokyo Vehicular Technology*, volume 2, pages 1300–1305, 2000. doi: 10.1109/VETECS.2000.851335. (Cited on pages 21 and 41.)

A. Gusmão, R. Dinis, and N. Esteves. On frequency-domain equalization and diversity combining for broadband wireless communications. *IEEE Transactions on Communications*, 51(7):1029–1033, 2003. ISSN 0090-6778. (Cited on pages 18, 22, and 46.)

A. Gusmão, P. Torres, R. Dinis, and N. Esteves. A Turbo FDE Technique for Reduced-CP SC-Based Block Transmission Systems. *IEEE Transactions on Communications*, 55(1): 16–20, 2007. doi: 10.1109/TCOMM.2006.887482. (Cited on pages 44, 45, 46, 96, 101, and 103.)

I. M. I. Habbab, M. Kavehrad, and C.-E. W. Sundberg. ALOHA with capture over slow and fast fading radio channels with coding and diversity. *IEEE Journal on Selected Areas in Communications*, 7(1):79–88, 1989. doi: 10.1109/49.16846. (Cited on pages 69 and 80.)

Z. Hadzi-Velkov and B. Spasenovski. Capture effect in IEEE 802.11 basic service area under influence of Rayleigh fading and near/far effect. In *The 13th IEEE International Symposium on Personal, Indoor and Mobile Radio Communications, 2002.*, volume 1, pages 172–176, 2002a. doi: 10.1109/PIMRC.2002.1046683. (Cited on page 81.)

Z. Hadzi-Velkov and B. Spasenovski. On the capacity of IEEE 802.11 DCF with capture in multipath-faded channels. *International journal of wireless information networks*, 9(3): 191–199, 2002b. (Cited on page 81.)

Z. Hadzi-Velkov and B. Spasenovski. Capture effect with diversity in IEEE 802.11b DCF. In *Proceedings. Eighth IEEE International Symposium on Computers and Communication,* *2003 (ISCC 2003).*, pages 699–704, 2003. doi: 10.1109/ISCC.2003.1214199. (Cited on page 81.)

Joachim Hagenauer. Rate-compatible Punctured Convolutional Codes (RPCP Codes) and Their Applications. *IEEE Transactions on Communications*, 36:389–400, Apr. 1988. (Cited on pages 32, 33, and 34.)

B. Hajek, A. Krishna, and R. O. LaMaire. On the capture probability for a large number of stations. *IEEE Transactions on Communications*, 45(2):254–260, 1997. doi: 10.1109/26.554374. (Cited on pages 75 and 81.)

L. Hanzo. *OFDM and MC-CDMA for broadband multi-user communications, WLANs, and broadcasting.* Wiley-IEEE Press, 2003. (Cited on pages 3, 17, and 18.)

B. A. Harvey and S. B. Wicker. Packet combining systems based on the Viterbi decoder. *IEEE Transactions on Communications*, 42(234):1544–1557, 1994. doi: 10.1109/TCOMM.1994.582838. (Cited on pages 38 and 39.)

Jianwei Huang, R. A. Berry, and M. L. Honig. Wireless scheduling with hybrid ARQ. *IEEE Transactions on Wireless Communications*, 4(6):2801–2810, 2005. doi: 10.1109/TWC.2005.858025. (Cited on page 39.)

Wei Lan Huang, K. Ben Letaief, and Ying Jun Zhang. Joint Channel State Based Random Access and Adaptive Modulation in Wireless LANs with Multi-Packet Reception. *IEEE Transactions on Wireless Communications*, 7(11):4185–4197, 2008. doi: 10.1109/T-WC.2008.070151. (Cited on page 83.)

IEEE. 802.11-2007 IEEE Standard for Information Technology-Telecommunications and Information Exchange Between Systems-Local and Metropolitan Area Networks-Specific Requirements - Part 11: Wireless LAN Medium Access Control (MAC) and Physical Layer (PHY) Specifications, June 2007. URL `http://www.ieee802.org/11/`. (Cited on pages 12 and 74.)

Information Sciences Institute. Ns-2 network simulator (version 2.31). Software Package retrieved from http://www.isi.edu/nsnam/ns/, 2007. (Cited on pages 42, 60, and 118.)

Ray Jennings and Andrew Hartline. Disjunction - Stanford Encyclopedia of Philosophy [cited 29 January of 2012]. Website, Jan 2001. URL `http://plato.stanford.edu/entries/disjunction/`. (Cited on page 34.)

Ming Jiang and L. Hanzo. Multiuser MIMO-OFDM for Next-Generation Wireless Systems. *Proceedings of the IEEE*, 95(7):1430–1469, 2007. doi: 10.1109/JPROC.2007.898869. (Cited on pages 16 and 76.)

S. Kallel. Analysis of a type II hybrid ARQ scheme with code combining. *IEEE Transactions on Communications*, 38(8):1133–1137, Aug 1990. ISSN 0090-6778. (Cited on pages 32, 34, and 52.)

S. Kallel. Sequential decoding with an efficient incremental redundancy ARQ scheme. *IEEE Transactions on Communications*, 40(10):1588–1593, 1992. doi: 10.1109/26.168789. (Cited on page 32.)

S. Kallel. Efficient hybrid ARQ protocols with adaptive forward error correction. *IEEE Transactions on Communications*, 42(234):281–289, 1994. doi: 10.1109/TCOMM.1994.577031. (Cited on page 32.)

S. Kallel. Complementary punctured convolutional (CPC) codes and their applications. *IEEE Transactions on Communications*, 43(6):2005–2009, 1995. doi: 10.1109/26.387438. (Cited on pages 33 and 34.)

S. Kallel and D. Haccoun. Generalized type II hybrid ARQ scheme using punctured convolutional coding. *IEEE Transactions on Communications*, 38(11):1938–1946, 1990. doi: 10.1109/26.61474. (Cited on page 32.)

S. Kallel and C. Leung. An adaptive incremental redundancy selective-repeat ARQ scheme for finite buffer receivers. In *Proc. Tenth Annual Joint Conf. of the IEEE Computer and Communications Societies. Networking in the 90s. IEEE INFOCOM '91*, pages 791–796, 1991. doi: 10.1109/INFCOM.1991.147585. (Cited on page 32.)

V. Kawadia and P. R. Kumar. A cautionary perspective on cross-layer design. *IEEE Wireless Communications Magazine*, 12(1):3–11, 2005. doi: 10.1109/MWC.2005.1404568. (Cited on page 80.)

L. Kleinrock and F. Tobagi. Packet Switching in Radio Channels: Part I–Carrier Sense Multiple-Access Modes and Their Throughput-Delay Characteristics. *IEEE Transactions on Communications*, 23(12):1400–1416, 1975. (Cited on page 73.)

R. Knopp and P. A. Humblet. Information capacity and power control in single-cell multiuser communications. In *Proc. IEEE Int Communications ICC '95 Seattle, 'Gateway to Globalization' Conf*, volume 1, pages 331–335, 1995. doi: 10.1109/ICC.1995.525188. (Cited on page 83.)

M. Kopbayashi, J. Boutros, and G. Caire. Successive interference cancellation with SISO decoding and EM channel estimation. *IEEE Journal on Selected Areas in Communications*, 19(8):1450–1460, 2001. doi: 10.1109/49.942508. (Cited on page 79.)

H. Krishna and S. Morgera. A New Error Control Scheme for Hybrid ARQ Systems. *IEEE Transactions on Communications*, 35(10):981–990, 1987. doi: 10.1109/TCOM.1987.1096688. (Cited on pages 31, 32, and 34.)

C. T. Lau and C. Leung. Capture models for mobile packet radio networks. *IEEE Transactions on Communications*, 40(5):917–925, 1992. doi: 10.1109/26.141457. (Cited on page 80.)

A. Le Duc, C. Le Martret, and P. Ciblat. Delay and jitter closed-form expressions for cross-layer hybrid ARQ schemes. In *IEEE Vehicular Technology Conference (VTC)-Fall*, September 2009a. (Cited on page 40.)

A. Le Duc, C.J. Le Martret, and P. Ciblat. Packet error rate and efficiency closed-form expressions for cross-layer hybrid arq schemes. In *IEEE Workshop on Signal Processing Advances in Wireless Communications (SPAWC'09)*, June 2009b. (Cited on page 40.)

J.S. Lee and L.E. Miller. *CDMA systems engineering handbook*, volume 50. Artech House London, 1998. (Cited on page 16.)

Pierre Lescuyer and Thierry Lucidarme. *Evolved Packet System (EPS) - The LTE and SAE Evolution of 3G UMTS*. John Wiley & Sons, Ltd., 2008. (Cited on pages 1, 5, and 40.)

N. Likhanov, E. Plotnik, Y. Shavitt, M. Sidi, and B. Tsybakov. Random access algorithms with multiple reception capability and N-ary feedback channel. *Problemy Peredachi Informatsii*, 29(1):82–91, 1993. (Cited on page 82.)

S. Lin and D.J. Costello. *Error control coding*. Prentice-Hall Englewood Cliffs, NJ, 1983. (Cited on pages 22, 23, 27, and 29.)

S. Lin and J. S. Ma. A Hybrid ARQ System with Parity Retransmission for Error Correction. Technical Report 7478(32232), IBM, Jan. 11 1979. (Cited on page 29.)

S. Lin and P. S. Yu. SPEC - An Effective Hybrid-ARQ Scheme. Technical Report 7591(32852), IBM, April, 4 1979. (Cited on page 29.)

Shu Lin and P. Yu. A Hybrid ARQ Scheme with Parity Retransmission for Error Control of Satellite Channels. *IEEE Transactions on Communications*, 30(7):1701–1719, Jul. 1982. ISSN 0090-6778. (Cited on pages 29, 30, 31, 32, and 34.)

Shu Lin, D. Costello, and M. Miller. Automatic-repeat-request error-control schemes. *IEEE Communications Magazine*, 22(12):5–17, 1984. doi: 10.1109/MCOM.1984.1091865. (Cited on pages 22, 26, 27, and 30.)

M. Lotfinezhad, Ben Liang, and E. S. Sousa. Adaptive Cluster-Based Data Collection in Sensor Networks with Direct Sink Access. *IEEE Transactions on Mobile Computing*, 7(7): 884–897, 2008. doi: 10.1109/TMC.2007.70769. (Cited on page 83.)

J. Luo and A. Ephremides. On the throughput, capacity, and stability regions of random multiple access. *IEEE Transactions on Information Theory*, 52(6):2593–2607, 2006a. doi: 10.1109/TIT.2006.874518. (Cited on page 82.)

Jie Luo and A. Ephremides. Power levels and packet lengths in random multiple access with multiple-packet reception capability. *IEEE Transactions on Information Theory*, 52(2): 414–420, 2006b. doi: 10.1109/TIT.2005.862076. (Cited on page 84.)

Wei Luo and A. Ephremides. Power levels and packet lengths in random multiple access. *IEEE Transactions on Information Theory*, 48(1):46–58, 2002. doi: 10.1109/18.971738. (Cited on pages 81 and 84.)

Wei Luo, K. Balachandran, S. Nanda, and K.K. Chang. Delay analysis of selective-repeat ARQ with applications to link adaptation in wireless packet data systems. *IEEE transactions on wireless communications*, 4(3):1017–1029, May 2005. ISSN 1536-1276. (Cited on page 40.)

A. B. MacKenzie and S. B. Wicker. Stability of multipacket slotted Aloha with selfish users and perfect information. In *Proc. INFOCOM 2003. Twenty-Second Annual Joint Conf. of the IEEE Computer and Communications. IEEE Societies*, volume 3, pages 1583–1590, 2003. doi: 10.1109/INFCOM.2003.1209181. (Cited on page 81.)

M. Madueno and J. Vidal. Joint physical-MAC Layer design of the broadcast protocol in ad hoc networks. *IEEE Journal on Selected Areas in Communications*, 23(1):65–75, 2005a. (Cited on page 92.)

M. Madueno and J. Vidal. PHY-MAC performance of a MIMO network-assisted multiple access scheme. In *Proc. IEEE 6th Workshop Signal Processing Advances in Wireless Communications*, pages 875–879, 2005b. doi: 10.1109/SPAWC.2005.1506265. (Cited on page 93.)

D. Mandelbaum. An adaptive-feedback coding scheme using incremental redundancy (Corresp.). *IEEE Transactions on Information Theory*, 20(3):388–389, 1974. doi: 10.1109/TIT.1974.1055215. (Cited on pages 28 and 32.)

Guglielmo Marconi. Improvements in Transmitting Electrical Impulses and Signals, and in Apparatus therefor, 1897. URL http://www.earlyradiohistory.us/1901fae.htm. (Cited on page 2.)

JL Massey. *Collision-Resolution Algorithms and Random-Access Communications*. Springer-Verlag, 1981. (Cited on page 72.)

J. Metzner. On Improving Utilization in ALOHA Networks. *IEEE Transactions on Communications*, 24(4):447–448, 1976. doi: 10.1109/TCOM.1976.1093317. (Cited on page 80.)

J. Metzner. Improvements in Block-Retransmission Schemes. *IEEE Transactions on Communications*, 27(2):524–532, 1979. doi: 10.1109/TCOM.1979.1094392. (Cited on page 35.)

J. Metzner and D. Chang. Efficient Selective Repeat ARQ Strategies for Very Noisy and Fluctuating Channels. *IEEE Transactions on Communications*, 33(5):409–416, 1985. doi: 10.1109/TCOM.1985.1096317. (Cited on page 36.)

V.A. Mikhailov and B.S. Tsybakov. Upper bound for the capacity of a random multiple access system. *Problemy Peredachi Informatsii*, 17(1):90–95, 1981. (Cited on pages 72 and 85.)

M. Miller and Shu Lin. The Analysis of Some Selective-Repeat ARQ Schemes with Finite Receiver Buffer. *IEEE Transactions on Communications*, 29(9):1307–1315, 1981. doi: 10.1109/TCOM.1981.1095141. (Cited on page 26.)

M. J. Miller. *ARQ Systems*. PhD thesis, Electrical Engineering Department, University of Hawaii, Honolu, HI, 1982. (Cited on page 30.)

M. Moher. An iterative multiuser decoder for near-capacity communications. *IEEE Transactions on Communications*, 46(7):870–880, 1998. doi: 10.1109/26.701309. (Cited on page 79.)

P. Montezuma and A. Gusmao. A pragmatic coded modulation choice for future broadband wireless communications. In *Proc. VTC 2001 Spring Vehicular Technology Conf. IEEE VTS 53rd*, volume 2, pages 1324–1328, 2001. doi: 10.1109/VETECS.2001.944600. (Cited on page 18.)

S. D. Morgera and V. K. Oduol. Soft-decision decoding applied to the generalized type-II hybrid ARQ scheme. *IEEE Transactions on Communications*, 37(4):393–396, 1989. doi: 10.1109/26.20121. (Cited on pages 31, 32, and 34.)

J. Mosely and P. Humblet. A Class of Efficient Contention Resolution Algorithms for Multiple Access Channels. *IEEE Transactions on Communications*, 33(2):145–151, 1985. doi: 10.1109/TCOM.1985.1096261. (Cited on page 72.)

S. Moshavi. Multi-user detection for DS-CDMA communications. *IEEE Communications Magazine*, 34(10):124–136, 1996. doi: 10.1109/35.544334. (Cited on page 77.)

H.G. Myung and D.J. Goodman. *Single carrier FDMA: a new air interface for long term evolution*, volume 9. Wiley, 2008. (Cited on page 17.)

Young-Han Nam, P. K. Gopala, and H. El-Gamal. Resolving Collisions Via Incremental Redundancy: ARQ Diversity. In *Proc. INFOCOM 2007. 26th IEEE Int. Conf. Computer Communications. IEEE*, pages 285–293, 2007. doi: 10.1109/INFCOM.2007.41. (Cited on page 93.)

K. R. Narayanan and G. L. Stuber. A novel ARQ technique using the turbo coding principle. *IEEE Communications Letters*, 1(2):49–51, 1997. doi: 10.1109/4234.559361. (Cited on pages 33 and 34.)

V. Naware, G. Mergen, and L. Tong. Stability and delay of finite-user slotted ALOHA with multipacket reception. *IEEE Transactions on Information Theory*, 51(7):2636–2656, 2005. doi: 10.1109/TIT.2005.850060. (Cited on pages 76 and 81.)

G. D. Nguyen, A. Ephremides, and J. E. Wieselthier. Comments on "Capture and Retransmission Control in Mobile Radio". *IEEE Journal on Selected Areas in Communications*, 24(12):2340–2341, 2006. doi: 10.1109/JSAC.2006.887338. (Cited on page 81.)

Loutfi Nuaymi. *WiMAX: Technology for Broadband Wireless Access.* John Wiley & Sons, Ltd., 2007. ISBN 978-0-470-02808-7. (Cited on page 5.)

A. Nyandoro, L. Libman, and M. Hassan. Service Differentiation Using the Capture Effect in 802.11 Wireless LANs. *IEEE Transactions on Wireless Communications*, 6(8):2961–2971, 2007. doi: 10.1109/TWC.2007.051030. (Cited on page 81.)

Rodolfo Oliveira, Luis Bernardo, and Paulo Pinto. The Influence of Broadcast Traffic on IEEE 802.11 DCF Networks. *Elsevier Computer Communications*, 32(2):439–452, 2009. (Cited on page 74.)

B. Ozgul and H. Delic. Wireless access with blind collision-multiplicity detection and retransmission diversity for quasi-static channels. *IEEE Transactions on Communications*, 54(5):858–867, 2006. doi: 10.1109/TCOMM.2006.873989. (Cited on page 91.)

P. Patel and J. Holtzman. Analysis of a simple successive interference cancellation scheme in a DS/CDMA system. *IEEE Journal on Selected Areas in Communications*, 12(5):796–807, 1994. doi: 10.1109/49.298053. (Cited on page 78.)

A. J. Paulraj, D. A. Gore, R. U. Nabar, and H. Bolcskei. An overview of MIMO communications - a key to gigabit wireless. *Proceedings of the IEEE*, 92(2):198–218, 2004. doi: 10.1109/JPROC.2003.821915. (Cited on page 76.)

M. Pereira, L. Bernardo, R. Dinis, R. Oliveira, P. Carvalho, and P. Pinto. A MAC Protocol for Half-Duplex Multi-Packet Detection in SC-FDE Systems. In *Proc. IEEE 69th Vehicular Technology Conf. VTC Spring 2009*, pages 1–6, 2009a. doi: 10.1109/VETECS.2009.5073306. (Cited on pages 97 and 145.)

M. Pereira, L. Bernardo, R. Dinis, R. Oliveira, P. Carvalho, and P. Pinto. Delay Optimization on a p-Persistent MAC Protocol for a Multi-Packet Detection in SC-FDE System. In *Proc. IEEE Wireless Communications and Networking Conf. (WCNC)*, pages 1–6, Apr. 2010a. (Cited on pages 97 and 145.)

M. Pereira, L. Bernardo, R. Dinis, R. Oliveira, P. Carvalho, and P. Pinto. Performance of Packet Combining ARQ Error Control in a TDMA SC-FDE System. In *Proc. IEEE Wireless Communications and Networking Conf. (WCNC)*, pages 1–6, Apr. 2010b. (Cited on pages 42 and 145.)

M. Pereira, L. Bernardo, R. Dinis, R. Oliveira, and P. Pinto. Performance Comparison of Diversity Combining ARQ Error Control Schemes with Multi-Packet Detection Schemes. In *Proc. 19th Int Computer Communications and Networks (ICCCN) Conf*, pages 1–7, 2010c. doi: 10.1109/ICCCN.2010.5560156. (Cited on pages 127 and 145.)

Miguel Pereira, Luis Bernardo, Rui Dinis, Rodolfo Oliveira, Paulo Carvalho, and Paulo Pinto. MAC Optimization for Half-Duplex Multi-Packet Detection in SC-FDE Saturated Systems. In *IASTED International Conference on Parallel and Distributed Computing* *and Networks (PDCN)*, Innsbruck, Austria, February 16-18 2009b. (Cited on pages 97 and 145.)

Miguel Pereira, Luis Bernardo, Rui Dinis, Rodolfo Oliveira, Paulo Carvalho, and Paulo Pinto. Delay Analysis for TDMA Schemes with Packet Recombining. In Luis Camarinha-Matos, Pedro Pereira, and Luis Ribeiro, editors, *Emerging Trends in Technological Innovation*, volume 314 of *IFIP Advances in Information and Communication Technology*, pages 263–272. Springer Boston, 2010d. doi: 10.1007/978-3-642-11628-5_28. (Cited on pages 42 and 145.)

Miguel Pereira, Luis Bernardo, Rui Dinis, Rodolfo Oliveira, Paulo Carvalho, and Paulo Pinto. Performance of diversity combining ARQ error control in a TDMA SC-FDE system. *IEEE Transactions on Communications*, 60(3):735–746, March 2012. (Cited on pages 42 and 145.)

E.C. Posner. *Combinatorial structures in planetary reconnaissance*, chapter Error Correcting Codes, pages 15–46. New York: Wiley, 1968. (Cited on page 32.)

M. B. Pursley and S. D. Sandberg. Incremental-redundancy transmission for meteor-burst communications. *IEEE Transactions on Communications*, 39(5):689–702, 1991. doi: 10.1109/26.87160. (Cited on page 32.)

X. Qin and R. Berry. Exploiting multiuser diversity for medium access control in wireless networks. In *Proc. INFOCOM 2003. Twenty-Second Annual Joint Conf. of the IEEE Computer and Communications. IEEE Societies*, volume 2, pages 1084–1094, 2003. doi: 10.1109/INFCOM.2003.1208945. (Cited on page 83.)

T.S. Rappaport. *Wireless communications: principles and practice.* Prentice Hall communications engineering and emerging technologies series. Prentice Hall PTR, 2002. ISBN 9780130422323. URL `http://books.google.pt/books?id=TbgQAQAAMAAJ`. (Cited on page 15.)

T.S. Rappaport *et al.* *Wireless communications: principles and practice*, volume 207. Prentice Hall PTR New Jersey, 1996. (Cited on page 74.)

M. C. Reed, C. B. Schlegel, P. D. Alexander, and J. A. Asenstorfer. Iterative multiuser detection for CDMA with FEC: near-single-user performance. *IEEE Transactions on Communications*, 46(12):1693–1699, 1998. doi: 10.1109/26.737408. (Cited on page 79.)

L.G. Roberts. ALOHA packet system with and without slots and capture. *ACM SIGCOMM Computer Communication Review*, 5(2):28–42, 1975. (Cited on page 68.)

D. N. Rowitch and L. B. Milstein. On the performance of hybrid FEC/ARQ systems using rate compatible punctured turbo (RCPT) codes. *IEEE Transactions on Communications*, 48(6):948–959, 2000. doi: 10.1109/26.848555. (Cited on pages 33 and 34.)

H. R. Sadjadpour, Zheng Wang, and J. J. Garcia-Luna-Aceves. The capacity of wireless ad hoc networks with multi-packet reception. *IEEE Transactions on Communications*, 58(2): 600–610, 2010. doi: 10.1109/TCOMM.2010.02.080175. (Cited on page 5.)

B. Saeki and I. Rubin. An Analysis of a TDMA Channel Using Stop-and-Wait, Block, and Select-and-Repeat ARQ Error Control. *IEEE Transactions on Communications*, 30(5): 1162–1173, 1982. ISSN 0090-6778. (Cited on page 59.)

R. Samano-Robles, M. Ghogho, and D. C. McLernon. P-Persistent Stabilisation for Wireless Network Diversity Multiple Access Protocols. In *Proc. IEEE 7th Workshop Signal Processing Advances in Wireless Communications SPAWC '06*, pages 1–5, 2006. (Cited on page 90.)

R. Samano-Robles, M. Ghogho, and D. C. McLernon. Quality of Service in Wireless Network Diversity Multiple Access Protocols Based on a Virtual Time-Slot Allocation. In *Proc. IEEE Int. Conf. Communications ICC '07*, pages 5843–5848, 2007. doi: 10.1109/ICC.2007.968. (Cited on page 90.)

R. Samano-Robles, M. Ghogho, and D. C. McLernon. Cooperative and sequential user detection for wireless network diversity multiple access protocols. In *Proc. 3rd Int. Symp. Wireless Pervasive Computing ISWPC 2008*, pages 189–193, 2008a. doi: 10.1109/ISWPC.2008.4556194. (Cited on pages 90 and 91.)

R. Samano-Robles, M. Ghogho, and D. C. McLernon. An infinite user model for random access protocols assisted by multipacket reception and retransmission diversity. In *Proc. IEEE 9th Workshop Signal Processing Advances in Wireless Communications SPAWC 2008*, pages 111–115, 2008b. doi: 10.1109/SPAWC.2008.4641580. (Cited on page 90.)

R. Samano-Robles, M. Ghogho, and D. C. McLernon. A multiaccess protocol assisted by retransmission diversity and multipacket reception. In *Proc. IEEE Int. Conf. Acoustics, Speech and Signal Processing ICASSP 2008*, pages 3005–3008, 2008c. doi: 10.1109/ICASSP.2008.4518282. (Cited on page 90.)

R. Samano-Robles, M. Ghogho, and D. C. McLernon. Wireless Networks With Retransmission Diversity and Carrier-Sense Multiple Access. *IEEE Transactions on Signal Processing*, 57(9):3722–3726, 2009. doi: 10.1109/TSP.2009.2021455. (Cited on pages 90 and 91.)

J. Sant and V. Sharma. Performance analysis of a slotted-ALOHA protocol on a capture channel with fading. *Queueing systems*, 34(1):1–35, 2000. (Cited on page 81.)

H. Sari, G. Karam, and I. Jeanclaude. Transmission techniques for digital terrestrial TV broadcasting. *IEEE Communications Magazine*, 33(2):100–109, 1995. doi: 10.1109/35.350382. (Cited on pages 18 and 22.)

M. Sawahashi, K. Higuchi, H. Andoh, and F. Adachi. Experiments on pilot symbol-assisted coherent multistage interference canceller for DS-CDMA mobile radio. *IEEE Journal on Selected Areas in Communications*, 20(2):433–449, 2002. doi: 10.1109/49.983369. (Cited on page 78.)

S. Sesia, G. Caire, and G. Vivier. Incremental redundancy hybrid ARQ schemes based on low-density parity-check codes. *IEEE Transactions on Communications*, 52(8):1311–1321, Aug. 2004. ISSN 0090-6778. (Cited on pages 33 and 34.)

S. Sesia, I. Toufik, and M. Baker. *LTE–The UMTS Long Term Evolution*. John Wiley & Sons Ltd., 2011. (Cited on page 3.)

S. Shakkottai, T. S. Rappaport, and P. C. Karlsson. Cross-layer design for wireless networks. *IEEE Communications Magazine*, 41(10):74–80, 2003. doi: 10.1109/MCOM.2003.1235598. (Cited on page 80.)

M. Sidi and I. Cidon. Splitting protocols in presence of capture. *IEEE Transactions on Information Theory*, 31(2):295–301, 1985. doi: 10.1109/TIT.1985.1057019. (Cited on page 80.)

P. Silva and R. Dinis. Frequency-Domain Multiuser Detection for CP-Assisted DS-CDMA Signals. In *Proc. VTC 2006-Spring Vehicular Technology Conference IEEE 63rd*, volume 5, pages 2103–2108, 2006. (Cited on page 103.)

P. Sindhu. Retransmission Error Control with Memory. *IEEE Transactions on Communications*, 25(5):473–479, 1977. doi: 10.1109/TCOM.1977.1093857. (Cited on pages 5, 28, 31, 33, 34, 36, and 39.)

N. Souto, R. Dinis, J. C. Silva, and P. Carvalho. Iterative multipacket detection for high throughput transmissions in OFDM systems. *IEEE Transactions on Communications*, 58(2):429–432, 2010. (Cited on page 101.)

V. Srivastava and M. Motani. Cross-layer design: a survey and the road ahead. *IEEE Communications Magazine*, 43(12):112–119, 2005. doi: 10.1109/MCOM.2005.1561928. (Cited on pages 2 and 80.)

L. Takács. *Introduction to the Theory of Queues*. Greenwood Press, 1982. (Cited on page 110.)

F. Tobagi and L. Kleinrock. Packet Switching in Radio Channels: Part II–The Hidden Terminal Problem in Carrier Sense Multiple-Access and the Busy-Tone Solution. *IEEE Transactions on Communications*, 23(12):1417–1433, 1975. (Cited on pages 73 and 74.)

Lang Tong, Qing Zhao, and G. Mergen. Multipacket reception in random access wireless networks: from signal processing to optimal medium access control. *IEEE Communications Magazine*, 39(11):108–112, 2001. doi: 10.1109/35.965367. (Cited on page 81.)

Lang Tong, V. Naware, and P. Venkitasubramaniam. Signal processing in random access. *IEEE Signal Processing Magazine*, 21(5):29–39, 2004. (Cited on pages 76, 77, and 81.)

M.K. Tsatsanis, Ruifeng Zhang, and S. Banerjee. Network-assisted diversity for random access wireless networks. *IEEE Transactions on Signal Processing*, 48(3):702 –711, Mar. 2000. ISSN 1053-587X. (Cited on pages 6, 87, 89, 91, 92, 95, 96, and 101.)

D. N. C. Tse and S. V. Hanly. Multiaccess fading channels. I. Polymatroid structure, optimal resource allocation and throughput capacities. *IEEE Transactions on Information Theory*, 44(7):2796–2815, 1998. doi: 10.1109/18.737513. (Cited on page 83.)

David Tse and Pramod Viswanath. *Fundamentals of Wireless Communication*. Cambridge University Press, Jun. 2005. ISBN 0521845270. URL `http://www.amazon.com/exec/obidos/redirect?tag=citeulike07-20&path=ASIN/0521845270`. (Cited on pages 1, 4, 10, 12, 15, 16, and 18.)

B.S. Tsybakov and V.A. Mikhailov. Ergodicity of a slotted ALOHA system. *Problemy Peredachi Informatsii*, 15(4):73–87, 1979. ISSN 0555-2923. (Cited on page 69.)

B.S. Tsybakov and V.A. Mikhailov. Random multiple access of packets: Part and try algorithm. *Probl. Peredachi Inf*, 16:65–79, 1980. (Cited on page 72.)

M. Tuchler and J. Hagenauer. Turbo equalization using frequency domain equalizers. In *Allerton Conference*. Citeseer, Oct. 2000. (Cited on page 96.)

M. Tuchler and J. Hagenauer. Linear time and frequency domain turbo equalization. In *Proc. VTC 2001 Fall Vehicular Technology Conf. IEEE VTS 54th*, volume 4, pages 2773–2777, 2001. doi: 10.1109/VTC.2001.957266. (Cited on page 96.)

C. van der Plas and J.-P. M. G. Linnartz. Stability of mobile slotted ALOHA network with Rayleigh fading, shadowing, and near-far effect. *IEEE Transactions on Vehicular Technology*, 39(4):359–366, 1990. doi: 10.1109/25.61357. (Cited on page 81.)

W. van Etten. Maximum Likelihood Receiver for Multiple Channel Transmission Systems. *IEEE Transactions on Communications*, 24(2):276–283, 1976. doi: 10.1109/TCOM.1976.1093265. (Cited on page 76.)

M. K. Varanasi. Group detection for synchronous Gaussian code-division multiple-access channels. *IEEE Transactions on Information Theory*, 41(4):1083–1096, 1995. doi: 10.1109/18.391251. (Cited on page 78.)

M. K. Varanasi and B. Aazhang. Multistage detection in asynchronous code-division multiple-access communications. *IEEE Transactions on Communications*, 38(4):509–519, 1990. doi: 10.1109/26.52662. (Cited on page 78.)

N. Varnica, E. Soljanin, and P. Whiting. LDPC code ensembles for incremental redundancy hybrid ARQ. In *Information Theory, 2005. ISIT 2005. Proceedings. International Symposium on*, pages 995–999, Sept. 2005. (Cited on page 33.)

S. Vembu and A. J. Viterbi. Two different philosophies in CDMA-a comparison. In *Proc. IEEE 46th Vehicular Technology Conf. 'Mobile Technology for the Human Race'*, volume 2, pages 869–873, 1996. doi: 10.1109/VETEC.1996.501435. (Cited on page 77.)

S. Verdu. Minimum probability of error for asynchronous Gaussian multiple-access channels. *IEEE Transactions on Information Theory*, 32(1):85–96, 1986. doi: 10.1109/TIT.1986.1057121. (Cited on page 77.)

S. Verdu. Demodulation in the presence of multiuser interference: progress and misconceptions. *Intelligent Methods in Signal Processing and Communications*, pages 15–44, 1997. (Cited on page 77.)

Sergio Verdu. *Multiuser Detection.* Cambridge University Press, 1998. (Cited on pages 1, 5, 77, and 78.)

E. Visotsky, Y. Sun, V. Tripathi, ML Honig, and R. Peterson. Reliability-based incremental redundancy with convolutional codes. *IEEE Transactions on Communications*, 53(6):987–997, June 2005. ISSN 0090-6778. (Cited on pages 33 and 34.)

P. Viswanath, D. N. C. Tse, and R. Laroia. Opportunistic beamforming using dumb antennas. *IEEE Transactions on Information Theory*, 48(6):1277–1294, 2002. doi: 10.1109/TIT.2002.1003822. (Cited on page 83.)

A. J. Viterbi. Very low rate convolution codes for maximum theoretical performance of spread-spectrum multiple-access channels. *IEEE Journal on Selected Areas in Communications*, 8(4):641–649, 1990. doi: 10.1109/49.54460. (Cited on pages 77 and 78.)

A. J. Viterbi, J. K. Wolf, E. Zehavi, and R. Padovani. A pragmatic approach to trellis-coded modulation. *IEEE Communications Magazine*, 27(7):11–19, 1989. doi: 10.1109/35.31452. (Cited on page 114.)

A.J. Viterbi *et al. CDMA: Principles of spread spectrum communication*, volume 3. Addison-Wesley Norwell, MA, 1995. (Cited on pages 5 and 16.)

Branka Vucetic and Jinhong Yuan. *Turbo Codes: Principles and Applications.* Kluwer Academic Publishers, 2002. (Cited on page 46.)

N.D. Vvedenskaya and M.S. Pinsker. Non-optimality of the part-and-try algorithm. In *Abstracts Intl. Workshop Conv. Codes, Multiuser Comm*, pages 141–148, 1983. (Cited on page 72.)

Xiaodong Wang and H. V. Poor. Iterative (turbo) soft interference cancellation and decoding for coded CDMA. *IEEE Transactions on Communications*, 47(7):1046–1061, 1999. doi: 10.1109/26.774855. (Cited on page 79.)

Xin Wang and J. J. Garcia-Luna-Aceves. Embracing Interference in Ad Hoc Networks Using Joint Routing and Scheduling with Multiple Packet Reception. In *Proc. INFOCOM 2008. The 27th Conf. Computer Communications. IEEE*, pages 843–851, 2008. doi: 10.1109/INFOCOM.2008.136. (Cited on page 84.)

Xin Wang, Yingqun Yu, and G. B. Giannakis. A Robust High-Throughput Tree Algorithm Using Successive Interference Cancellation. *IEEE Transactions on Communications*, 55(12):2253–2256, 2007. (Cited on page 86.)

Xin Wang, Yingqun Yu, and G. B. Giannakis. Design and Analysis of Cross-Layer Tree Algorithms for Wireless Random Access. *IEEE Transactions on Wireless Communications*, 7(3):909–919, 2008a. doi: 10.1109/TWC.2007.060658. (Cited on page 86.)

Yu-Ming Wang and Shu Lin. A Modified Selective-Repeat Type-II Hybrid ARQ System and Its Performance Analysis. *IEEE Transactions on Communications*, 31(5):593–608, 1983. doi: 10.1109/TCOM.1983.1095873. (Cited on pages 30, 31, 34, and 36.)

Z. Wang, H. Sadjadpour, and J.J. Garcia-Luna-Aceves. The capacity and energy efficiency of wireless ad hoc networks with multi-packet reception. In *Proceedings of the 9th ACM international symposium on Mobile ad hoc networking and computing*, pages 179–188. ACM, 2008b. (Cited on page 84.)

S. B. Wicker. Adaptive rate error control through the use of diversity combining and majority-logic decoding in a hybrid-ARQ protocol. *IEEE Transactions on Communications*, 39(3):380–385, 1991. doi: 10.1109/26.79277. (Cited on pages 38 and 39.)

S. B. Wicker. Reed-Solomon error control coding for Rayleigh fading channels with feedback. *IEEE Transactions on Vehicular Technology*, 41(2):124–133, 1992. doi: 10.1109/25.142771. (Cited on page 28.)

S. B. Wicker and M. J. Bartz. Type-II hybrid-ARQ protocols using punctured MDS codes. *IEEE Transactions on Communications*, 42(234):1431–1440, 1994a. doi: 10.1109/TCOMM.1994.582793. (Cited on pages 32 and 34.)

S. B. Wicker and M. J. Bartz. The design and implementation of type-I and type-II hybrid-ARQ protocols based on first-order Reed-Muller codes. *IEEE Transactions on Communications*, 42(234):979–987, 1994b. doi: 10.1109/TCOMM.1994.580206. (Cited on pages 32 and 34.)

R.W. Wolff. *Stochastic modeling and the theory of queues*, volume 14. Prentice hall Englewood Cliffs, NJ, 1989. (Cited on page 39.)

J.M. Wozencraft and M. Horstein. Digitalised communication over two-way channels. In *Proceedings of Fourth London Symposium Information Theory*, 1960. (Cited on page 27.)

J.M. Wozencraft and M. Horstein. *Coding for two-way channels*. Massachusetts Institute of Technology, Research Laboratory of Electronics, 1961. (Cited on page 27.)

H. Yamamoto and K. Itoh. Viterbi decoding algorithm for convolutional codes with repeat request. *IEEE Transactions on Information Theory*, 26(5):540–547, 1980. doi: 10.1109/TIT.1980.1056245. (Cited on page 28.)

S.C. Yang. *OFDMA System Analysis and Design.* Artech House Publishers, 2010. (Cited on page 17.)

Junliang Yao, Xiaoniu Yang, Jiandong Li, Zhao Li, and Yan Zhang. Blind Collision Resolution Using Cooperative Transmission. In *Proc. IEEE 71st Vehicular Technology Conf. (VTC 2010-Spring)*, pages 1–5, 2010. doi: 10.1109/VETECS.2010.5493749. (Cited on page 92.)

M. R. Yazdani and A. H. Banihashemi. On construction of rate-compatible low-density Parity-check codes. *IEEE Communications Letters*, 8(3):159–161, 2004. doi: 10.1109/LCOMM.2004.825728. (Cited on page 33.)

R. Yim, N. B. Mehta, A. F. Molisch, and Jinyun Zhang. Dual power multiple access with multipacket reception using local CSI. *IEEE Transactions on Wireless Communications*, 8(8):4078–4088, 2009. doi: 10.1109/TWC.2009.080425. (Cited on page 87.)

P. Yu and Shu Lin. An Efficient Selective-Repeat ARQ Scheme for Satellite Channels and Its Throughput Analysis. *IEEE Transactions on Communications*, 29(3):353–363, 1981. doi: 10.1109/TCOM.1981.1094999. (Cited on page 26.)

Yingqun Yu and G. B. Giannakis. High-Throughput Random Access Using Successive Interference Cancellation in a Tree Algorithm. *IEEE Transactions on Information Theory*, 53(12):4628–4639, 2007. (Cited on page 84.)

Yingqun Yu and G.B. Giannakis. SICTA: a 0.693 contention tree algorithm using successive interference cancellation. In *Proc. IEEE 24th Annual Joint Conf. of the IEEE Computer and Communications Societies INFOCOM'05*, volume 3, pages 1908 – 1916, Mar. 2005. (Cited on pages 5 and 84.)

Q. Zhang and S.A. Kassam. Hybrid ARQ with selective combining for fading channels. *IEEE Journal on Selected Areas in Communications*, 17(5):867–880, May 1999. ISSN 0733-8716. (Cited on pages 33 and 34.)

Qian Zhang, T.F. Wong, and J.S. Lehnert. Performance of a type-II hybrid ARQ protocol in slotted DS-SSMA packet radio systems. *IEEE Transactions on Communications*, 47 (2):281–290, Feb 1999. ISSN 0090-6778. (Cited on pages 33 and 34.)

Ruifeng Zhang and M. K. Tsatsanis. A random multiple access scheme with network assisted diversity and ARQ control. In *Proc. IEEE Int. Conf. Communications ICC 2000*, volume 1, pages 154–158, 2000. (Cited on page 90.)

Ruifeng Zhang and M. K. Tsatsanis. Network-assisted diversity multiple access in dispersive channels. *IEEE Transactions on Communications*, 50(4):623–632, 2002. ISSN 0090-6778. (Cited on pages 89, 95, and 96.)

Ruifeng Zhang, N. D. Sidiropoulos, and M. K. Tsatsanis. Collision resolution in packet radio networks using rotational invariance techniques. *IEEE Transactions on Communications*, 50(1):146–155, 2002. (Cited on page 91.)

Ying Jun Zhang. Multi-round contention in wireless LANs with multipacket reception. *IEEE Transactions on Wireless Communications*, 9(4):1503 –1513, april 2010. ISSN 1536-1276. (Cited on page 83.)

Ying Jun Zhang, Peng Xuan Zheng, and Soung Chang Liew. How Does Multiple-Packet Reception Capability Scale the Performance of Wireless Local Area Networks? *IEEE Transactions on Mobile Computing*, 8(7):923–935, 2009. doi: 10.1109/TMC.2008.169. (Cited on page 83.)

Ying Jun Zhang, Soung Chang Liew, and Da Rui Chen. Sustainable Throughput of Wireless LANs with Multipacket Reception Capability under Bounded Delay-Moment Requirements. *IEEE Transactions on Mobile Computing*, 9(9):1226–1241, 2010. doi: 10.1109/TMC.2010.86. (Cited on pages 83 and 84.)

Qing Zhao and L. Tong. A multiqueue service room MAC protocol for wireless networks with multipacket reception. *IEEE/ACM Transactions on Networking*, 11(1):125 – 137, Feb. 2003. ISSN 1063-6692. (Cited on page 82.)

Qing Zhao and L. Tong. A dynamic queue protocol for multiaccess wireless networks with multipacket reception. *IEEE Transactions on Wireless Communications*, 3(6):2221–2231, 2004. doi: 10.1109/TWC.2004.837654. (Cited on page 82.)

Qing Zhao and Lang Tong. Semi-blind collision resolution in random access wireless ad hoc networks. *IEEE Transactions on Signal Processing*, 48(10):2910–2920, 2000. (Cited on page 81.)

M Zorzi. Mobile radio slotted ALOHA with capture, diversity and retransmission control in the presence of shadowing. *Wireless Networks*, 4(5):379–388, 1998. ISSN 1022-0038. (Cited on page 81.)

M. Zorzi and R. R. Rao. Capture and retransmission control in mobile radio. *IEEE Journal on Selected Areas in Communications*, 12(8):1289–1298, 1994. doi: 10.1109/49.329345. (Cited on pages 75 and 81.)

M. Zorzi and R. R. Rao. Reply to “Comments on ‘Capture and Retransmission Control in Mobile Radio”’. *IEEE Journal on Selected Areas in Communications*, 24(12):2341–2342, 2006. doi: 10.1109/JSAC.2006.887340. (Cited on page 81.)