

**DEPARTMENT OF ELECTRONIC AND
ELECTRICAL ENGINEERING
THE UNIVERSITY OF SHEFFIELD**



**Performance Evaluation and
Optimisation of the DVB/DAVIC Cable
Modem Protocol**

Victor Rangel Licea

June 2002

Submitted for the degree of Doctor of Philosophy
Department of Electronic and Electrical Engineering
The University of Sheffield

DECLARATION

No portion of the work referred to in this dissertation has been submitted in support of an application for another degree or qualification of this or any other university or any other institution of learning.

Mr. Victor Rangel

June 2002

ABSTRACT

The growing demand for the next generation of communications services, including high-speed Internet access, work at home, voice and video over IP, and in general broadband services has motivated the extensions of communications services in residential areas. There are several technologies currently entering the home, such as Community Antenna Television (CATV), Digital Subscriber Loop (DSL), Fibre, Wireless and Satellite Technologies. CATV technology has high penetration and excess unused bandwidth in the downstream direction promotes them as suitable candidates for the delivery of high-speed communications services.

CATV technology was designed to broadcast TV signals. Such systems have recently been enhanced by the addition of communication from the station to the headend (upstream direction). Shared access of users in the system imposes very challenging constraints on the CATV Media Access Control (MAC) protocol that must efficiently exploit the limited bandwidth available in the upstream channel.

This dissertation focuses on how best to share the upstream channel between all active users (cable modems, stations or nodes) based on the European Digital Video Broadcasting (DVB) / Digital Audio Visual Council (DAVIC) cable communication protocol (widely known as the DVB/DAVIC protocol). Three different analyses are presented.

The first part of the analysis examines the fundamental performance properties of the DVB/DAVIC protocol, such as maximum system throughput; delay bounds; maximum number of stations (or streams) supported; effects of increasing the station's buffer capacity; effects of reducing the signalling frame period; effects of varying the bandwidth for contention, reservation and fixed-rate access for data transmissions; effects of changing the packet size for data transmissions; analysis of mean access packet delays and the maximum throughput achievable per station. This performance evaluation is given for realistic traffic loads (e.g. Internet traffic, VoIP streams and isochronous streams).

The model used herein is based on the core structure of the Common Simulation Framework (CSF) running in OPNET. In order to validate the results derived from the simulation model, an analytical model was formulated based on an M/G/1 queuing system.

In the second part of the analysis, performance optimisation and comparison of the two contention resolutions algorithms adopted by the DVB/DAVIC is written up. Here, the *backoff windows* for the *exponential backoff algorithm* and *Entry spreading* factors for the *splitting tree algorithm* that provide optimum system performance are presented.

Furthermore, three enhanced contention slot allocators are introduced (*Simple-CSA*, *Forced-CSA* and *Variable-CSA*). These mechanisms dynamically adjust the bandwidth used for contention access, significantly increasing the system performance when different bounds and traffic loads are considered.

The last part of the analysis presents novel improvements and new reservation request techniques, which will enable the DVB/DAVIC MAC protocol to provide the delay requirements that are optimal for the delivery of delay sensitive services and high-speed Internet traffic. Three novel reservation request techniques (namely *Reserved-Request*, *Unsolicited Grant Slot* and *Enhanced-Pure Reservation Request*) are introduced. Such techniques can reduce or avoid the increased risk of collisions that degrade system performance during periods of high congestion.

More specifically, in this part of the analysis we also approach the transmission of delay sensitive services, using Quality of Service (QoS) to help guarantee delivery. A prioritisation mechanism that provides reduced access delays for isochronous streams (during contention periods) is described. Additionally, the effect of including header suppression in the delivery of VoIP traffic is examined. We conclude this dissertation with a performance comparison between the DVB/DAVIC and the DOCSIS protocols, where major characteristics and fundamental performance properties are evaluated.

ACKNOWLEDGEMENTS

My most special thanks go to my wife ADRIANA, for inspiring me and supporting my efforts throughout the period of my studies.

I would like to thank all the people who have influenced me during my postgraduate studies. To all of you, thank you very much. I would like to extend a very special thanks to a number of people that have made this dissertation possible.

I especially wish to thank the '*Universidad Nacional Autónoma de México*' (UNAM) for sponsoring my research at the University of Sheffield.

I am indebted to my academic supervisor Dr. Rob Edwards from the EEE Department, at University of Sheffield for his guidance and support during the last two years of my research. He taught me how to do research without losing the practical prospective.

I would also thank my Mexican tutor Dr. Salvador Landeros Ayala and Mtra. Artemisa Pedroza de Gortari from the *Facultad de Ingenieria*, UNAM for helping me to renew my scholarship several times at the UNAM.

A very special thanks to my first academic supervisor Prof. Colin Smythe from the Computer Science Department, at the University of Sheffield for supporting me and providing the stimulation research environment in which this work has been carried out. I also thank him for accepting to review this dissertation.

I acknowledge my tutor Prof. Graham Rees and my second supervisor Dr. Mohammed Benaissa for providing general comments about my research.

I need to specially thank Dr. Klaus-Dieter Schunke from the Institute for Communications Technology, Braunschweig Technical University, Germany for providing me with technical information for this research and his careful reviews of some of my papers.

I would like also to thank the students from the *Centre For Mobile Communications Research* (C4MCR) group, University of Sheffield, who have made this a pleasant time.

My special gratitude to my parents and brothers for their great help in backing me up psychologically in Mexico throughout the period of this project.

LIST OF PUBLICATIONS

Journal Papers

- (1) V. Rangel, R. Edwards, P. Tzerefos and K. Schunke. "Delivery of Low Bit Rate Isochronous Streams over the Digital Video Broadcasting/Digital Audio-Video Council international cable television protocol", Accepted for publication in May 2002, *IEEE Tran. on Broadcasting*, Ref. No. BTS-00-68.
- (2) V. Rangel, R. Edwards, and K. Schunke, "Contention Resolution Algorithms for CATV Networks Based on the DVB/DAVIC Cable Modem Protocol Specification (ETS EN 200 800)", Accepted for publication, *Cable Telecommunication Engineering (CTE)*, Sep. 2002.
- (3) V. Rangel and R. Edwards, "Performance Evaluation of CATV Networks Based on the European Cable Communications Protocol: DVB/DAVIC", Submitted for publication, *IEEE Tran. on Broadcasting*, Jun. 2002.
- (4) V. Rangel and R. Edwards, "Performance Analysis of QoS over CATV Networks Based on the DVB/DAVIC Protocol", Submitted for publication, *CTE*, Jun. 2002.
- (5) V. Rangel and R. Edwards. "Performance Evaluation of Adaptive Contention Slot Allocators for CATV Networks based on the European Cable Communications Protocol". *Submitted to IEEE GLOBECOM-2002*, Mar. 2002.

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TABLE OF CONTENTS

LIST OF FIGURES	XIII
LIST OF TABLES	XVI
CHAPTER 1: INTRODUCTION	1-1
1.1 STATEMENT OF THE PROBLEM	1-1
1.2 MODELLING OF COMMUNICATIONS NETWORKS AND RESEARCH RELEVANT TO PERFORMANCE EVALUATION OF CATV PROTOCOLS	1-4
1.2.1 PERFORMANCE CONSIDERATIONS OF COMPUTER NETWORKS	1-4
1.2.2 MODELLING TECHNIQUES	1-5
1.2.3 RELATED WORK TO PERFORMANCE EVALUATION OF CATV PROTOCOLS	1-6
1.2.3.1 Analytical modelling of CATV protocols	1-6
1.2.3.2 Analysis of the stability of contention resolution algorithms	1-7
1.2.3.3 Simulation modelling of CATV protocols	1-9
1.3 ORIGINAL CONTRIBUTIONS	1-16
1.4 OVERVIEW OF THIS THESIS	1-18
CHAPTER 2: OVERVIEW OF CURRENT CATV NETWORKS	2-1
2.1 INTRODUCTION	2-1
2.2 EVOLUTION OF CATV NETWORKS AND CABLE MODEM TECHNOLOGY	2-2
2.3 CATV NETWORKS	2-3
2.3.1 CATV SPECTRUM ALLOCATION	2-4
2.3.2 UPGRADE OF CATV NETWORKS TO BI-DIRECTIONAL HFC ARCHITECTURES	2-5
2.4 CABLE MODEM TECHNOLOGY	2-7
2.4.1 CABLE MODEM CONFIGURATIONS	2-7
2.4.2 NEXT GENERATION SERVICES	2-9
2.5 STANDARDS FOR CATV NETWORKS	2-10
2.6 CERTIFIED CABLE MODEMS	2-13
2.7 COMPETING TECHNOLOGIES	2-14
2.7.1 FTTX TECHNOLOGIES	2-15
2.7.2 XDSL TECHNOLOGIES	2-16
2.8 CONCLUSIONS	2-20
CHAPTER 3: THE DVB/DAVIC PROTOCOL	3-1
3.1 INTRODUCTION	3-1
3.2 DVB PROJECT OVERVIEW	3-1
3.3 EUROMODEM	3-2
3.4 DVB/DAVIC REFERENCE MODEL	3-3
3.5 DVB/DAVIC MAC OPERATION	3-4

3.5.1 RESERVATION ACCESS MODE OPERATION	3-6
3.5.2 EXPONENTIAL BACKOFF ALGORITHM OVERVIEW	3-8
3.5.3 SPLITTING TREE ALGORITHM OVERVIEW	3-9
3.5.4 UPSTREAM SLOT STRUCTURE	3-11
3.6 CONCLUSIONS	3-13

CHAPTER 4: SIMULATION AND ANALYTICAL MODELLING OF DVB/DAVIC **4-1**

4.1 INTRODUCTION	4-1
4.2 SIMULATION MODELLING TOOLS	4-1
4.2.1 OPNET MODELLING	4-3
4.2.2 COMMON SIMULATION FRAMEWORK (CSF)	4-4
4.3 DVB/DAVIC SIMULATION MODEL	4-6
4.3.1 DVB/DAVIC NETWORK DESCRIPTION	4-7
4.3.2 NETWORK INTERFACE UNIT DESIGN	4-8
4.3.3 INTERACTIVE NETWORK ADAPTOR DESIGN	4-12
4.4 ANALYTICAL MODELLING	4-14
4.4.1 MEAN ACCESS DELAY FORMULATION FOR A SINGLE NODE SCENARIO	4-19
4.4.2 THROUGHPUT FORMULATION FOR A SINGLE NODE SCENARIO	4-28
4.4.3 BOUNDS ON MAXIMUM SYSTEM THROUGHPUT	4-30
4.5 ACCESS DELAY AND SYSTEM THROUGHPUT VALIDATION	4-31
4.6 CONCLUSIONS	4-35

CHAPTER 5: UPSTREAM CHANNEL CAPACITY AND CHARACTERISATION **5-1**

5.1 INTRODUCTION	5-1
5.2 CASE STUDIES	5-2
5.3 TRAFFIC TYPE CHARACTERISTICS AND SYSTEM PARAMETERS	5-3
5.3.1 INTERNET TRAFFIC	5-3
5.3.2 VOICE OVER IP (VoIP)	5-4
5.3.3 MIXED TRAFFIC (INTERNET +VoIP)	5-6
5.3.4 ISOCHRONOUS STREAMS	5-6
5.3.5 SYSTEM PARAMETERS AND ASSUMPTIONS	5-8
5.4 PERFORMANCE CHARACTERISATION OF THE DVB/DAVIC PROTOCOL	5-9
5.4.1 OFFERED LOAD SCALABILITY	5-9
5.4.2 CAPACITY IN TERMS OF ACTIVE STATIONS	5-12
5.4.3 EFFECTS OF CONTENTION ACCESS FOR DATA TRANSMISSION	5-14
5.4.4 EFFECTS OF RESERVATION REQUEST SIZE	5-18
5.4.5 EFFECTS OF BUFFER CAPACITY	5-23
5.4.6 EFFECTS OF INCREASING THE NUMBER OF SIGNALLING FRAMES	5-27
5.4.7 EFFECTS OF PACKET SIZE FOR SINGLE AND MULTIPLE NODE SCENARIOS	5-29
5.4.7.1 Single node scenario	5-30
5.4.7.2 Multiple node scenario	5-32
5.5 CONCLUSIONS	5-36

CHAPTER 6: OPTIMISATION OF CRA ALGORITHMS USING ADAPTIVE CSA	6-1
6.1 INTRODUCTION	6-1
6.2 CONTENTION SLOT ALLOCATOR	6-2
6.2.1 HOW MANY CONTENTION-SLOTS PER SIGNALLING FRAME?	6-2
6.2.2 SIMPLE-CSA: PERFORMANCE OPTIMISATION OF THE USE OF CONTENTION SLOTS	6-3
6.2.2.1 <i>Exponential backoff algorithm</i> performance using a Simple-CSA	6-3
6.2.2.2 <i>Splitting tree algorithm</i> performance using a Simple-CSA	6-6
6.2.3 ENHANCED - CSAS	6-9
6.2.3.1 Forced-CSA used with the <i>exponential backoff algorithm</i>	6-10
6.2.3.2 Variable-CSA used with the <i>exponential backoff algorithm</i>	6-11
6.2.3.3 Performance comparison of Enhanced-CSAs	6-12
6.3 DYNAMICS OF THE DVB CONTENTION RESOLUTION ALGORITHMS	6-17
6.3.1 EXPONENTIAL BACKOFF ALGORITHM OPTIMISATION	6-17
6.3.2 SPLITTING TREE ALGORITHM OPTIMISATION	6-20
6.3.3 PERFORMANCE COMPARISON BETWEEN THE EXPONENTIAL BACKOFF ALGORITHM AND THE SPLITTING TREE ALGORITHM	6-23
6.4 CONCLUSIONS	6-29

**CHAPTER 7: PERFORMANCE OPTIMISATION FOR THE SUPPORT
OF TCIS AND A PERFORMANCE COMPARISON OF
DVB/DAVIC AND DOCSIS** **7-1**

7.1 INTRODUCTION	7-1
7.2 ENHANCED-RESERVATION-REQUEST MECHANISMS	7-2
7.2.1 RESERVED REQUEST (RR)	7-2
7.2.2 CONTINUOUS RESERVED REQUEST (CRR)	7-3
7.2.3 ENHANCED PURE RESERVATION ACCESS (EPRA)	7-3
7.2.4 PIGGYBACK REQUEST (PG)	7-4
7.2.5 CONTINUOUS PIGGYBACK REQUEST (CPG)	7-5
7.2.6 UNSOLICITED GRANT SLOT (UGS)	7-6
7.2.7 PERFORMANCE COMPARISON OF ENHANCED-RESERVATION-REQUEST MECHANISMS	7-7
7.3 QUALITY OF SERVICE (QoS)	7-13
7.3.1 PRIORITISATION IN QoS	7-14
7.3.2 RESERVATION (FIXED-RATE ACCESS) IN QoS	7-17
7.4 COMPARISON: DVB/DAVIC vs. DOCSIS	7-24
7.4.1 TECHNICAL COMPARISONS	7-25
7.4.1.1 MAC layer comparisons	7-25
7.4.1.2 PHY layer comparison	7-26
7.4.2 PERFORMANCE COMPARISONS	7-27
7.5 CONCLUSIONS	7-36

CHAPTER 8: FINAL CONCLUSIONS	8-1
8.1 INTRODUCTION	8-1
8.2 GENERAL DISCUSSIONS	8-2
8.3 FUTURE WORK	8-18
REFERENCES	R-I
APPENDIX A: GLOSSARY	A-I
APPENDIX B: DVB/DAVIC PROTOCOL STACK AND PACKET FORMATS	A-VI
B.1 PROTOCOL STACK	A-VI
B.2 PACKET FORMATS	A-VII
APPENDIX C: CATV PENETRATION PER COUNTRY	A-XI
APPENDIX D: VERIFICATION TEST FOR THE EXPONENTIAL BACKOFF ALGORITHM	A-XI
APPENDIX E: RESULTS FOR A 6 MBPS UPSTREAM CHANNEL	A-XIV
APPENDIX F: GUIDE TO CD-ROM	A-XVI
F.1 DVB SIMULATION MODEL	XVI
F.2 PUBLICATIONS	XVI
F.4 SPECIFICATIONS	XVI
F.5 THESIS DOCUMENTS	XVII
F.6 TOOLS	XVII

LIST OF FIGURES

FIGURE 2.1 – EXAMPLE BLOCK DIAGRAM OF A TYPICAL CATV NETWORK.	2-3
FIGURE 2.2 – SPECTRUM ALLOCATION ON A CATV NETWORK.	2-4
FIGURE 2.3 – MODERN CATV NETWORK WITH HFC ARCHITECTURE.	2-6
FIGURE 2.4 – CABLE MODEM CONFIGURATIONS.	2-7
FIGURE 3.1 – DVB SYSTEM REFERENCE MODEL.	3-3
FIGURE 3.2 – DVB/DAVIC INITIALISATION AND REGISTRATION PROCESS.	3-5
FIGURE 3.3 – CONTENTION-RESOLUTION-GRANT CYCLE.	3-7
FIGURE 3.4 – BLOCK DIAGRAM OF THE <i>EXPONENTIAL BACKOFF ALGORITHM</i> .	3-8
FIGURE 3.5 – BLOCK DIAGRAM OF THE <i>SPLITTING TREE ALGORITHM</i> .	3-10
FIGURE 3.6 – SLOT STRUCTURE.	3-11
FIGURE 4.1 – STAGE DIAGRAM OF OPNET SIMULATION MODELS.	4-3
FIGURE 4.2 – GENERIC CSF MODEL.	4-5
FIGURE 4.3 – NETWORK TOPOLOGY.	4-7
FIGURE 4.4 – NIU NODE DESCRIPTION.	4-8
FIGURE 4.5 – INTERNET PACKET DISTRIBUTION.	4-8
FIGURE 4.6 – NIU MANAGER PROCESS – FSM.	4-9
FIGURE 4.7 – NIU CHILD PROCESS – FSM.	4-10
FIGURE 4.8 – PROTO-C CODE OF THE STATE: TX. OPPORTUNITY.	4-11
FIGURE 4.9 – INA NODE DESCRIPTION.	4-12
FIGURE 4.10 – INA MANAGER PROCESS – FSM.	4-13
FIGURE 4.11 – INA TX CONTROLLER - FSM.	4-14
FIGURE 4.12 – UPSTREAM CHANNEL MODEL WITH VIRTUAL CONTENTION RESERVATION AND FIXED-RATE CHANNELS.	4-15
FIGURE 4.13 – UPSTREAM TIMING DIAGRAM.	4-19
FIGURE 4.14 – MEAN SERVICE TIME WHEN THE SYSTEM IS IN IDLE STATE.	4-22
FIGURE 4.15 – SCHEDULER LOOK AHEAD DELAY.	4-23
FIGURE 4.16 – VARIANCE OF THE MEAN SERVICE TIME.	4-28
FIGURE 4.17 – TIME OF BUSY CYCLE.	4-28
FIGURE 4.18 – SYSTEM THROUGHPUT: SIMULATION AND THEORETICAL RESULTS FOR DIFFERENT PACKET SIZES.	4-32
FIGURE 4.19 – MEAN ACCESS DELAY: SIMULATION AND THEORETICAL RESULTS FOR DIFFERENT PACKET SIZES.	4-33
FIGURE 5.1 – MEAN ACCESS DELAY, SYSTEM THROUGHPUT AND UTILISATION VS. OFFERED LOAD.	5-10
FIGURE 5.2 – CUMULATIVE PROBABILITY VS. MEAN ACCESS DELAY.	5-11
FIGURE 5.3 – SYSTEM THROUGHPUT AND UTILISATION VS. NO. OF ACTIVE STATIONS, <i>EXPONENTIAL BACKOFF</i> AND <i>SPLITTING TREE ALGORITHM</i> .	5-12
FIGURE 5.4 – MEAN ACCESS DELAY VS. NO. OF ACTIVE STATIONS, <i>EXPONENTIAL BACKOFF</i> AND <i>SPLITTING TREE ALGORITHM</i> .	5-13
FIGURE 5.5 – MEAN ACCESS DELAY VS. NO. OF ACTIVE STATIONS FOR DIFFERENT CONTENTION MESSAGES.	5-15
FIGURE 5.6 – SYSTEM THROUGHPUT VS. NO. OF ACTIVE STATIONS FOR DIFFERENT CONTENTION MESSAGES.	5-16
FIGURE 5.7 – DATA TRANSMITTED IN THE CONTENTION-BASED REGION VS. NO. OF ACTIVE STATIONS FOR DIFFERENT CONTENTION MESSAGES.	5-17
FIGURE 5.8 – BANDWIDTH CONSUMED BY COLLISIONS VS. NO. OF ACTIVE STATIONS FOR DIFFERENT CONTENTION MESSAGES.	5-17
FIGURE 5.9 – MEAN ACCESS DELAY VS. NO. OF ACTIVE STATIONS FOR DIFFERENT REQUEST SIZES.	5-19
FIGURE 5.10 – SYSTEM THROUGHPUT VS. NO. OF ACTIVE STATIONS FOR DIFFERENT REQUEST SIZES.	5-20
FIGURE 5.11 – RESERVATION REQUEST BANDWIDTH CONSUMED VS. NO. OF ACTIVE STATIONS.	5-21
FIGURE 5.12 – MEAN ACCESS DELAY VS. NO. OF ACTIVE STATIONS FOR VOIP AND IP TRAFFIC.	5-22
FIGURE 5.13 – MEAN ACCESS DELAY VS. NO. OF ACTIVE STATIONS FOR INCREASED BUFFER SIZE.	5-24
FIGURE 5.14 – BUFFER CAPACITY VS. OFFERED LOAD FOR INCREASED BUFFER SIZE.	5-25
FIGURE 5.15 – DISCARDED PACKETS VS. OFFERED LOAD FOR INCREASED BUFFER SIZE.	5-26
FIGURE 5.16 – MEAN ACCESS DELAY VS. NO. OF ACTIVE STATIONS FOR DIFFERENT MCI VALUES.	5-28

FIGURE 5.17 – SYSTEM THROUGHPUT VS. NO. OF ACTIVE STATIONS FOR DIFFERENT MCI VALUES.	5-29
FIGURE 5.18 – SYSTEM THROUGHPUT/UTILISATION VS. PACKET SIZE.	5-30
FIGURE 5.19 – SYSTEM THROUGHPUT IN KBPS AND PACKETS/S VS. PACKET SIZE.	5-31
FIGURE 5.20 – SYSTEM THROUGHPUT VS. NO. OF ACTIVE STATIONS FOR INCREASED PACKET SIZE.	5-32
FIGURE 5.21 – MEAN ACCESS DELAY VS. NO. OF ACTIVE STATIONS FOR INCREASED PACKET SIZE.	5-33
FIGURE 5.22 – BANDWIDTH CONSUMED BY COLLISIONS VS. NO. OF ACTIVE STATIONS FOR INCREASED PACKET SIZE.	5-34
FIGURE 6.1 – MEAN ACCESS DELAY VS. NO. OF ACTIVE STATIONS. <i>EXPONENTIAL BACKOFF ALGORITHM</i> WITH A SIMPLE-CSA.	6-4
FIGURE 6.2 – SYSTEM THROUGHPUT VS. NO. OF ACTIVE STATIONS. <i>EXPONENTIAL BACKOFF ALGORITHM</i> WITH A SIMPLE-CSA.	6-4
FIGURE 6.3 – AVERAGE NO. OF CSS PER RESERVATION REQUEST VS. NO. OF ACTIVE STATIONS. <i>EXPONENTIAL BACKOFF ALGORITHM</i> WITH A SIMPLE-CSA.	6-5
FIGURE 6.4 – MEAN ACCESS DELAY VS. NO. OF ACTIVE STATIONS. <i>SPLITTING TREE ALGORITHM</i> WITH A SIMPLE-CSA.	6-6
FIGURE 6.5 – SYSTEM THROUGHPUT VS. NO. OF ACTIVE STATIONS. <i>SPLITTING TREE ALGORITHM</i> WITH A SIMPLE-CSA.	6-7
FIGURE 6.6 – AVERAGE NO. OF CSS PER RESERVATION REQUEST VS. ACTIVE STATIONS. <i>SPLITTING TREE ALGORITHM</i> WITH A SIMPLE-CSA.	6-8
FIGURE 6.7 – SYSTEM THROUGHPUT VS. NO. OF ACTIVE STATIONS. <i>EXPONENTIAL BACKOFF ALGORITHM</i> WITH VARYING CSA APPLIED.	6-14
FIGURE 6.8 – AVERAGE NO. OF CSS PER RESERVATION REQUEST VS. NO. OF ACTIVE STATIONS. <i>EXPONENTIAL BACKOFF ALGORITHM</i> WITH VARYING CSA APPLIED.	6-15
FIGURE 6.9 – MEAN ACCESS DELAY VS. NO. OF ACTIVE STATIONS. <i>EXPONENTIAL BACKOFF ALGORITHM</i> WITH VARYING CSA APPLIED.	6-15
FIGURE 6.10 – CUMULATIVE PROBABILITY VS. NO. OF ACTIVE STATIONS. <i>EXPONENTIAL BACKOFF ALGORITHM</i> WITH VARYING CSA APPLIED. LOAD = 46% PRODUCED BY 34 STATIONS.	6-16
FIGURE 6.11 – CUMULATIVE PROBABILITY VS. NO. OF ACTIVE STATIONS FOR DIFFERENT BACKOFF WINDOWS AND OFFERED LOAD OF 46% AND MIXED TRAFFIC.	6-19
FIGURE 6.12 – MEAN ACCESS DELAY VS. NO. OF ACTIVE STATIONS FOR DIFFERENT WINWOWS AND VOIP TRAFFIC.	6-19
FIGURE 6.13 – SYSTEM THROUGHPUT VS. NO. OF ACTIVE STATIONS FOR DIFFERENT BACKOFF WINWOWS AND VOIP TRAFFIC.	6-20
FIGURE 6.14 – MEAN ACCESS DELAY VS. NO OF ACTIVE STATIONS FOR DIFFERENT ENTRY SPREADING FACTORS AND MIXED TRAFFIC.	6-21
FIGURE 6.15 – MEAN ACCESS DELAY VS. NO. OF ACTIVE STATIONS FOR DIFFERENT ENTRY SPREADING FACTORS AND VOIP TRAFFIC.	6-22
FIGURE 6.16 – SYSTEM THROUGHPUT VS. NO. OF ACTIVE STATIONS FOR DIFFERENT ENTRY SPREADING FACTORS AND VOIP TRAFFIC.	6-23
FIGURE 6.17 – SYSTEM THROUGHPUT. <i>EXPONENTIAL BACKOFF</i> VS. <i>SPLITTING TREE ALGORITHM</i> FOR DIFFERENT TRAFFIC CONFIGURATIONS.	6-24
FIGURE 6.18 – MEAN ACCESS DELAY. <i>EXPONENTIAL BACKOFF</i> VS. <i>SPLITTING TREE ALGORITHM</i> FOR DIFFERENT TRAFFIC CONFIGURATIONS.	6-25
FIGURE 6.19 – COMPONENTS OF DELAY, <i>EXPONENTIAL BACKOFF</i> VS. <i>SPLITTING TREE ALGORITHM</i>	6-26
FIGURE 6.20 – UPSTREAM SLOT USAGE. <i>EXPONENTIAL BACKOFF</i> VS. <i>SPLITTING TREE ALGORITHM</i>	6-27
FIGURE 7.1 – FUNCTIONALITY OF THE <i>RR</i> MECHANISM FOR DIFFERENT PACKET SIZES IN THE TRANSMISSION QUEUE.	7-2
FIGURE 7.2 – FUNCTIONALITY OF THE <i>CRR</i> MECHANISM FOR DIFFERENT PACKET SIZES IN THE TRANSMISSION QUEUE.	7-3
FIGURE 7.3 – FUNCTIONALITY OF THE <i>EPRA</i> MECHANISM FOR DIFFERENT PACKET SIZES IN THE TRANSMISSION QUEUE.	7-4
FIGURE 7.4 – DATA SLOT STRUCTURE WITH A PIGGYBACK REQUEST.	7-4
FIGURE 7.5 – FUNCTIONALITY OF THE <i>PG</i> MECHANISM FOR DIFFERENT PACKET SIZES IN THE TRANSMISSION QUEUE.	7-5
FIGURE 7.6 – FUNCTIONALITY OF THE <i>CPG</i> MECHANISM FOR DIFFERENT PACKET SIZES IN THE TRANSMISSION QUEUE.	7-5

FIGURE 7.7 – MEAN ACCESS DELAY VS. NO. OF ACTIVE STATIONS FOR VOIP AND MIXED TRAFFIC. <i>EXPONENTIAL BACKOFF ALGORITHM WITH ENHANCED-RESERVATION-REQUEST MECHANISMS.</i>	7-8
FIGURE 7.8 – SYSTEM THROUGHPUT VS. NO. OF ACTIVE STATIONS FOR VOIP AND MIXED TRAFFIC. <i>EXPONENTIAL BACKOFF ALGORITHM WITH ENHANCED-RESERVATION-REQUEST MECHANISMS.</i>	7-10
FIGURE 7.9 – BANDWIDTH CONSUMED BY COLLISIONS VS. NO. OF ACTIVE STATIONS FOR VOIP AND MIXED TRAFFIC. <i>EXPONENTIAL BACKOFF ALGORITHM WITH ENHANCED-RESERVATION-REQUEST MECHANISMS.</i>	7-11
FIGURE 7.10 – BANDWIDTH CONSUMED BY COLLISIONS VS. NO. OF ACTIVE STATIONS FOR VOIP AND MIXED TRAFFIC. <i>SPLITTING TREE ALGORITHM WITH ENHANCED-RESERVATION-REQUEST MECHANISMS.</i>	7-12
FIGURE 7.11 – IP HEADER STRUCTURE.	7-14
FIGURE 7.12 – MEAN ACCESS DELAY VS. NO. OF ACTIVE STATIONS FOR VOIP AND INTERNET TRAFFIC. <i>SPLITTING TREE ALGORITHM WITH AND WITHOUT PRIORITISATION.</i>	7-15
FIGURE 7.13 – SYSTEM THROUGHPUT VS. NO. OF ACTIVE STATIONS FOR VOIP AND INTERNET TRAFFIC. <i>SPLITTING TREE ALGORITHM WITH AND WITHOUT PRIORITISATION.</i>	7-16
FIGURE 7.14 – CONNECTION-SETUP FOR A FIXED-RATE ACCESS CONNECTION.	7-17
FIGURE 7.15 – MEAN ACCESS DELAY AND SYSTEM THROUGHPUT VS. NO. OF ACTIVE STATIONS. <i>SPLITTING TREE ALGORITHM AND MIXED TRAFFIC.</i>	7-19
FIGURE 7.16 – MEAN ACCESS DELAY AND SYSTEM THROUGHPUT. <i>SPLITTING TREE ALGORITHM AND VOIP TRAFFIC (G.711 AT 64 KBPS).</i>	7-22
FIGURE 7.17 – MEAN ACCESS DELAY AND SYSTEM THROUGHPUT. <i>SPLITTING TREE ALGORITHM AND VOIP TRAFFIC (G.723.1 AT 5.3 KBPS).</i>	7-23
FIGURE 7.18 – UPSTREAM VOLUMETRIC DATA AND EFFICIENCY.	7-29
FIGURE 7.19 – DVB/DAVIC UPSTREAM PDU STRUCTURES.	7-29
FIGURE 7.20 – DOCSIS UPSTREAM PDU STRUCTURES.	7-30
FIGURE 7.21 – MAXIMUM CHANNEL CAPACITY FOR DVB/DAVIC AND DOCSIS FOR ONE NODE.	7-31
FIGURE 7.22 – MEAN ACCESS DELAY AND MAXIMUM CHANNEL CAPACITY, FOR DVB/DAVIC AND DOCSIS, 12 KBPS STREAMS WITH PACKET SIZE = 1518 BYTES.	7-32
FIGURE 7.23 – MEAN ACCESS DELAY AND SYSTEM THROUGHPUT, FOR DVB/DAVIC AND DOCSIS WITH 32 KBPS ISOCRONOUS STREAMS AND DIFFERENT PACKET SIZES.	7-34
FIGURE 8.1 – FIXED FIELDS FOR RTP/UDP/IP HEADERS.	8-19
FIGURE 8.2 – SIMILARITIES OF CATV WITH FIXED/MOBILE TERRESTRIAL AND SATELLITE WIRELESS ARCHITECTURE.	8-21
FIGURE B.1 – PROTOCOL STACK.	A-VI
FIGURE B.2 – UPSTREAM SLOT AND MINISLOT FRAME STRUCTURE OF THE DVB/DAVIC PROTOCOL.	A-VII
FIGURE B.3 – UPSTREAM PDU STRUCTURE OF THE DVB/DAVIC PROTOCOL.	A-VIII
FIGURE B.4 – DOWNSTREAM FRAME STRUCTURE.	A-IX
FIGURE E. 1 – PERFORMANCE OPTIMISATION AND COMPARISON OF THE <i>EXPONENTIAL BACKOFF ALGORITHM</i> AND THE <i>SPLITTING TREE ALGORITHM</i> , FOR DIFFERENT TRAFFIC PATTERS AND 6 MBPS UP. CHANNEL.	A-XIV
FIGURE E. 2 – PERFORMANCE ANALYSIS OF ISOCRONOUS STREAMS FOR DIFFERENT PACKET SIZES, USING THE <i>EXPONENTIAL BACKOFF ALGORITHM</i> AND A 6 MBPS UPSTREAM CHANNEL.	A-XV

LIST OF TABLES

TABLE 2.1 –	NEXT GENERATION OF CATV SERVICES.	2-9
TABLE 2.2 –	CERTIFIED CABLE MODEM VENDORS (FROM CABLELABS CERTIFIED/QUALIFIED PRODUCT, INFORMATION AS OF JANUARY 17, 2002 [126] AND CABLE DATACOM NEWS [125]).	2-13
TABLE 2.3 –	BROADBAND ACCESS TECHNOLOGIES.	2-19
TABLE 4.1 –	SYMBOLS AND ABBREVIATIONS.	4-18
TABLE 4.2 –	MAXIMUM THEORETICAL THROUGHPUT FOR DIFFERENT PACKET SIZES.	4-32
TABLE 5.1 –	PACKET SIZE DISTRIBUTION AND CHARACTERISATION.	5-3
TABLE 5.2 –	VOIP ENCAPSULATION WITH AND WITHOUT HEADER SUPPRESSION.	5-5
TABLE 5.3 –	EFFECTIVE BANDWIDTH IN ISOCHRONOUS STREAMS (KBPS).	5-7
TABLE 5.4 –	SIMULATION PARAMETERS.	5-8
TABLE 5.5 –	MAXIMUM SYSTEM PERFORMANCE AND BANDWIDTH CHARACTERISATION.	5-35
TABLE 6.1 –	MAXIMUM SYSTEM PERFORMANCE FOR INTERNET AND MIXED TRAFFIC SCENARIOS.	6-9
TABLE 6.2 –	MAXIMUM SYSTEM PERFORMANCE FOR INTERNET TRAFFIC AND DIFFERENT CSAS.	6-12
TABLE 7.1 –	(OPTIMISED) SIMULATION PARAMETERS.	7-7
TABLE 7.2 –	VOIP CODECS: G.711 AND G.723.1	7-21
TABLE 7.3 –	FREQUENCY ALLOCATION AND MODULATION CHARACTERISTICS.	7-26
TABLE 7.4 –	TECHNICAL CHARACTERISTICS.	7-27
TABLE 7.5 –	SIMULATION PARAMETERS.	7-28
TABLE 7.6 –	NUMBER OF STREAMS SUPPORTED FOR DVB/DAVIC AND DOCSIS.	7-35
TABLE 8.1 –	SUMMARY OF MAXIMUM SYSTEM PERFORMANCE AND BANDWIDTH CHARACTERISATION.	8-7
TABLE 8.2 –	SUMMARY OF MAXIMUM SYSTEM PERFORMANCE FOR DIFFERENT CSAS.	8-9
TABLE 8.3 –	PRIORITIES FOR CONNECTIONS.	8-19
TABLE C. 1 –	CABLE TV PENETRATION IN EUROPE, NORTH AMERICA AND ASIA. MAIN SOURCES FROM ECCA ANNUAL MEETING 2001 [7] AND EUROPEAN UNION [30].	A-X

Chapter 1

INTRODUCTION

1.1 Statement of the problem

The growing demand for high-speed Internet access and broadband services has motivated the extensions of communications services in residential areas [99]. There are several technologies currently arriving in homes [124], such as Digital Subscriber Loop (DSL)¹, Fibre, Wireless and Satellite Technologies. However, Community Antenna Television (CATV) is also one of those technologies with high penetration into communities (a share of cable TV households subscribers on homes passed by cable) [99]. Belgium, Luxembourg, the Netherlands, Sweden and Switzerland are countries where CATV networks are the most widespread, [7] and [30]. The aggregate number of cable subscribers in these countries is about 15 million, with a cable penetration between 90 and 100% (the highest percentage in the world), [7]. Germany has more subscribers with 22 million (at 85% penetration).

Historically, cable penetration is weak in Greece, the UK, Portugal, France, Spain and Italy, between 7 and 42% penetration [7]. The reason why the United Kingdom is an exception to this trend is due to satellite penetration. In other parts of the world, the USA and China have a high cable subscription with 67 million (65% penetration) [128] and 90 million (80% penetration) [135], respectively. Table C.1 (Appendix C) shows an approximation of the number of cable subscriptions and penetration (in %) for countries in Europe, North America and Asia.

¹ A complete list of abbreviations is provided in Appendix A.

Because of high existing penetration combined with high channel capacity, CATV networks are at the cutting edge in high-speed communications in homes. For the delivery of digital information, CATV networks use a spectrum of about 550-870 MHz, that supports about 40 downstream channels in Europe and 50 channels in North America. In Europe, the maximum data rate per channel can be up to 50 Mbps, which results in an aggregate bandwidth of 2 Gbps. With this capacity, CATV networks can support all of the popular communication services (e.g. video and audio on demand, high-speed Internet access, etc) that only require a limited bandwidth in the upstream channel for their operation.

The available spectrum for the upstream direction is small, ranging from 5-65 MHz in Europe and 5–42 MHz in North America. The maximum number of upstream channels is reduced to approximately 15 channels (4-MHz channel with a capacity of 6.17 Mbps) in Europe and 11 channels (3.2-MHz channel with a capacity of 10.24 Mbps) in North America. This corresponds to a maximum aggregate bandwidth of ≈ 92 and 112 Mbps, respectively.

Therefore, the available bandwidth in the upstream direction, makes CATV networks limited for the support of an increased number of users and particularly when interactive services demand a considerable network capacity (e.g. videoconferencing, video-telephony, Voice over IP-VoIP and interactive multimedia).

Several technologies help to overcome the problem of limited bandwidth. One is to make more efficient use of the existing bandwidth by sending signals in the guard bands that separate each channel from those next to it. A second method is to compress the signals. The last strategy to increase the efficient use of the upstream bandwidth is to optimise or enhance the functionality of the CATV MAC protocol used. In this thesis, we focus on the last technique, which has been the focus of research for the past few years.

Currently, there are three different MAC protocols for CATV networks. They are the IEEE 802.14, Data Over Cable Service Interface Specification (DOCSIS) and the Digital Video Broadcasting (DVB) / Digital Audio-Visual Council (DAVIC), which have been produced by the IEEE 802.14 Working Group, Multimedia Cable Networks

System (MCNS) Holdings, and the DVB/DAVIC Committee, respectively. Section 2.5 presents a detailed description of the standardisation process and the main characteristics of each standard.

Most of the research found in the literature (described shortly) focuses on performance evaluation, improvements and optimisations for the IEEE 802.14 and DOCSIS MAC protocols. However, research based on the European cable communication standard (DVB/DAVIC) is scarce. This protocol has a strong support from the European Cable Communication Association (ECCA), EuroCableLabs (ECL) and European cable operators or Multiple Service Operators (MSOs) as they are more commonly referred to. The major issue for the DVB/DAVIC Committee is the support of IP telephony over cable. This would allow European MSOs to compete with Telephone Companies (Telcos) in the local loop for telephony services and, in general, for high-speed data communications services over their existing coaxial infrastructures.

The first two versions of the DVB/DAVIC have been finalised (DVB/DAVIC 1.0 [32] and 1.2 [34]). Extensions to the protocol are being added. The major revision came in DVB/DAVIC 1.2, which addressed a *splitting tree algorithm* (used to resolve collisions), minislots (used to send shortened reservation requests) and particularly Quality of Service (QoS) support. More recently, a third version of the DVB/DAVIC protocol was realised in October 2001 [35], which includes new *enhanced reservation request mechanisms* (e.g. *piggyback* and *continuous piggyback*) and header suppression for Voice over IP (VoIP).

Although DOCSIS modems, with European PHY spectrum (referred as EuroDOCSIS [2]) are an alternative for European MSOs, these modems are not preferred for use in Europe [64]. This is because a number of manufacturers are putting investment into the DVB/DAVIC cable modem solutions both for end user and for the interactive network adaptors in the cable headend.

A first step in the provision of data communications services and particularly IP Telephony over the DVB/DAVIC protocol is the implementation of a MAC model, in order to address capacity planning and quantification of the services that can be provided to users.

The research presented in this dissertation focuses on the DVB/DAVIC MAC protocol and the upstream channel, which combines elements of random access, collision detection via the downstream channel, bandwidth allocation for contention access and reservation methods that are not an issue on the downstream direction. This thesis approaches the performance of the DVB/DAVIC protocol and provides several enhancements to the MAC layer that improve the efficiency of this protocol for the support of timing critical interactive services and high-speed data transmissions.

1.2 Modelling of communications networks and research relevant to performance evaluation of CATV protocols

It was not until the evolution of Local Area Networks (with the developments of Ethernet and Token Ring networks) that the performance of communications networks became an interesting and important topic [74]. Performance issues arise from a need to support multimedia applications on computer networks. This demand has led to the introduction of techniques and protocols that are designed to handle both constant and variable bit rate applications simultaneously with high quality [74].

1.2.1 Performance considerations of computer networks

The required measurements for the performance analysis of communications networks include, among others: *Mean access delay*; the average delay that occurs between the time a packet is ready for transmission from a node, until the packet is transmitted and received at the final destination. *Throughput*; the rate of user data being transmitted between nodes. The analysis is commonly done in terms of the total number of bits transferred (kbps, Mbps or as a percentage of the channel capacity) and the letter “S” is normally used as an abbreviation. *Utilisation of the network medium*; the fraction of the total channel capacity being used, (including data, protocol overheads, reservation requests, retransmissions and collisions).

These are the most important performance considerations according to [114]. Results for such parameters are generally plotted either as a function of the offered load, which is the actual load or traffic demand presented in the network, or as a function of the number of (active) stations transmitting traffic to the network. In addition, there are several other properties that can affect performance. These are:

1. *Channel capacity*
2. *Propagation delay*
3. *Number of bits per frame*
4. *Local network protocol*
5. *Offered load*
6. *Number of stations*

The first three factors listed above, can be seen as the parameters that characterise the network and are generally treated as constants. The ***local network protocol*** is the focal point of the design effort and consists mainly of the *medium access* and *physical* layers. The *physical* layer is not likely to be much of a factor. Generally, at this layer data information is transmitted with little delay. The *medium access* layer however does have a significant effect on network performance and is discussed in length in this thesis. The last two factors are concerned with determining the performance as a function of *offered load* or *number of active stations*.

One factor that was not listed above is the error rate of the channel. With error correction techniques used by communications protocols, such as CATV protocols [52], [22] and [34], link errors are not a significant factor in performance [114]. Therefore, the error rate of the channel will not be mentioned again.

1.2.2 Modelling techniques

Initial techniques used to estimate the performance analysis of communications protocols relied on mathematical models using stochastic processes based on probability and queuing theory [120]. The high complexity involved in the solutions for queuing networks led to the formalisation of approximation methods such as Mean Value Analysis (MVA) [89], [90], convolution [91] and linear programming [65] and [66]. Such models have made several assumptions, examples of which are random rates of packet arrival and fixed number of stations. According to [71], the random arrival

assumption is incorrect and the traffic in computer communications networks is bursty with distinguishable patterns that are repeated in specific time intervals.

Therefore, whilst useful, a random arrival assumption may not be totally accurate in the analysis of computer communication networks [115].

However, even if packet arrivals times could be accurately predicted, network performance evaluation is far from solved due to the hybrid access of random access protocols.

Simulation is another technique used for performance analysis, which has been used to analyse communications protocols with great success during the last decade [70]. The use of simulation must be due in part to the large number of networks in existence. Simulation packages designed specially for communications systems, which reduce model development and analysis time considerably, have better accuracy and benefit from a continuous increase in available processing power. In general, simulation models are designed to study more complex scenarios than analytical techniques [4].

Errors can be introduced during the design stages and as a check to ameliorate this, simulation results are compared against results obtained from other methods of performance evaluation. Methods may include mathematical analysis and/or benchmarking.

A drawback of benchmarking in validation is that this requires an existing system that can be measured. Complex network architectures are difficult to benchmark, (e.g. when trying to predict performance of geographically remote networks).

1.2.3 Related work to performance evaluation of CATV protocols

1.2.3.1 Analytical modelling of CATV protocols

Analytical models for the performance evaluation of CATV protocols are few in number. A difficulty is that CATV protocols are too complex to be modelled analytically without making an unreasonable number of assumptions. Therefore, it is uncertain whether such analytical methods are useful. CATV protocols, including DOCSIS, IEEE 802.14 and DVB/DAVIC are random access protocols based on the

TDMA technique, which could not consequently be accurately modelled with existing models for either TDMA or random access mechanisms.

In the area of this dissertation few works with an analytical approach have been attempted. An exception was the work carried out by Sriram² [113]. He proposed a simplified analytical model to estimate the mean packet access delay of the IEEE 802.14 and DOCISIS MAC protocols.

In order to make a reliable estimation of mean access delay, issues such as contention access delay (D_c , the time a station takes to transmit a reservation request successfully) and queuing delays (W_q) should be addressed.

For the approximation of queuing delays, according to Gross and Harris [45], the reliable result is given by the method of Pollaczek-Khintchine (PK). The author of [113] used the PK formula for queuing delays and for contention access delays (D_c)³, in his work he used a simplified estimation given by a high percentile value of the mean plus five times the standard deviation. The same method, presented also in Sriram [112], was used to calculate the end-to-end packet delay (D_{ete})⁴ as the mean access delay plus five times the standard deviation.

We have used a similar approach to estimate D_{ete} of the DVB/DAVIC protocol, and it was found that the maximum number of users supported is considerably overestimated due to the high percentile values used. For this reason, it is uncertain whether the method of [113] can produce reliable results.

1.2.3.2 Analysis of the stability of contention resolution algorithms

In the literature, we have found that studies carried out in the past focused mainly on the stability of contention resolution algorithms (including the *exponential backoff*,

² The related work presented in this section will be cited the first time by the surname of the authors.

³ $D_c = F \cdot (2 + 3 \cdot \rho)$, where F and ρ are the upstream frame duration and channel utilisation.

⁴ $D_{ete} = D_c + W_q + T_x + 5 \cdot \sigma_D$, where T_x is the transmission delay of a packet and σ_D the standard deviation of access delay.

polynomial backoff and the *splitting tree algorithm*) rather than their performance characteristics (e.g. access delay, system throughput and utilisation).

a) Exponential backoff algorithm: In studies involving the *exponential backoff algorithm*, Aldous [3] has proved that the *exponential backoff algorithm* is always unstable in the infinite model for any positive arrival rate (λ)⁵. Studies carried out by Håstad et al. [48] also demonstrated that the binary *exponential backoff algorithm* is unstable for any λ above 0.568 even for a system with a finite numbers of stations. However, a recent study carried out by Goldberg et al. [39] suggested that the *stability of the exponential backoff algorithm* can possibly be as long as $\lambda < 1/e$ in the infinite model and with a finite expected delay (the average waiting time of messages in the system).

b) Polynomial backoff algorithm: Raghavan and Upfal [83] and the authors of [48] proved that this algorithm is always stable for any $\lambda < 1$ and for a finite number of stations. However for the infinite model, Kelly and MacPhee [62] proved that the *polynomial backoff algorithm* is always unstable.

b) Splitting tree algorithm: Most of the work related to the *splitting tree algorithm* focuses on stability and the time required to resolve collisions, rather than estimating the total mean access delay. As an example, Greenberg et al. [44] used a hybrid algorithm based on a base 2 estimation algorithm (access probability = 2^{-i}) and the *splitting tree algorithm* of Capetanakis [14] and Tsybakov [119] to find an estimation of the multiplicity of conflicts ‘ n ’ and prove that the time to resolve conflicts of multiplicity is stable, achieving a maximum throughput = 0.4015. Cidon and Sidi [17] extended the idea of Greenberg and obtained an improved algorithm that reduces the time to resolve conflicts maintaining stability for all λ up to 0.487.

Most of these approaches work under the assumption that the number of nodes $\rightarrow \infty$ in order to support the independence of the transmission probability assumption. However, even if the technique for an infinite number of nodes was to be used, it has been proved

⁵ λ is defined as the arrival rate per unit of time. In this section the unit of time is one slot.

that such solutions could not be applicable for the approximation of networks with a finite number of nodes. Therefore, infinite model results have limited relevance to finite systems like the ones studied in this text.

Particularly, the CATV protocols require approximations of both *exponential backoff* and *splitting tree algorithms*, which cannot be accurately modelled using these existing analytical models due to the variable bandwidth assigned to the contention and reservation access regions from cycle to cycle. Therefore, the high degree of complexity and the need for accurate results indicate the use of simulation techniques for the performance evaluation and optimisation of CATV protocols.

1.2.3.3 Simulation modelling of CATV protocols

A number of papers reviewed in the literature have focused on simulation techniques to analyse specific characteristics and performance issues of the IEEE 802.14 and DOCSIS protocols based on HFC networks in general, and as such might not be directly applicable to the DVB/DAVIC protocol. The issues addressed include Contention Resolution Algorithms (CRA), Contention Slot Allocators (CSA), scheduling and prioritised mechanisms, performance comparison of CATV protocols, registration after power up and performance evaluation and comparison of CATV protocols. Below is a review of the research performed in each field. Special attention is paid to issues regarding CRAs, CSAs and prioritised mechanisms.

a) Contention Resolution Algorithms (CRA): the research carried out in this field deals with how best to optimise the time to resolve collisions between reservation request transmissions. Citta et al. [18], Sala et al. [95], [97] and Golmie et al. [40], [41] analysed and compared the performance of two CRAs (*p-persistence* and *splitting tree algorithm*), for the IEEE 802.14 protocol.

The *p-persistence algorithm* resolves collisions by restricting the contending users to transmit in the next contention minislot with probability p . Thus, when a collision occurs only a portion of the users involved in the collision transmits in the next contention minislot and eventually the collision is resolved.

In general, the exact value of contending users is not known. Sala used the pseudo-Bayesian estimator proposed by Rivest [92] to estimate the number of contending users in each contention minislots.

The *splitting tree algorithm* is more complex than that of the *p-persistence*, due to the feedback and allocation information transmitted in the downstream channel in every signalling frame. With this algorithm, all nodes involved in a collision split into a number of subsets as introduced in [14]. In series, the first subset transmits, followed by the second and then the remaining subsets. The chances of future collisions are reduced by forcing stations that collided in the same slot to retransmit requests in different slots in the future. A detailed description of this algorithm is presented in Section 3.4.3.

The authors of [41] studied the performance of the *splitting tree algorithm* using four different strategies. These were called *Free-access*, *Blocked-access*, *R-access* and *T_{bound}-access*. In the *Free-access* strategy the first transmission of requests are allowed to take place on the same minislots used to retransmit collided requests. New arrival requests are mixed with old or retransmitted requests. In *Blocked access* new requests are not allowed in the minislots used to resolve current collisions. This is illustrated by a contention interval that is split into two regions. One region is reserved for ongoing collision resolution and the other, denoted as the newcomer minislots region, is open for newcomer requests.

In *R-Access*, which is also referred to as *Adaptive p-persistence* in [40], only a portion of newcomer stations are allowed to transmit in the newcomer region. In *T_{bound}-access*, the headend sets a value (termed as *T_{bound}*) every cycle and a station is allowed to transmit in the newcomer region only if the arrival time of the new message is less than *T_{bound}*. Golmie found that by restricting the contention access of newcomer arrivals (with *R-Access* and *T_{bound}-access*), the performance of the IEEE 802.14 could be improved.

It was found in [97] that performance difference between the studied algorithms is very small, with a slight increase in performance in favour to the *p-persistence* algorithm over the *splitting tree algorithm*. This difference was attributed mainly to

the upstream slot structure. In [41], the structure was fixed (the minislots for contention access were allocated at the beginning of each signalling frame), while the slot structure in [97] was variable (the minislots for contention access were spread over the signalling frame).

b) Contention Slot Allocators (CSA): here the research focuses on the optimisation of the number of contention minislots (or request minislots) that should be allocated for contention access. For the IEEE 802.14 protocol, Sala et al. [96], [98], Lin et al. [73] and the authors of [41] proposed different mechanisms to dynamically allocate the number of request minislots for contention access according to the traffic load. For the DOCSIS protocol, Cho et al. [15] optimised the number of request minislots by changing the size of the signalling frame (referred to as MAC Management Access - MAP in DOCSIS), and the number of contention minislots per MAP.

In [96] and [98] two mechanisms were studied, *Simple* and *Forced Minislots CSA*, which are applied only to the *p-persistence* CRA. In the *simple-CSA* mechanism the un-scheduled slots in every signalling frame are allocated as request minislots. In the *Forced-CSA*, the author suggests that in order to obtain optimum system performance, the number of contention minislots per reservation request is given by the maximum throughput of the Slotted Aloha system [116].

In [41] two mechanisms were also introduced for contention allocation, *Fixed* and *Variable* CSA. In the *Fixed-CSA* technique the number of request minislots in each signalling frame remains the same, whilst in the *Variable-CSA* technique the number of contention minislots is dynamically adjusted according to the offered load, maximum request size, number of minislots in a data slot and the frame structure. The results obtained in [41] revealed that the *Variable-CSA* performs better than the *Fixed-CSA* in terms of lower access delays at higher loads and higher throughput.

The authors of [73] have studied several techniques based on the IEEE 802.14 and the DOCSIS protocol. For the IEEE 802.14 protocol, four different techniques, namely *Fix3-Var*, *Fix3-Fix3*, *Load-Fix3* and *Load-Var* were analysed. The first term of each technique (e.g. *Fix3* of *Fix3-Var*) refers to the number of contention

minislots to be allocated for new arrivals and the second term (e.g. *Var*) refers to the number of contention minislots to be allocated to resolve collisions.

In the *Fix3* technique the headend allocates 3 contention minislots in each signalling frame, in the *Var* technique the headend allocates 3 contention minislots for every collision registered in the previous signalling frame.

In the *Load* technique, the headend allocated N contention minislots, where N depends on the current traffic load. From these techniques the author concluded that the dynamics of the *Load-Var* technique allowed the system to perform better than the other allocation strategies.

For the DOCSIS protocol, the researchers of [73] also proposed different strategies to allocate the optimum number of contention slots in each signalling frame (namely S , E , Dbl , Exp , $MeanSE$ and SE), where S and E represent the minimum and maximum number of contention minislots to be allocated, respectively. With Dbl the headend allocates ' $\max(2 \cdot No_Collisions, E)$ ' contention minislots in the following signalling frame. With Exp the number of contention minislots is given by ' $\max(2^{No_Collisions}, E)$ ' and with SE the headend allocates S contention minislots when there are no collisions, and otherwise E . From all these strategies the best performance is obtained when the E , $MeanSE$ and SE techniques are used. In the same reference, [73], the authors also approached the number of contention minislots to be allocated in each signalling frame by changing the backoff window of the *exponential backoff algorithm*.

The researchers of [15] used a different approach to obtain the optimum number of contention minislots. The authors studied the performance of the DOCSIS protocol when the MAP size was changed from 1 to 10 ms and the number of contention minislots was ranged from 2 to 16 minislots per MAP. Optimum results were found when the MAP size was set to 2 ms (which contains 40 minislots of 16 bytes each), and the number of contention minislots per MAP was set to 6.

c) *Prioritisation and scheduling mechanisms:* this field of research focuses on the classification of data packets (by assigning different levels of priority) to gain access to the upstream channel at the station premises. At the headend, reservation requests

are granted according to the scheduling mechanism adopted. Classification of data packets (or requests) is needed to support multimedia applications and QoS services, which require relatively low access delays.

Limb and Sala [72] studied the performance of a Centralised Priority Reservation (CPR) mechanism for the transmission of multimedia traffic for the IEEE 802.14 protocol. In this mechanism a station maintains separate queues for each priority of data traffic, and the priority is indicated in the type field of the request message. Requests with higher priority have precedence over requests with lower priority. The headend maintains separate queues for each priority and schedules in the first place higher priority requests before serving a queue of a lower priority.

Sdralia et al. [102] also studied the performance of a prioritised-scheduling algorithm (at the headend), using up to 8 levels of priorities for the DOCSIS protocol. The main implication of this prioritised-scheduling mechanism is that high-priority traffic will always take precedence over low-priority traffic. More priority can be given to low-priority traffic by scheduling it to have more than one slot into the future. This technique is referred to as Scheduling Advance (SA) in [72] and Ivanovich and Zukerman [60].

Different techniques can also be used. For example, Nichols and Laubach [78] proposed a scheduler algorithm with three levels of priorities, for the transmission of Constant Bit Rate (CBR), Committed Information Rate (CIR) and Best Effort traffic (BE). Each of these traffic types has preference over the others with a fixed frequency (e.g. 20% of channel capacity for CBR, 80% for CIR and 5% for BE traffic).

Sala et al. [94] studied a scheduler algorithm with a Self Clock Fair Queuing (SCFQ) discipline, in which every reservation request is attached with a finish service time and placed in a queue with increasing order of service time. This finish time is defined as the deadline time to serve (or grant) the request and is computed according to the traffic type characteristics (e.g. station's data rate, packet size, arrival time) and the round trip delay.

The authors of [73] analysed a scheduler algorithm for the IEEE 802.14 protocol, using three different strategies. In the first strategy (SJF) the shortest jobs (with minimum bandwidth request) are served first. In the second strategy (LJF) the largest jobs take precedence over short jobs, and in the last one (MSJF), the shortest jobs are served first but the data minislots are allocated into several bursts. From these three strategies it was found that by adopting SJF lower end-to-end access delays are produced, but a larger delay for contention access is obtained.

Research in this area focuses on the implementation or analysis of scheduling algorithms at the headend. Corner et al. [20] argue that the headend scheduling algorithm is not sufficient and that a system at the station's premises integrated also with a scheduler algorithm is needed to efficiently support multimedia traffic and QoS services. The authors proposed a new priority scheme for the IEEE 802.14 protocol, which separates and resolves collisions in a priority order.

In this new system, a priority usage for each contention minislot is allocated by the headend. Upon packet arrival, a station initially transmits in minislots exactly matching their priority level, so the headend knows that all stations participating in a particular collision are of the same priority level. The headend allocates three slots in the next signalling frame for each collided request. These slots are reserved for requests of the same priority as with the first collision.

Golmie et al. [43] used the same idea of Corner to evaluate the performance of a priority scheme for the DOCSIS protocol. In this prioritised scheme, Golmie proposed a slight modification to the *exponential backoff algorithm* (used by the DOCSIS protocol), where the backoff value was set equal to the number of contention slots reserved for high priority stations.

Both mechanisms ensure that in case of collision, high priority stations retransmit requests in a timely manner.

d) *Registration after power-up disruption*; In this area of research several strategies have been proposed to optimise the recovery time after a service disruption (caused by power failures or maintenance services). In Sdralia et al. [103] and [104] an evaluation of the recovery of the DOCSIS 1.0 protocol after a large-scale power

failure is presented, using the default ranging algorithm of the DOCSIS protocol. This algorithm adjusts the transmission power every time a station fails to range with the headend. The same authors in [106] and [105] proposed a persistent ranging algorithm, where stations try to retransmit using the same transmission power level for a number of attempts. Only after this limit is reached are the power parameters modified and the process started again. Sdralia showed that the persistence ranging algorithm performs better than the default ranging algorithm of the DOCSIS 1.0 protocol, where a reduction of $\approx 38\%$ was achieved in the recovery time.

For the IEEE 802.14 protocol, Sala et al. [93] studied the performance of three CRAs (*p-persistence*, *binary exponential backoff* and *splitting tree algorithm*) in case of registration on power-up. The authors found that even in the worst case, the *p-persistence* algorithm performs better than the *splitting tree algorithm* up to ≈ 2000 stations.

e) Performance comparisons of CATV protocols; An investigation of protocol issues and performance comparisons between IEEE 802.14 vs. DOCSIS can be found in Golmie et al. [42] and Smythe et al. [108]. Performance comparisons between DOCSIS vs. DVB/DAVIC are reviewed in Smythe et al. [109], [110], [111] and for comparisons between “DVB/DAVIC vs. IEEE 802.14 vs. DOCSIS can be found in Rangel et al. [84]. Here, the comparisons focus mainly on the capacity of the upstream channel to deliver several traffic types and packet sizes, using different parameters of configuration.

From the literature overview, there is a lack of studies that address the scalability, performance and optimisation of the DVB/DAVIC protocol. The first to report a brief analysis of the performance of the DVB/DAVIC protocol was Schunke [100]. This analysis focused on the performance of the three access modes of the DVB/DAVIC protocol (contention, reservation and fixed-rate) for the support of CBR (64-kbps with 48-byte packets) and bursty traffic (64 kbps with 1.6-Kbytes), using a network size of 50 stations and simulation time of 5 seconds. Simulation results revealed that system performance directly depends of the access mode that is chosen for the current traffic load. The contention access mode is only suitable for a light traffic load, whereas fixed-

rate access mode handles latency-sensitive traffic with a constant data rate. The reserved access mode only handles the transmission of heavy burst data rate traffic.

The performance comparison between DOCSIS vs. DVB/DAVIC introduced in [109], [110] and [111] for the DVB/DAVIC part) is based on the analysis presented in [100]. In general, the study carried out by Schunke, whilst valuable, is limited and only presents a brief analysis of the DVB/DAVIC protocol (in terms of mean access delays), using a fixed network size of 50 stations and two different traffic types (CBR and bursty traffic). In addition, with such a short simulation period (5 seconds for each simulation run) the steady state may not be obtained for some configurations, which could lead to inaccurate results.

Hence, there is an increased need to estimate the performance of the DVB/DAVIC protocol, under realistic traffic scenarios and configurations that allow reliable analysis of the main characteristics of this protocol.

In summary, the lack of studies of the DVB/DAVIC protocol, combined with the increased demand for high-speed communications services, motivate the analysis of this protocol in such detail.

1.3 Original contributions

This research provides a rigorous evaluation of the DVB/DAVIC protocol and presents fundamental performance characteristics. These characteristics address: the maximum upstream channel capacity; mean access delay; system throughput and utilisation bounds; maximum number of stations (or streams) supported; effects of increasing the station's buffer capacity; effects of reducing the signalling frame period; effects of varying the bandwidth for contention, reservation and fixed-rate access for data transmissions; effects of changing the packet size for data transmissions; analysis of mean access packet delays; and the maximum throughput achievable per station. All these performance characteristics are given for realistic traffic loads (including Voice over IP, Internet traffic, isochronous streams and mixed traffic scenarios), as well as different protocol configurations.

This dissertation also focuses on a rigorous analysis of the two adopted contention resolutions algorithms of the DVB/DAVIC protocol (*exponential backoff algorithm* and *splitting tree algorithm*). Here, we study in detail performance characteristics, optimisation and implementation issues for each algorithm. Furthermore, three enhanced contention slot allocators are introduced (*Simple-CSA*, *Forced-CSA* and *Variable-CSA*), which dynamically adjust the bandwidth used for contention access, significantly increasing the system performance when different bounds and traffic loads are considered.

The major contributions of this research are the introduction of novel improvements and new techniques (presented below), which will enable the DVB/DAVIC MAC protocol to provide the delay requirements optimally for the delivery of delay sensitive services and high-speed Internet traffic.

In this dissertation, we introduce three novel reservation request techniques, namely *Reserved-Request*, *Unsolicited Grant Slot* and *Enhanced-Pure Reservation Request* that reduce or avoid the increased risk of collisions during large periods of congestion. In addition, we compare the performance of these three mechanisms with the enhanced reservation request strategies introduced in Schunke [101] (*Continuous Reservation Request*, *Piggyback Request* and *Continuous Piggyback Request*).

A second approach for the delivery of delay sensitive services involves Quality of Service (QoS) with a guaranteed delivery. In this research, a prioritised mechanism that provides reduced access delays for isochronous streams (during contention access) is described. Additionally, the effect of considering header suppression for the delivery of VoIP traffic is approached.

We also present a performance comparison between the DVB/DAVIC and the DOCSIS protocol, where the major characteristics and the fundamental performance properties of each protocol are evaluated. For each analysis we provide saturation points, maximum number of streams supported and reasons for inefficiencies.

1.4 Overview of this thesis

Chapter 2 presents relevant theory that includes the evolution and description of CATV networks. It describes traditional and modern cabling infrastructures, their differences and similarities, and presents the changes that have to be made to upgrade CATV networks, for the support of bi-directional digital communications. In addition, background material included is the introduction of *cable modem technology* that shows different cable modem configurations, and identifies the requirements for the next generation of communications services. This chapter also outlines the main characteristics of protocols proposed from different standardisation bodies, such as DOCSIS, IEEE 802.14 and DVB/DAVIC, and concludes with an overview of alternative technologies for high-speed digital access.

Chapter 3 describes the main characteristics as well as the architecture, MAC operation, and a description of the *exponential backoff algorithm* and the *splitting tree algorithm* adopted by the DVB/DAVIC protocol specification.

Chapter 4 presents the structure of OPNET simulation models and describes the Common Simulation Framework (CSF) model, which is the foundation for an advanced model. A comprehensive description of the simulation model for the DVB/DAVIC MAC protocol is presented and the theoretical model to be used for analysing the DVB/DAVIC performance and validating the simulation model results is formulated.

Chapter 5 looks in detail at the performance of the DVB/DAVIC protocol using the simulation model presented in chapter four. A number of protocol configurations are evaluated for different traffic situations. The main issues addressed in this chapter are the network capacity, scalability in terms global offered load and number of stations, buffer capacity, signalling frame period and packet size variation, as well as the effects of changing the bandwidth for contention and reservation access. Results focus on system throughput, mean packet access delays and system utilisation when the load generated by all stations is increased up to maximum network capacity.

Chapter 6 covers a detailed analysis of the dynamics of the two CRAs adopted by the DVB/DAVIC protocol. The analysis focuses on performance optimisations when different *backoff bounds* for the *exponential backoff algorithm* and different values for the *Entry-Spreading* factor of the *splitting tree algorithm* are considered. In addition, a performance comparison between these two algorithms is presented. This chapter also introduces three enhanced CSAs (*Simple-CSA*, *Forced-CSA* and *Variable-CSA*) that will further optimise the performance of the DVB/DAVIC for different traffic configurations (e.g. Internet traffic, VoIP traffic and mixed traffic).

In **Chapter 7**, three novel reservation request mechanisms are proposed (*Reserved-Request*, *Unsolicited Grant Slot* and *Enhanced-Pure Reservation Request*) for the support of timing critical interactive services, and their performance is compared with the default reservation request mechanism and the three techniques presented in [101]. This chapter also describes a prioritisation mechanism that will further reduce the delay caused by contention access. In addition, the effects of using fixed-rate access and header suppression for delay sensitive applications are studied, especially for the delivery of VoIP streams. Finally a performance comparison between DVB/DAVIC and DOCSIS protocol is addressed.

Chapter 8 presents a comprehensive analysis of the key findings, and shows how the results could be used by either vendors or operators in order to improve network performance. Finally, issues of further research are outlined.

Chapter 2

OVERVIEW OF CURRENT CATV NETWORKS

2.1 Introduction

The main function of traditional Community Antenna Television (CATV) networks – distribution of TV and radio programs- has rapidly been extended in the last few years. Modern cable systems are being designed to deliver a variety of communications services with support for current and future applications

An increased growth of the Internet has created a demand for broadband access. Telephone companies (Telcos) have until recently provided data communications channels. They were slow and relatively expensive.

CATV networks are considered as an alternative bearer of new interactive services, because they have large unused bandwidth. Advances in CATV network technology and in particular the introduction of cable modem technology over Hybrid Fibre Coax (HFC), allows for diverse range of data communications services, such as interactive television, broadband Internet and IP telephony.

This chapter presents an overview of CATV networks. The physical characteristics of traditional cable networks are shown and the structure of modern CATV networks highlighted. It also presents some of the most important protocol specifications. Special attention is paid to cable modem technology. Finally, an outline of high-speed DSL and fibre technologies is presented.

2.2 Evolution of CATV networks and cable modem technology

Community Antenna Television (CATV) began in 1948 in Astoria, Oregon. It was created as a way to improve television reception for people who lived in remote or hilly areas, where good television signal reception was difficult. Since that time, cable television has become a common source of video entertainment for more than 500 million viewers around the world [125]. Early interest in cable modem technology emerged in the 1980s and was focused around the IEEE 802.4 token bus over CATV networks.

At the same time, research institutions and universities began using the CATV infrastructure for two-way campus data networking. These institutional networks were called I-Nets and were mainly used to connect institutional Local Area Networks (LAN). Unfortunately, such cable modem products were relatively costly to manufacture and maintain and the resulting broadband data networks were expensive to operate [63]. Technical obstacles combined with a lack of financial incentive restricted development. In essence, large-scale investment was required to serve sufficient customers for good return.

The first organisation to partly overcome these technical difficulties was LANcity Corporation of Andover, Mass. USA in 1990, which became a commercial success with the development of cable modem technology. LANcity's cable modems became an essential device as it allowed the CATV networks to extend communications services to entire cities. Soon after LANcity developed its cable modem, Zenith introduced a similar type of cable modem. By 1992, both of these companies were offering "Symmetrical LAN over cable" [63]. Since then, diverse groups have worked together to help cable modem technology and cable data networks standards. Section 2.5 presents a number of the CATV standards that support this technology.

2.3 CATV networks

Traditional CATV networks are based on a tree and branch network architecture as illustrated in Figure 2.1. The main components are: *headend*, *trunk cables*, *amplifiers*, *feeder* or *distribution cables*, *splitters*, *drop cables*, *taps* and *terminal equipment*. Signals emanate from a headend location, which receives programming on TV channels from a variety of sources such as satellites, broadcast transmission and local television studios. From the *headend* the signals are delivered to subscribers, first down a *trunk* (tree) *cable*, that carry the signals to residential areas. This *trunk cable* was historically a high quality coax cable and was intended to cover large distances, often well over 10 miles and with ≈ 20 -40 *amplifiers* in cascade from the *headend* to subscriber [24].

Along the coaxial *trunk* route, signals would need to be amplified to retain quality. *Amplifiers* are required approximately every 0.4 miles depending on the bandwidth of the system. The more splits there are in a cable network and the greater the distance from the *headend*, the more *amplifiers* are needed.

The *distribution* or *feeder cable* expands around the neighbourhood. The length of this cable can be up to 1 mile. This limitation is due to the fact that Radio Frequency (RF) energy is tapped off to feed homes.

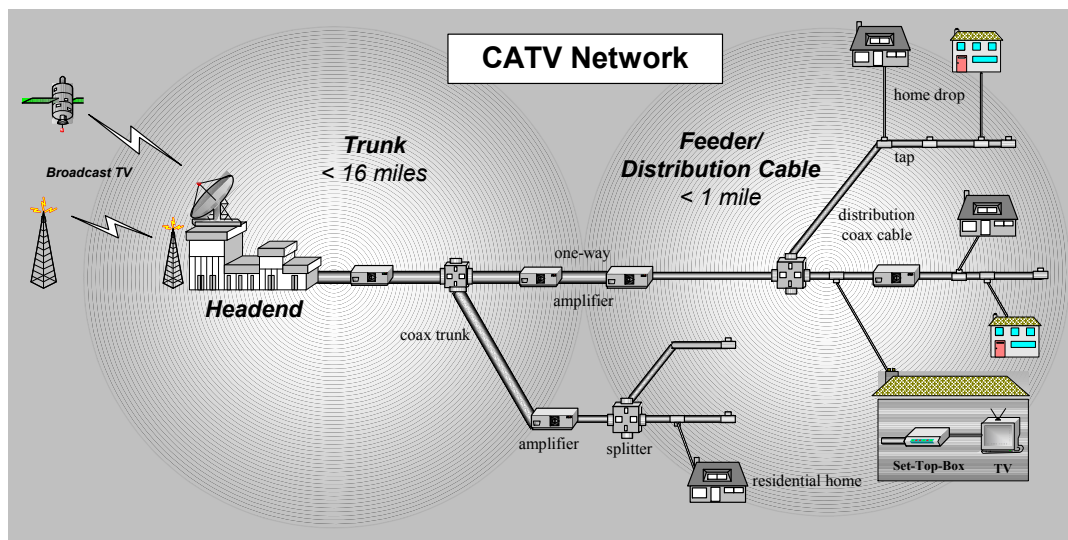


Figure 2.1 – Example block diagram of a typical CATV network.

Splitters are used at every junction to distribute the signal between branches. The interface between *truck cables* and *feeders* is done by *bridge amplifiers*. More sophisticated *amplifiers* are used in the distribution loop because noise levels are higher in the *feeder cables* than in the *truck* portion. These *amplifiers* are called *line extenders* and operate at higher power levels in this part of the system. *Line extenders* are required approximately every 0.25 miles and are restricted to a maximum of three [99].

Taps to connect subscribers are placed approximately every 150 feet. The *drop cable* connects the subscriber to the *tap* and its maximum length is approximately 400 feet. *Terminal equipment* is the last component of the system. They are the receivers that use the signal.

As the service delivered to subscribers was originally designed exclusively to accommodate television programming, the service was unidirectional. The signals delivered were analogue and replicas of the one a broadcaster sends through the airwaves. Because of this there was no need to modify the television set.

2.3.1 CATV spectrum allocation

The spectrum for CATV networks is divided into the downstream and upstream frequencies. Figure 2.2 shows the basic downstream and upstream spectrum allocations and the slight variations in the frequency ranges for Europe (EU), United States (US) and Japan (JP) [109].

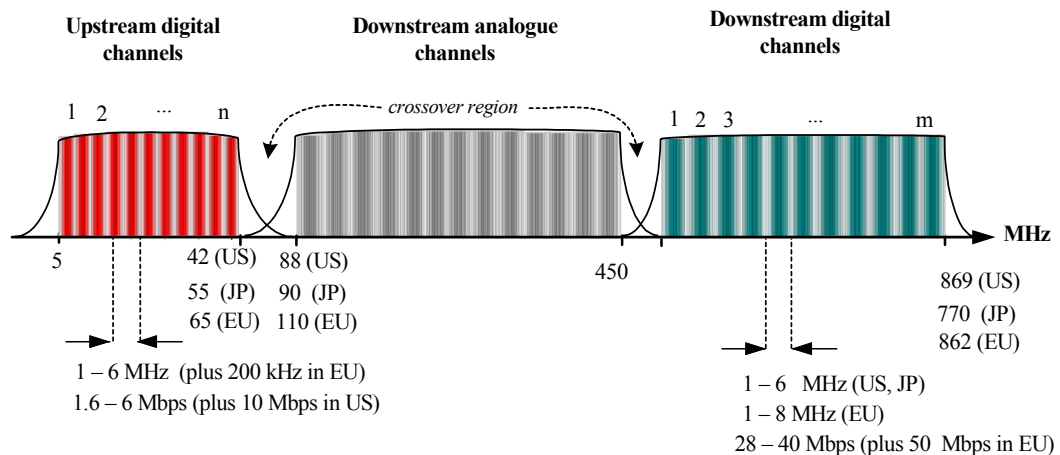


Figure 2.2 – Spectrum allocation on a CATV network.

The downstream channels support the legacy analogue broadcast television (80-450 MHz, the frequency depends on the country), and multiples of 1-6 MHz or 1-8 MHz channels in the 450-900 MHz region for the transmission of digital data. The upstream channels are also divided into 1-6 MHz channels, but the high level of noise means that data capacity is only 1-6 Mbps per channel in EU and 1-10 Mbps in US and Japan as opposed to the 28-40 Mbps available in each downstream channel in US and Japan. A data rate of up to 50 Mbps can also be supported in EU.

In terms of modulation schemes, most manufacturers have implemented 64 and 256 Quadrature Amplitude Modulation (QAM) for the downstream and Quaternary Phase Shift Keying (QPSK) for the upstream channels. Japan and US have also implemented a 16-QAM modulation. These modulation techniques are described in [6].

2.3.2 Upgrade of CATV networks to bi-directional HFC architectures

In the last few years, CATV networks have been upgraded to improve the quality of the signal transmission and to increase available capacity. Modern CATV networks are initially built with Hybrid Fibre Coax technology [64]. The gradual evolution has shown the technical and economical viability of two-way communication in a CATV network.

A combined fibre coaxial CATV network is referred to as an HFC System. Fibre links are used to transport subcarrier multiplexed signals typical to a group of between 500 and 2000 subscribers. Such networks are now standard and provide a bandwidth up to 750 MHz (in most systems) in the downstream direction for digital and analogue transmissions [27]. This bandwidth will be up to 1 GHz in the future [12].

In order to deliver data over HFC networks, laser transmitters convert a fixed frequency (6 MHz wide in the US and 8 MHz in Europe) sent from the *headend* into optical signals. At the outskirts of a community, a laser receiver, named as Optical Network Unit (ONU), reconverts the signals so that they can again be transmitted over the coaxial cable, which goes into each individual house. At the Customer Premise Equipment (CPE) a receiver (set-top box or cable modem) tunes to the appropriate fixed frequency in order to receive downstream signals.

The support of digital signals requires the use of the appropriate modulation technique. The most widely used digital modulations for HFC networks are QAM, QPSK and Vestigial Side Band (VSB).

A modern CATV network for analogue video and digital services is presented in Figure 2.3. The *truck cables* from traditional CATV networks are replaced with high-reliable, low-attenuation fibre links. This implies that a large number of the analogue amplifiers along the tree trunk are no longer needed. Hence, the reliability of the system increases and the quality of the signal improves, since there exist fewer active components between the headend and the subscribers [12].

In addition, one-way (*line extender*) amplifiers from traditional CATV networks need also to be replaced by two-way amplifiers to allow upstream transmission from subscribers to the headend. Controllers at the *headend*, (referred as Cable Modem Terminations Systems -CMTS or Interactive Network Adaptor –INA) and cable-modems need to be installed at the headend and customer premises, respectively.

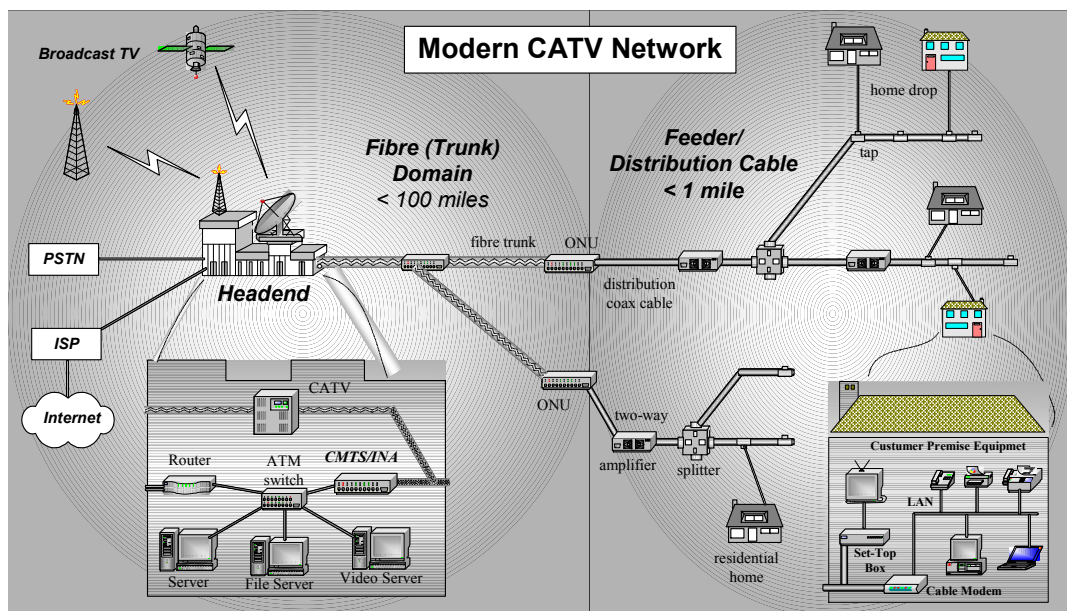


Figure 2.3 – Modern CATV network with HFC architecture.

2.4 Cable modem technology

Digital data signals are carried over Radio Frequency (RF) carrier signals on a cable system. Digital data utilises **cable modems**, devices that convert digital information into a modulated RF signal and convert RF signals back to digital information. The conversion is performed at the subscriber's premises, and again by the headend equipment handling multiple subscribers.

Cable modems have some advantages over other competing technologies, such as dial-up modem and the digital subscriber line (DSL). Although cable modems can operate at speeds up to 50 Mbps downstream and 12 Mbps upstream [22] and [35], they are normally programmed to operate at about 500 kbps - 2 Mbps upstream and \approx 500 kbps - 10 Mbps downstream. Subscribers are always connected, eliminating the call set-up times. Some of the disadvantages are that cable modems are less secure than DSL modems because the line is shared with others in the neighbourhood and data speeds vary according to the number of active users.

2.4.1 Cable modem configurations

Currently there are three configurations for cable modems: *external cable modems*, *internal cable modems*, and *interactive set-top boxes* (STB), as illustrated in Figure 2.4.

- *External cable modem* configurations are common and there are several combinations. Four examples are given here. The first combination is to have a PC,

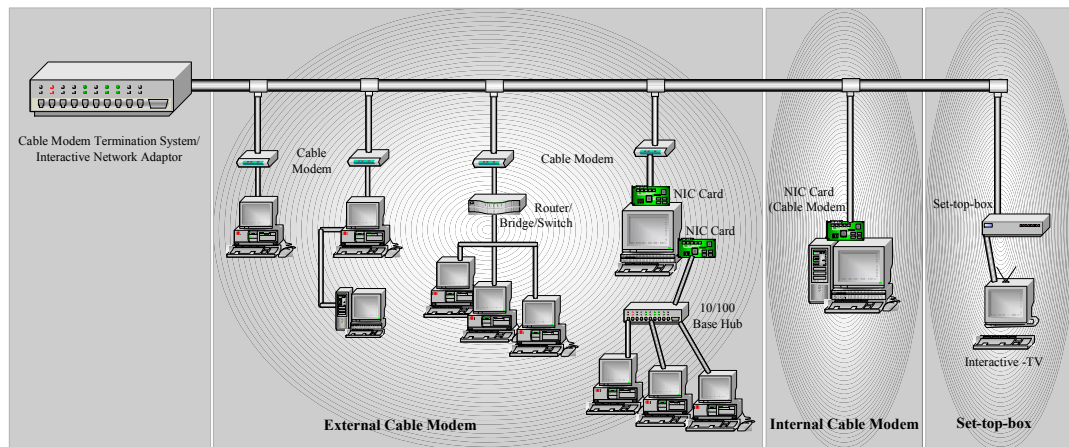


Figure 2.4 – Cable modem configurations.

with a cheap Network Interface Card (NIC) inside, directly connected to a cable modem through an Ethernet connection (10BaseT) or Universal Serial Bus (USB). The second alternative is to have two PCs connected to a single cable modem. Most providers have instructions on how to accomplish this and require a home user to download additional software (such as Network Address Translator-NAT [107]) to accommodate the dual connection on a single modem [9]. This means that the MAC address of the packets coming from a PC (without cable modem connection) should be changed to the MAC address of the PC's NIC card with cable modem connectivity. This is because most cable modems only forward traffic to the CATV network from a single MAC address.

The third alternative is to have a router, bridge or switch connected to the cable modem. With this configuration, the cable modem is attached to the network and the router/bridge/switch is responsible for forwarding traffic onto the cable system using its MAC address. This is an attractive solution but it is more expensive because a costly device is required.

A last configuration presented is to have two NIC cards in a single PC and the corresponding software (NAT) to forward traffic from other PCs. With one NIC card this PC has connectivity with the cable modem. With the other NIC card, a 10/100 Base Hub can be connected, which in turn interacts with others PCs. This is an inexpensive solution and allows multiple PCs to have interconnectivity with the outside world.

According to [9], the use of the PC with the two NIC cards is not necessary since the hub can be connected directly to a cable modem. The hub will act as a bridging function onto the cable modem, and concentrate the traffic through the individual devices.

- *Internal cable modems* are cable modems that are built into a PCI add-on card. It may be the cheapest implementation possible, but there are some drawbacks. The design only allows a cable modem to connect to one computer. In addition, internal cable modems do not isolate the connection to the cable network from the AC power system. This can cause major problems in some cable systems.

- The *Interactive STB* configuration serves a very different function to the first two configurations. The STB is not really designed to connect to a computer, although it functions in a similar way to a cable modem. Its primary role is to provide more television channels on the same limited number of frequencies. This is made possible by using Digital Video Broadcasting (DVB) technology. STBs also provide two-way communication. This two-way communication allows the user access to Enhanced-TV, Interactive TV, Web-browsing, email, etc., directly on the TV screen.

2.4.2 Next generation services

The services that modern CATV networks would provide are listed in Table 2.1 (Sources from [16], [36] and [25]). Not all of these services have to be supported in the near future, but the network infrastructure should be able to support all of them as and when demand rises. Most of these new services are graphics, audio and video oriented, and contrary to the simple text-oriented first generation application, they demand a high-speed communications infrastructure.

In addition, the requirement of ‘*real-time*’ interactivity makes the need for multi-megabit-per-second high-speed communications even greater. Since voice and video transmission are inherently time sensitive (isochronous) applications, the network has to exhibit deterministic behaviour, and there has to be an upper limit for the transmission

Table 2.1 – Next generation of CATV services.

Analogue and Digital TV and Audio	Telecommunications Services	Multimedia Services	Local Services
Pay-per-channel	Telephony	Online-services	Local-channels
Pay-per-view	Video Phone Services	Interactive games	Local TV
Audio-on-demand	Videoconferencing	Distance Learning	Local Radio
Video-on-demand	Fax	Interactive Video	Local online services
High-definition TV	Telemetry	Interactive Multimedia	
Information Services	Shopping and bill paying services	Remote-control and monitoring	
Near-video-on-demand	High-speed Internet access		
Tele-banking/shopping	LAN interconnection (Leased Lines D1/E1)		
Tele-teaching learning	Gambling		

delay of chunks of audio and video. Bounded transmission delay is vital, since it dictates buffer size within the receiver and transmitter as well as the quality of the transmission.

For instance, for a videoconferencing service, the maximum tolerable (one-way) mean access delay for 95% of all video frames transmitted is ≈ 300 ms in order to guarantee a good quality of human interaction [38]. However, voice streams require lower access delays. As an example, high quality VoIP frames using a G.711 codec require a delay of between 1-20 ms and controllers for acoustic and electric echo are not needed, [118]. This delay can be more flexible for VoIP using codec G.723.1, which can be up to 150 ms for a good quality call. Here echo control is required but does not compromise the effective interaction between the users. If the delays are in the range 200 to 400 ms, the effectiveness of the interaction is lower but can be still acceptable, [118].

2.5 Standards for CATV networks

The development of standards and common protocols is essential to any wide-spread acceptance of similar technology. Standards are needed for new services so that different vendors' products can interoperate. Standards will provide economies of scale for a mass market resulting in lower cost products, because many internal elements would have the same specifications and could be manufactured in huge quantities

To make this vision true, industry groups, including the Digital Audio Visual Council (DAVIC) [129], Digital Video Broadcasting (DVB) [130], European Telecommunications Standards Institute (ETSI) [132], Institute of Electrical and Electronics Engineering (IEEE 802.14), ATM Forum [5], Multimedia Cable Network Systems (MCNS) Holdings [127], and the Society of Cable Telecommunications Engineers (SCTE) [134], created specification with the goal of increasing interoperability between equipment.

Alliances between the different groups led one organisation to adopt the other standards. As an example, the SCTE adopted the MCNS specification and the DVB group adopted DAVIC 1.2, while the IEEE 802.14 decided to work with the ATM Forum for the implementation of ATM over HFC networks.

Common goals within the above groups have resulted in the development of several international standards for CATV networks. A comprehensive description of these standards can be found in [84] and [109]. We list the main protocols for cable systems here.

- DVB/DAVIC: Digital Video Broadcasting (DVB) which has adopted the Digital Audio-Visual Council (DAVIC) recommendations with respect to CATV, has been responsible for the development of the European standard ETSI ES 200 800 [34], widely known as the DVB/DAVIC protocol specification (DVB-RCC). Chapter 3 presents a detailed description of this protocol.
- DOCSIS: The Data Over Cable Service Interface Specification (DOCSIS) standard emerged from the work of MCNS, a consortium consisting of Comcast, Time Warner, TCI, and Cox Communications. MCNS in turn partnered with MediaOne (previously Continental Cablevision), Rogers Cablesystems (Canada) and with CableLabs [127] who have now administered the initiative since 1997. This partnering of interests represents operators that provide cable service to a majority of North American cable subscribers, with connected homes in excess of 60 million. (85% in the US, 80% in Canada and 12% in Mexico), [8].

CableLabs has produced the Data Over Cable Service Interface Specifications, which comprise DOCSIS 1.0 [21], DOCSIS 1.1 [22], DOCSIS 1.2 [23] and EuroDOCSIS [2], on behalf of the North American Cable industry and using cable modem technology.

DOCSIS 1.0 is an evolved LANcity-based protocol, targeted at residential, low cost, off-the-shelf cable modems with certified interoperability between vendors, [69]. The initial release was in December 1996. The basic architecture of this protocol is a single, large Ethernet-based bridged LAN with primarily a best-effort Internet access system. It was not designed for QoS support.

DOCSIS 1.1 is based upon the required needs of the *PacketCable Audio/Video Codecs Specification* [80]. The PacketCable project [133] is an internal project within CableLabs and its main task is the support of voice and video applications.

The DOCSIS 1.1 version added substantial protocol support to provide dynamic QoS features for packet voice services, in addition to packet data services. Other enhancements include, for example, baseline privacy and multicast support.

DOCSIS 1.2 is a detailed technical specification of a next-generation physical layer technology (PHY) for integration into the Data-Over-Cable Service Interface Specification. The protocol specification is based on Terayon's Synchronous Code Division Multiple Access (S-CDMA) and Broadcom's advanced Frequency Agile TDMA (FA-TDMA), which were first integrated into the IEEE 802.14 protocol specification [54]. These modulation techniques will allow cable modems to support an upstream data rate up to 30 Mbps. This version recently became officially accepted as the DOCSIS 1.2 protocol specification [23], in December 2001.

Today, CableLabs runs a DOCSIS vendor certification process for cable modems. In January 2002, more than 100 cable modems were DOCSIS 1.0 certified from approximately 80 manufacturers.

- IEEE 802.14: The IEEE 802 was responsible for the development of LAN/MAN protocol standards. In May 1994, the IEEE 802.14 sub-committee was established to develop a MAC and PHY specification to support cable networks. This effort resulted in the development of the IEEE 802.14 protocol specification. In July 1997 the committee released the first internal draft for the MAC and PHY layers [51]. However, MCNS' effort undermined the IEEE 802.14 group's work and was able to define a specification much quicker than the IEEE. Despite the launch of the North American initiative, the IEEE 802.14 committee continued its work with the objective of creating an international, rather than national, standard.

Unfortunately, in September 1999, the joint effort was ceased, followed by the disbanding of the IEEE 802.14 Working Group in November 1999. This was the result of the slow standardisation process within the IEEE, which failed to observe the time to market constraint and lost the support of industry [84]. The latest draft specification, Draft 3, Revision 3 [52], will remain as a proposed specification within IEEE for one year after which time, if there is no interest by any other group or body, not necessarily within IEEE, it will be withdrawn.

In an effort to acknowledge the IEEE 802.14 Working Group's work, CableLabs have implemented IEEE 802.14's advanced PHY specification [54] within DOCSIS 1.2 [23]- a partial victory for the group.

2.6 Certified cable modems

The first organizations that offered proprietary cable modems and helped to release the cable modem's potential were LANcity, Zenith and Intel in 1993. Two years later, Motorola, Hybrid, Com21 and Westend joined in the development of this technology [125]. Today, approximately 80 vendors have entered the cable modem revolution, which is now a multi-billion-dollar industry. Of these vendors only ≈ 70 have obtained at least one DOCSIS CM or CMTS product certified by CableLabs [127].

Table 2.2 shows the current cable modem manufacturers with certification from CableLabs. This table also presents the organisations manufacturing DVB/DAVIC equipment.

Table 2.2 - Certified cable modem vendors (from CableLabs Certified/Qualified Product, information as of January 17, 2002 [127] and Cable Datacom News [126]).

DOCSIS-Products	DOCSIS-Vendors with CableLabs Certification
DOCSIS 1.0: CM (0)	3COM(0,2,45) , Aastra(0), Accton(0), ADC(2,5), Alcatel(0),
DOCSIS 1.1: CM (1)	Ambit Microsystem(0,1), Arris(0,1,2,3,4,5) , Askey(0,4), Asustec(0,4),
DOCSIS 1.0: CMTS (2)	BAS(2), BestData(0), CIS Technology(0), Cadant(3), CastleNet(0,4),
DOCSIS 1.1: CMTS (3)	Cisco Systems(0,2,3,5) , Com21(0), Coresma(0), Correlant(0),
EuroDOCSIS 1.0: CM (4)	D-Link Systems(0,4), DX Antenna(0), Dakos(0), Dassault(0,4),
EuroDOCSIS 1.0:CMTS (5)	DeltaKbel(0), ElsaAG(0,4), Ericsson(0,1,4) , Future Networks(0), General Instrument (0), GVC(0), Global Telemann(0), Hauppauge(0), HighSpeed Surfing(0), Hitron Technologies (0), Infinite(0), JooHong(0), LG Innotec(0), LinkSys(0), Lucent(Delta-Kbel)(0), Maspro Denkoh (0), Matsushita Panasonic(0), Motorola(0,2,4,5) , NetGame(0), NetGear(0), Net & Sys Co.(0,4), Nortel(0), Ole Communications(0), Pacific BB Comms(2), Panasonic(0), Philips(0), Powercom (0), RiverDelta(2), RiverStone(2,5), SMC Networks(0), SOHWARE(0), Saejin (0), Samsung (0,4), Scientific-Atlanta(0,1,4) , Sony(0), TCE(0), Techno Trend (4), Tellabs(0,1,2,5) , Terayon(0,2,5) , Texas Instrumets(0,1), Thomson(0,4), Toshiba (0,1,4) , TriGem(0), TurboComm(0), U.S. Robotics(0), Zoom Telephonics (0), ZyXEL (0).
DVB/DAVIC-Products	DVB/DAVIC -Vendors
DVB/DAVIC: 1.0 EM (0)	Alcatel (1), Cisco Systems(1+DDIC), Com21 (0+DDIC), DeltaKbel(0),
DVB/DAVIC: 1.0 INA (1)	HB Telecom(0), Hughes Network Systems (0), RiverDelta(1), Terayon(0+DDIC), The Industree (0+DDIC), Thomson (1+DDIC)

CATV vendors in bold have obtained more than two cable modem devices certified by CableLabs.

However, not all DVB/DAVIC vendors have obtained the verification from the DVB/DAVIC Interoperability Consortium (DDIC). Moreover, the number of organisations producing DOCSIS equipment is considerably large when compared with the number of DVB/DAVIC vendors (≈ 10).

Most cable modems, currently deployed as a commercial service, are external cable modems (with Ethernet or/and USB ports). Some cable modems have advanced capabilities. For instance, Motorola and Com21 have manufactured external cable modem routers that have a 4-port hub, built-in NAT, Dynamic Host Configuration Protocol (DHCP) server and firewall for creating home networks. *Interactive, Toshiba, Tellabs* and *Cisco Systems* have produced external cable modems with PacketCable Technology [80] that allows VoIP calls (and may be capable of video over IP transmissions in the future).

Moreover, *Terayon* is the first organisation that has already produced cable modems compliant to DOCSIS 1.2 [23], which includes advanced PHY capabilities and dual modulation techniques, S-CDMA and FA-TDMA. DOCSIS 1.2 triples the upstream capacity of DOCSIS 1.1 (up to 30 Mbps with 64-QAM modulation) enabling operators to create new services for residential and business markets, such as video conferencing and peer-to-peer applications.

Currently, the number of cable modems shipped worldwide has reached the top-mark of 15 million modems, with *Motorola, Toshiba* and *Thomson* leading the market with 39%, 17% and 12% of the sales, respectively [126].

2.7 Competing technologies

Multiple broadband competing technologies are being developed to provide high-speed access to small office and residential areas. Digital Subscriber Line (xDSL), and several forms of passive optical networks (FTTx) are direct competitors to HFC infrastructures and the challenge is to work out the least expensive and most efficient solution, while taking into account the different boundary conditions. In this section, a review of these competing technologies is addressed.

2.7.1 FTTx technologies

FTTx is a series of networking technologies that connect the users and the central office via optical fibre links. Although the cost of optical fibre is not much different from coaxial cable, optical equipment is very expensive. There are two main forms of FTTx technology:

- **FTTC:** Fibre-to-the-Curb (FTTC) is also referred as Switched Digital Video (SDV). This technology makes use of fibre optic to connect the headend to Optical Network Units (ONU) 'at the curb' and serves small groups of homes (between 8 and 24 homes) [67]. Twisted pair and coax cable is used to connect the ONU to individual homes. Twisted pair supports POTS while coax supports analogue video and new digital services can be supported by either.

While TDM is used at present to multiplex signals for delivery to individual homes, ATM switches will most likely replace the ONU in the near future [67]. Telephony, video and digital services are multiplexed for transport over the FTTC network and are de-multiplexed by the ONUs. The maximum downstream bandwidth supported by FTTC networks is 51 Mbps using coax cable. Twisted pair can also be used to support the same bandwidth if the length is not greater than 500 feet.

- **FTTH:** Fibre-to-the-Home (FTTH). This technology is a point-to-point network architecture that uses only fibre links. The cost of deployment of a FTTH network is considered high compared to FTTC and HFC. Many people believe that FTTC technology is more cost-effective than FTTH due to maintenance and the large number of ONUs. However the opposite is true, [76]. Although FTTH networks avoid the use of ONUs, the high number of laser transceivers used and the cost of the fibre links, which run down to the last mile into each home, makes them a costly solution. However, the increased bandwidth available to each fibre out of the *Headend* might be as high as 155 Mbps (OC-3c), allowing each home to become a service provider by attaching servers within the subscriber's premises [67]. FTTH is the topology that offers the highest capacity due to its all-fibre structure and is an attractive solution for Telco's. Some Telco's have already installed FTTH networks, such as Nippon Telegraph and Telephone, (NTT) [122] and Deutsche Telecom [123].

Due to a high installation cost involved, the initial structure may not be point-to-point. However, using a Passive Optical Network (PON) and passive optical couplers to split fibres from the *Headend* to the home into multiple fibres reduces the number of laser transceivers required for every home, and in turn reduces the cost of deployment. Bi-directional communication over FTTH networks is achieved by a MAC protocol that shares bandwidth on the PON structure among different home servers [75].

- **SuperPONs:** Super Passive Optical Networks (SuperPONs) is another alternative technology in its early stages, which can be applied to FTTC and FTTH networks [81]. This technology is considered as the next generation PONs, designed to run greater distances (about 60 miles) and can support a larger number of users (up to 2048 ONUs). SuperPONs provide shared bandwidth up to 2.5G bps downstream using TDM and 311 Mbps upstream with TDMA. The major technical challenge in such architectures is the development of a MAC protocol that could allow fair access to the network by such a large number of users.

2.7.2 xDSL technologies

xDSL is a series of Digital Subscriber Line technologies which allows for the transmission of information (voice, video and data), over existing copper telephone lines at high speeds. The xDSL technology is attractive, since there exists a huge installed base of twisted pair lines (800-million) worldwide [26]. Optical fibre will carry signals from a central office to a neighbourhood node, which in turn converts the signals and puts them into a telephone line to the user. In some cases, a direct copper line can be used if the distance from the central office to the user is not too large (usually under 4 miles). DSL has a similar problem to cable modem technology; upstream bandwidth is much smaller than downstream (except its symmetric system which is much more expensive). There are various forms of DSL, referred to as xDSL.

- **ADSL:** Asymmetric Digital Subscriber Line (ADSL) is quickly becoming the most popular form of xDSL. ADSL can support up to 8 Mbps bandwidth for downloading and up to 640 kbps for uploading [6].

The asymmetrical nature of ADSL technology makes it ideal for Internet/Intranet surfing, video-on-demand and remote LAN access.

ADSL requires a voice/data splitter, commonly called a POTS Splitter (Plain Old Telephone Service) to be installed at the consumer's home or business location. The splitter separates voice from data transmissions. For simultaneous use of the telephone and data access, additional phone wires may need to be installed at the location. Full rate ADSL provides service up to a maximum range of 12,000 feet (about 2 miles) from the provider company's central office to the end-user. For distances up to 15,000–18,000 feet (about 2.8-3.4 miles) the data rate decreases to 1.5 Mbps downstream and 64 kbps upstream [68].

- *G.Lite*: This technology, often called ADSL Lite, Splitterless or Universal ADSL and now also known as G.992.2 [58], does not require a POTS splitter to be installed at the consumer's home or business. ADSL Lite provides bandwidth downstream up to 1.5 Mbps and upstream up to 512 kbps. ADSL Lite provides service up to a maximum range of 12,000 feet (about 2.0 miles) from the central office. Under good home conditions and loop quality the range can be extended up 18,000 feet (\approx 3.4 miles) [68]. This technology will be primarily targeted towards residential customers, for combined data and circuit/IP voice services over a single twisted copper loop pair. In addition, it is expected that the data rates of this standard will meet the needs of the average consumer for some time to come [9].
- *SDSL*: Symmetrical Digital Subscriber Line (SDSL) delivers high-speed data networking over a single-pair of copper phone lines, at the same speed in both the upstream and downstream directions. Speed ranges achieved by this technology are 384 kbps, 768 kbps, 1 Mbps, 1.544 Mbps (T1 service over two copper pairs) or up to 2.048 Mbps (for E1 streams over three copper pairs) at a maximum range of 12,000 feet (about 2.3 miles) [11]. SDSL is ideal for business applications that require identical downstream and upstream speeds, such as video conferencing or collaborative computing as well as similar applications appropriate for ADSL technology. SDSL uses either CAP modulation or the same kind of line-modulation technique employed in ISDN, known as 2B1Q.

- *HDSL*: High-data-rate DSL (HDSL) is a two or three copper pair technology that achieves symmetrical data rate transmissions, conforming to T1 (1.544 Mbps) or E1 (2.048 Mbps) standards, respectively [37]. This technology uses either baseband 2B1Q or passband CAP modulation schemes and the distance from the central office is limited to 12,000 feet (\approx 2.3 miles). According to [9], the use of two or three copper pairs is no longer needed, because the most recent versions of HDSL architectures use only one pair of wires, and it is expected to be more accepted by the providers.
- *VDSL*: Very high bit-rate Digital Subscriber Line (VDSL) is the fastest xDSL technology. Several VDSL formats have been proposed and trailed. However, standardisation is still in process at ETSI. This technology emerged in order to deliver multi-megabit data rates over short spans of copper wire, such as the distribution of digital TV programming to the neighbourhood node for FTTC applications.

VDSL delivers up to 52 Mbps downstream and from 1.5 to 2.3 Mbps upstream over a single pair of copper wires [25]. This technology is limited to a maximum range of 1,000 to 4,500 feet ($<$ 1 mile) from the central office, depending upon the speed. Modulation schemes proposed for VDSL include DMT and CAP. Moreover, a symmetrical version of VDSL is under study to operate in the range from 6.5 to 26 Mbps [11].

Table 2.3 presents a comparison of xDSL technologies along with FTTx and HFC structures.

Table 2.3 – Broadband access technologies.

Technology	Bandwidth & Medium	Modulation	Max. Distance	Application
HFC (1988)	42 Mbps down 10 Mbps and possibly to 30Mbps up fibre/coax cable	QAM, QPSK VSB	100 miles fibre trunk 1 mile coax cable	High-speed Internet, VoIP access, motion video, VoD, remote LAN access, etc.
FTTC (End 1980s)	51 Mbps down and up fibre/coax/twisted pair	TDM down TDMA up	100 miles fibre trunk 1 mile coax cable 500 feet copper	Video broadcasting, HDTV Interactive multimedia, ATM traffic
FTTH (End 1980s)	155 Mbps down and up	TDM down TDMA up	100 miles only fibre links	Video broadcasting, HDTV Interactive multimedia, ATM traffic
SuperPONs	2.3 Gbps down 311 Mbps up	TDM down TDMA up	60 miles	Video broadcasting, HDTV Interactive multimedia, ATM traffic
ADSL (1995)	1.5-8.2 Mbps down 16-640 kbps up single twisted pair	DMT	3.4 miles at 1.5 Mbps 3.0 miles at 2.0 Mbps 2.3 miles at 6.3 Mbps 1.7 miles at 8.2 Mbps	Telephony, high-speed Internet access, motion video, VoD, remote LAN access.
G.Lite, DSL-Lite or Splitterless ASDL (1997)	1.5 Mbps down 512 kbps up single twisted pair	DTM	3.4 miles	Telephony, IP voice services over a single twisted copper loop pair
HDSL (1991)	1.5 Mbps duplex on two twisted-pair lines, up and down 2.0 Mbps duplex on three twisted-pair lines, up and down	2B1Q CAP	2.3 miles	T1/E1 service between server and phone company or within a company; WAN, LAN, server access
SDSL (1996)	1.5 Mbps duplex, up and down 2.0 Mbps on a single duplex line up and down	2B1Q CAP	2.3 miles	Same as for HDSL
VDSL (1995)	13 to 53 Mbps down, 1.5 to 2.3 Mbps up single twisted pair	DMT, CAP	0.9 miles at 13 Mbps 0.6 miles at 26 Mbps 0.2 miles at 53 Mbps	Same services as ADSL + Video broadcasting, HDTV Interactive multimedia, ATM traffic over Fiber to the Neighbourhood.

2.8 Conclusions

The increased need of Internet and broadband services has created an awareness of and demand for high-speed access at mass-market prices. CATV networks evolved as an alternative to satisfy these needs. However, the transition of CATV networks, from analogue broadcast unidirectional to a high-speed bi-directional digital medium, requires the reduction of noise and the availability of cable modem technology, which will provide bi-directional communication over the shared medium. The introduction of fibre links in the cable plant, combined with progress in optical laser technology, has made possible the reduction of noise. Cable modem manufacturers first introduced proprietary equipment with limited capabilities in terms of multimedia applications and low QoS support. Later, with the alliance of influential groups, such as the DVB-DAVIC-ETSI, MCNS-SCTE and IEEE 802.14 - ATM Forum, several cable modem standards have emerged. However, only the DOCSIS protocol and the DVB/DAVIC protocol achieved the short time standardisation process demanded by Multiple Service Operators. Such protocols contribute to a wider acceptance of CATV networks.

Telco's, with its DSL technology, have entered the race for high-speed digital services over their point-to-point switched network. Although this technology (from telephone companies) is slightly more expensive than cable modem technology, DSL has the advantage of a dedicated bandwidth for each node/service. Moreover, Fibre To The Curb/Home is a promising technology, which can achieve very high data transmissions in the order of 150 Mbps. Unfortunately, this technology is still in its early stages and at the current time is too expensive for a residential or small business to afford.

In the race for supplying multimedia broadband services and high-speed Internet access to residential customers, CATV networks outdistance the competitors of digital subscriber line, fixed-point wireless, and fibre optic to the home systems. The race is ongoing and for the foreseeable future it appears likely that cable and DSL technology will lead the race to the home for supplying television, video on demand, home shopping, video games, telephony, high quality video telephony, high-speed data services, and eventually hundreds of residential applications being envisioned or to be invented.

Chapter 3

THE DVB/DAVIC PROTOCOL

3.1 Introduction

Digital Video Broadcasting (DVB) has become one of the most exciting developments in the area of consumer electronics at the end of the twentieth century [88]. The DVB Project emerged from a group called the Launching Group of European broadcasters, consumer electronics manufacturers and radio regulatory bodies in 1992. Since then a number of protocol specifications have emerged. This chapter presents a review of the DVB project and the main standards produced by this group. Special attention is paid to the DVB/DAVIC protocol specification, which is the one that will be analysed, optimised and improved in subsequent chapters. Here the main characteristics, as well as the architecture, MAC operation, and a description of the *exponential backoff algorithm* and the *splitting tree algorithm* adopted by the DVB/DAVIC protocol are described.

3.2 DVB project overview

The focal point of the European DVB project is the development of standards for the delivery of digital TV over satellite DVB-S, terrestrial DVB-T and more recently delivery of digital data over cable links DVB-C for the downstream channel and DVB-RCC (or DVB/DAVIC) for the upstream channel or return channel.

The DVB-S (EN 300 421 [29]) is a satellite specification designed to operate within a range of transponder bandwidths (26 MHz to 72 MHz). However, the cable network on the downstream direction DVB-C (ETS 300 429 [31]) has the same core as the satellite system, but the modulation system is based on QAM rather than QPSK.

Initially DVB recommendations did not cater for bi-directional communications. However, the implementation of interactive TV will require data in the reverse direction. Therefore, the DVB group is turning to other standardising bodies in order to produce a specification with a wider range of applications, which will span beyond digital TV broadcast such as high-speed Internet access, VoIP, video-telephony and videoconferencing, among other services.

At the initial stages DVB was evaluating which standard would be the most suitable for the delivery of Motion Picture Experts Group (MPEG-2) audio/video streams over CATV networks. At its meeting in July 1997, DVB announced it would adopt the DAVIC 1.2 specification. Thereafter, both the UK and the European Cable Communication Associations (ECCA) announced that they were going to support the DVB/DAVIC standard.

In order to accomplish compatibility with the other standards under development, DVB requested from the IEEE 802.14 group to optionally include “DAVIC 1.2 Part 8” in its specification, which dealt with lower layer protocols and physical interfaces for coaxial cable (including the MAC functionality). The European Telecommunication Standard ETS 300 800 [32] (also known as DVB/DAVIC) has now been produced as the baseline specification of the interaction channel for CATV distribution systems. Shortly thereafter, the European Telecommunication Standards Institute ETSI officially accepted this standard, which became the European Standard “ETSI ES 200 800” [34], in April 2000.

3.3 EuroModem

Meanwhile the process of standardisation of the ETSI ES 200 800 protocol specification, the European Cable Communication Association (ECCA) [131] and EuroCableLabs (ECL), as the centre of competence of the European cable operators, decided in January 1999 to push and co-ordinate the activities for the market introduction of the DVB/DAVIC compliant cable modems (based on the standards ETS 300 800 and ETSI ES 200 800). Thus, the EuroModem project emerged and created the technical specification of an external cable modem that fulfils the requirements of the

European cable operators. The final EuroModem specification [36] was published after approval by the ECLs in May 1999.

Two different types of EuroModem devices have been defined. The class “A” EuroModem is the basic version and is used mainly for high-speed Internet access. This has functionality similar to that of a DOCSIS 1.0 compliant modem. Using a class A EuroModem a secure data transmission is possible due to the defined encryption technique. The class “B” EuroModem is the enhanced version supporting some additional features. For instance, it is possible to deliver high-quality telephony services or Integrated Services Digital Network (ISDN) connections and also IP telephony (VoIP) with a guaranteed Quality-of-Service (QoS) level. A common telephony interface allows connection of a telephone device directly to the class B EuroModem. This modem has functionality similar to that of the DOCSIS 1.1 compliant cable modem. For a further comparison of these three protocol specifications the readers are referred to [84] and [109].

3.4 DVB/DAVIC reference model

The system reference model for interactive services for the DVB/DAVIC protocol specification is shown in Figure 3.1. In the system model there are two channels established between the service provider and the user.

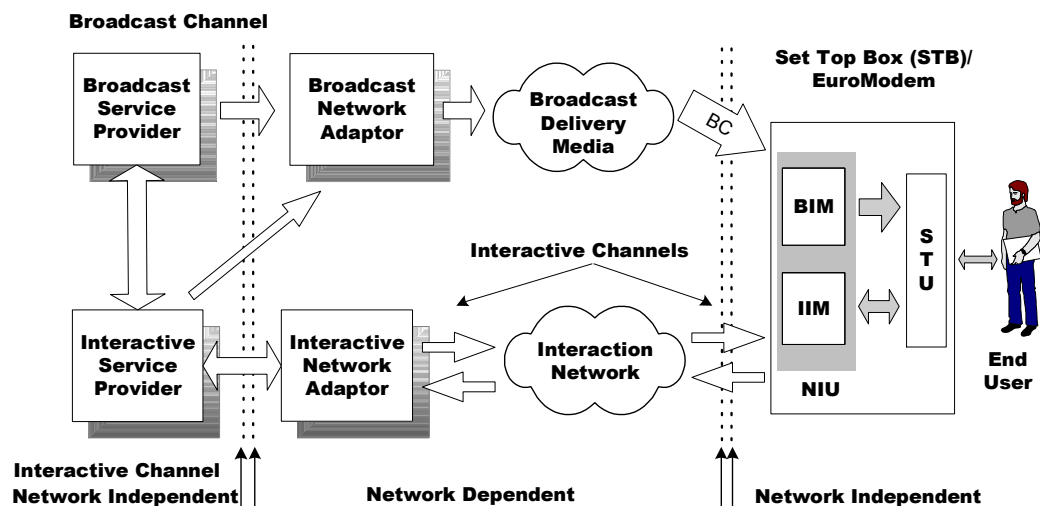


Figure 3.1 – DVB system reference model.

They are the Broadcast Channel (BC) and the Interaction Channel (IC). The BC is a unidirectional broadband broadcast channel including video, audio and data. The IC is a bi-directional interaction channel, which is established between the service provider and the user for interaction purposes. It is formed by the following paths:

- *Return interaction path* (also referred to as upstream channel): from the user to the service provider. It is used to make requests to the service provider, transmit user data or to answer questions.
- *Forward interaction path* (also referred to as downstream channel): from the service provider to the user. It is used to provide information generated by the service provider for the user and any other required communication for interaction service provision.

Data connectivity is achieved via STB or EuroModem, which contain a Network Interface Unit (NIU) for interfacing with the cable network. The NIU consists of the Broadcast Interface Module (BIM) and the Interactive Interface Module (IIM). The user terminal provides an interface for both the broadcast and interaction channels. The interface between the user terminal and the interaction network is via the IIM.

In the upstream channel, a 64-byte (upstream) slot structure is used and in the downstream channel, the packet structure is based on 188-byte MPEG2 TS frames. Appendix B presents a description of packet formats, PDU structures, signalling methods and protocol configurations.

3.5 DVB/DAVIC MAC operation

One downstream channel can manage up to eight upstream channels by using MAC control information (referred here as MCI, see Figure B.4). This MCI field plays a vital role in the operation of this communication system, because it contains synchronisation information of the upstream slots. Its main functionality is to co-ordinate the usage, assign access modes, and indicate if reception of contention-based slots was successful. Each slot is assigned one of the following four classifications from the INA: ranging (for synchronisation and calibration purposes), contention (for light traffic load and

MAC control message transmissions), reservation (for bursty or high traffic load) or fixed slots (for constant bit rate traffic).

These MPEG2 frames with MCI information are transmitted in the downstream channel (at least) once every 3 ms when the upstream data rate is 6.176 Mbps [grade D], 3.088 Mbps [grade C] or 1.544 Mbps [grade B], and every 6 ms for 256 kbps [grade A]). The MCI field describes up to 36, 18, 9 and 3 upstream slots for grades D, C, B, and A, respectively. The limits between access regions allow the EuroModem (which is also referred to as EM, NIU or station) to know when to send data on contention at a time that risk of collision with data of reservation or fixed-rate regions does not exist.

Previously in [101] and also in [34] the authors have reported that several functions are performed by the MAC protocol for connection control and data transmission as depicted in Figure 3.2. On power-on or reset, the initialisation and provisioning procedure makes sure that an NIU is capable of tuning to the correct channel in the upstream and downstream directions and that it can receive the basic network parameters.

Initialisation and Registration Process

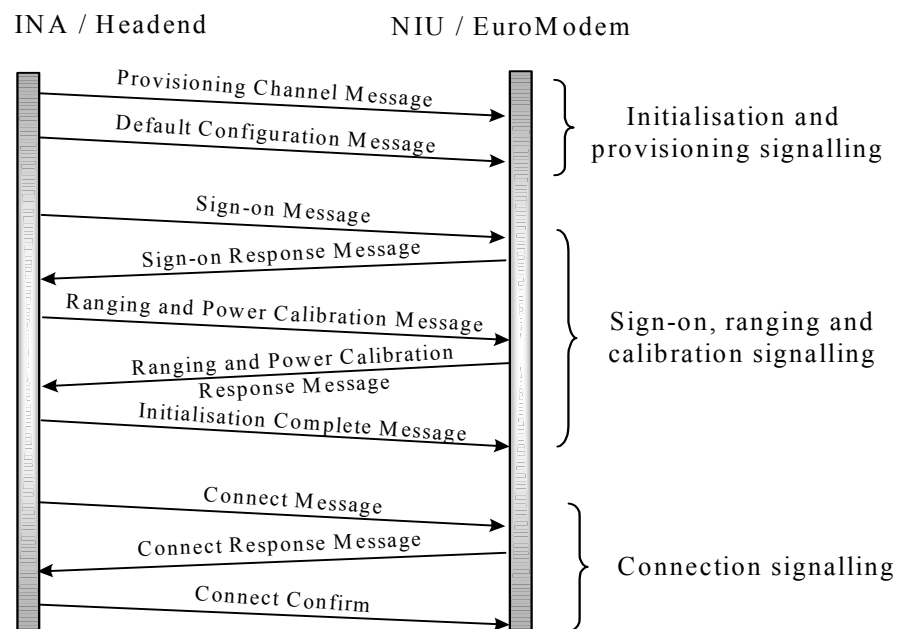


Figure 3.2 – DVB/DAVIC initialisation and registration process.

Then, the sign-on, ranging and calibration are performed in order to adjust the internal clock and the transmission power of the NIU according to the specific transmission delay and cable attenuation. The initial connection is also established by default. The MAC protocol also carries out the establishment and release of logical connections and allows readjustment of parameters, as well as performing an exchange of keys and establishment of a secure connection. Here Diffie-Hellman and Data Encryption System (DES) are used.

Once an NIU has initialised and registered with the INA, bandwidth reservation is provided by both per-packet and per-session (or connection) basis. The former is known as **Reservation Access** and is provided by sending reservation requests to the INA. The first version of EuroModems (class A) will support this functionality as a mandatory access mode. The latter is known as **Fixed-rate access**, where the INA provides either a finite amount of slots (in the fixed-rate access region) to a specific NIU or a given bit rate requested by an NIU until the INA stops the connection on NIU's demand. This access mode will be an optional functionality for EuroModem Class A, and mandatory for EuroModem Class B.

3.5.1 Reservation access mode operation

The reservation-access mode with its dynamic slot-allocation feature is the main access mode of the DVB MAC protocol for the transmission of data packets via the upstream channel. For this access mode, the DVB/DAVIC group has adopted two contention resolution algorithms, which are used to resolve collisions: the **exponential backoff algorithm** and **splitting tree algorithm**. The *splitting tree algorithm* takes advantage of the *exponential backoff algorithm* in the sense that feedback and allocation information allow a station, (with new incoming arrivals) to compete for contention-based slots without risk of collision with *backlogged* stations. In addition, this algorithm makes use of minislots, which decreases the risk of collisions, since one contention-based slot is divided into three minislots of 21-bytes long transferring shortened reservation request messages, increasing the probability of successful request transmissions and consequently improving the system performance.

The reservation-access mode uses the following ‘*Contention-Resolution-Grant Cycle*’, (CRGC) for data transmissions, as shown in Figure 3.3:

- 1) Wait until a ‘*Reservation_ID*’ has been received (which allows an NIU to send a reservation request).
- 2) When a packet arrives, send a reservation request message, using the *Reservation_ID* assigned, in a contention-based slot.
- 3) Wait until the INA sends the following MPEG2 frame with signalling information (MCI) and check in the reception indicator field whether the reservation was received successfully.
- 4) If the request resulted in a collision, use the CRA selected (*exponential backoff algorithm* or the *splitting tree algorithm*) to retransmit the reservation request as long as collisions are resolved.

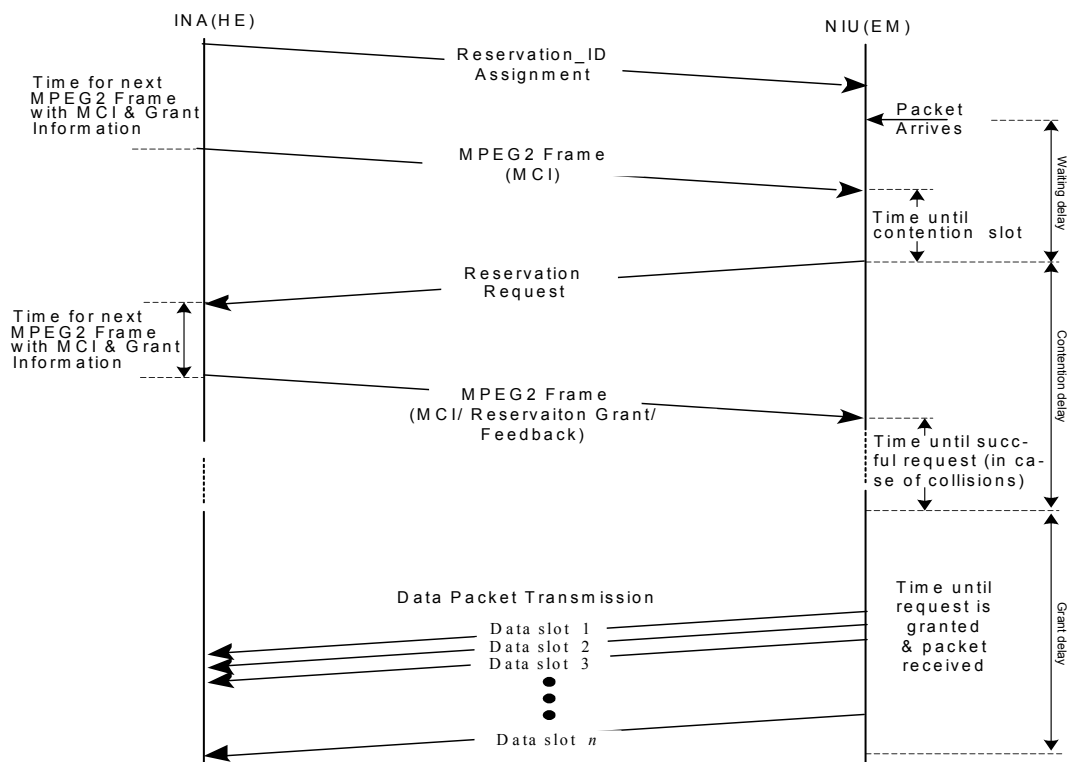


Figure 3.3 – Contention-Resolution-Grant Cycle.

- 5) If the request was transmitted successfully, wait until the INA grants the request to transmit the data packet.
- 6) Finally, after transmitting the current data packet in the reserved slots assigned by the INA, if there are more data packets to transmit repeat the process starting in 1).

In the following sections, a description of the operation and dynamic that each contention resolution algorithm uses to resolve collisions is presented.

3.5.2 Exponential backoff algorithm overview

In this sub-section we present a review of the operation of the *exponential backoff algorithm* adopted by the DVB/DAVIC protocol. Furthermore, in Section 6.3 we approach a performance analysis and optimisation of this algorithm.

When an NIU wants to transmit a reservation request in a contention slot, it chooses arbitrarily one contention slot in the next group of contention slots described by the signalling information field, as depicted in Figure 3.4. In case of collision, this algorithm defines how many cells a station needs to let pass before it can transmit. This number of cells is computed as a uniform random integer variable in the range of $[0-2^{backoff}]$.

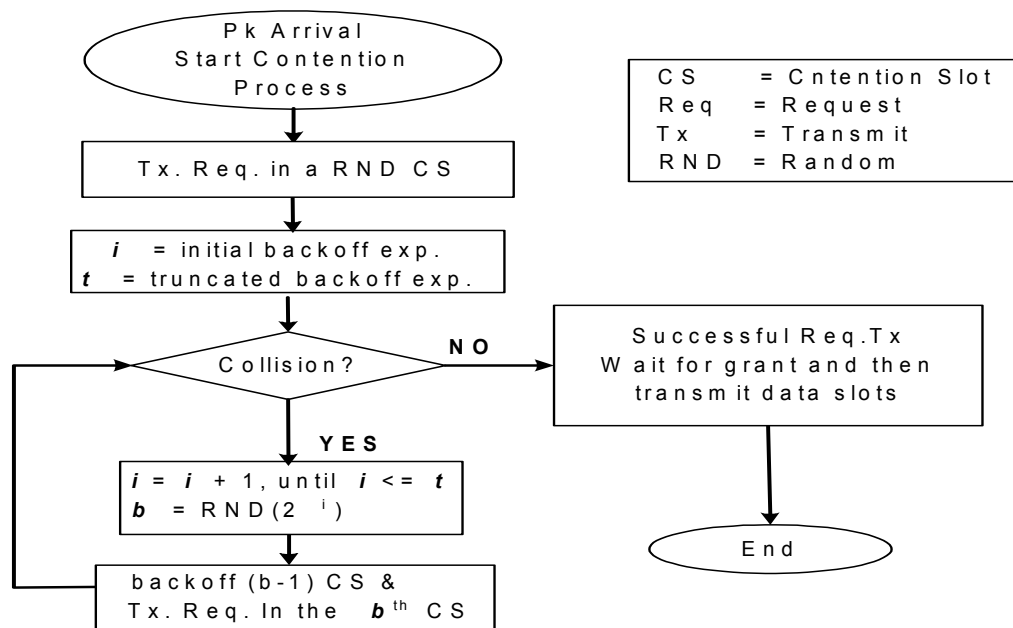


Figure 3.4 – Block diagram of the *exponential backoff algorithm*.

The *backoff* parameter is first initialised with an *initial backoff* exponent (also termed as ‘*Minimum Backoff Exponent*’). This parameter is updated according to the reception indicator received from transmission. The *truncated backoff* parameter (also referred to as ‘*Maximum Backoff Exponent*’) is the maximum allowable *backoff*. If a collision occurs, the *backoff* value is incremented by one. Once the *backoff* value reaches the maximum number determined by the *truncated backoff*, the *backoff* remains at this value regardless of the number of subsequent collisions. After every successful transmission, the *backoff* value is reset to the *initial backoff* value. The values of *initial* and *truncated backoff* are sent by the INA to the stations at the beginning of the connection and remain the same until the NIUs stop the connection with the INA. These two values are equal for all the NIUs as opposed to the *backoff* value that is different at each station.

3.5.3 Splitting tree algorithm overview

As defined in Appendix B.2, minislots are only used to transmit reservation request when this algorithm is used. After an NIU sends a reservation request in a contention-based minislot, it waits for feedback information from the INA. In the case of collision, the NIU enters the contention resolution cycle and activates its resolution state.

The resolution is carried out according to an INA controlled *splitting tree algorithm* as shown in Figure 3.5. All necessary information to resolve collisions is provided in the Reservation Grant Message, which contains minislot feedback (such as ‘Feedback Offset’ and ‘Feedback Collision Number’ 1, 2 and 3) and minislot allocation information (Such as ‘Stack Entry’, ‘Entry Spreading’, ‘Number of Allocations’, ‘Allocation Offset’ and ‘Allocation Collision Number’).

In the current Reservation Grant Message, If ‘Stack Entry’ is not set, an NIU may enter the contention process only when the ‘Allocation Collision Number’ is equal to zero. If ‘Stack Entry’ is set, the NIU may enter the contention resolution in any of the contention-based minislots, independent of the value of ‘Allocation Collision Number’. In both cases the random number for the minislot selection in the range between 0 and ‘Entry Spreading’ should be in the window from 0 to 2 before sending the request.

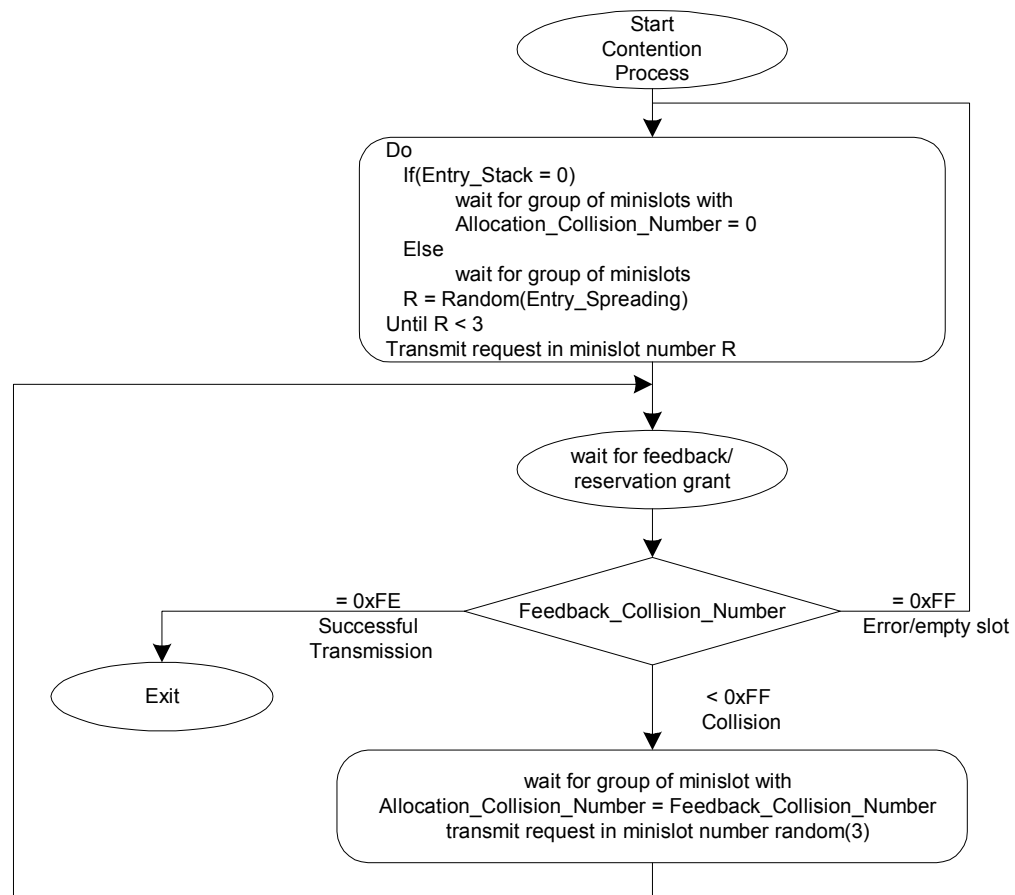


Figure 3.5 – Block diagram of the *splitting tree algorithm*.

The ‘Feedback Collision Number’ is equal to 0xFF or 0xFE for idle and successful transmission, respectively.

All other values of the ‘Collision Number’ are numbered as collisions and used to select the retransmission minislots. The NIU should retransmit in a minislot having an ‘Allocation Collision Number’ equal to ‘Collision Number’. The retransmission of the collided request takes place in a minislot that is randomly selected from among the group of three minislots with the corresponding ‘Allocation Collision Number’

3.5.4 Upstream slot structure

In Section 3.4, we have stated that the upstream channel is divided into discrete basic slots called upstream slots. A fixed number of upstream slots are grouped to form a signalling frame. A generic slot structure is shown in Figure 3.6 (for a 3.088 Mbps upstream channel), where the INA determines the slot structure format (and reservation capacity) by setting the number of ranging, contention, reservation and fixed slots in each signalling frame, an example of which is depicted in Figure 3.6a. They are used for the following purposes:

- Ranging slots: These slots are used in the sign-on, ranging and calibration signalling process to measure and adjust the time delay and the power.

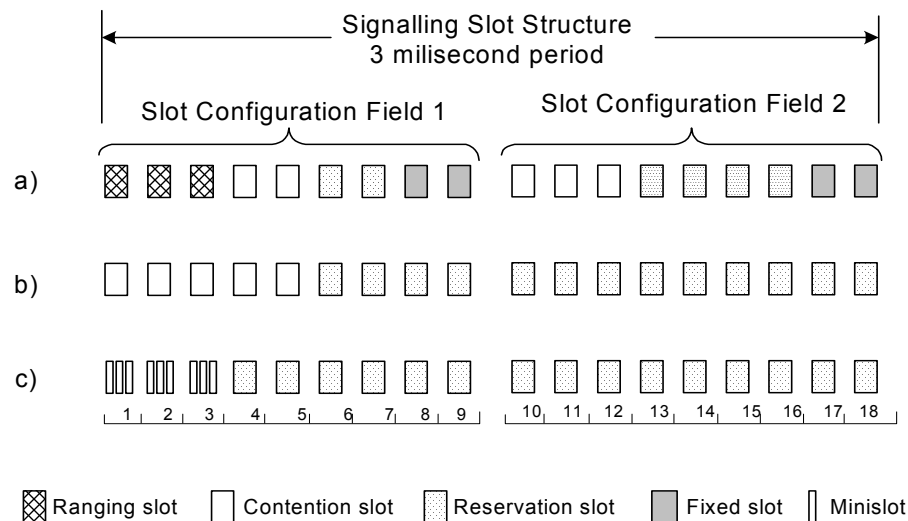


Figure 3.6 – Slot structure.

- Contention slots: are accessible for every station to send reservation requests collisions are possible and solved by a contention resolution algorithm. These slots can also be used to send user data on low traffic loads, if this is enabled by the INA.
- Reservation slots: the user sends control information announcing his demand for transmission capacity. He gets grants for the use of slots.
- Fixed slots: are used for CBR traffic, these slots provides the best results in QoS regarding delays, jitters and loss of upstream data units.

The slot structure is based on two Slot Configuration Fields (SCF). Each SCF describes 9 slots in the order presented in Figure 3.6a (ranging, contention, reservation and fixed slots). The number of ranging slots transmitted during a long period of time (e.g. 60 seconds) is insignificant, compared to the number of contention, reservation and fixed slots transmitted, and the performance of the network does not change if the ranging slots are not used at all. Thus, the use of ranging slots will not be considered in the signalling frame structure for performance analysis.

Furthermore, since fixed slots may or may not be supported in the first version of EuroModem (class A), the performance analysis to be addressed in the following chapters will be based mainly on signalling slot structures that consist of contention and reservation slots, as depicted in Figure 3.6b. In this structure the contention and reservation slots have been rearranged so that all contention slots are described in the first SCF, and in the second SCF when SCF1 describes only contention slots. This new arrangement decreases slightly the processing times of the INA and NIUs.

As pointed out in Section 3.5.3, if the *splitting tree algorithm* is used to resolve collisions, one contention-based slot is divided into three independent minislots that carry shortened request messages, as depicted in Figure 3.6c. When this algorithm is used, the number of contention slots described in the first SCF (or SCFs) can be decreased.

In general, the slot structure presented in Figures 3.6b and 3.6c will be used when the *exponential backoff algorithm* and the *splitting tree algorithm* are selected, respectively.

3.6 Conclusions

An overview of the evolution of the European DVB/DAVIC Cable Communication System has been provided. In the early stages of the cable data networks, the DVB group in collaboration with the European Cable Communication Association brought a significant value for the broadcast data-services organisations by providing the set of specifications for the return channel (ETS 300 800, ETS 200 300, EuroModem, EuroBox) that will enable cable network operators to provide the next generation of interactive services.

In this chapter we have also presented the network architecture and the major constituent components of the DVB/DAVIC protocol, which represent the most important elements that describe this standard. Here the basic operation at the physical layer and the Media Access Control layer were described along with the functionality of the contention resolution algorithms adopted.

The definition of the MAC layer was important, since spectrum of cable networks available for upstream transmissions from subscribers to the headend is scarce. As a final remark, the DVB/DAVIC protocol fits optimally into a cable environment and provides an integrated solution for full-interactive service provision.

Chapter 4

SIMULATION AND ANALYTICAL MODELLING OF DVB/DAVIC

4.1 Introduction

As reviewed earlier in Section 1.2.3.2, the DVB/DAVIC protocol requires approximations of both the *exponential backoff algorithm* and the *splitting tree algorithm*, which cannot be accurately modelled using existing analytical models due to the variable bandwidth assigned to the contention and reservation access regions from (MCI) cycle to cycle. Therefore, the high degree of complexity and the need for accurate results indicate the use of simulation techniques for the performance evaluation and optimisation of the DVB/DAVIC protocol. The use of a simulation model will allow us to examine a wide range of configurations needed for drawing general conclusions, and to test new enhanced mechanisms that would increase the performance of the DVB/DAVIC protocol in a relatively short amount of time.

In this context, the aim of this chapter is to present both the simulation model developed for the DVB/DAVIC protocol and the analytical model formulated for the validation of the simulation results.

4.2 Simulation modelling tools

According to [70] there are three main simulation languages oriented to the modelling of communications networks. The first language is the Block Oriented Networks Simulator (BONeS) DESIGNER, which is a graphically-oriented, general-purpose simulation language that contains many features for modelling communications networks.

The major data blocks are data structures and block diagrams. In order to build a model in BONEs, one first defines a data structure that corresponds roughly to a message and its associated data field. The user then develops a block diagram that describes how the data structures flow through the networks.

The second language is SES/workbench, which is also a graphically-oriented, general-purpose simulation language that contains many features for modelling computer systems and communications networks. The major building blocks are nodes, arcs and transactions. To build a model in workbench, one defines a transaction that corresponds to a message. The user then develops a directed graph consisting of nodes and arcs, which describes how transactions flow through the network.

The last simulation language is the OPTimised Network Engineering Tool (OPNET) Modeller. This software is a communication simulation language specially oriented toward the modelling of communications networks that uses *Network*, *Nodes*, and *Process Editors* to build a simulation model.

From these three simulation languages, OPNET Modeller (v6.0 [79]) has been used as the simulation tool for the performance characterisation and optimisations of the DVB/DAVIC protocol because this simulation package contains an extensive set of features designed to support network modelling and provides an increased flexibility to develop detailed custom models.

4.2.1 OPNET modelling

In order to have an insight of how simulation models are implemented in OPNET, in this section an overview of the hierarchical design used by this simulation package is presented. OPNET models are based on a hierarchical three-level structure as illustrated in Figure 4.1.

- Network domain:** The top-most level is the *Network domain* (see Figure 4.1a). The role of this domain is to define the topology of the communication network, the communication entities called nodes and their interconnection (using bus, point-to-point or radio links). Based on these basic building blocks, more complex models can be developed.
- Node domain:** The next level is called the *Node domain* (Figure 4.1b). This level defines the functionality of each communication device that can be deployed and interconnected to the network, (e.g. routers, bridges, terminals, switches, etc). Each node consists of *traffic source generators*, *processors*, *queues* and various transmitters or receivers allowing a node to be attached to communication links in the network.

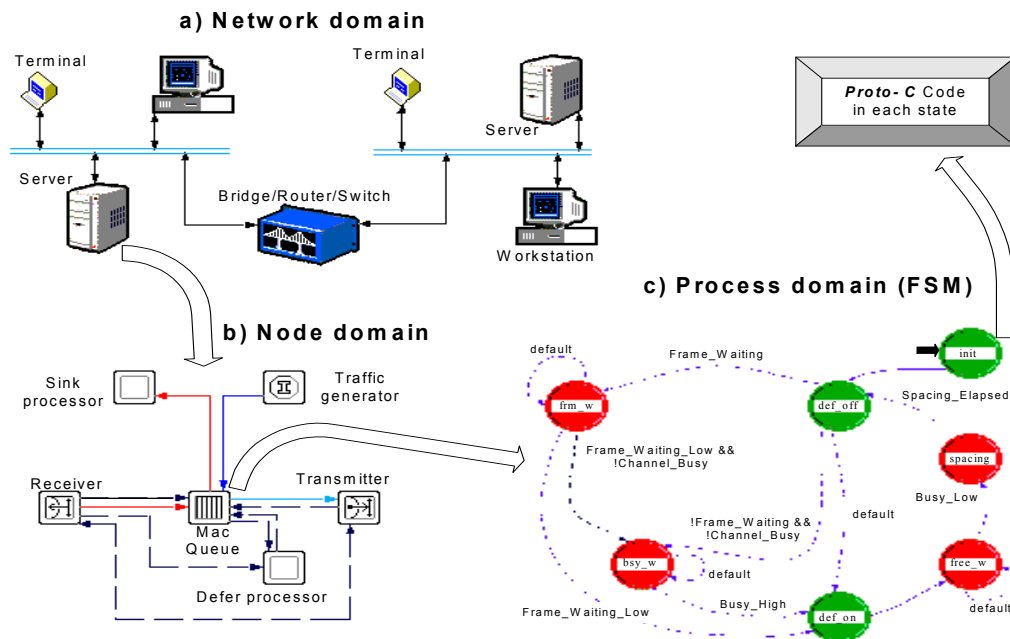


Figure 4.1 – Stage diagram of OPNET simulation models.

- *Traffic generators* are built-in objects used for simulating traffic sources. The generated traffic stream has packet length and a packet inter-arrival distribution, which is selected from a set of supported probability distributions. *Queues* and *Processors* are objects and are highly specifiable. *Queues* are used to simulate buffers of communications links and protocol behaviour. Example statistics that a queue can collect are current/average queue size, queuing delay and queue overflow. Each queue object might spawn multiple sub-queues in which packets can be stored. *Processors* have the capability of packet manipulation but no ability to store packets and can be used as packet sinks or as traffic generators, when the traffic streams cannot be modelled using the predefined probability distribution.
- **Process domain:** The last level is called the *Process domain* (Figure 4.1c). At this level the functionality of each queue or processor object is defined using a Finite State Machine (FSM). States and transitions graphically define the progression of a process in response to events. In general, each FSM can define private state variables and can make calls to code in user provided libraries. FSM are dynamic and can be spawned by other FSMs during simulation in response to specific events. Dynamics of FSM dramatically simplify specification of protocol that manages a scalable number of resources or sessions. Finally, each state of a process model contains ‘Proto-C’ code, supported by an extensive library of functions designed for protocol programming.

4.2.2 Common Simulation Framework (CSF)

Although OPNET models are accompanied by an extensive library of predefined communications protocols covering all seven layers of the Open System Interconnection (OSI) Reference Model, no MAC protocols are available that are suitable for modelling bi-directional CATV networks. Thus, the high demand for evaluating emerging protocols such as the IEEE 802.14 and DOCSIS, have motivated MIL3 (now OPNET Technologies Inc.) and Cablelabs to develop the Common Simulation Framework (CSF) [77], which provides the basic building blocks of a typical CATV network for delivering data applications without defining the functionality for the MAC or layers above.

This framework not only includes statistics collection models and interfacing with other network technologies (e.g. Ethernet, ATM, etc), but also the appropriate links between the Headend (HE) equipment and cable modems for upstream and downstream transmissions. The latest version of the CSF (v.13) covers the basic functionality of the MAC and PHY layers of a DOCSIS v.1.0 compliant cable modem and Cable Modem Terminations System (CMTS). Some additional functions, such as prioritisation [102], committed information rate scheduler algorithm [120] and two schemes to reduce the network's recovery time [105] and [106] were added at the University of Sheffield.

The network architecture of the generic CSF model is presented in Figure 4.2. It consists of the Headend node, Statistics Collection block, Network cloud, and Cable Modems.

- a) **Headend:** The main purpose of this node is to receive messages from the upstream channel and to relay them to either the downstream channel or the Network cloud. A traffic source module within the Headend node models incoming traffic from the Network side interface. Additionally, some other functions performed by this node (at the MAC layer) are CM initialisation, registration, scheduling of upstream transmission and routing of incoming frames to the Statistics Collection module.
- b) **Cable Modem:** This node connects both the upstream and downstream unidirectional channels. With the use of the specific MAC protocol, it generates traffic that is sent to the upstream channel. The CSF provides different types of source traffic, such as ON-OFF exponential distributed bursty traffic, 53-byte isochronous streams and World Wide Web (WWW) traffic flows.

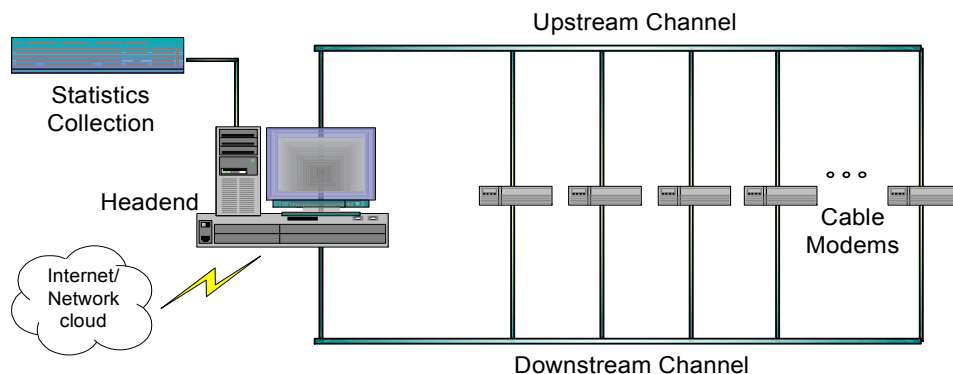


Figure 4.2 – Generic CSF model.

Each CM can have more than one type of traffic source active as well as more than one source of the same type.

- c) **Upstream and Downstream channels:** These channels model the pair of frequencies allocated to a set of CMs for bi-directional communications with each other and the backbone network/internet. The data rates and capacity of these links are parameters defined by the specific protocol and can be either symmetrical or asymmetrical. There are additional link level statistics provided, such as throughput (which includes protocol overhead), utilisation, collision events, multiplicity etc.
- d) **Statistics Collection:** This module interfaces with the HE node and collects frames received from the upstream channel in order to produce performance analysis statistics, such as mean access packet delay, system throughput, jitter, utilisation, global offered load, etc.
- e) **Network Cloud:** This module interfaces with the HE node and is used to model communications and traffic load originating from a backbone, which is not part of the CATV network. The generic CSF does not include any sample architecture for this module.

4.3 DVB/DAVIC simulation model

The simulation model for the DVB/DAVIC protocol is based on the main network topology of the CSF (v.13). All the functionality of the MAC and PHY layers of the DVB/DAVIC protocol (including the INA and NIUs) have been developed and incorporated to the CSF, replacing the functionality of the MAC and PHY layers of the DOCSIS protocol, respectively. Suitable changes were also applied to the Statistics Collection module to support new performance statistics. Furthermore, one traffic source generator was added to this model in order to emulate Internet traffic as proposed by the IEEE 802.14 Working Group [53].

4.3.1 DVB/DAVIC network description

The DVB/DAVIC simulation model uses a similar network topology as proposed by Narayanaswamy [77] for the CSF. This network topology is presented in Figure 4.3a. It consists of two logical buses (upstream and downstream) and Network Interface Units (subscribers) which are connected to the upstream and downstream channels to communicate with the Interactive Network Adaptor (HE). It also contains an INA module, which processes upstream data, evaluates upstream requests and sends data and signalling information to all NIUs listening downstream.

The corresponding representation into an OPNET model is as illustrated in Figure 4.3b. Several network models were created containing different number of nodes (ranging from 6 to 700 nodes), according to the performance analysis addressed.

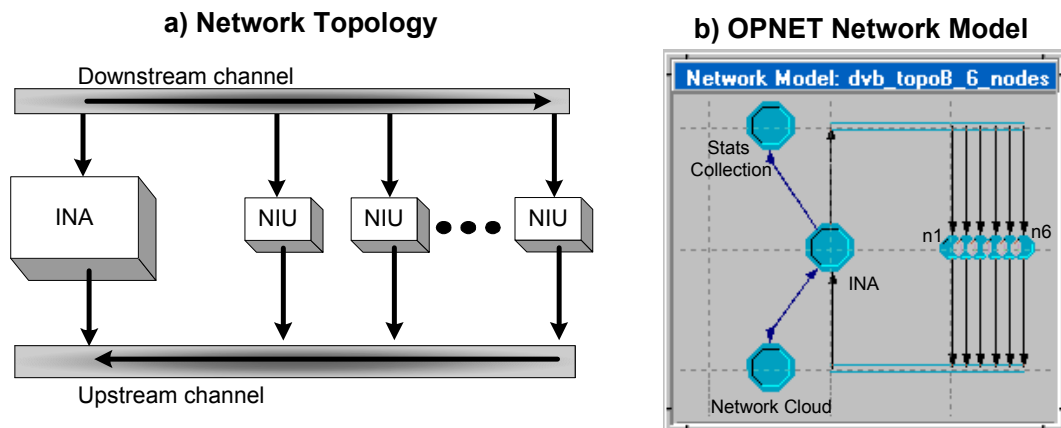


Figure 4.3 – Network topology.

4.3.2 Network Interface Unit design

The interdependent components of the NIU system consist of the traffic sources, a traffic sink (to record statistics), the *NIU MAC* and transceivers, as shown in Figure 4.4a. The traffic sources generate packets that are delivered to the *NIU MAC* using a packet stream connection. This is represented by the *Packet_In* link in the figure. The *NIU MAC* transfers and receives packets from the transmitter/receiver system. These actions are represented by the *Packet_Out* and *Packet_In* connections in Figure 4.4a, respectively. All user data coming from the downstream channel is forwarded to the traffic sink for collecting statistics using the *Packet_Out* connection.

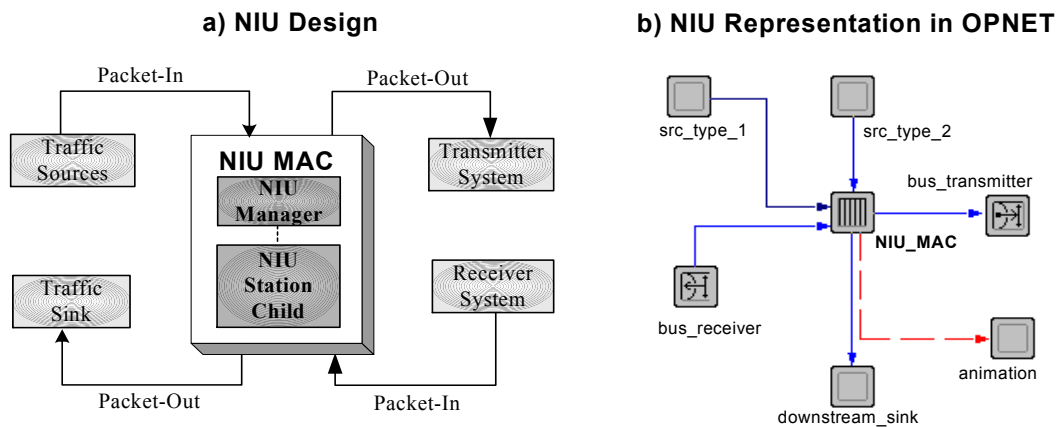


Figure 4.4 – NIU node description.

The NIU representation into a node model in OPNET is as depicted in Figure 4.4b. Two traffic sources were used to emulate Internet traffic and isochronous streams

- **Traffic Source 1:** This traffic source emulates Internet traffic according to a custom Probability Density Function (PDF), as recommended by the IEEE 802.14 Working Group [53]. The distribution message size is shown in Figure 4.5.

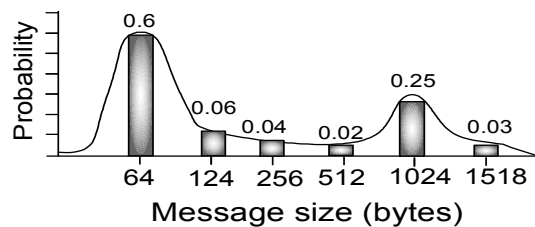


Figure 4.5 – Internet packet distribution.

The average size is ≈ 368 bytes or 8.3 ATM cells and the message inter-arrival time is exponential distributed with mean $T = 1/\lambda$, where λ varies according to the offered load (e.g. 32 or 64 kbps). The minimum packet size of 64 bytes corresponds to an acknowledgement packet that is exchanged during a TCP session and the packet size of 1518 bytes corresponds to the maximum size of an Ethernet packet that is mainly

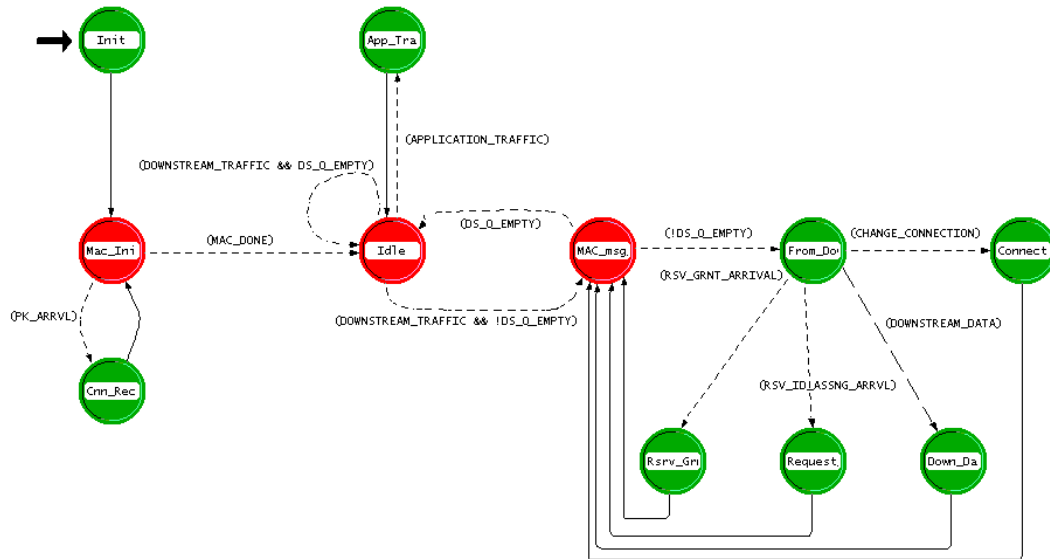


Figure 4.6 – NIU Manager process – FSM.

used in PC Networks.

- Traffic Source 2:** This traffic type emulates a CBR stream. It is a modified implementation of the ON-OFF source provided by the CSF, which is used to generate isochronous streams. All sources create packets of constant length, variable between simulation runs from 20 to 1518 bytes to cover a large range of applications with a constant inter-arrival rate. The time, t , that each source starts generating packets was exponentially distributed with a mean of 1 second. This prevents NIUs from issuing the first reservation request simultaneously and causing an excessive number of collisions at the beginning of the simulation.

The most important block in the NIU node is the *NIU MAC* (Figure 4.4a). This block is based on two asynchronous processes: The *NIU Manager* process handles upstream /downstream traffic and performs the initialisation and registration process depicted in Figure 3.2.

The finite state machine used for the *NIU Manger* process is as illustrated in Figure 4.6. Packets from the traffic source generators and signalling frames that describe the use of the upstream channel are transferred to the *NIU Child* process (see Figure 4.7). The *Child* process performs bandwidth request, collision detection/resolution and packet transmissions.

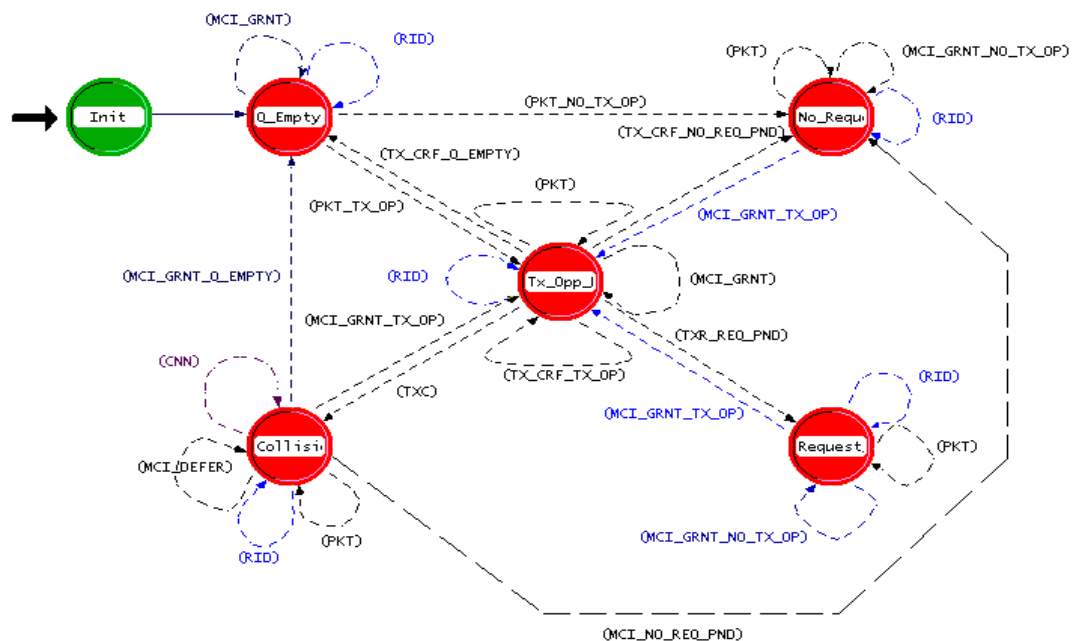


Figure 4.7 – NIU Child process – FSM.

Each state of these process models contains ‘Proto-C’ code, which is supported by an extensive library of functions designed for protocol programming. An example of which is presented in Figure 4.8 for the state “Transmit opportunity”.

```

Microsoft Visual C++ - [Tx_opportunity_proto_C_code.c *]
File Edit View Insert Project Build Tools Window Help
/*****
 * Tx_Opp_Pending (exit). Jan 31, 2000 - UoS * Initial Development
 *****/
int slot_type, num_frames_in_pkt;
handle_event(); //handle the new event: Tx.oppport. data or MPEG pk.
if (event_type==TXR_EVENT|event_type==TXC_EVENT|event_type==TXF_EVENT)
{num_frames_in_pkt = 0;
/*****
 * Transmit depending on upstream slot type on current slot opportunity
 *****/
switch (tx_slot_type)
{case CONTENTION:
  if (niu_use_fxd_slots_for_pdus == 1)
  if (Fixed_Connection_Set == 0)
  tx_rsc_request (1);
  else
  {/*****
   * Get a PDU data message from the current queue
   *****/
   frm_ptr = op_pk_copy(op_subq_pk_access (0, OPC_QPOS_HEAD));
   op_pk_nfd_get(frm_ptr, "Slot_Type", &slot_type);
   op_pk_nfd_get(frm_ptr, "Fragment_Count", &num_frames_in_pkt);
   op_pk_destroy(frm_ptr);
   if (num_frames_in_pkt > Max_Cont_Access_Msg_Length)
   if (Rsv_ID_received==0) tx_rsc_request (0); else tx_rsv_re
  }
  break;
case RESERVATION:
  if (Rsv_ID_received == 1 && grant_tx_opp)
  {if (!rsc_rec_ind_found)
   tx_grant();
   else
   {tx_rsv_request ();
    rsc_rec_ind_found = 0;
    if (op_prg_odh_ltrace_active("mci_info"))
    print_queue_pks();
   }
  }
  else if (!rsc_req_pending && !niu_use_cont_us_for_rsc_req)
  tx_rsv_request ();
  else
  tx_rsc_request (0);
  break;
case FIXED:
  tx_fixed_data ();
  break;
}
}
Ready Ln 2, Col 68 REC COL OVR READ

```

Figure 4.8 – Proto-C code of the state: Tx. Opportunity.

4.3.3 Interactive Network Adaptor design

The Interactive Network Adaptor (INA) node consists of the *INA MAC* and four link access points to interact to the CATV network, the Network Cloud and Statistics Collection node as pointed out in Figure 4.9a.

The representation of the INA into an OPNET node is shown in Figure 4.9b. Here the *INA MAC* was connected to the downstream bus access point via a statistical wire, thus preventing the *INA MAC* sending another frame to the bus. Currently, the traffic generator (Downstream source) has been disabled, thus no data packet would be transmitted downstream, since the main research focuses on the performance analysis of the upstream channel, which is the only critical network resource because of the reservation access mechanism and its marked asymmetry. In addition, the upstream scheduler is part of the *INA Manager* process.

The *INA MAC* node is also modelled as two asynchronous processes, the *INA Manager* and the *INA Transmitter (TX)* process. The *INA Manager* process handles the MAC Initialisation (which consists of initialisation, provisioning, ranging, calibration and connection signalling), receipt of upstream packets and downstream traffic. The *INA Tx Controller* handles transmission of data and MAC signalling messages generated by the *INA Manager* process.

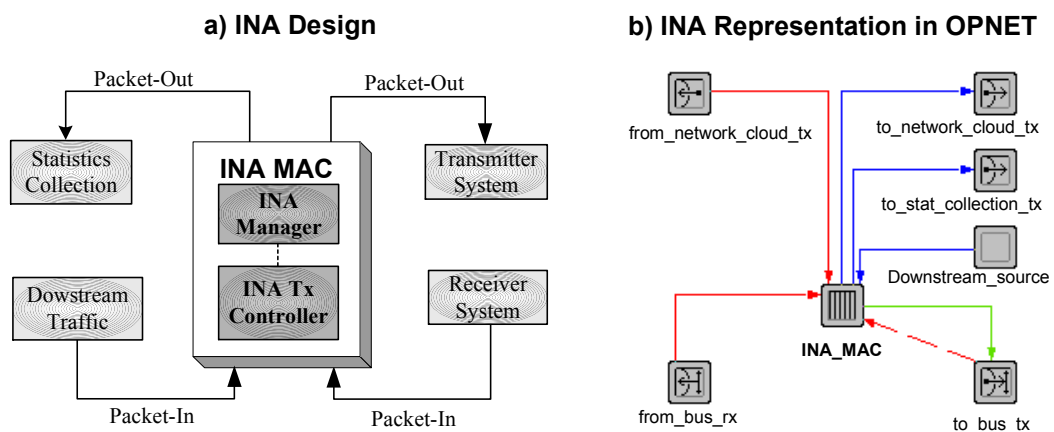


Figure 4.9 – INA node description.

The finite state machine of the *NIU Manager* process is depicted in Figure 4.10. The operation of this state machine is as follows. Upon simulation start the state variables are initialised. The MAC Initialisation procedure makes sure that an NIU can synchronise with the INA and that it can receive the basic network parameters. When all NIUs have established a connection with the INA, an initial frame containing signalling information is sent to the cable network.

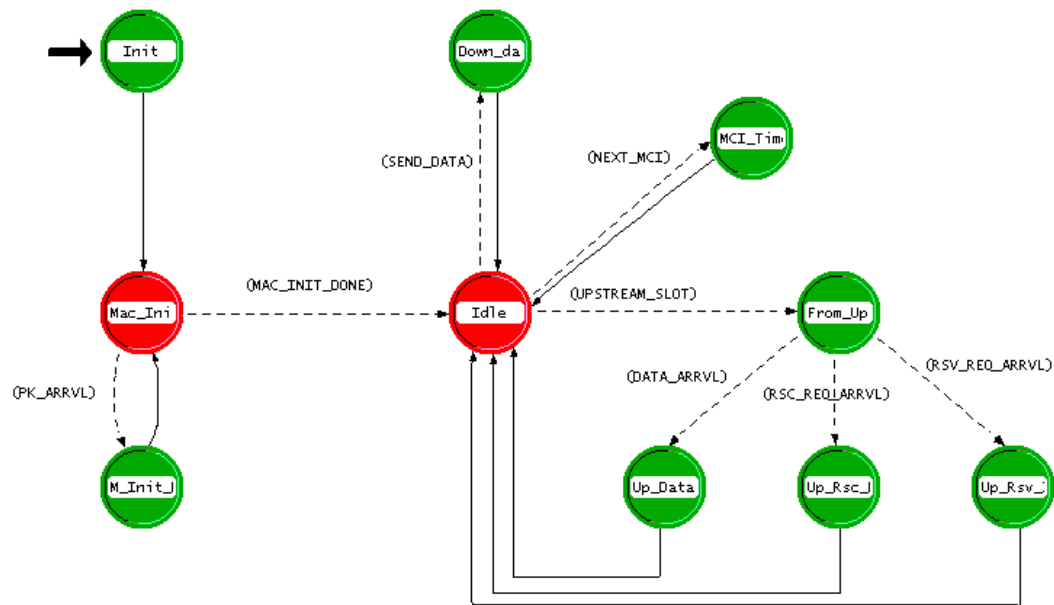


Figure 4.10 – INA Manager process – FSM.

The state machine then switches from the idle state to one of the five other possible states (that were considered for this study), representing the following events: 1) data packet arrival from the upstream channel, 2) resource request message arrival from the upstream channel, 3) reservation request message arrival from the upstream channel, 4) MCI timer expired, and, 5) data packet arrival from the traffic generator. For each of these events, the appropriate MAC message or data packet is created and passed to the *INA TX Controller* process for further transmission on the downstream channel.

The *INA TX Controller* is a spawned state machine (see Figure 4.11) or child process to the INA Manager. Its purpose is to receive either MAC messages or data messages from its parent and transmit them on the downstream channel as MPEG2 frames. The *INA TX Controller* process interacts with the bus object (*to_bus_tx*) via a statistical wire that informs when the bus is free (Figure 4.9b). When the downstream bus is idle and there are MAC messages or data packets to transmit, the *INA TX Controller* serves first the MAC messages in a First In First Out (FIFO) order, and assembles up to three MAC messages into an MPEG2 frame, which is then delivered to the downstream channel.

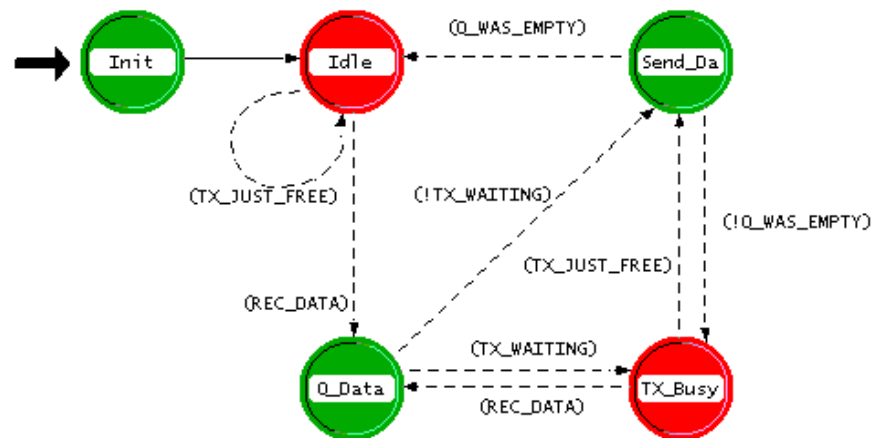


Figure 4.11 – INA TX Controller - FSM.

4.4 Analytical modelling

The analytical evaluation of CATV protocols, and by implication DVB/DAVIC, proves very difficult due to the increased complexity involved in such protocols, as reviewed in Section 1.2.3.2. The downstream channel, which is a unidirectional broadcast channel used only by the INA, can be easily modelled as a simple delay queue with a constant service rate. However, the modelling of the upstream channel is complicated due to hybrid multiple random access-reservation MAC.

A DVB/DAVIC upstream channel can be modelled as a triplet of virtual channels as depicted in Figure 4.12. A portion of the upstream channel is dedicated to reservation and fixed-rate access in the reservation channel (**R**-channel) and the fixed channel (**F**-channel) respectively, while reservation requests are placed in the contention channel (**C**-channel). There is a strong correlation between the traffic patterns in the three virtual channels. This is because traffic in the **C**-channel accounts for reservation/resource request, which if successful, trigger transmissions in the **R** or **F** channels.

According to [10] it is possible to derive simple analytical models for such a system using the **C**-channel and considering the **R** and **F** channels as a single virtual channel. The throughput derived (S) for the generalised multi-access reservation system, in accordance with [10] results in:

$$S = \frac{1}{1 + \frac{v}{S_c}} \quad (4.1)$$

where v is the No. of time units required to transmit one request when a data packet requires one time unit, and S_c is the normalised throughput of the **C**-channel. This method requires that the throughput of the **C**-channel can be calculated.

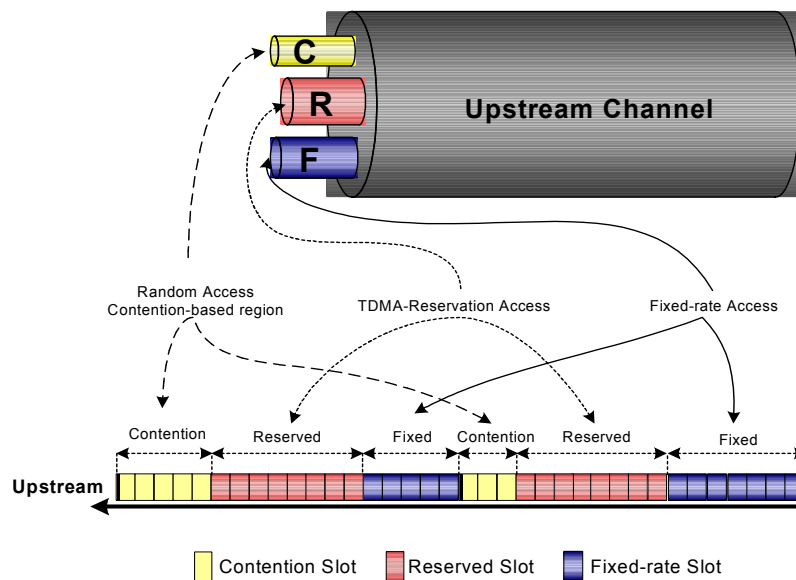


Figure 4.12 – Upstream channel model with virtual contention reservation and fixed-rate channels.

For certain shared access methods such as the Slotted Aloha, splitting tree or TDMA the maximum normalised throughput is $1/e$. 0.478 and 1 respectively [10].

At the time of writing this dissertation no accurate analytical model for CATV protocols, which deal with variable reservation access (queues with contention), had been proposed. When modelling the hybrid DVB/DAVIC protocol, two areas require special consideration:

- Analysis of the contention resolution algorithm. We have focused here on the stability of contention resolution algorithms and the time to resolve collisions.
- Variable bandwidth of the **C**-channel. This effect is a direct consequence on the above feature and the CRA adopted. Depending on the offered load,

$$c \in \left[\frac{MCI_{CSs}}{MCI_{max_slots}} \cdot CC, CC \right] \quad (4.2)$$

where c is the bandwidth of the **C**-channel, MCI_{CSs} is the minimum number of contention slots described per MCI frame, and MCI_{max_slots} is the maximum number of slots described in each MCI frame. The upper bound is realised when the cable network is idle and every MCI frame consists only of contention slots, thus $c = CC$. The lower bound is realised when the cable network is in congestion and the bandwidth of the **C**-channel becomes the *minimum number of contention slots per MCI frame* times the number of MCI frames per second. Thus, the bandwidth of the **R** and **F** channels becomes,

$$r + f = MCI_{max_slots} - MCI_{CSs} \quad (4.3)$$

By implication, the bandwidth of the **R** and **F** channels varies and is correlated to c . This variation is a function of the offered load and the efficiency of the contention resolution algorithm, which specifies the number of successful transmissions in the **C**-channel. Therefore, even if the **C**-Channel was approximated by the Slotted Aloha or TDM access algorithm, the throughput calculation proves problematic, as the channel bandwidth is variable.

Considering the above limitation, the formulation of an accurate analytical model becomes highly complex. Despite the high level of complexity associated with the production of an analytical model, a form of validation process is essential, even of the basic functionality of the DVB/DAVIC protocol in order to verify correctness of the simulation model.

The analytical model formalised in Sections 4.4.1 to 4.4.3, address some questions that concern performance evaluation of a new protocol, (i.e. the knowledge of packet access delays and maximum system throughput achieved per stations). This is one of the first issues that participant vendors and cable network manufacturers have to address, as there is a significant delay associated with transmissions in the upstream channel that has to do with the *CRGC*, even where there are not contending users in the cable network. The analytical model presented in Section 4.4.1 and Section 4.4.2 is based in the following reasonable assumptions:

- The bandwidth of the **F**-channel for fixed-rate access is not considered. This is an advanced functionality included in the second version of EuroModems [36].
- In accordance with the DVB/PDAVIC protocol, at least one MCI frame should be transmitted in the 3 ms period. We consider that the DVB/DAVIC transmits 4 MCI frames per 3 ms period. Thus the number of slots described per MCI frames is of 9 slots, with a 6.17 Mbps upstream channel.

The following formulation addresses issues of mean packet access delay and maximum system throughput achieved for a single upstream channel. The quantities and abbreviations that will be used throughout the analysis have been defined in Table 4.1.

Table 4.1 – Symbols and abbreviations.

Symbol	Description	Value or range if applicable
λ	Packet arrival rate in Pk/s (also referred to as offered load in kbps)	
μ	Average number of packets transmitted per second	-
\bar{X}	Mean packet service time	-
ρ	Utilisation factor	-
X_{idle}	Service time when the system is idle	-
X_{busy}	Service time when the system is busy	-
$\sigma_{\bar{x}}^2$	Variance of the mean service time	0
L	Mean number of packets in the system (including the packet in service)	-
Pk_{size}	Length of the packet to be transmitted in bytes	[64 – 1518]
Pk_{slot}	Number of slots required to transmit a packet	-
Pk_{mci}	Number of MCI frames required to transmit a data packet	
$AAL5_{header}$	Protocol overhead added because of the ATM-AAL5 encapsulation	8 bytes
$AAL5_{PDU}$	Length of the ATM-AAL5 Packet Data Unit	48 bytes
\bar{W}	Mean average waiting time	-
\bar{D}_{ete}	Mean access end-to-end delay from the NIU to the HE	-
MCI_{pk}	Number of MCI frames required to transmit a packet	-
MCI_{max_slots}	Maximum number of slots described in each MCI frame	9 slots
MCI_{CSs}	Minimum number of contention slots described per MCI frame	1 slot
MCI_t	MAC Control Information interval	0.750 ms
D_{prop}	Propagation delay from the NIU to the HE	0.051 ms
D_{sl_tx}	Slot transmission delay	0.083 ms
CC	Upstream channel capacity in Mbps	6.176
P_0	Probability that the system is in an idle state	
P_1	Probability that the system is in a busy state	
\bar{T}_{ip}	Average time of the idle period	
\bar{T}_{pb}	Average time of the busy period	
\bar{T}_{bc}	Average time of the busy cycle	
\bar{L}_{bc}	Average packets in queue per busy cycle	
D_{lah}	Scheduler look-ahead delay	
D_{tx_MPEG}	Downstream transmission delay of an MPEG2 TS frame	
$D_{interleave}$	Delay incurred to enable the correction of burst noise induced errors.	
D_{NIU_proc}	NIU processing delay	
DS_size	Total size of the MPEG2 TS frame to be transmitted downstream including FEC information.	204 bytes
$D_{ws_bit_rate}$	Downstream data rate	42 Mbps
L_x	Average number of packets served per second	
S	Throughput	
$S_{max_large_packets}$	Maximum system throughput for large packet sizes	
$S_{max_exp_backoff}$	Max. throughput for different packet sizes using the <i>exp. backoff algorithm</i>	
$S_{max_split_tree}$	Max. throughput for different packet sizes using the <i>splitting tree algorithm</i>	

4.4.1 Mean access delay formulation for a single node scenario

For the upstream direction, the *Reservation Access* mechanism of the DVB/DAVIC MAC protocol can be modelled as a queuing system. The upstream timing representation, depicted in Figure 4.13, permits a graphical view of the dynamic of our queuing system. This particular figure is shown for a first-come-first-serve order service. In this timing diagram the horizontal line just below the upstream channel represents the queue (at the NIU) and the horizontal line above the channel represents the service facility. An arrow approaching the queue from below indicates that an arrival of a data packet has occurred (represented by A_{pk}). Conversely, arrows emanating from the server indicate the departure of a data packet from the queue (D_{pk}).

In this queuing system there are mainly two types of delays involved in the transmission of a data packet, namely waiting delay in queue (W) and service delay (X). Additionally, when a packet departs from the server, there is a constant delay involved, which is caused by the transmission delay and the propagation delay before the data packet reaches its final destination. Here, only the transmission delay of the last slot of the current data packet is only considered. The transmission delay of the other data slots form part of the service delay. Hence, the general equation to calculate the total mean access end-to-end delay, \bar{D}_{ete} , is given by:

$$\bar{D}_{ete} = \bar{X} + \bar{W} + D_{Sl_Tx} + D_{prop} \quad (4.4)$$

where \bar{X} and \bar{W} are the mean of service and waiting delay, respectively.

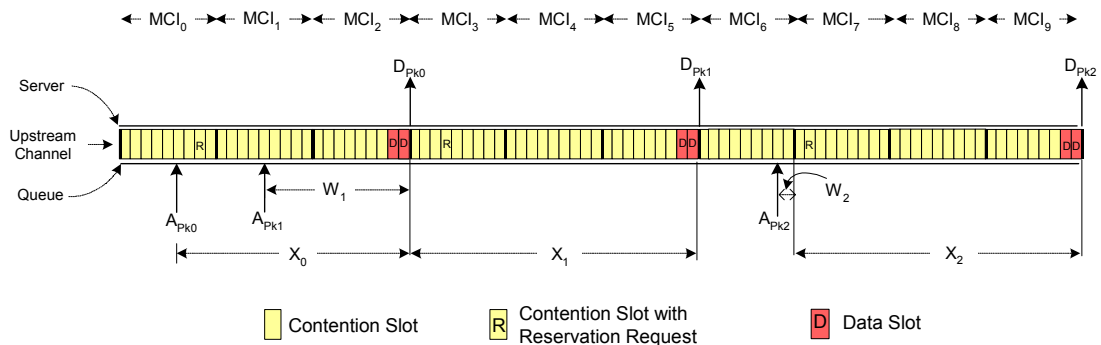


Figure 4.13 – Upstream timing diagram.

The transmission delay of a data slot and the propagation delay are given by Equations 4.5 and 4.6, respectively.

$$D_{sl_Tx} = \frac{512}{CC} = 83\mu s \quad (4.5)$$

$$D_{prop} = \frac{l}{0.67 \cdot c} = 50\mu s \quad (4.6)$$

In Equation 4.5, CC corresponds to the upstream channel capacity set to 6.176 Mbps and l in Equation 4.6 corresponds to the maximum length of the HCF network, from the INA to the further NIU. For this analysis l was set to 10Km and c is the speed of light constant ($\approx 3 \times 10^8$ m/s).

The solution of this queuing system depends on the arrival and service distribution of the data packets. Assuming that the data packets follow a Poisson distribution (e.g exponential distributed inter-arrival times) and by obtaining the mean service time (\bar{X}) of Figure 4.13, we can approach the solution by using an M/G/1 queuing system. Such system is characterised by a Poisson arrival process (at a mean rate of λ arrivals per second with a mean inter-arrival rate of $1/\lambda$) and with a general service time distribution (with a mean service time of \bar{X} seconds).

According to [45], a well-known result for the M/G/1 system is given by the Pollaczek-Khintchine (PK) formula presented by Equation 4.7.

$$L = \rho + \frac{\rho^2 + \lambda^2 \cdot \sigma_{\bar{X}}^2}{2 \cdot (1 - \rho)} \quad (4.7)$$

This formula gives the average number of packets in the system L , where ρ is the utilisation factor given by $\rho = \frac{\lambda}{\mu} = \lambda \cdot \bar{X}$, $\rho < 1$, and $\sigma_{\bar{X}}^2$ is the variance of the service-time distribution. From this formula the expected waiting time \bar{W} in queue can be obtained by using the well-known Little's theorem, $L = \lambda \cdot T$ [10], where T is the mean waiting time in the system given by:

$$T = \bar{X} + \bar{W} \quad (4.8)$$

Thus, the waiting time in the queue results in:

$$\overline{W} = \frac{\rho + \frac{\rho^2 + \lambda^2 \cdot \sigma_{\overline{X}}^2}{2 \cdot (1 - \rho)}}{\lambda} - \overline{X} = \frac{\rho^2 + \lambda^2 \cdot \sigma_{\overline{X}}^2}{2 \cdot \lambda \cdot (1 - \rho)} \quad (4.9)$$

Then, by substituting the utilisation factor $\rho = \lambda \cdot \overline{X}$ in Equation 4.9, we can obtain the formula for the mean waiting time in the system, as:

$$\overline{W} = \frac{\lambda \cdot \overline{X}^2 + \lambda \cdot \sigma_{\overline{X}}^2}{2 \cdot (1 - \rho)} \quad (4.10)$$

The mean and the variance of the service time (\overline{X} and $\sigma_{\overline{X}}^2$ respectively) are needed also. From analysing Figure 4.13, we can see that the mean service time has three values, represented by X_0 , X_1 and X_2 . A service time of X_0 is given to packets that find the system idle. In other words, the packets start being processed immediately and do not have to wait in queue to be served. A service time of X_1 is given to packets that find the system busy and have to wait in queue before they are processed.

The service time X_2 ($=X_1$) is for special cases when packets find the system idle and have to wait for a very short period of time before being processed. This is because such packets arrive at the end of the current MCI frame and the DVB/DAVIC reservation mechanism is unable to start service immediately, due to the time required for the propagation and the slot transmission delay. In other words, when a packet arrives within the last two contention slots of the current MCI frame, in order to start service immediately (by sending or scheduling a request) the following condition should be satisfied,

$$Current_time + D_{Sl_Tx} + D_{prop} \leq CSn_{Tx_time} \quad (4.11)$$

where $Current_time$ is the time when the packet arrives and CSn_{Tx_time} is the time at which the n^{th} contention starts. If this condition is not satisfied, the DVB/DAVIC reservation mechanism retries with the contention slots of the following MCI frame and then selects randomly one contention slot from the set of contention slots that satisfy Equation 4.11.

First we analyse the case when the service times of the data packets are either X_0 or X_1 , and consider for the moment that packets that arrive at the end of the MCI cycle are given a service time of X_0 (instead of X_1). Once we have derived a formula for the mean service time, we can easily obtain a similar equation, but now taking into account the third case of service time $X_2 (=X_1)$.

The initial mean service time, let's say \bar{X}' , is given by the probability that a data packet finds the system idle, P_0' , multiplied by the service time when the system is idle, $X_{idle} = X_0$, plus the probability that a data packet finds the system busy, P_1' , multiplied by $X_{busy} = X_1$. Thus the initial service time is given by:

$$\bar{X}' = P_0' \cdot X_{idle} + P_1' \cdot X_{busy} \quad (4.12)$$

where the probability of finding the system busy or idle is given by Equations 4.13 and 4.14, respectively.

$$P_1' = \frac{\lambda}{\mu} \quad (4.13)$$

$$P_0' = 1 - P_1' \quad (4.14)$$

The service time when the system is in idle state can be obtained by making a closer analysis in Figure 4.13, as depicted in Figure 4.14. Here X_{idle} consists of three different delays, as indicated by Equation 4.15.

$$X_{idle} = X_{t1} + X_{t2} + X_{t3} \quad (4.15)$$

The first delay, X_{t1} , is a variable delay but can be approached by using the mean inter-arrival time in the range $[MCI_{0_start} - MCI_{0_end}]$ given by Equation 4.16.

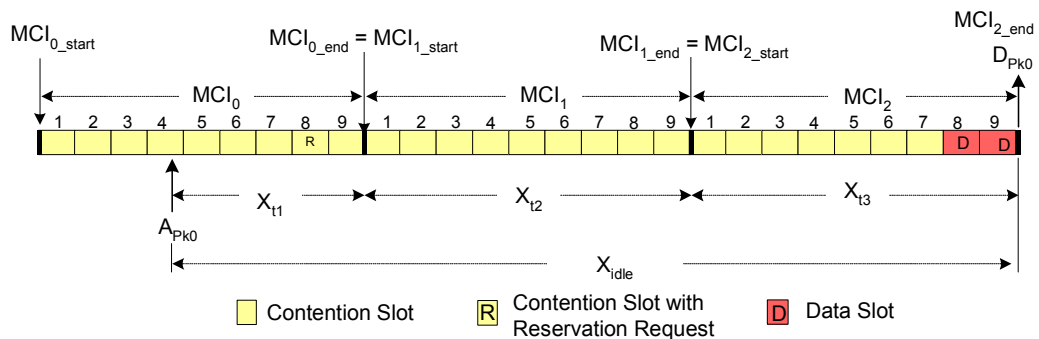


Figure 4.14 – Mean service time when the system is in idle state.

$$X_{t1} = \frac{MCI_{0_end} - MCI_{0_start}}{2} = \frac{MCI_t}{2} \quad (4.16)$$

where MCI_t is the duration of a complete MCI frame (= 0.75 ms) that describes 9 contention slots.

The second delay, X_{t2} , is referred to as ‘scheduler look-ahead’ time. The number of MCI frames, N_{MCI} , that the INA scheduler will have transmitted between subsequent grants is directly related to the time that an MCI frame is scheduled before the current MCI frame ends. This delay is given by:

$$X_{t2} = N_{MCI} \cdot MCI_t \quad (4.17)$$

As illustrated in Figure 4.15, the INA scheduler does not wait until the current MCI frame expires before it transmits the next one. This is to be expected since there is a delay associated with the transmission and propagation delay of the MCI frame, as well as a processing delay for parsing the MCI frame in each NIU. Consequently, the INA must make sure that the next MCI frame is sent soon enough in order to reach the NIUs, and that it is processed before the time described by the previous MCI frame elapses.

In order to achieve this functionality the scheduler needs to take into account the following possible delays that an MCI frame will experience in the worst case scenario:

- Downstream transmission delay of an MPEG2 data frame (D_{tx_MPEG}). This delay is incurred if a data message has just started being transmitted before the MCI frame is scheduled and cannot be cancelled. For a 42 Mbps downstream channel $D_{tx_MPEG} =$

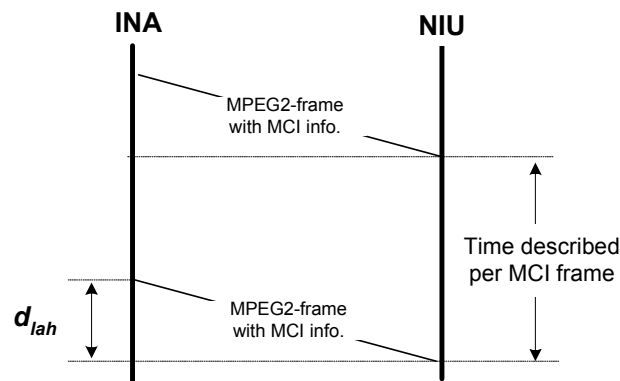


Figure 4.15 – Scheduler look ahead delay.

$$204 \cdot 8 / 42 \text{M} = 38.9 \mu\text{s}.$$

- Transmission delay of an MPEG2 frame containing an MCI frame (D_{Tx_MPEG}). This allows the scheduler to take into account transmission delay of the forthcoming MCI frame.
- Interleave delay ($D_{interleave}$). This delay is incurred to enable the correction of burst noise induced errors, protecting against a burst of symbols errors from being sent to the RS decoder.
- Roundtrip propagation delay ($2 \cdot D_{prop}$). This is the delay incurred to transmit a single bit of information from the INA to the furthest NIU and vice-versa.
- NIU processing delay (D_{NIU_proc}). This is the delay that it takes to the NIU to parse the MCI frame and translate it into transmission opportunities.

Based on the above delays, the *scheduler look-ahead* delay, D_{lah} , is given by:

$$D_{lah} = 2 \cdot D_{Tx_MPEG} + D_{interleave} + 2 \cdot D_{prop} + D_{NIU_proc} \quad (4.18)$$

According to [34], the interleave delay is given by the following equation,

$$D_{interleave} = \frac{DS_size \cdot 8 \cdot (interleave_depth - 1)}{D_{ws_bit_rate}} \quad (4.19)$$

where DS_size (=204 bytes) is the total size of the MPEG-2 TS frame to be transmitted downstream including FEC information and the $interleave_depth$ is a constant set to 12 for downstream IB modulation. $D_{ws_bit_rate}$ is the downstream data rate set to 42 Mbps. With these values indicated above, the interleave delay results in 427 μs .

In order to get the complete *scheduler look-ahead* time (given by Equation 4.18), the processing time of the MCI frame at the NIU (D_{NIU_proc}) is needed. Such delay is hardware specific, but according to the specification [34], this value should be under 500 ms. We first consider a short processing delay. Assuming that this delay is about 100 ms, then the *scheduler look-ahead* delay becomes:

$$D_{lah} = 2 \cdot 38.9 + 427 + 2 \cdot 50 + 100 \approx 705 \mu\text{s}$$

The number of MCI frames, N_{MCI} (needed in Equation 4.17) that would be scheduled before the current MCI frame expires is then calculated as D_{lah} over the length of the subsequent MCI frames (MCI_t). Thus:

$$N_{MCI} = \begin{cases} \frac{D_{lah}}{MCI_t} & \text{when } D_{lah} \bmod MCI_t = 0 \\ \frac{D_{lah}}{MCI_t} + 1 & \text{otherwise} \end{cases} \quad (4.20)$$

For this analysis (when $D_{NIU_proc} \approx 100$ ms), the *scheduler look-ahead* delay results in $D_{lah} = 705$ μ s, hence $N_{MCI} = 705 \cdot 10^{-6} / 750 \cdot 10^{-6} \approx 1$. Consequently, the second component of delay of the service time in idle state results in $X_{t2} = 1 \cdot MCI_t$.

On the other hand, if we consider a large processing delay (e.g $D_{NIU_proc} \approx 500$ ms), then the second component of delay of the service time becomes:

$$X_{t2} = 1105 \cdot 10^{-6} / 750 \cdot 10^{-6} \approx 2 \cdot MCI_t$$

In order to get a complete formula for X_{idle} we now need to calculate the third component of delay, X_{t3} , which is related to the number of MCI frames required to transmit a data packet (Pk_{mci}) and is given by:

$$X_{t3} = Pk_{mci} \cdot MCI_t \quad (4.21)$$

where Pk_{mci} can be calculated by using the following equations:

$$Pk_{slot} = \begin{cases} \frac{Pk_{size} + AAL5_{header}}{AAL5_{PDU}} & \text{when } Pk_{size} + AAL5_{header} \bmod AAL5_{PDU} = 0 \\ \frac{Pk_{size} + AAL5_{header}}{AAL5_{PDU}} + 1 & \text{otherwise} \end{cases} \quad (4.22)$$

In this equation, Pk_{slots} gives the number of upstream slots required to transmit a data packet, Pk_{size} is the length of the packet size in bytes to be transmitted, $AAL5_{header}$ (= 8 bytes) corresponds to the protocol overhead caused by the encapsulation of PDU messages into ATM cells, $AAL5_{PDU}$ (= 48 bytes) is the payload of an ATM cell.

Then, the number of MCI frames required to transmit a data packet is Pk_{slots} over the maximum number of reservation slots per MCI frame. Thus:

$$Pk_{mci} = \begin{cases} \frac{Pk_{slot}}{MCI_{max_slots} - MCI_{CSs}} & \text{when } Pk_{slot} \bmod MCI_{max_slots} - MCI_{CSs} = 0 \\ \frac{Pk_{slot}}{MCI_{max_slots} - MCI_{CSs}} + 1 & \text{otherwise} \end{cases} \quad (4.23)$$

where MCI_{max_slots} and MCI_{CSs} correspond to the maximum number of upstream slots described per MCI frame and the minimum number of contention slots per MCI frame, respectively.

By substituting X_{t1} , X_{t2} and X_{t3} from Equations 4.16, 4.17 and 4.21 respectively in Equation 4.15, the service time in an idle state now becomes:

$$X_{idle} = \frac{MCI_t}{2} + N_{MCI} \cdot MCI_t + Pk_{mci} \cdot MCI_t = (0.5 + N_{MCI} + Pk_{mci}) \cdot MCI_t \quad (4.24)$$

The service time in a busy state (X_{busy}) is similar to X_{idle} , but the only difference is that new packets, finding the system busy, spend a complete MCI frame looking for a contention slot in which to place a reservation request instead of $MCI_t/2$. Thus, the calculation for X_{busy} is given by:

$$X_{busy} = MCI_t + X_{t2} + X_{t3} = (1 + N_{MCI} + Pk_{mci}) \cdot MCI_t \quad (4.25)$$

Up to this point, we have calculated all the elements of the initial mean service time. Thus by substituting Equations 4.13, 4.14, 4.24 and 4.25 in Equation 4.12, \bar{X}' is then calculated as:

$$\bar{X}' = \left(1 - \frac{\lambda}{\mu}\right) \cdot (0.5 + N_{MCI} + Pk_{mci}) \cdot MCI_t + \frac{\lambda}{\mu} \cdot (1 + N_{MCI} + Pk_{mci}) \cdot MCI_t$$

Hence,

$$\bar{X}' = \left(Pk_{mci} + 0.5 + N_{MCI} + \frac{\lambda}{2 \cdot \mu} \right) \cdot MCI_t \quad (4.26)$$

We can now derive the real mean service time (\bar{X}) when the third case of service time (X_2) is considered. Since not all newcomer packets finding the system in idle state are

given a service time of X_{idle} , the probability P_0 that a service time of X_{idle} is actually given, needs to be calculated. This probability is obtained by the following equation:

$$P_0 = P_0' \cdot \frac{(MCI_t - D_{Sl_Tx} - D_{prop})}{MCI_t} \quad (4.27)$$

Thus, the probability P_1 that a data packet is given a service time of X_{busy} is then

$$P_1 = 1 - P_0 \quad (4.28)$$

Therefore, the real mean service time results in:

$$\begin{aligned} \bar{X} = P_0 X_{idle} + P_1 X_{busy} &= (1 - \frac{\lambda}{\mu}) \cdot \left(\frac{MCI_t - D_{Sl_Tx} - D_{prop}}{MCI_t} \right) \cdot (0.5 + N_{MCI} + Pk_{mci}) \cdot MCI_t + \\ &+ \left(1 - (1 - \frac{\lambda}{\mu}) \cdot \left(\frac{MCI_t - D_{Sl_Tx} - D_{prop}}{MCI_t} \right) \right) \cdot (1 + N_{MCI} + Pk_{mci}) \cdot MCI_t \end{aligned}$$

Hence,

$$\bar{X} = (D_{Sl_Tx} + D_{prop}) \cdot (0.5 - \frac{\lambda}{2\mu}) + (0.5 + N_{MCI} + Pk_{mci} + \frac{\lambda}{2\mu}) \cdot MCI_t \quad (4.29)$$

Once we have calculated the mean service time (\bar{X}), we can now obtain the variance (σ_X^2), needed in Equation 4.10. It is well-defined that the variance is the average squared deviation from the mean, given by the following formula:

$$\sigma_X^2 = \frac{\sum_{j=1}^N (X_j - \bar{X})^2}{N} \quad (4.30)$$

If we consider that all packets receive a service time either of X_{idle} or X_{busy} , then the variance can be re-calculated as:

$$\sigma_X^2 = \rho \cdot (X_{busy} - \bar{X})^2 + (1 - \rho) \cdot (X_{idle} - \bar{X})^2 \quad (4.31)$$

In this new equation, the variance only depends on the value of the utilisation factor ($\rho = \frac{\lambda}{\mu} < 1$). Thus, the variance follows the distribution depicted in Figure 4.16.

From this figure we can appreciate that the highest value for the variance is when $\rho = 0.5$, which results in $\sigma_{\bar{X}}^2 = 0.000000036 \approx 0$. By substituting $\sigma_{\bar{X}}^2 = 0$ in Equation 4.10, the mean waiting time in queue is then given by:

$$\bar{W} = \frac{\lambda \cdot \bar{X}^2}{2 \cdot (1 - \rho)} \quad (4.32)$$

By substituting Equations 4.29 and 4.32 in Equation 4.4, the derivation of the end-to-end packet delay is complete and is given by:

$$\bar{D}_{ete} = (D_{Sl_Tx} + D_{prop}) \cdot (1.5 - \frac{\lambda}{2 \cdot \mu}) + (0.5 + N_{MCI} + Pk_{mci} + \frac{\lambda}{2 \cdot \mu}) \cdot MCI_t + \frac{\lambda \cdot \bar{X}^2}{2 \cdot (1 - \frac{\lambda}{\mu})} \quad (4.33)$$

4.4.2 Throughput formulation for a single node scenario

Having formulated the total end-to-end packet access delay, the throughput, S , is given by the number of packets serviced per second (L_x) multiplied by the packet size, thus:

$$S = L_x \cdot Pk_{size} \quad (4.34)$$

In order to obtain L_x , we need to calculate first the mean number of packet serviced per busy cycle, \bar{L}_{bc} . Then, the number of packets serviced per second can be easily obtained by dividing \bar{L}_{bc} between the average time of a busy cycle, \bar{T}_{bc} , thus:

$$L_x = \frac{\bar{L}_{bc}}{\bar{T}_{bc}} \quad (4.35)$$

From analysing Figure 4.17 a busy cycle (\bar{T}_{bc}) can be defined as the sum of a busy

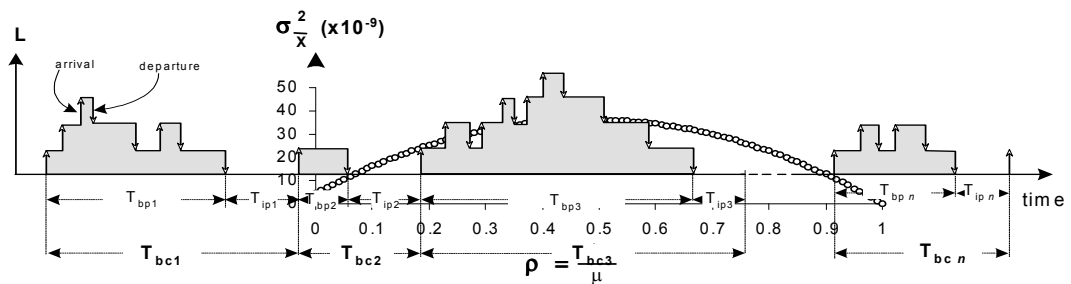


Figure 4.17 – Time of busy cycle.
Figure 4.16 – Variance of the mean service time.

period, \bar{T}_{bp} , plus an adjacent idle period, \bar{T}_{ip} , this gives:

$$\bar{T}_{bc} = \bar{T}_{bp} + \bar{T}_{ip} \quad (4.36)$$

In Equation 4.36, the busy period (\bar{T}_{bp}) is defined to begin with the arrival of a data packet to an idle channel and ends when the channel next becomes idle. Accordance to Gross [23], this busy period can be found by using the ratio:

$$\frac{\bar{T}_{bp}}{\bar{T}_{ip}} = \frac{\text{Probability that the system is busy}}{\text{Probability that the system is idle}} = \frac{\frac{\lambda}{\mu}}{1 - \frac{\lambda}{\mu}}$$

Since the arrivals are assumed to follow a Poisson distribution, the idle period is exponential with mean $\bar{T}_{ip} = \frac{1}{\lambda}$. Thus, the busy period is given by:

$$\bar{T}_{bp} = \frac{1}{\lambda} \cdot \frac{\frac{\lambda}{\mu}}{1 - \frac{\lambda}{\mu}} = \frac{1}{\mu \cdot (1 - \frac{\lambda}{\mu})} = \frac{1}{\mu - \lambda} \quad (4.37)$$

By substituting Equation 4.37 in Equation 4.36, the busy cycle results in:

$$\bar{T}_{bc} = \bar{T}_{bp} + \bar{T}_{ip} = \frac{1}{\mu - \lambda} + \frac{1}{\lambda} = \frac{\mu}{\lambda \cdot (\mu - \lambda)} = \frac{1}{\lambda \cdot (1 - \frac{\lambda}{\mu})} \quad (4.38)$$

The number of packets served in a busy cycle is now considered. This value can be obtained by making the following analysis. If a data packet is served in \bar{X} units, then the number of packets served in the busy cycle is given by dividing the average time of the busy period between the average time that it takes a packet to be served, so:

$$\bar{L}_{bc} = \frac{\bar{T}_{bp}}{\bar{X}} = \frac{\frac{1}{\mu - \lambda}}{\bar{X}} = \frac{1}{\bar{X} \cdot (\mu - \lambda)} = \frac{1}{1 - \frac{\lambda}{\mu}} \quad (4.39)$$

Finally, the number of packets served per second is then calculated as:

$$L_x = \frac{\bar{L}_{bc}}{\bar{T}_{bc}} = \frac{\frac{1}{1 - \frac{\lambda}{\mu}}}{\frac{1}{\lambda (1 - \frac{\lambda}{\mu})}} = \lambda \quad (4.40)$$

Therefore our formulation now is complete and the throughput can be obtained by:

$$S = \lambda \cdot Pk_{size}, \quad \forall \rho < 1 \quad (4.41)$$

4.4.3 Bounds on maximum system throughput

So far, we have formulated the mean access delay and the throughput for a single node configuration. However, for a larger number of nodes we have derived some bounds that can be used to estimate the maximum system throughput sustainable by the DVB/DAVIC's upstream channel. These bounds depend mainly on the traffic types being delivered and on the maximum bandwidth assigned for contention and reservation access, as well as the contention resolution algorithm used. In this section we present bounds for three different configurations. These bounds will play a vital role to validate more complex traffic configurations in the following chapters.

A simple bound on the maximum upstream throughput would be when only data slots are being transmitted, so that 48 bytes out of 64 bytes are transmitting payload for an efficiency of $(48/64=)$ 75%. This would be approached in theory if very large packets were transmitted, so that the capacity of the contention-based access region was negligible. But in practice, part of the bandwidth is reserved for contention access, and the efficiency heavily depends on the bandwidth assigned to this access mode.

Therefore, the maximum system efficiency can be estimated by Equation 4.42.

$$S_{\max_large_packets} = \frac{48 \cdot RS}{64 \cdot (RS + CS)} \quad (4.42)$$

where RS s and CS s are the number of reservation and contention access slots described per signalling frame, respectively.

The simple bound of the maximum upstream throughput presented in Equation 4.42 cannot be used to calculate the maximum throughput of variable packet sizes. This is because such estimation was based only on large packets being transmitted when most of the bandwidth was allocated to the reservation-based access region and did not take into account the additional bandwidth that should be allocated to the contention-based access region so as to resolve collision. In [96] and [98] a more efficient calculation to estimate the maximum throughput of a CATV network (based on the IEEE 802.14

standard) was proposed. By making slight changes we can use the same estimation applied to the DVB/DAVIC standard as shown in Equation 4.43.

$$S_{\max_exp_backoff} = \frac{Payload}{Payload + Overhead + eContention_Slots} = \frac{Pk_{size}}{Pk_{slots} + eCSs} \quad (4.43)$$

For the *splitting tree algorithm*, reservation requests are now transmitted using a minislot (MSs) of 21 bytes instead of a complete contention slot of 64 bytes. Thus, the maximum theoretical system throughput that can be achieved with this algorithm is given by Equation 4.44.

$$S_{\max_split_tree} = \frac{Pk_{size}}{Pk_{slots} + eMSs} \quad (4.44)$$

4.5 Access delay and system throughput validation

In this section we validate the results yielded by the simulation model for different packet sizes and variable offered load. In order to get the maximum throughput, we set the ‘*Minimum number of contention slots per MCI frame*’ to 1 slot. The arrival rate of the packets was exponentially distributed and the data rate for the upstream channel was at 6.17 Mbps. Table 4.2 presents the maximum throughput sustainable for different packet sizes ranging from 64 to 1518 bytes.

In this table the maximum throughput is given by Equation 4.45.

$$S_{\max} = \frac{Pk \cdot 8}{\bar{X}} \quad (4.45)$$

where \bar{X} corresponds to the maximum service time given by Equation 4.29.

Table 4.2 – Maximum theoretical throughput for different packet sizes.

Packet Size (bytes)	Max. Service Time \bar{X} (ms)	Max. Throughput (S_{max})
64	2.24	228 kbps (≈ 4 % of the cc)
128	2.24	456 kbps (≈ 7 % of the cc)
256	2.24	912 kbps (≈ 15 % of the cc)
512	2.99	1368 kbps (≈ 22 % of the cc)
1024	3.74	21876 kbps (≈ 35 % of the cc)

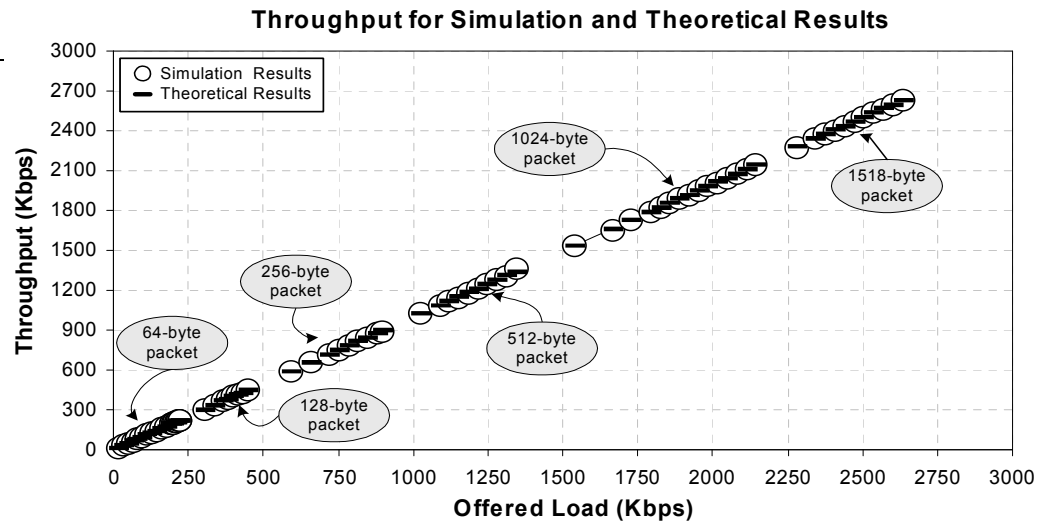


Figure 4.18 – System throughput: simulation and theoretical results for different packet sizes.

Figures 4.18 and 4.19 show the average end-to-end delays and throughput for simulation and analytical results, respectively, as the offered load is increased for the six different packet sizes presented in Table 4.2. For each packet, the offered load (λ) was increased from a relatively low rate at 16 kbps, up to approximately 98% of the maximum theoretical throughput. Offered loads close to 100% of the maximum throughput were not considered, since the stability of the M/G/1 systems [10] requires $\lambda < 1$.

By analysing these figures, we can verify that the results obtained using the simulation model for the DVB/DAVIC protocol are in good agreement with theoretical results. For example, for the worst case scenario that is when a packet size of 64 bytes is used. From simulation results (Figure 4.18), the maximum throughput that a single node can achieve resulted in 224 kbps. From analytical results (Table 4.2), this number resulted in 228 kbps. This extremely low performance of the DVB/DAVIC protocol (equivalent to $\approx 4\%$ of the channel capacity) is due to the two-phase data transmission cycle, which requires a request to be issued in the upstream channel and the data to be transmitted after a grant message is received in the downstream channel.

The delay between the time when a request is issued and the time when it is granted is another parameter, and needs to be analysed and accurately calculated. This delay, referred here as the ‘scheduler-look ahead (D_{lah})’, has proven to be one of the major delay elements in the transmission cycle, which dramatically reduces the throughput. When the packet size was changed to maximum size (1518 bytes) the maximum throughput sustainable per station resulted in ≈ 2630 kbps (equivalent to 43% of the channel capacity).

In general, results for mean access delays (Figure 4.19) revealed that for all packets analysed, the maximum deviation between simulation and analytical results was found

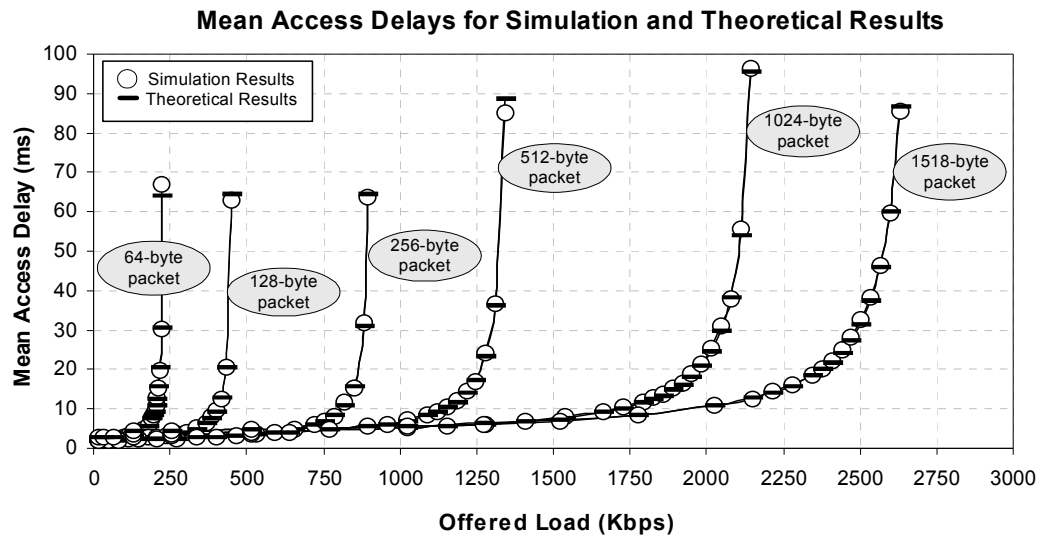


Figure 4.19 – Mean access delay: simulation and theoretical results for different packet sizes.

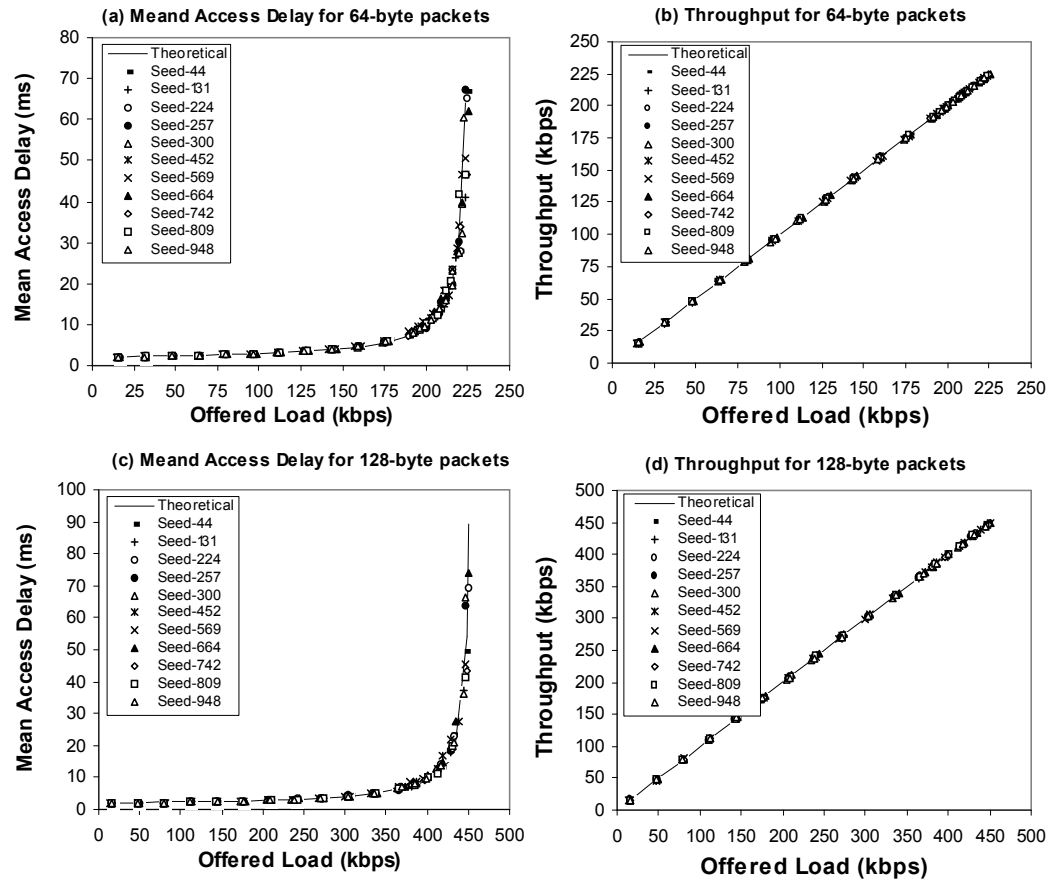


Figure 4.20 – Mean access delay and throughput for different seeds.

to be under 3% with an offered load up to $\approx 98\%$ of the maximum channel capacity. This deviation was found to be under $\approx 2\%$ for system throughput (Figure 4.18).

Simulation results presented in the previous two figures were obtained with a seed of 257. We also used different seeds for simulation results in order to verify the accuracy of the simulation model. Figure 4.20 shows the performance analysis when ten random seeds were used (selected in the range from 0 to 1000) for 64 and 128 bytes packet sizes. Results for mean access delays revealed that all seeds follow the same distribution as that of the theoretical results. For offered loads under 95% of the maximum theoretical throughput, the maximum deviation was still under 3%. For offered loads close to saturation (from 96 to 99%) the deviation was seen to be about 4%. However, the maximum deviation in terms of throughput was unchanged at 2%.

We have proved that simulated performance of the system does not depend on the seed used, therefore it will not be mentioned again and all simulation results presented in the following chapters will be yielded using a seed of 257.

For a larger number of nodes, we also carried out a rigorous verification test of the operation of the DVB/DAVIC simulation model. For this test, we used a mixed traffic pattern consisting of 32 kbps IP traffic and 9.7 kbps VoIP traffic (these traffic types are described in Section 5.4), using the *exponential backoff algorithm*. We ran a simulation for 60 seconds and captured the sequence of events between the 59th and the 60th second. Appendix D shows this sequence for 30 stations. From analysing these events we can verify that every station transmitted all packets received and used the contention resolution algorithm correctly. On average, every station received $(1/(0.092 \text{ IP inter-arrival} + 0.12 \text{ VoIP inter-arrival})) = 19.6$ packets per seconds, which corresponds to the number of packets received and transmitted in Appendix D.

4.6 Conclusions

The DVB/DAVIC protocol is a relatively new specification and little information about its performance and scalability is known. In this chapter, the suitable simulation and theoretical models have been presented. These models will be used in subsequent chapters for the performance evaluation, optimisations and enhancements of the DVB/DAVIC protocol.

The simulation model implemented for this research was based on the main network topology of the CSF (v.13), developed by MIL3, Cablelabs and the University of Sheffield. This model has been significantly modified to support the basic functionality of the MAC and PHY layers of the DVB/DVIC protocol, including the functionality of the two contention resolution algorithms adopted by the DVB/DAVIC protocol. In addition, new features for further optimisations were also incorporated in the simulation model, such as contention slot allocators, enhanced reservation requests mechanisms, and a prioritised scheduler, which will be used in subsequent chapters.

Existing analytical models from the literature were insufficient to model the DVB/DAVIC protocol, as they did not take into account the variable bandwidth of the

contention-based access region and the performance of the *exponential backoff algorithm* or the *splitting tree algorithm*. A less complicated theoretical model has been formulated for the purposes of validation of the simulation model. Such a model was based on an M/G/1 queuing system for a single node scenario. This model addressed one of the fundamental performance properties of computer communications protocols, which relates to the mean packet access delays and maximum sustainable throughput of a station.

Results obtainable using the simulation model for the DVB/DAVIC protocol were in good agreement with theoretical results, with a maximum deviation of 4% for mean access delays and 2% for throughput. We also found that the DVB/DAVIC protocol is highly inefficient when short data packets are transmitted, achieving only ≈ 224 kbps ($\approx 4\%$ of the channel capacity) when 64-byte packets are delivered.

For a larger number of nodes, we have also provided some theoretical bounds that could be used to estimate the maximum system throughput sustainable by the DVB/DAVIC's upstream channel for different traffic loads and protocol configurations.

Chapter 5

UPSTREAM CHANNEL CAPACITY AND CHARACTERISATION

5.1 Introduction

The design and implementation of communications protocols require knowledge of fundamental performance characteristics. Such characteristics include channel capacity, system throughput and utilisation bounds, maximum number of nodes (or streams) supported, mean access packet delay and the maximum throughput achievable per node under different traffic loads and configurations, among others. Knowledge of these characteristics allows service providers or network operators to plan the offered services, make cost and tariffing predictions and implement the network topologies. In addition, there are protocol operational parameters for which only recommended values are given in the specification. Such parameters should be optimised for different traffic conditions in order to achieve optimum system performance.

From the literature survey presented in Section 1.2.3.3, it may be postulated that there is a lack of studies that address the scalability, performance issues and optimisation of the DVB/DAVIC protocol explicitly. Therefore, the objective of this chapter is to present a comprehensive estimation of the performance characteristics of the upstream channel, when the contention and reservation access modes are used. In order to study complex analysis and address issues of capacity and scalability for the DVB/DAVIC protocol, the simulation model described in Section 4.3 will be used. The structure of the analysis to follow is presented in the following section.

5.2 Case studies

Having introduced and validated the OPNET simulation model in Chapter 4, a number of analyses are presented here to define the fundamental properties and scalability of the upstream channel. Specific issues of the case studies address the following performance analysis given below.

- ***Offered load scalability***: how the protocol scales against increasing the offered load generated by a number of active stations, and what the delay and system throughput characteristics are.
- ***Capacity in terms of active stations***: how the protocol scales against an increasing number of active stations, each generating a predefined traffic load.
- ***Station's buffer capacity***: how the performance characteristics change if the stations' buffer capacity is altered.
- ***Effects of contention slots for data transmission***: what the delay and system throughput characteristics are when transmitting a determined number of cells in the '*contention-based access region*'.
- ***Effects of varying the maximum reservation request message per station***: how the protocol scales against an increase in the maximum '*reservation request message*'.
- ***Effects of changing the number of MCI frames***: performance gain when the signalling frame cycle is reduced.
- ***Effects of varying the packet size in isochronous streams***: how the protocol scales against changing the packet size and inter-arrival times when delivering isochronous data rates.

These configurations are examined because they demonstrate the type of results that could be obtained using both the simulation and analytical model, and also help to identify key issues and suggest performance optimisation methods with a wide range of applications. Results, such as maximum network capacity, can be used for capacity

planning as well as identifying the limits of the different levels of service that can be provided to subscribers.

For all the configurations defined above we need to find saturation points and the reasons for inefficiencies, as well as how optimal features such as the use of the *splitting tree algorithm* to resolve collisions could possibly improve the overall system performance.

5.3 Traffic type characteristics and system parameters

As stated in Section 4.3.2, each station has been enabled with two traffic sources, the first traffic source generates Internet traffic and the second generates either VoIP streams or isochronous streams. In this chapter we base our analysis on these two traffic sources to examine the impact and trade-offs on system performance of four different topical traffic types: Internet traffic, Voice over IP (VoIP), mixed traffic, and isochronous streams, (described below) and at the same time providing a performance characterisation of the DVB/DAVIC protocol.

5.3.1 Internet traffic

This generator emulates Internet traffic. From available traffic measurement studies [46] and [53], it is known that the frequency of Ethernet packets is as indicated in Table 5.1, which also presents the number of ATM cells required per packet and the total packet size to be transmitted. It is expected that most upstream packets will be acknowledgements of the higher bandwidth downstream packets. Hence, the majority of packets generated will be 64 bytes long and packet sizes of 1518 bytes correspond to the maximum size of Ethernet packets that are mainly used by File Transfer Protocol (FTP) applications.

Table 5.1- Packet size distribution and characterisation.

Packet Size (bytes)	Probability of occurrence	Number of ATM cells	Transmitted packet size with overhead	Number of signalling frames required
64	0.6	2	128	1
128	0.06	3	192	1
256	0.04	6	384	1
512	0.02	11	704	1
1024	0.25	22	1408	2

1518

0.03

32

2048

2

Packets are generated with exponential distributed inter-arrival times, with a mean value selected to produce the desired average data rate as defined in [53]. In this chapter the inter-arrival times are set in such a way that the resulting mean offered load per active station is 64 kbps, 32 kbps, 26.6 kbps or 16 kbps according to the configuration analysed. The average packet size (Pk_{bytes}) for this traffic type is ≈ 368 bytes, which computes to 8.3 ATM cells (Pk_{slots}) per message.

5.3.2 Voice over IP (VoIP)

Today voice is transmitted either in analogue or digital form in circuit-switched networks (PSTN, ISDN and GSM). Between two end users a reserved (virtual) channel exists. A constant data rate is used to transmit the voice data (e.g. 64 kbps in the European ISDN, 56 kbps in the USA ISDN) with a service charge. In future, voice calls may be transmitted via existing IP networks with the same or better levels of quality.

We have selected VoIP traffic (from the set of services of ‘*IP Telephony*’) to analyse the performance characterisation of the upstream channel, because this technology will play a key role in future telecommunications networks. In general, *IP Telephony* refers to communications services (e.g. VoIP, fax and voice-messaging applications) that are transported via the Internet in digital form using discrete packets rather than in the traditional circuit-committed protocols of the Public Switched Telephone Network (PSTN). The VoIP traffic type, presented here, emulates a speech codec ‘G.723.1’ [57], which according to the ITU, IETF and the VoIP Forum is the preferred codec for Internet telephony applications [101] and [28].

This codec generates a data rate of 5.3 kbps or 6.3 kbps depending on the mode. In this research codecs of 5.3 kbps will be used. This codec generates and encodes a 20-byte data frame every 30 ms. In a study carried out in [28], it was found that as the number of audio frames per packet increases, the packet overhead decreases. The overhead decrease is explained by more information bytes (audio frames) being included in a packet with a fixed header (RTP+UDP+IP). Conversely, the latency increases as the number of frames per packet increases. This is expected since more audio frames

require more time to be captured and buffered. Hence, a trade-off exists between packet overhead and local latency for audio packet transfer.

The optimal point for the number of audio frames per packet is between seven and eight for G.723.1 codecs. According to [28], these values should be used by terminals that target packet efficiency as well as low latency for audio packets. However, sometimes, the audio local latency is reduced at the expense of packet overhead, in order to achieve latencies acceptable to users. In fact, a few applications are willing to sacrifice some protocol overhead for achieving better audio latency. Some H.323 terminals use a value of 3 or 4 for the number of frames per audio packet in order to reduce this latency [28].

Therefore, in order to be consistent with these figures for H.323 terminals, 4 audio frames will be used per packet. Then, the following protocol overheads are added in order to yield a complete VoIP packet.

Each audio packet has a Real Time Protocol (RTP) header of 12 bytes that carries sequence numbers, a synchronisation source identifier and time-stamps. A User Datagram Protocol (UDP) header of 8 bytes is needed to carry a UDP with an unreliable transmission service. In addition a 20-byte IP header is needed to transfer routing information. A 3-byte Logical Link Control (LLC) and a 5-byte SubNetwork Attachment Point (SNAP) headers are used to carry a PDU with a connectionless service and to identify the type of the bridged media, respectively. Finally, an 18-byte MAC header is used to transmit the PDU to its final destination over the shared media.

Thus, by adding the complete headers one obtains an improved VoIP stream of 9.7 kbps as indicated in Table 5.2. Hence, in our performance analysis, VoIP streams of 9.7 kbps will be considered.

Table 5.2- VoIP encapsulation with and without header suppression.

Frame/Header	9.7 kbps Streams	12.4 kbps Streams
Voice frame	80 bytes	20 bytes
RTP header	6 bytes	
UDP header	12 bytes	
IP header	8 bytes	20 bytes
LLC header	3 bytes	
SNAP header	5 bytes	
Ethernet MAC header	18 bytes	
Total Size	146 bytes (4 ATM cells)	40 bytes (1 ATM cells)

Another novel and topical VoIP stream that will be analysed is that of 12.4 kbps, where every 30 ms a voice frame is generated and encoded using header compression, as illustrated also in Table 5.2.

In order to support VoIP traffic, according to [118], there is a need for an overall end-to-end packet delay of less than 150 ms for a high-quality call and up to 400 ms for a low-service quality call. In our experiments we consider delays under 50 ms from the NIU to the headend for the support of VoIP streams, leaving an extra 100 or 350 ms delay for the final destination according to the expected quality of the call.

5.3.3 Mixed traffic (Internet +VoIP)

This traffic type emulates a combined traffic situation, where each station is generating a VoIP data stream of 9.7 kbps, as introduced in Section 5.3.2. Additionally, some Internet traffic, as presented in Section 5.3.1, is multiplexed into the data stream and is transmitted via the upstream channel. The mean data rate per active station is set to 32 kbps (consisting of 9.7 kbps of VoIP traffic and 22.3 kbps of Internet traffic) or 41.7 kbps (consisting of 9.7 kbps of VoIP traffic and 32 kbps of Internet traffic) according to the case study analysed.

5.3.4 Isochronous streams

Isochronous streams are time-dependent and exist with processes where data must be delivered within certain time constraints. For example, most multimedia streams require an isochronous transport mechanism to ensure that data is delivered as fast as it is displayed and that the audio is synchronised with the video. This traffic type emulates isochronous streams with data rates of 12 kbps, 32 kbps, 64 kbps and 128 kbps, suitable for timing-critical interactive services (e.g. low quality video, compressed/uncompressed voice telephony and audio). Different packet sizes (64, 128, 256, 512, 1024 and 1518 bytes) were considered for the delivery.

All isochronous streams used in the simulations included the higher layer protocol overhead. For example, the most likely protocol for isochronous streams is TCP/IP with a Direct IP or Ethernet bridge solution. The latter has (at the MAC layer) a 61-byte overhead comprising of 20-bytes (TCP header) + 20-bytes (IP header) + 3-bytes (LLC

header) and 18-bytes (Ethernet MAC header/trailer). Therefore, not all of the stream capacity is available to isochronous applications. For instance, in the worse case scenario when 64-byte packets are delivered, for a 12 kbps isochronous stream the effective bandwidth (excluding higher layer protocol overhead = 20-bytes TCP + 20-bytes IP + 3-bytes LLC + 18-bytes Ethernet MAC = 61-bytes), is dramatically reduced to 0.56 kbps when the TCP/IP protocol with an Ethernet bridge solution is considered.

Such overheads should be taken into account when considering the delivery of applications with specific bandwidth requirements. The effective throughput for the different streams as a function of the packet size and protocol stack (see Figure B.1) used for these simulations is given in Table 5.3, (shaded lines in the table present the effective bandwidth when the TCP/IP with an Ethernet bridge solution is considered, which is the most popular solution in CATV networks).

It is evident from Table 5.3, that although lower packet sizes provide quicker interaction, smaller packet sizes reduce the available bandwidth due to a proportional increase in overhead. In order to compensate for the considerable reduction in bandwidth, in the last version of the of the DVB/DAVIC protocol [35] (recently

Table 5.3 - Effective bandwidth in isochronous streams (kbps).

Isochronous Streams (kbps)		Packet size (bytes)					
		64	128	256	512	1024	1518
12	TCP/IP	3.94	7.97	9.98	10.99	11.50	11.66
	UDP/IP	6.19	9.09	10.55	11.27	11.64	11.75
	PPP	9.38	10.69	11.34	11.67	11.84	11.89
	TCP/MAC Eth	0.56	6.28	9.14	10.57	11.29	11.29
	UDP/MAC Eth	2.81	7.41	9.70	10.85	11.43	11.61
32	TCP/IP	10.50	21.25	26.63	29.31	30.66	31.09
	UDP/IP	16.50	24.25	28.13	30.06	31.03	31.35
	PPP	25.00	28.50	30.25	31.13	31.56	31.70
	TCP/MAC Eth	1.50	16.75	24.38	28.19	30.09	30.71
	UDP/MAC Eth	7.50	19.75	25.88	28.94	30.47	7.50
64	TCP/IP	21.00	42.50	53.25	58.63	61.31	62.19
	UDP/IP	33.00	48.50	56.25	60.13	62.06	62.69
	PPP	50.00	57.00	60.50	62.25	63.13	63.41
	TCP/MAC Eth	3.00	33.50	48.75	56.38	60.19	61.43
	UDP/MAC Eth	15.00	39.50	51.75	57.88	60.94	15.00
128	TCP/IP	42.00	85.00	106.50	117.25	122.63	124.37
	UDP/IP	66.00	97.00	112.50	120.25	124.13	125.39
	PPP	100.00	114.00	121.00	124.50	126.25	126.82
	TCP/MAC Eth	6.00	67.00	97.50	112.75	120.38	122.86
	UDP/MAC Eth	30.00	79.00	103.50	115.75	121.88	30.00

released in October 2001), a new advanced mechanisms is used for header suppression. By using this feature, all constant components of the different headers can be suppressed, thus the effective data rate of isochronous streams (e.g. multiplexed voice, audio and/or video) is remarkably increased.

5.3.5 System parameters and assumptions

In all simulations, one upstream channel with a capacity of 3.088 Mbps and one downstream channel with 42 Mbps were considered. Another important simulation parameter is the choice of the network size. According to [100] the number of active subscribers per upstream channel on European cable networks is approximately 50 users in peak hours of traffic.

In our analysis we use cluster sizes of up to 70 stations per upstream channel in order to analyse the performance of the DVB/DAVIC protocol during high periods of congestion. Some scenarios will approach an increased number of stations (up to 340 stations), with a reduced traffic load to analyse the maximum number of nodes that can be supported. The complete set of simulation parameters used in order to produce the results presented in this chapter are summarised in Table 5.4.

Table 5.4 – Simulation parameters.

Simulation Parameter	Value
Upstream data rate (QPSK)	3.088 Mbps
Downstream rate (64-QAM)	42 Mbps
Downstream signalling	In Band
Maximum number of active NIUs (EuroModems)	340*
Minimum and maximum backoff values for the <i>exponential backoff algorithm</i>	3 and 5*
<i>Entry spreading</i> factor of the <i>splitting tree algorithm</i>	6*
Minimum contention-based slots per signalling frame	3 slots*
Buffer capacity per NIU	3000 ATM cells*
Transmission time (cycle) of the signalling frame	3 ms*
Simulation time for each run	60s
Maximum reservation request message length	32 ATM cells (slots)*
Maximum contention access message length	0 ATM cells (slots)*
Distance from nearest/farthest NIU to the Headend	10-16 Km
Spacing between closest and farthest NIU	Randomly distributed
Headend and NIU processing delay	2 microseconds each
Propagation delay (coax and fibre)	5 microseconds/Km

* These parameters will be ranged as indicated by the traffic scenarios

The values selected for the different parameters are the defaults defined in the specification where applicable. All references to packet sizes in the analysis that follows refer to the size of the packet as it enters the system from a PC station. This would be the packet that the NIU is going to submit for delivery over the cable network and does not include DVB MAC and PHY overheads. However, those include 18-byte Ethernet, 20-byte IP, 20-byte TCP, 8-byte UDP and 12-byte RTP (for VoIP only) headers. These protocol overheads should be taken into account when the results presented here are used to evaluate application performance. In addition, simulations will be run using both contention resolution algorithms adopted by the DVB/DAVIC protocol (*exponential backoff algorithm* and *splitting tree algorithm*) as defined by the scenarios.

Finally, for the tests defined, the INA utilised a simple First In First Out (FIFO) scheduler. A more complex scheduler was not necessary since all the streams are treated evenly.

5.4 Performance characterisation of the DVB/DAVIC protocol

5.4.1 Offered load scalability

In this section we analyse the system performance under increased offered load. The simulated network examined in this scenario consists of a small network size of 20 active stations, each generating a variable amount of traffic load made up from maximum Ethernet packets of 1518 bytes. The aim of this analysis is to find the highest achievable system performance in terms of system throughput and mean access delays rather than the effect of packet size, which is addressed in Section 5.7.

As mentioned in Section 4.4.3, a bound for the maximum upstream throughput is given by Equation 4.42. In this analysis, we have reserved at least 2 slots (out of 18 slots) per signalling frame for contention access, $CSs = 2$ and $RSs = 16$. Therefore, the maximum theoretical efficiency achieved is $\approx 67\%$, which we may expect from our simulation analysis (as shown below).

$$S_{\max} = \frac{48 \cdot RSs}{64 \cdot (RSs + CSs)} = \frac{48 \cdot 16}{64 \cdot (16 + 2)} \approx 66.7\%$$

The network configuration for this analysis was as follows: the upstream channel capacity (cc) was 3.088 Mbps and the offered load was ranged from a relatively low 243 kbps up to 4.1 Mbps with increments of 243 kbps for each simulation run. The contention resolution algorithm (used to resolve collisions) was the *exponential backoff algorithm*.

Figure 5.1 presents the mean access delay, throughput and utilisation versus offered load. The maximum system throughput of approximately 65% was obtained with an offered load equivalent to the full cc (3 Mbps), which is quite close to the maximum theoretical throughput estimated (of 67%). The deviation of 2% was accounted for by additional slots added to the contention access region. In average, 2.3 slots were allocated for contention access due to collisions and the inability of the allocation algorithm (at the headend) to schedule all the data slots.

The maximum channel utilisation achieved was $\approx 91\%$. The difference between maximum throughput and utilisation of 26% is attributed to the ATM protocol overhead, DVB Physical layer protocol overhead, reservation request transmissions, collisions and retransmissions. The remaining bandwidth of 9% of the cc was wasted due to unused contention slots. In the following scenarios and particularly is Section 5.4.7, we are going to analyse this difference in a broader context.

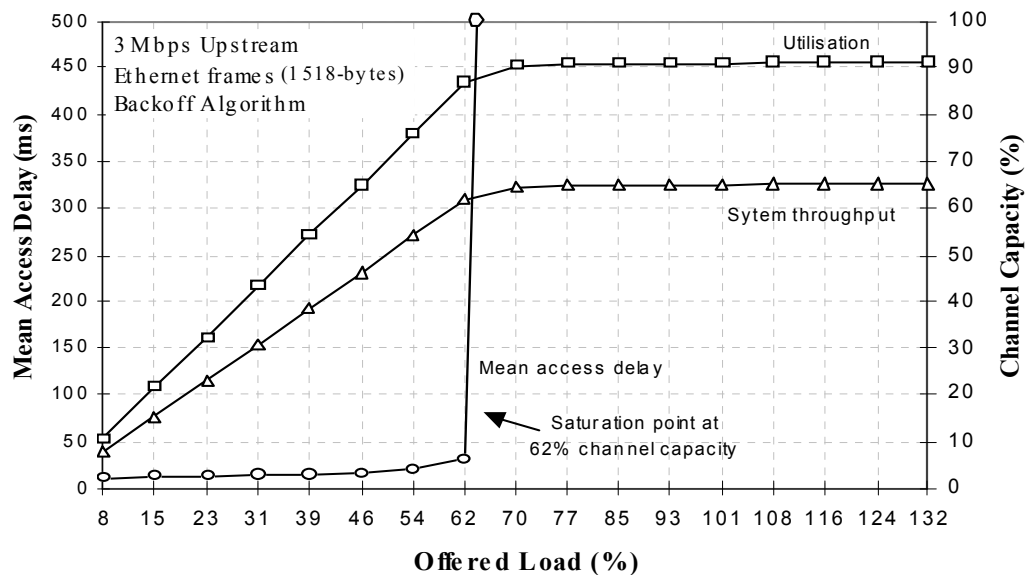


Figure 5.1 - Mean access delay, system throughput and utilisation vs. offered load.

The mean packet delay curve is stable (producing considerably low mean access delays under ≈ 33 ms) up to the point of saturation, which was found to be at approximately 61% of the cc , lower than the capacity achieved by the system throughput. From this point, even a slight increase in offered load may result in system instability, as we can observe with an offered load approximately of 62% of the cc .

Figure 5.2 gives a better insight into the delay characteristics of the system by plotting the number of frames that experienced delays of a certain value. In this figure at 61% of the cc , 75% of the Ethernet packets were transmitted in less than 33 ms and the other 25% under 67 ms. At 62% traffic load, the saturation point is evident, and access delays have been considerably increased, since only 11% of the packets are now transmitted under 67 ms. Hence, the system cannot maintain maximum throughput whilst providing bounded delay characteristics. Therefore, the maximum system throughput sustained is 1.88 Mbps (61% cc). In the next scenario we address the number of stations that can be supported.

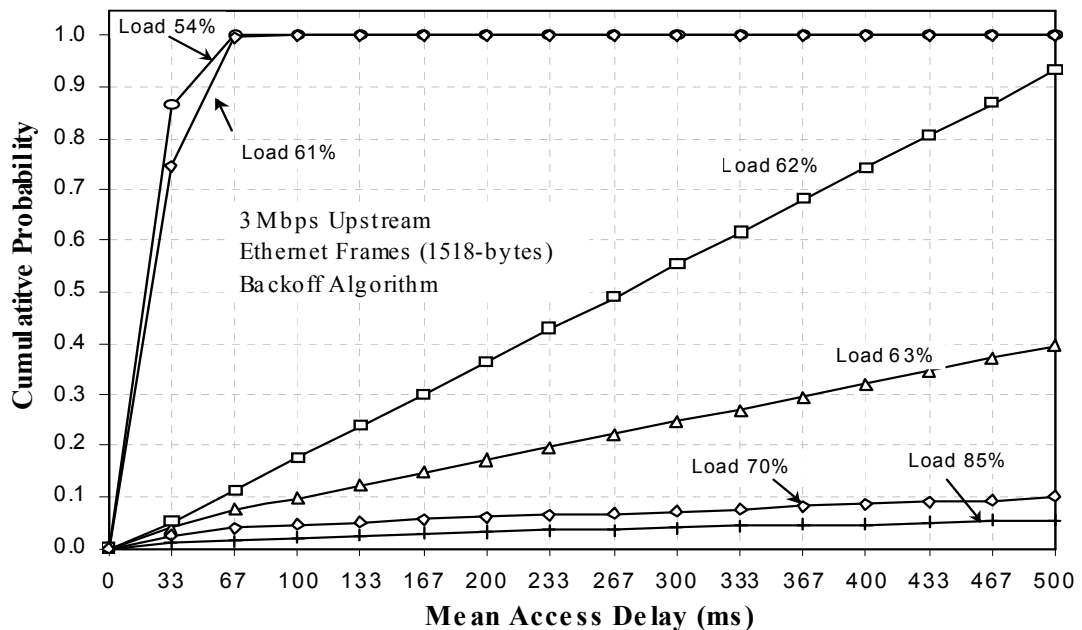


Figure 5.2 - Cumulative probability vs. mean access delay.

5.4.2 Capacity in terms of active stations

Here, an analysis of the scalability of the cable network in terms of number of active stations is presented. For this analysis we have now considered a low traffic configuration per station to find out the maximum number of stations that could be supported. The analysis starts with a small network size of 20 stations, then we increase the number of stations (in steps of 20) until the maximum network capacity has been exceeded.

The traffic load generated by each station was a single-Ethernet packet of 1518 bytes with an exponential distributed inter-arrival rate of 1 pk/sec or 12.14 kbps. The simulations were performed for each contention resolution algorithm.

Figure 5.3 depicts a linear system throughput until the knee with respect to the number of active stations. We can appreciate a slight increase in system performance when using the *splitting tree algorithm*. The maximum system throughput achieved was $\approx 61\%$ and $\approx 65\%$ of the *cc* for the *exponential backoff algorithm* and the *splitting tree algorithm*, respectively.

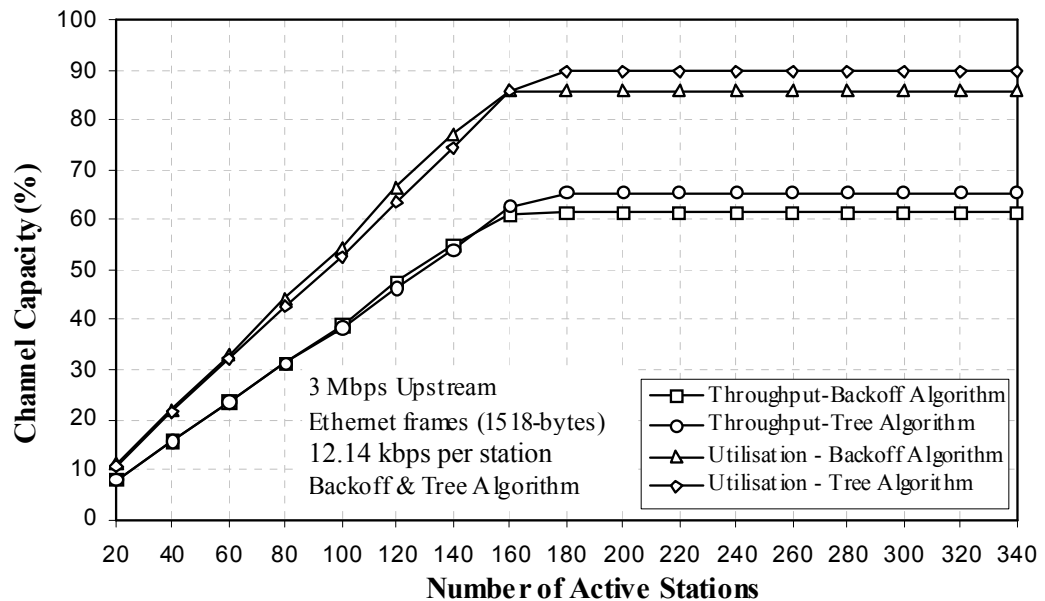


Figure 5.3 - System throughput and utilisation vs. No. of active stations, *exponential backoff* and *splitting tree algorithm*.

In terms of utilisation, the *exponential backoff algorithm* utilised approximately 86% of the link capacity in comparison with $\approx 90\%$ utilised by the *splitting tree algorithm*. In this analysis, which considered the maximum Ethernet packet size, the increase in system performance (throughput and utilisation) resulted in approximately of 4% of the *cc* when the *splitting tree algorithm* is selected. A bigger increase in system performance can be obtained when short packet data transmissions are used. In the following analysis we change the traffic pattern to appreciate this difference better.

Results for access delays, Figure 5.4 indicates that saturation points are presented beyond 140 stations (with mean packet delay of 39 ms) for the *exponential backoff algorithm* and 160 stations (with mean packet delay of 28 ms) for the and *splitting tree algorithm*. Past these points, a sharp delay increase is recorded, which grows higher than the mean inter-arrival rate (of 1s) and as a consequence mean access delays become asymptotically unbounded.

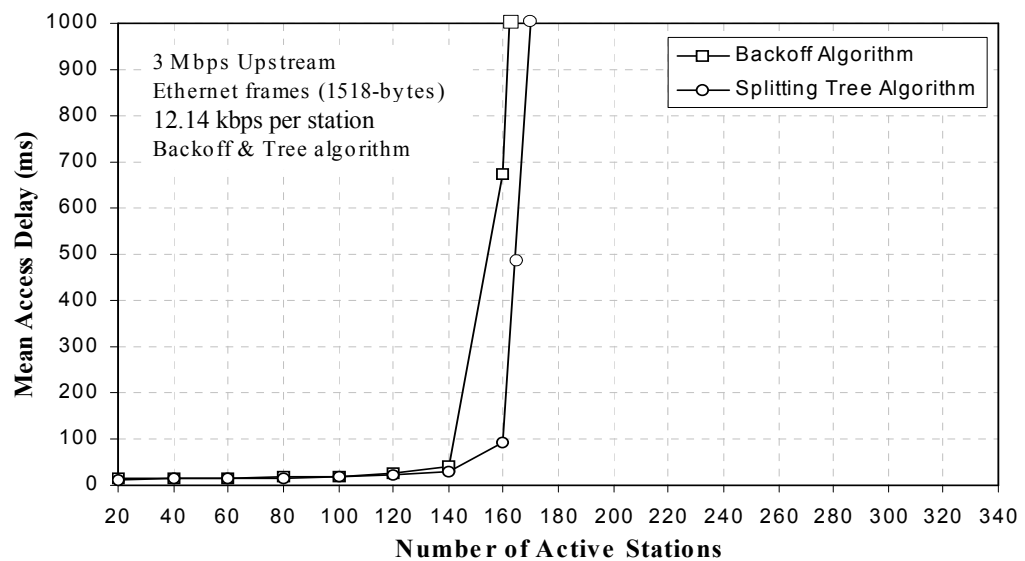


Figure 5.4 – Mean access delay vs. No. of active stations, *exponential backoff* and *splitting tree algorithm*.

In this section and also in Section 5.4.1, we have approached the highest system performance when transmitting maximum-size Ethernet packets in terms of offered load and maximum number of active stations supported. In typical networks, the packet sizes vary according to application services. Hence, in the following scenarios we address traffic patterns consisting mainly of Internet traffic, Voice over IP traffic, and mixed traffic configurations.

5.4.3 Effects of contention access for data transmission

As introduced in Section 3.5, the DVB/DAVIC protocol specification supports three different access modes for the transmission of data messages: contention, reservation and fixed-rate access. This section evaluates the system performance when stations are allowed to transmit small packets in the contention-based access region. The analysis presented here examines to what extent the system performance can be maximised by ranging the ‘*Maximum Contention Access Message Length*’ of a message, measured in ATM cells, that may be transmitted using contention access. Any message greater than this parameter is transmitted using reservation access.

The analysis addresses a worst case scenario of a novel and topical mixed traffic scenario (defined below), in which most of the packets are either minimum length Ethernet packets (in case of Internet traffic) or 40 bytes for VoIP streams. This traffic scenario was considered because small packet sizes benefit the response time of interactive applications. On the other hand, small packets put higher stress on the network. This is because more requests per volume of data have to be issued, which increases the protocol overhead and the probability of collisions.

The traffic load was created as follows: each active station generated 32 kbps Internet traffic as defined in Section 5.3.1. Additionally, 12.4 kbps Voice over IP traffic as introduced in Section 5.3.2, was multiplexed into the data stream. Thus, the mean data rate per active station was set to 44.4 kbps. The simulations were performed using the *exponential backoff algorithm*. Figures 5.5 to 5.8 present the results when the ‘*Maximum Contention Access Message Length*’ was ranged from 0 to 6 ATM cells.

Figure 5.5 shows that the lowest access delays can be obtained when stations only transmit messages comprised of 1 ATM cell in the contention access region. Low delays under 25 ms are produced with an offered load of 35% of the link capacity (created by 24 stations), which is more suitable for the support of both Internet applications and VoIP streams. The other values (2, 3 and 5 ATM cells) also offer good results, yielding mean access delays under 50 ms with the same traffic load. It can be appreciated that if none of the messages are allowed to be transmitted using contention access, higher delays are derived, decreasing the number of supported streams (to 22 stations).

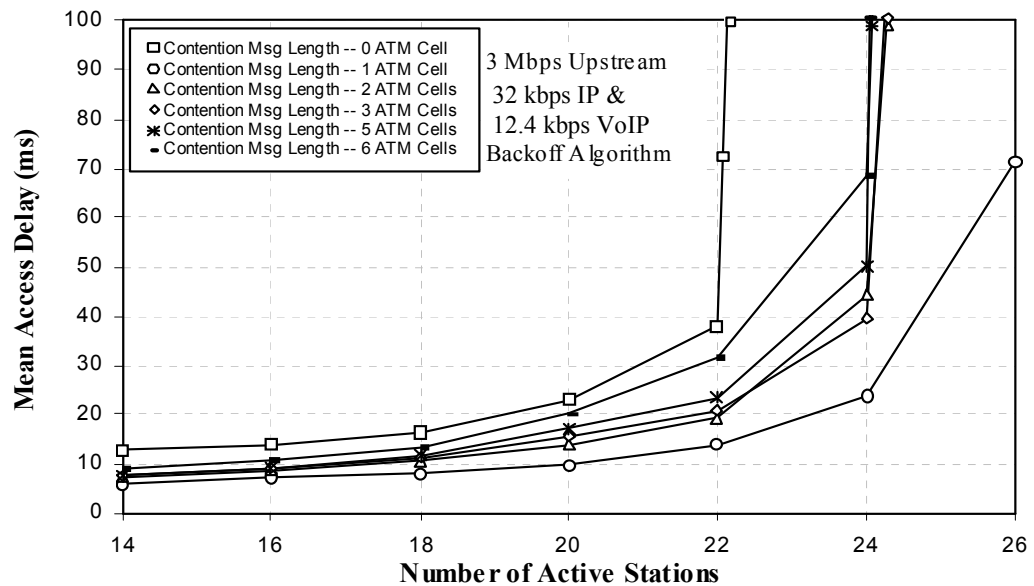


Figure 5.5 – Mean access delay vs. No. of active stations for different contention messages.

Figure 5.6 indicates that a maximum system throughput of $\approx (1.14 \text{ Mbps})$ 37% of the cc can be achieved also with the value of 1 ATM cell, in comparison to $\approx (1 \text{ Mbps})$ 32% of the cc yielded when all packets are transmitted using only reservation access. By allowing a station to transmit short packets in the contention-based region, not only a faster data transmission can be obtained (as appreciated in Figure 5.5) but also a higher system throughput can be achieved. This is because stations do not need to waste an extra slot to send a reservation request and then wait for its acknowledgement, thus improving the system performance. This increase in system performance can only be gained in small network sizes (under ≈ 50 stations). It has also been noticed that on heavy traffic loads there is a drawback. As the ‘*Maximum Contention Access Message Length*’ was increased, more data messages were transmitted using contention access. This in turn increased the collision risk with the transmissions of reservation requests sent by other stations and as a consequence system throughput is reduced. The effect is better appreciated in the next two figures.

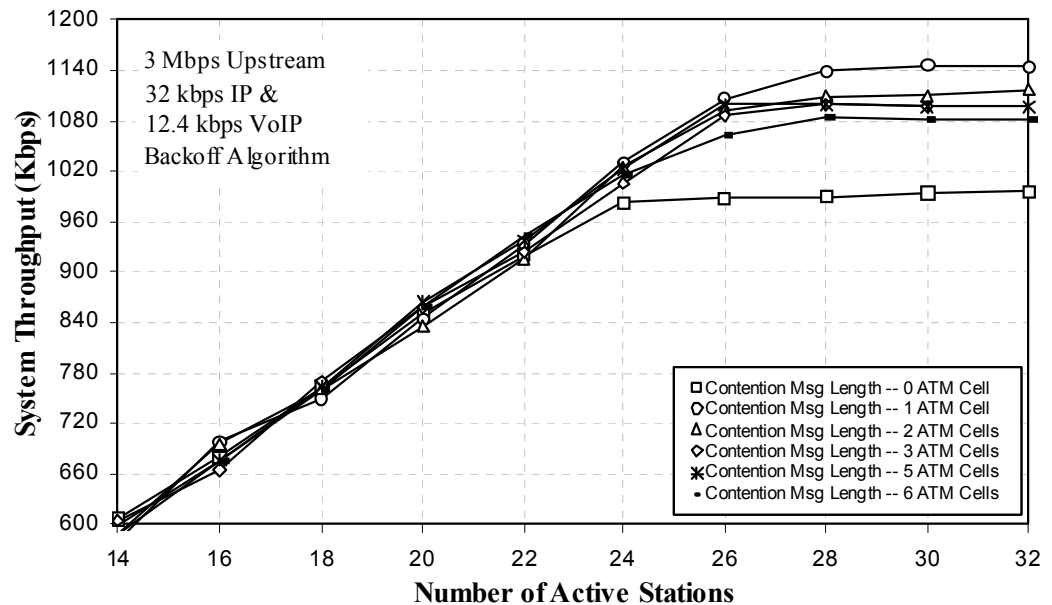


Figure 5.6 – System throughput vs. No. of active stations for different contention messages.

Figure 5.7 presents the total amount of data messages transmitted in the contention-based region, and Figure 5.8 shows the bandwidth consumed by collisions. Results from Figure 5.6 revealed that with a high offered load of 37% of the link capacity (generated by 26 stations) only 36% was successfully transmitted when using a value of 1 ATM

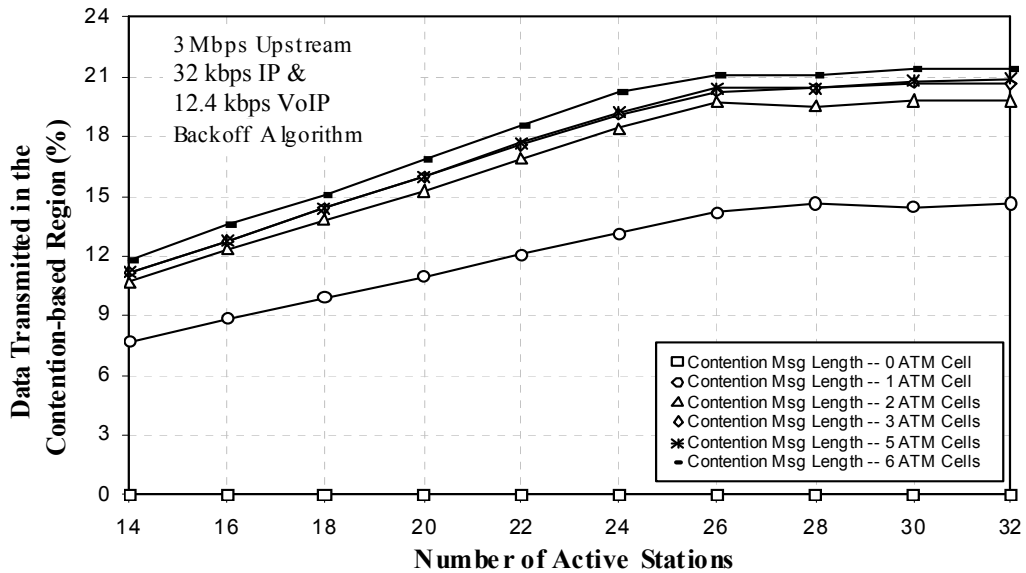


Figure 5.7 – Data transmitted in the contention-based region vs. No. of active stations for different contention messages.

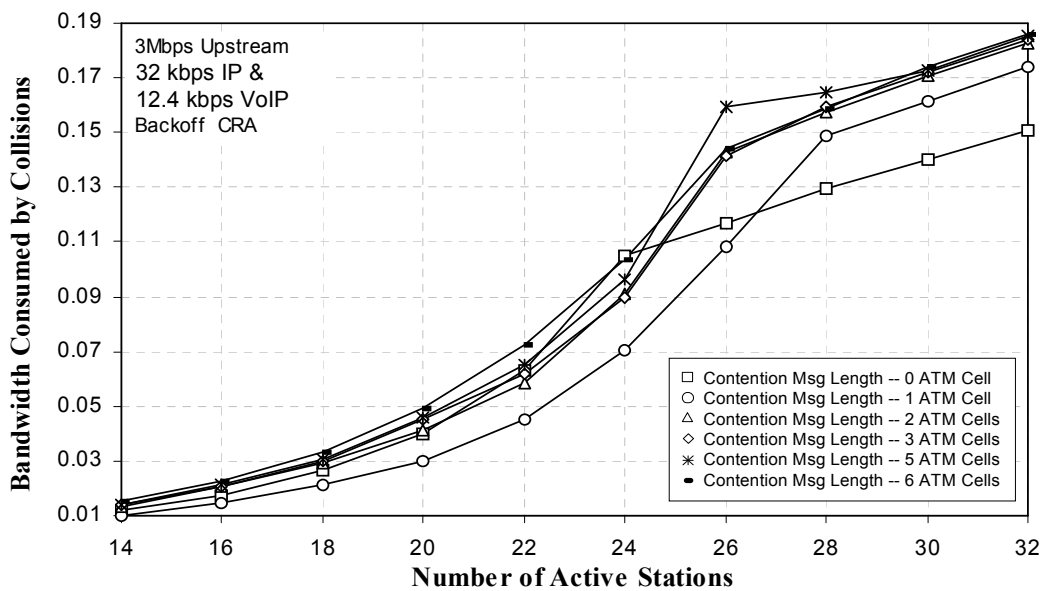


Figure 5.8 – Bandwidth consumed by collisions vs. No. of active stations for different contention messages.

cell for the '*Contention Access Message Length*' parameter (the other 1% was queued in the station's buffer). From this 36%, only 14.5% (Figure 5.7) of the offered load was transmitted in contention access, and the rest (21.5%) was sent using reservation access. In order to transmit the 1-ATM-cell messages and the reservation requests of the other messages using contention access, Figure 5.8 reveals that 10.8% of the bandwidth was consumed by collisions. With a value of 5-ATM cells for the same parameter, the bandwidth consumed by collisions was reported in the order of 16% of the *cc*, which results in a loss in system throughput as seen in Figure 5.6.

Thus, it can be noticed that as larger messages are allowed to be transmitted using contention access, the risk of collision is increased slightly. In the case that none of the messages were transmitted in contention access, it is obvious that with heavy traffic loads, the bandwidth consumed by collisions is minimised.

5.4.4 Effects of reservation request size

In this analysis, we now address the case when stations only use reservation access for the transmission of data messages. The analysis focuses on the impact on system performance when the *maximum request size* (or '*Maximum Reservation Access Message Length*' as named in the specification) is ranged from 6 to 32 ATM cells. This configuration parameter specifies the maximum length of a message that can be transmitted using a single reservation request access. Any message greater than this is transmitted by making multiple reservation requests.

When a station is enabled to use only reservation access for the transmission of data messages, as the offered load increases, short packets tend to accumulate in the station's buffer. This is true even at lower loads depending on the traffic type. Large packets for some applications may generate several (or many) ATM cells at once when segmented into 53-bytes chunks. If a station with a large packet is allowed to send only one request for the transmission of the complete packet, that may affect the transmissions of the other stations. This is because the headend serves the station's reservation requests in a FIFO order regardless of the traffic type, and the larger the request size the higher the channel access will be, which delays the other reservation request. This effect will be addressed at the end of this section.

In order to demonstrate the effects of an increased reservation request size, a mixed traffic configuration was analysed. The mean data rate per active station was set to 41.7 kbps, which consisted of 32 kbps Internet and 9.7 kbps Voice over IP traffic. The simulations were performed using now both CRAs. In Figures 5.9 to 5.11 we compare the results for different request limits of 6, 11, 22 and 32 ATM cells, which represent the transmission of packet sizes up to 256, 512, 1024 and 1518 bytes, respectively. Larger values were not considered, since we address the transmission of Internet and VoIP traffic that create packets under 32 ATM cells.

Figure 5.9 shows that the access delays can be significantly reduced when the *splitting tree algorithm* is selected. For both contention resolution algorithms the lowest mean access delay (around 50 ms) is yielded with request sizes of 22 and 32 ATM cells, which represent the transmission of 1024 and 1518 byte packets, respectively. It can be appreciated that for these request sizes, the *splitting tree algorithm* supported up to 8 stations more than the number supported by the *exponential backoff algorithm*. For a short request size up to 6 ATM cells, this difference was of 10 stations.

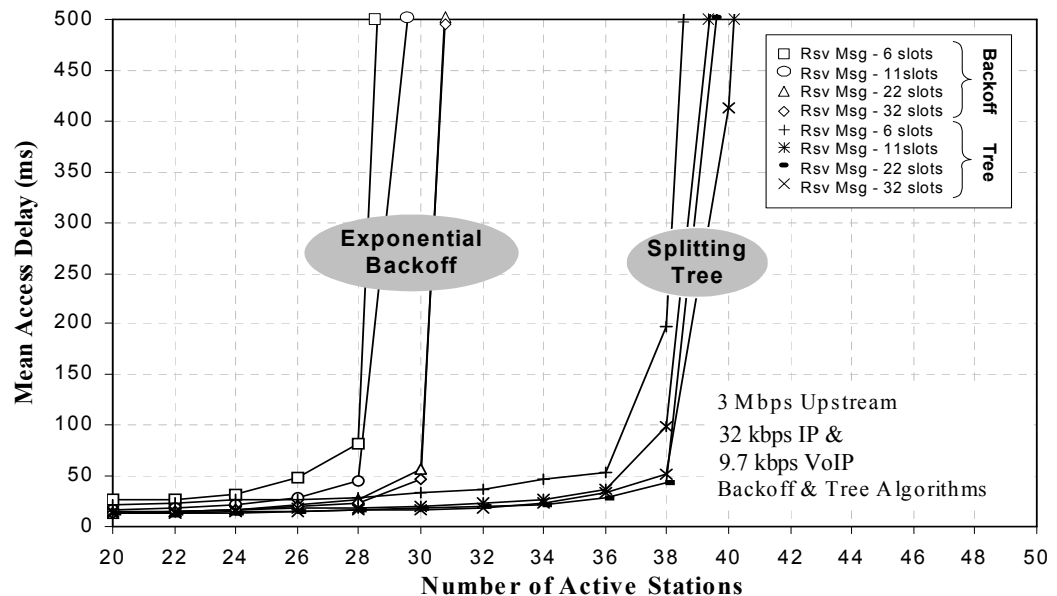


Figure 5.9 – Mean access delay vs. No. of active stations for different request sizes.

In terms of system throughput, as depicted in Figure 5.10 (for large request sizes of 22 and 32 ATM slots), the increase was approximately of 12% of the cc when the *splitting tree algorithm* was selected. Conversely, for short request sizes, the system performance increase can be increased up to $\approx 17\%$. Results for the *exponential backoff algorithm* revealed that once the maximum system throughput is reached (which is between 40 and 46% of the cc), as the offered load becomes higher, the throughput gradually declines. This is to be expected because of the increased likelihood of collisions, which consume more bandwidth as the cable network becomes congested.

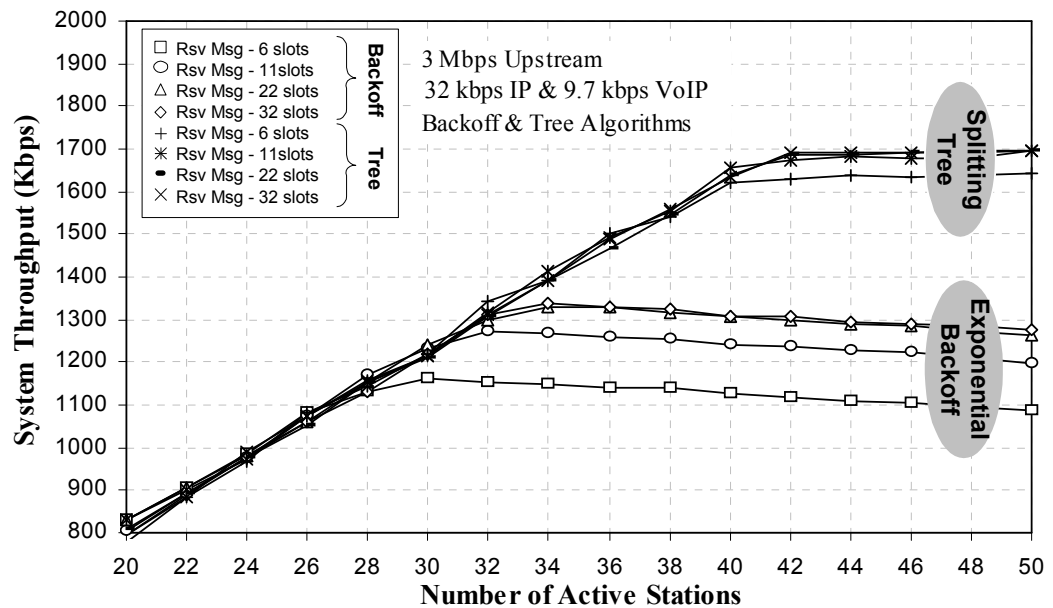


Figure 5.10 – System throughput vs. No. of active stations for different request sizes.

This effect can be clearly appreciated in Figure 5.11, which shows the total bandwidth consumed by only reservation requests. For the *exponential backoff algorithm*, with an offered load of 46% of the link capacity (produced by 34 stations) the bandwidth utilised by successful and unsuccessful requests ranged from 22 to 28.5% of the *cc* (accordingly to the request size), compared to 12.3 to 20.5% of the link capacity consumed by the *splitting tree algorithm*. This difference of approximately of 8% was accounted for increased collisions of reservation requests produced by the *exponential backoff algorithm*, which reduces the system throughput as the offered load increases.

Results presented in Figures 5.9 to 5.11 indicate that as the reservation request size increases, a gain in system performance is obtained. However, increasing the request size to its maximum value of 32-ATM cells, (for this analysis), may have its consequences as shown in the following analysis.

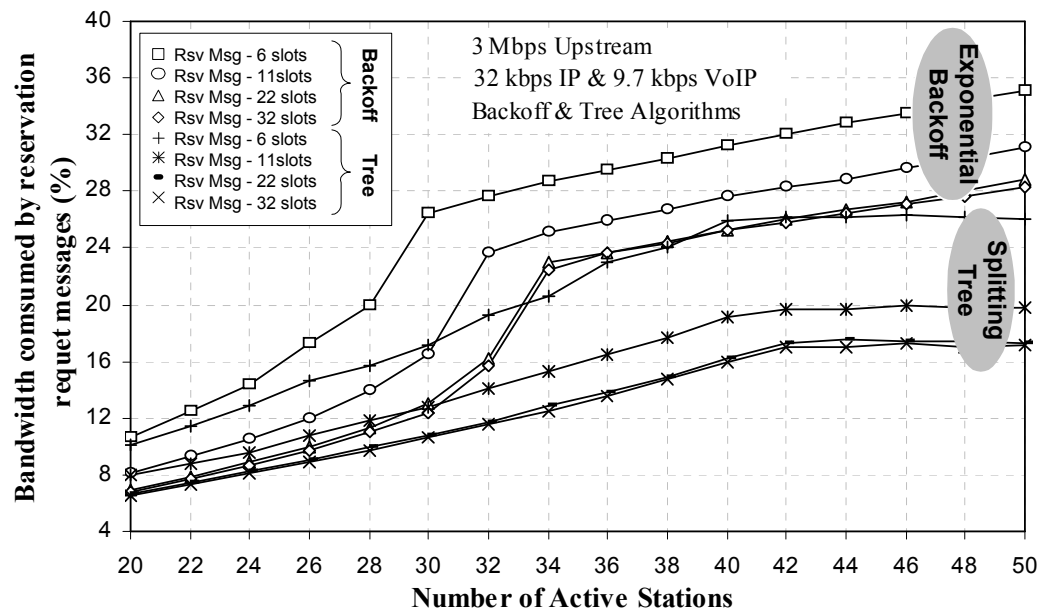


Figure 5.11 – Reservation request bandwidth consumed vs. No. of active stations.

Let us now analyse the case when every station was transmitting either 32kbps Internet traffic or 9.7 kbps VoIP traffic. For every simulation, half of the total active stations was transmitting Internet traffic and the other half was transmitting VoIP traffic. The packet distribution, packet sizes and packet inter-arrival times were the same as the previous analysis. The *splitting tree algorithm* was utilised for this study, since it provides a better system performance.

The main consequence of the use of the *splitting tree algorithm* is that with large request sizes (32 ATM cells), there is an increase in the access delay for Voice over IP traffic when the offered load becomes higher than 53% of the link capacity (produced by 80 active stations). Conversely, for Internet traffic, large request sizes may result in a reduction in access delay (regardless of the active number of stations) as shown in Figure 5.12.

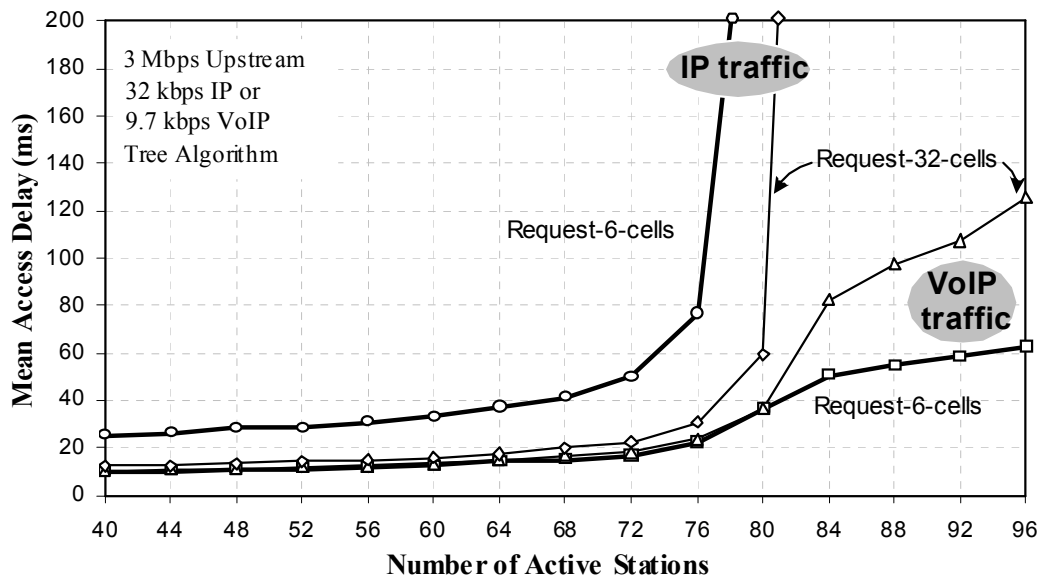


Figure 5.12 – Mean access delay vs. No. of active stations for VoIP and IP traffic.

A recommendation can now be made for CATV networks based on the DVB/DAVIC protocol. The *maximum request size* should be set as large as possible if the upstream channel is only used for Internet traffic. A value of 32 ATM cells for this parameter was found to provide the best system performance. However, if the upstream channel supports the transmission of both traffic sources (IP and VoIP), a higher interaction for VoIP streams can be obtained if the *maximum request size* is set as short as possible. Here a value of 6 ATM cells for this parameter was found to provide the fastest interaction for VoIP traffic.

5.4.5 Effects of buffer capacity

In most CATV networks, there are circumstances in which the externally offered load is larger than can be handled even with optimal transmissions. In these cases, if no measures are taken to restrict the entrance of traffic into the network, queue sizes at a station's buffer will grow and packet delays will increase, possibly violating maximum delay specifications. Furthermore, as queue sizes grow indefinitely, the buffer space at some stations may be exhausted. When this happens, some packets arriving at these stations will have to be discarded and later retransmitted, thereby wasting communication resources.

This analysis examines to what extent traffic congestion of CATV networks can be controlled by restricting the station buffer capacity in order to provide considerable low delays for the support of bursty applications (e.g. Internet traffic) and isochronous interactive services (e.g. Voice over IP traffic, audio and video). For this analysis, a mixed traffic configuration was chosen. The mean data rate per active station was set to 41.7 kbps, which consisted of 32 kbps Internet traffic and 9.7 kbps VoIP traffic, as introduced in Section 5.3.3. In order to demonstrate the effect of the buffer capacity, six different bounds were analysed (50, 100, 300, 500, 1000 and 3000 ATM cells). The simulations were performed using only the *splitting tree algorithm*.

It can be noticed in Figure 5.13 that the access delays increase as the maximum buffer capacity builds up. The results shown in this figure indicate that the maximum number of VoIP streams supported is 40 (which produce 54% of the link capacity), with mean access delays below 50 ms, regardless of the buffer size. On the other hand, buffer limits holding up to 100 ATM cells (38 Kbits of data) yield acceptable access delays (under 1 second) for the transmission of Internet traffic, with an offered load up to \approx 67% of the channel capacity, supporting up to 50 IP streams.

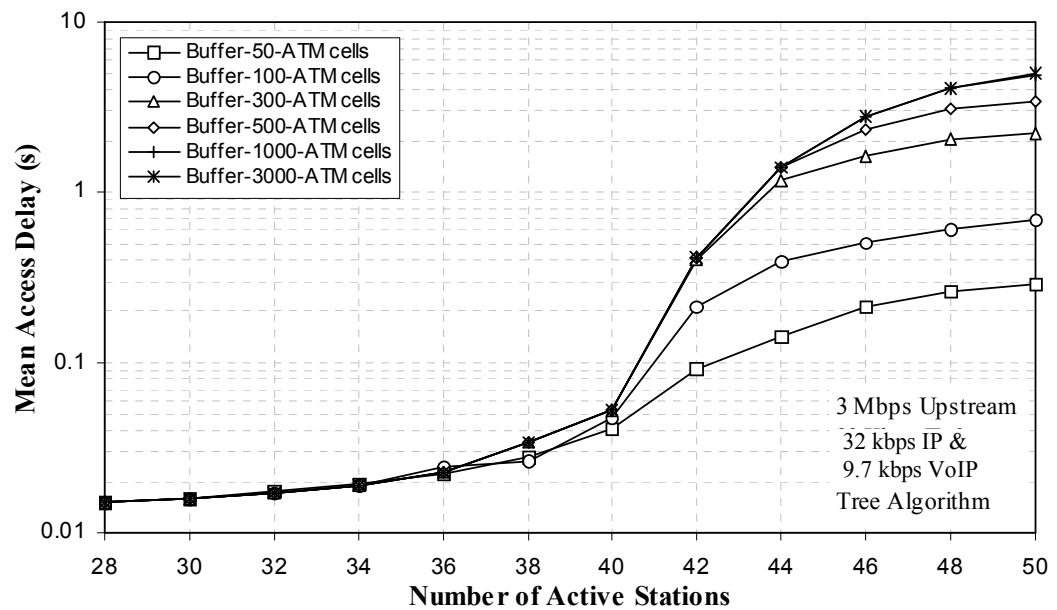


Figure 5.13 – Mean access delay vs. No. of active stations for increased buffer size.

In terms of buffer pooling capacity, Figure 5.14 indicates that beyond an offered load of 53% of the cc , the station's buffers start building up considerably up to its maximum holding capacity. With large buffer sizes (e.g. of 1000 or 3000-ATM cells) and with an offered load of $\approx 67\%$ of the link capacity, a considerable amount of bandwidth (over 10% of the cc) was held in the station's buffers.

We have found, on the other hand, that by using small buffer sizes (e.g. of 50 or 100-ATM cells) only a small fraction (below 1%) of the maximum channel capacity was held in the station's buffers. These results are more convenient for the support of VoIP streams, because packets are transmitted faster with a small buffer size than with a large buffer.

However, having a small buffer size may result in the following drawbacks. The first drawback is that when a station with no available buffer space receives a packet, it will be enabled to discard the packet, which may result in a reduction in the service quality if the number of discarded packets is considerably large. The second drawback is that discarded packets may cause a waste of bandwidth when they are retransmitted. Therefore, it is necessary to estimate the number of packet discarded per unit of time.

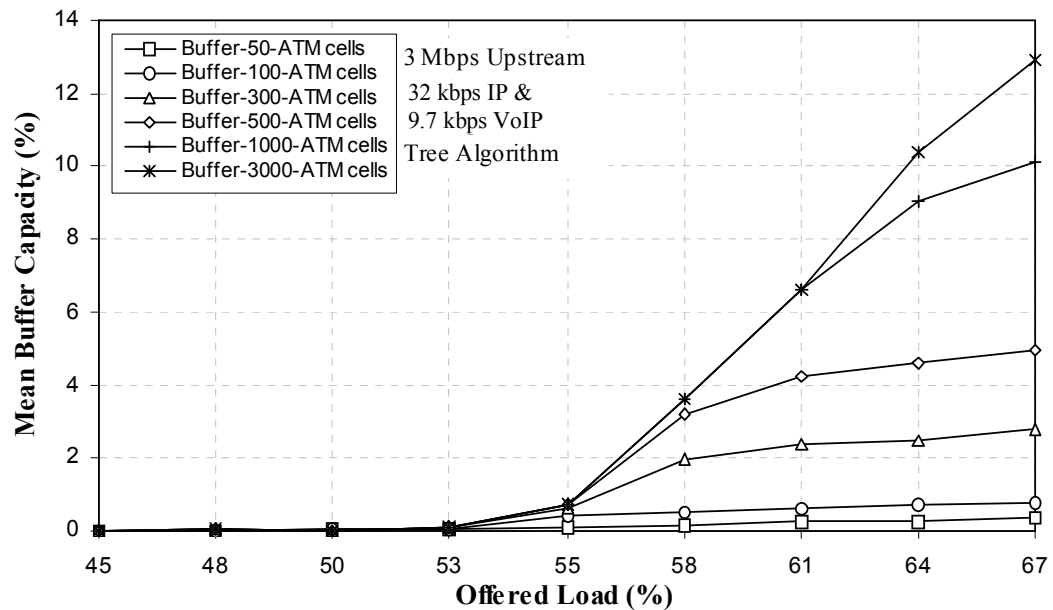


Figure 5.14 –Buffer capacity vs. offered load for increased buffer size.

Figure 5.15 shows the average number of packets that each station discarded. It can be appreciated that with an offered load beyond 53% of the link capacity, the shorter the buffer capacity the larger the number of packets discarded, but the lower the access delays produced. With a very large buffer space (e.g. 3000 ATM cells) none of the packets were discarded. However, large access delays in the order of 1 to 5 seconds were yielded, since a considerable amount of packets were held in the station's buffer waiting to be transmitted. Conversely, with a limited buffer capacity of 50 ATM cells, each station discards ≈ 3 packets per second.

Note that when the offered load is large, low access delays and buffer overflow can be controlled only by lowering the incoming traffic to the CATV network. Thus, there is a natural trade-off between giving sessions free access to the network and keeping delay at a level low enough so that interactive applications (e.g. VoIP, audio and video) are supported and retransmissions or other inefficiencies do not degrade the network performance.

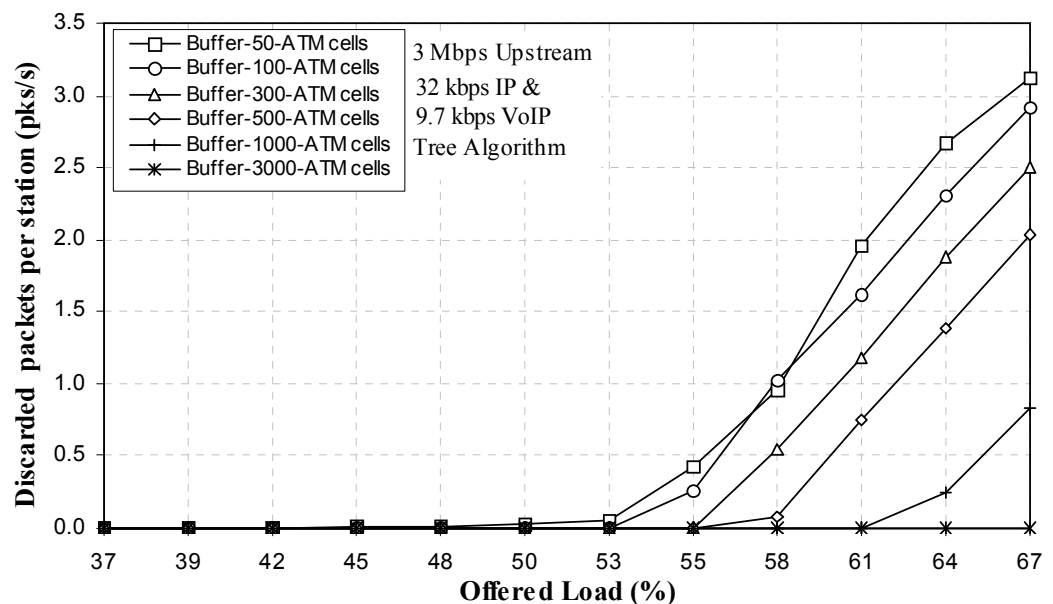


Figure 5.15 – Discarded packets vs. offered load for increased buffer size.

5.4.6 Effects of increasing the number of signalling frames

As introduced in Section 3.4, the description of upstream slots is contained in the signalling (or MCI) field of the MPEG2 TS frame. According to the DVB/DAVIC protocol specification, these frames with signalling information should be transmitted at least every 3 ms, when the upstream data rate is 6.176, 3.088 or 1.544 Mbps. The function of these signalling frames is to co-ordinate the usage of the upstream slots, assign access modes, and indicate whether the reception of contention-based slots was successful.

In this section we discuss the maximum gain in system performance that can be obtained by increasing the number of MPEG2 frames with signalling information that can be transmitted within the 3 ms period.

In a 3.088 Mbps upstream channel, the maximum number of MPEG2 frames (with signalling information) that can be transmitted within the 3 ms period is either 1 or 2. In a 6.176 Mbps upstream channel, this number is 1, 2 or 4. For this analysis, an Internet traffic configuration will be considered. The mean data rate per active station was set to 32 kbps (as defined in Section 5.3.1). The simulations were performed using both CRAs.

Results presented in Figure 5.16 indicate that by increasing to two the number of MPEG2 frames (with MCI information) within the 3 ms period (represented by the label ‘*MCI-2*’ in Figure 5.16) a reduction in access delay of approximately of 4.4 and 4.8 ms can be obtained for the *exponential backoff algorithm* and the *splitting tree algorithm*, respectively. This reduction can only be obtained with traffic loads under 47% (45 active users) and 53% (51 active users) of the *cc* for the *exponential backoff algorithm* and *splitting tree algorithm*, respectively.

However, for high traffic loads, lower access delays are produced with one signalling frame in every 3 ms period (represented by ‘*MCI-1*’). This change in access delay is because as the network load becomes higher than the point of saturation, there is an increased risk of collision of reservation requests with ‘*MCI-2*’ (according to simulation results), which degrades slightly the overall system performance as shown in Figure 5.16.

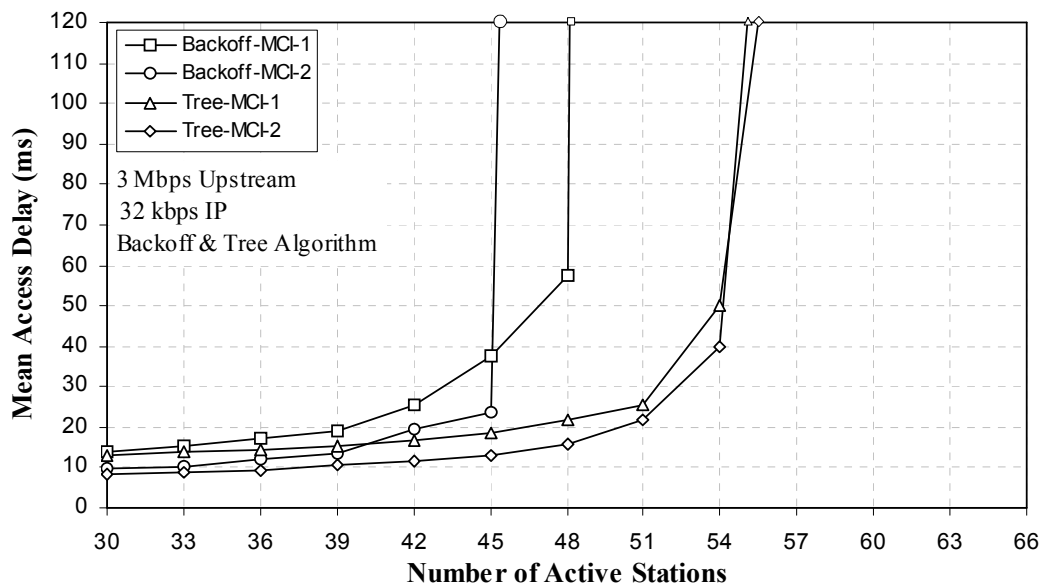


Figure 5.16 – Mean access delay vs. No. of active stations for different MCI values.

On the other hand, a loss of 6% in terms of system throughput was seen with the *exponential backoff algorithm* for ‘MCI-2’, as depicted in Figure 5.17. The *splitting tree algorithm* did not provide a loss for ‘MCI-2’, because this algorithm resolves collisions more efficiently than the *exponential backoff algorithm*, regardless of the number of MPEG2 frames (with MCI information) within the 3 ms period. This inefficiency of the *exponential backoff algorithm* results in a reduction in system throughput.

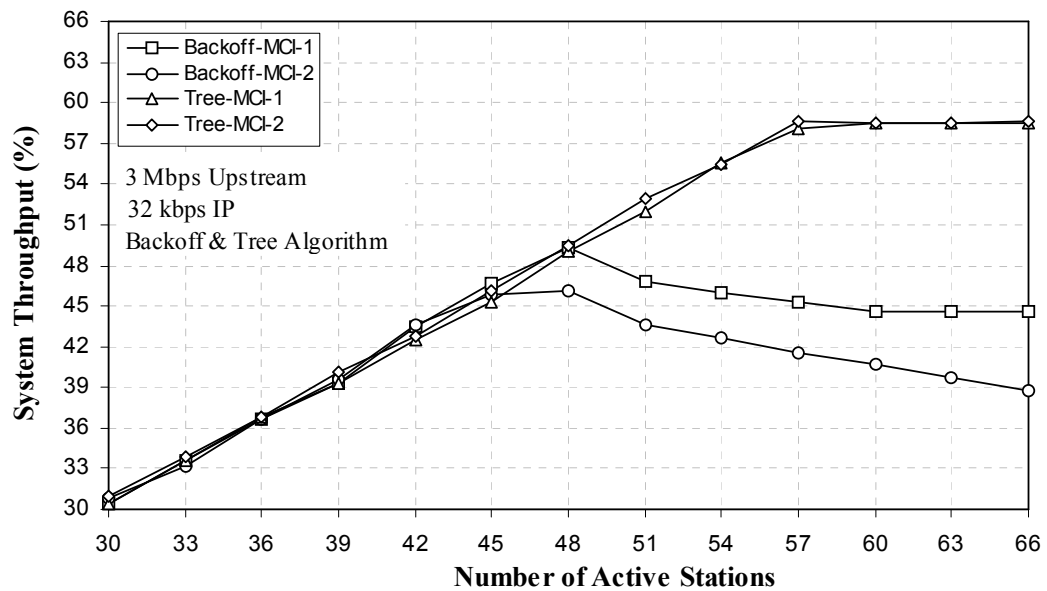


Figure 5.17 – System throughput vs. No. of active stations for different MCI values.

5.4.7 Effects of packet size for single and multiple node scenarios

Another critical parameter in the performance of any computer network is the packet size. From results in Section 4.5, packet size was proved by both analytical model and simulation (Figure 4.18) to have a significant impact on system performance, especially when small and large packets are considered for data transmissions.

5.4.7.1 Single node scenario

This analysis addresses the issue of maximum throughput per station that the communication protocol can achieve. For this particular case, a single node scenario network is considered and simulations were run for different Ethernet packet sizes (from 64 to 1518 bytes). Tests were performed with the *exponential backoff algorithm*. Results for the *splitting tree algorithm* reported the same system performance, because with only one station collisions are avoided. In order to obtain the maximum achievable throughput, the station was set to produce the maximum number of packets that it could receive from an Ethernet network, which was a 10 Mbps constant stream with variable number of packets per second depending on the packet size.

The system throughput and upstream channel utilisation versus the packet size generated is shown in Figure 5.18. The results revealed that the communication protocol is highly inefficient. The maximum throughput does not exceed 32% (=994 kbps) of the link capacity using maximum Ethernet packet size, even though the offered load was a constant 10 Mbps stream.

In this analysis it was found that the *scheduler-look ahead* is the major delay element in the transmission cycle, which dramatically reduces the throughput.

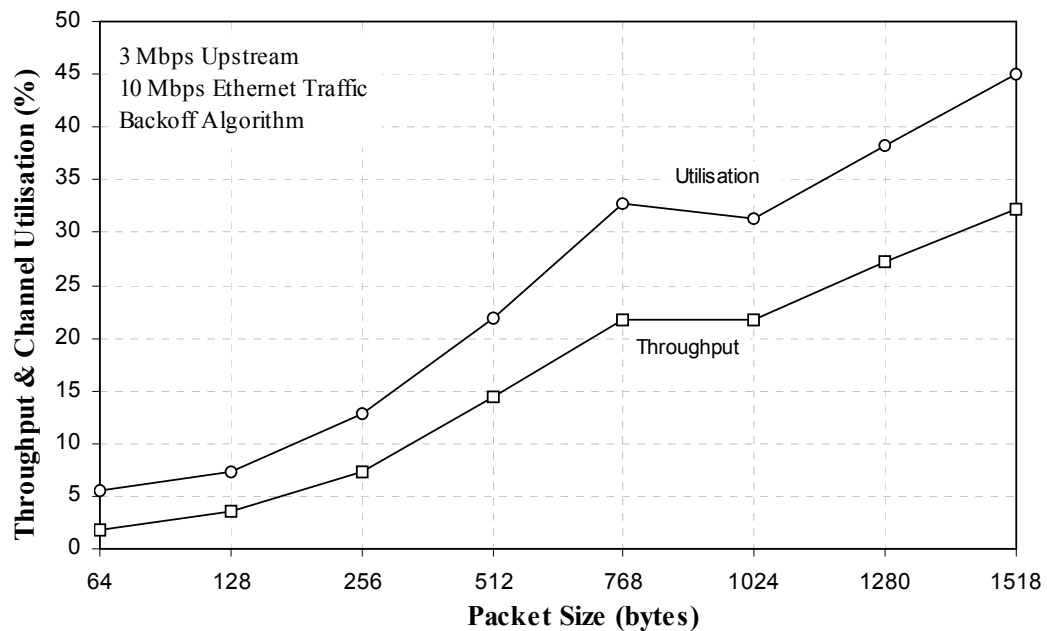


Figure 5.18 – System throughput/utilisation vs. packet size.

For example, in the worst-case scenario, when packet sizes of 64 bytes are transmitted, the maximum throughput that a single node can achieve resulted as low as 56 kbps, which corresponds to 1.84% of a 3.088 Mbps upstream channel.

A comparison between the throughput in kbps and the throughput in packets per second is presented in Figure 5.19. The increase in packet size compensates for the decrease of throughput in terms of packets per second. We can appreciate that after a 768-byte packet size, there is a considerable fall in the number of packets transmitted per second. This is reasonable, since the signalling frame only describes up to 18 slots per 3 ms period, and when all slots are used for data transmission, a packet size up to

$$18\text{Slots} \cdot 48\text{bytes}_{\text{payload}} = 864\text{bytes}$$

can be transmitted in one signalling frame. Larger packet sizes require two signalling frames. Regardless of the packet size, two more signalling frames are required for contention access, which results in $1/(3MCI_{frames} \cdot 3ms) = 111$ frames transmitted per second for packet sized under 864 bytes (1.8% deviation for simulation results) and $1/4 \cdot 3ms = 83$ frames transmitted per second for packet sized higher than 864 bytes (1.2% deviation for simulation results).

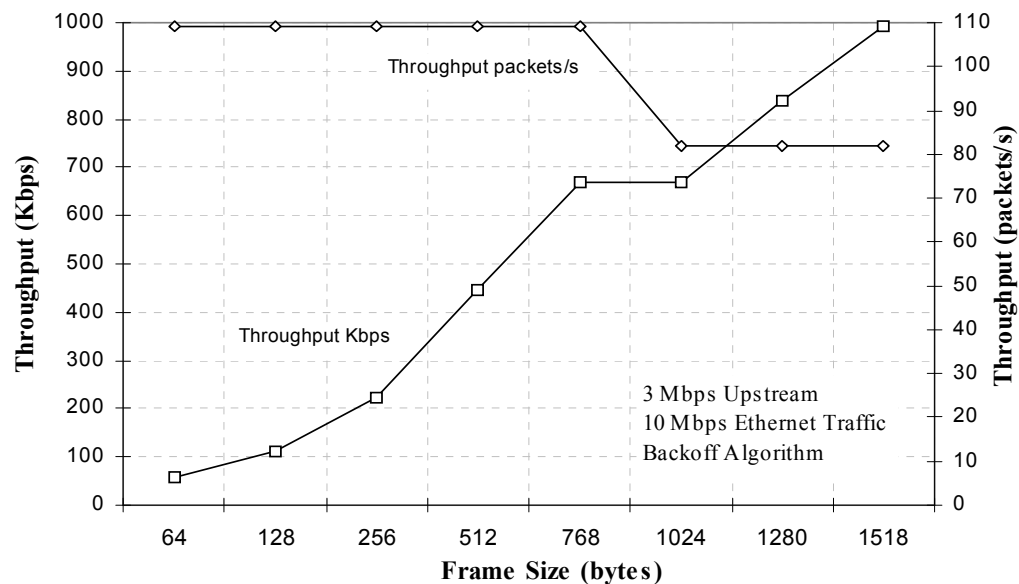


Figure 5.19 – System throughput in kbps and packets/s vs. packet size.

As a final remark for this analysis, it was seen that the reservation request mechanism of the DVB/DAVIC protocol was not optimised, since a poor performance is obtained when short messages are being transmitted. Therefore, it is evident that a performance optimisation or the incorporation of new mechanisms should be considered.

5.4.7.2 Multiple node scenario

In this section, we now consider a multiple-node network scenario to study the maximum performance gain that can be obtained by increasing the packet size. Here, isochronous streams of 32 kbps (produced by each station) were analysed and the inter-arrival rates were set according to the packet size. The simulations were carried out using only the *splitting tree algorithm* because of its increased performance over the *exponential backoff algorithm*. The packet sizes considered were 64, 128, 256, 512, 1024 and 1518 bytes that cover the whole range from the minimum to the maximum Ethernet packet size.

Figures 20 to 22 present a general performance overview when the packet size is increased. The delay and throughput results revealed that for small packet sizes, the communication protocol is also highly inefficient.

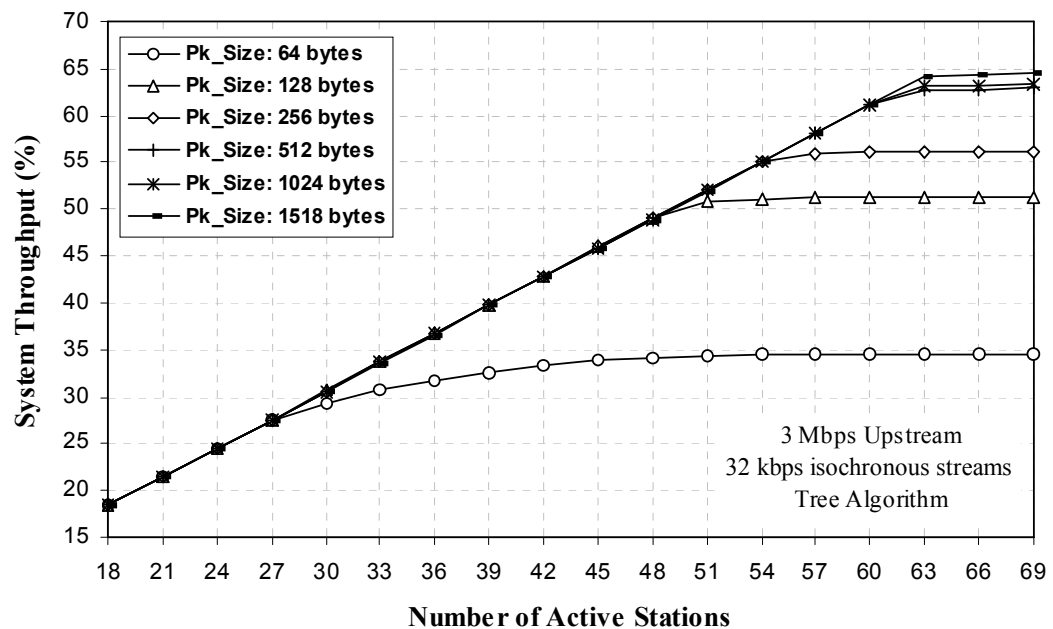


Figure 5.20 – System throughput vs. No. of active stations for increased packet size.

The maximum throughput (see Figure 5.20) yielded for minimum Ethernet packet size was as low as $\approx 34\%$ of the link capacity, which can only support up to 25 stations before delays became extremely large for the delivery of isochronous streams (as shown in Figure 5.21). Increasing the packet size can significantly improve the performance with maximum sustainable throughput up to $\approx 64\%$ of the cc . This increase in system performance was recorded for packet sizes of 512 bytes. Larger packet sizes have only marginal increase in system throughput and a small increase in packet access delay.

From Figure 5.21, for offered loads less than 47% of the cc (produced by up to 45 stations), access delays for short packet sizes (under 1024 bytes with the exemption of the 64-byte packet size) were relatively shorter than for larger packet sizes (e.g. 1024 and 1518 bytes). This is mainly due to the larger transmission delay incurred by long packets. On average, when a station has gained access to the reservation access region, it takes 1 signalling frame of 3 ms to transmit a packet up to 512 bytes and 2 signalling frames to transmit a packet of 1024 or 1518 bytes. Hence, a minimum difference of 3 ms between short and large packet sizes can be noticed in Figure 5.21.

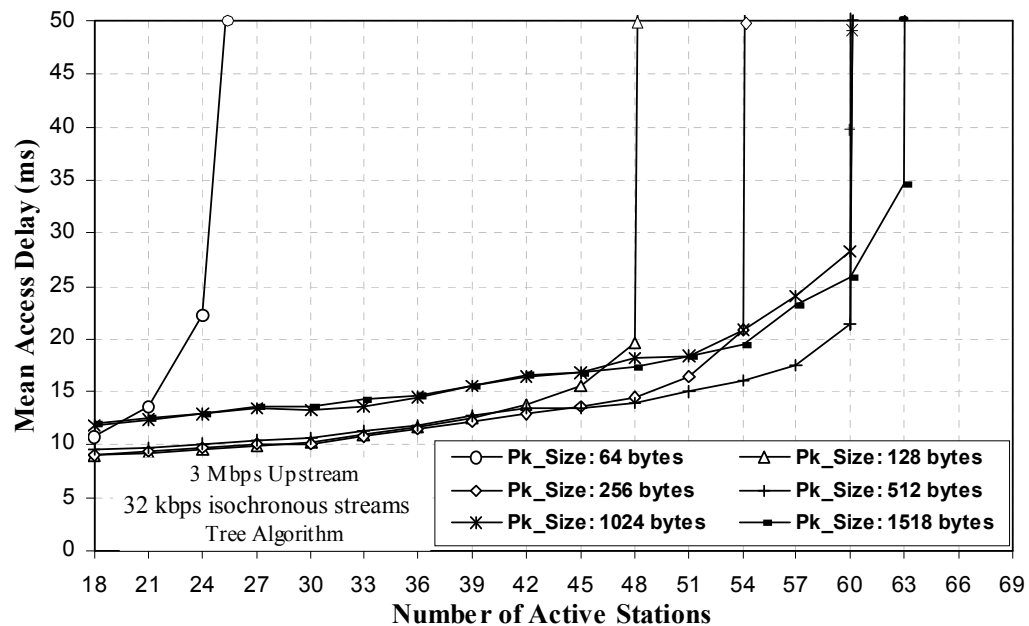


Figure 5.21 – Mean access delay vs. No. of active stations for increased packet size.

However, with higher offered loads (above $\approx 47\%$ of the cc), short packets cause more requests to be sent in the contention access region, resulting in more collisions (less throughput). This effect is better appreciated in Figure 5.22. This figure shows the percent of the bandwidth wasted by reservation requests, which resulted in collision. It is evident that with a 64-byte packet size, as the offered load increases, a considerable amount of bandwidth is wasted. For instance, with offered loads over 30% of the cc , approximately up to 8% was consumed only by collisions, which reduced considerably the bandwidth available for data packets. As a result of this reduction in the reservation access region, the stations are unable to transmit all data packets due to the high risk of collision among reservation request, which cause the station's buffers to build up almost unbounded. Thus, very large packet access delays are evident with an offered load as low as 28% of the cc (produced by 27 stations) as indicated in Figure 5.21.

With a 128-byte packet size, there was also a sizeable amount (up to $\approx 4\%$) wasted by collisions. However, with large packet sizes, (e.g. 1024 and 1518 bytes) the bandwidth wasted by collisions only represented a tiny portion of the link capacity (under 0.05% of

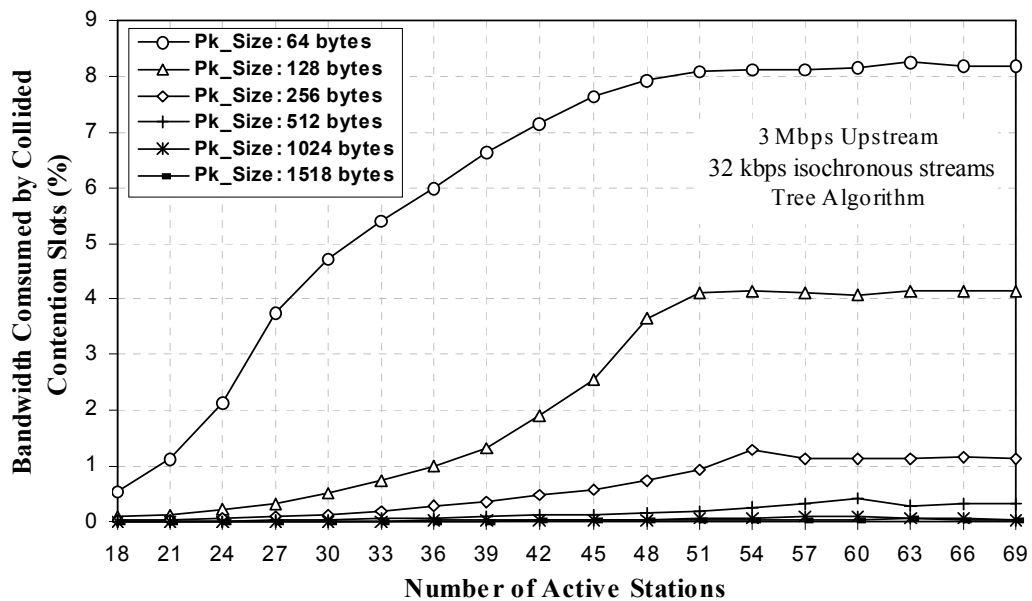


Figure 5.22 – Bandwidth consumed by collisions vs. No. of active stations for increased packet size.

the *cc*).

As a final analysis, Table 5.5 presents a summary of the maximum system performance and bandwidth characterisation for the six different packet sizes considered. In this table we can appreciate that regardless of the packet size, the maximum system utilisation is around 89-90% of the link capacity, with a maximum deviation between throughput/utilisation of $\approx 55\%$ and $\approx 24\%$ for minimum and maximum Ethernet packet sizes, respectively. There are several factors that may account for these deviations.

A major component is due to the relatively large amount of overhead involved at MAC (ATM encapsulation) and PHY (forward error correction and synchronisation) layers. This can be as large as 34% and 23% of the *cc* when delivering small and large packets of 64 and 1518 bytes, respectively.

Another factor is the amount of bandwidth used for the transmission of successful reservation requests. For 64-byte packets, this amount of bandwidth (of $\approx 12\%$) represents a significant portion of *cc*. However, for 1518-byte packets, this bandwidth (of $\approx 1\%$) represents a tiny fraction of the link capacity. This difference is because short messages require proportionally more reservation requests in order to transmit the same amount of data than with larger messages. For instance to transmit 32 kbps, a station will send 75 requests per second with a 64-byte packet size, in comparison to 2.64 requests per second with a 1518-byte packet size.

Table 5.5 - Maximum system performance and bandwidth characterisation.

Bandwidth Characterisation (%)	Packet size (bytes)					
	64	128	256	512	1024	1518
Maximum throughput	34.5	51.3	56.2	62.9	63.4	64.6
Maximum utilisation	89.1	90.1	90.5	89.8	88.9	88.5
Deviation	54.6	38.8	34.3	26.9	25.5	23.9
Supported streams	25.0	48.0	54.0	60.0	60.0	63.0
Request collided	8.3	4.1	1.3	0.4	0.1	0.1
Request successful	11.8	8.5	4.6	2.6	1.3	0.9
MAC&PHY overhead	34.5	26.2	28.4	23.9	24.1	23.0
Bandwidth unused	10.9	9.9	9.5	10.2	11.1	11.5

The next significant factor, as stated earlier, is the bandwidth wasted by unsuccessful (or collided) reservation request, which can result in a significant amount when short messages are delivered.

Finally, the other obvious performance issue is the inability of the system to reach the maximum channel capacity ($\approx 100\%$). The reason for this is the minimum number of contention slots allocated per signalling frame, which was set to 2 in this scenario (and accounted for $\approx 11\%$ *cc*).

For 1518-byte packets, some additional CSs were added to the signalling frame because of the inability of the headend to allocate all reservation slots for the transmission of data packets. Hence, the total bandwidth allocated to the contention access region (registered by simulations) accounted for $\approx 12\%$ of the link capacity, where only $\approx 1\%$ was used for successful and unsuccessful requests and the other bandwidth, of 11%, remained unused.

In general, the unused bandwidth can be reduced by decreasing the number of CSs allocated per signalling frame. However, this may be inefficient for short packet sizes, and particularly when the *exponential backoff algorithm* is used, as examined in the Sections 5.4.2 and 5.4.4.

We have also carried out a detailed performance analysis of isochronous streams (at 8kbps, 16 kbps, 32 kbps, 64 kbps and 128 kbps) for a 6 Mbps upstream channel and six different packet sizes (64, 128, 256, 512, 1024 and 1518 bytes). This analysis is reported in [85] and presented in Figure E.2 of Appendix E.

5.5 Conclusions

In this chapter, four novel and topical traffic types were used to analyse the fundamental performance characteristics of DVB/DAVIC compliant networks with the use of a simulation model. This data facilitates further optimisations and enhancements in the protocol development of DVB/DAVIC. We studied key issues on system performance using both contention resolution algorithms, such as: packet access delay and maximum throughput that can be achieved by a single station; packet size scalability, optimisation

of the maximum length of a message (measured in ATM cells) that can be transmitted using either contention access or a single reservation request; buffer capacity; scalability of the upstream channel in terms of increased offered load and node population.

The major factors affecting the system performance were seen to be the length of the packet being transmitted for delivery. Specifically it was demonstrated that regardless of the offered load and length of the packet size, a single station cannot achieve throughput higher than 32% of the maximum upstream channel capacity, even when the *splitting tree algorithm* is used. However, this figure can be as low as 1.8% when delivering minimum Ethernet packets (64 bytes) due to an excessive number of reservation requests, collisions, retransmissions and DVB MAC and PHY protocol overheads. Results for channel utilisation were higher than for throughput results with a range of 54% to 23% for minimum and maximum Ethernet packets, respectively. Protocol overhead was shown to be a major source of performance inefficiency. This can be as high as 35% of the upstream channel capacity for 64-byte packets.

Effects of contention access for data transmission revealed that by transmitting relatively short messages (in the order of 1-ATM cell) using only contention access, the system throughput can be increased and a quick interaction can be provided. Results for effects of reservation request size suggested that the *maximum request size* should be set as large as possible if the upstream channel is only used for Internet traffic. However, if the upstream channel supports the transmission of both traffic types (IP and VoIP), a higher interaction for VoIP packet can be obtained if the *maximum request size* is set as short as possible.

Results for the buffer capacity indicated that traffic congestion could be controlled by lowering the incoming traffic to the CATV network. Thus, there is a natural trade-off between giving sessions free access to the network and keeping delay at a level low enough so that interactive applications (e.g. VoIP, audio and Video) are supported and retransmission or other inefficiencies do not degrade the network performance.

Scalability of the upstream channel revealed that there is a gradual increase in the performance of the system with respect to increasing the cable population and offered load. Results also revealed that there are distinct saturation points after which

throughput and utilisation do not increase, meanwhile packet delays become unbounded. For all of these analyses, the maximum channel utilisation remained at approximately 90% of the link capacity.

Careful design should not allow a network population and offered load to exceed saturation points, since this would result in uncontrolled access delay.

Finally, given all these special characteristics of the protocol and the fact that isochronous applications (e.g. VoIP) are very sensitive to delay variations, it is necessary to further optimise the protocol.

Chapter 6

OPTIMISATION OF CRA ALGORITHMS USING ADAPTIVE CSA

6.1 Introduction

Contention resolution algorithms (CRA) define the set of rules used to resolve collisions. They play a vital role in the performance of a multi-access reservation protocol. This is because the faster they resolve collisions, the lower the access delay will be and the higher the system throughput will become. Initially, CRAs gained much interest in the early 1970s for usage in the transmission of radio packets, and especially during the development of the ALOHANET project [1]. Two major candidates were defined, ALOHA-based algorithms like ‘*exponential backoff* and *p-persistence*’ and *splitting tree algorithm*, as reported in Section 1.2.3.2. Since then, much research has been devoted to devising efficient contention resolution mechanisms for multi-access media for Local Area Networks (LANs), Metropolitan Area Networks (MAN), satellite networks, radio networks and CATV networks [41].

In this chapter, we study in detail performance, optimisation and implementation issues for the contention resolution algorithm adopted by the DVB/DAVIC protocol. We pay particular attention to the dynamics and operation of the *exponential backoff algorithm* and the *splitting tree algorithm*. Furthermore, special emphasis is paid to the design of adaptive mechanisms, called Contention Slot Allocators (CSA), which dynamically adjust the bandwidth used for contention access, significantly increasing the system performance when different bounds are considered.

6.2 Contention Slot Allocator

As introduced in [61], [96] and [98], the authors have pointed out that the performance of a multi-access reservation protocol depends more on the overall framing structure and the capacity assigned to the contention and reservation access modes than the details of the CRA adopted. In this section we focus on the performance impact when the reservation capacity is dynamically adjusted by the use of a *slot allocation mechanism*.

6.2.1 How many contention-slots per signalling frame?

After the INA has scheduled a number of reservation slots (RSs) to carry data packets, any number of contention slots (CSs) may then be allocated. When the load of the networks is low, very few CSs are required. On the other hand, since the load is low, there will be unused slots that could be used as CSs. As the offered load increases, depending on the length of the packets, more slots will need to be allocated as CSs.

The solution to determine how many CSs is rather simple: allocate all slots that are not being used for data as CSs. At low traffic loads, many more CSs will be allocated than those required. The surplus of CSs significantly decreases the risk of collision of reservation requests to a very low level, which in turn reduces the access delay for data packets.

This algorithm is a self-regulating mechanism, since if the number of CSs are too low, requests will not reach the INA and as a result more CSs will be automatically allocated. On the contrary, if the number of CSs is too high, more successful requests will reach the INA and the number of empty slots that can be used as CSs will decrease to a minimum threshold value, which guarantees that at least a few slots will be reserved for contention access. Thus, the performance of the network is highly dependant on the minimum number of CSs allocated in each signalling frame. In [61], [96] and [98] the authors did not consider the minimum number of contention slots that should be allocated in each signalling frame, since this would have led them to a low performance estimation. In the next section, we present a performance analysis for different values of the '*Minimum number of CSs per signalling frame*' parameter using both CRAs.

6.2.2 Simple-CSA: Performance optimisation of the use of contention slots

In order to demonstrate the effects of the use of a simple contention slot allocator, an Internet traffic situation was analysed. The mean data rate per active station was set to 64 kbps. The simulations were performed using both contention resolution algorithms. The *Min. No. of CSs per signalling frame* was ranged from 1 to 7 CSs for the *exponential backoff algorithm*. For the *splitting tree algorithm* this parameter was ranged from 0 to 6 CSs.

6.2.2.1 Exponential backoff algorithm performance using a Simple-CSA

We have used Equation 4.43, (introduced in Section 4.4.3) to estimate the maximum upstream throughput for the *exponential backoff algorithm*.

$$S_{\max_exp_backoff} = \frac{Pk_{size}}{Pk_{slots} + eCSs}$$

The term ‘*eCSs*’ in this equation, indicates that on average we need ‘*e*(=2.718)’ contention slots to transmit a reservation request successfully, in order to get the optimum system throughput, as suggested in [10]. Idles occur with a probability of $1/e \approx 0.368$, success occurs with a probability of $1/e$ and collisions occur with a probability of $1-2/e \approx 0.264$. Thus in summary, 1 CSs for idle, 1 CSs for success and 0.718 CSs for collisions are needed.

On average the packet size for Internet traffic is 368.1 bytes [53], and the number of data slots requested per packet is 8.3. So using Equation 4.43 we have,

$$S_{\max} = \frac{368.1}{8.3 * 64 + 2.718 * 64} = 52.2\% \approx 52\%$$

Therefore, the maximum theoretical system throughput that can be achieved is $\approx 52\%$ when large numbers of stations are transmitting.

From simulation results presented in Figures 6.1 and 6.2, it is apparent that with at least 3 or 4 CSs per signalling frame (represented by CSs-3 and CSs-4 in these figures), not only the lowest access delays are yielded, but also the highest system throughput is achieved.

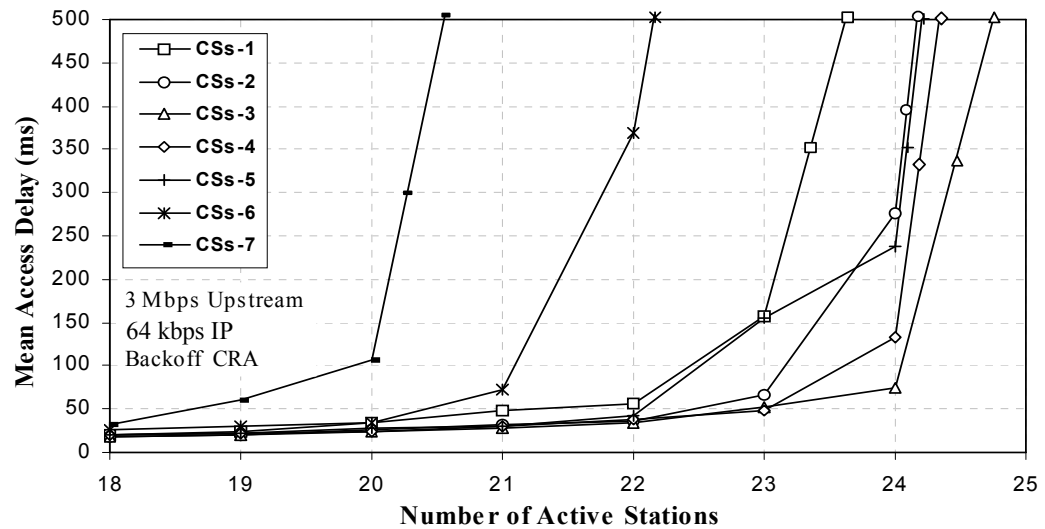


Figure 6.1 – Mean access delay vs. No. of active stations.
Exponential backoff algorithm with a Simple-CSA.

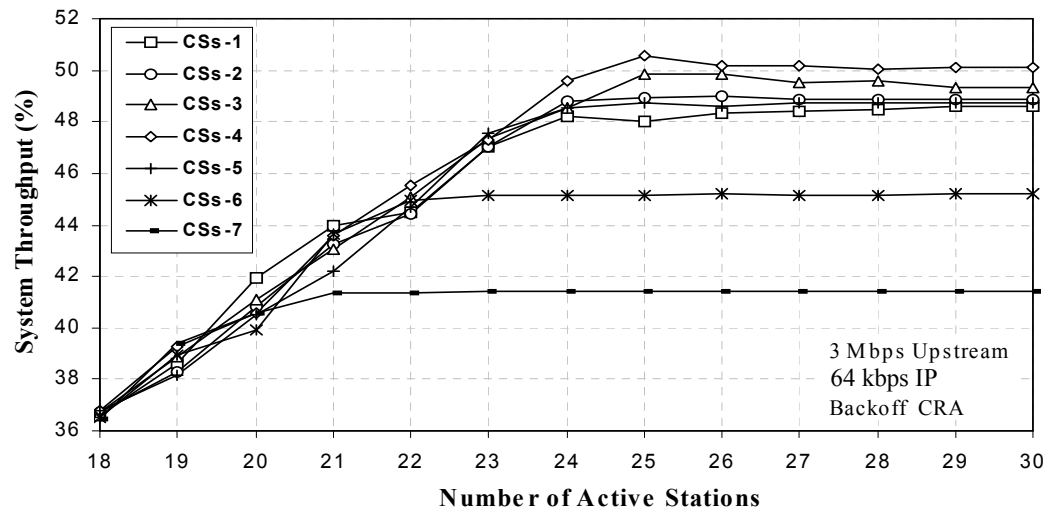


Figure 6.2 – System throughput vs. No. of active stations.
Exponential backoff algorithm with a Simple-CSA.

Larger values, (above 4 CSs per signalling frame) cause a waste of bandwidth, since the CSA tends to allocate more contention slots than those needed. This in turn results in a reduction in system throughput. Furthermore, this causes increased access delay as stations transmit reservation requests more frequently. Such requests at the headend have to wait longer for grant since there is a reduction in the bandwidth assigned for reservation access.

From Figure 6.3, it is evident that on high traffic offered loads, beyond 50% of the cc , (generated by at least 24 stations), the average number of contention slots needed per request ranged from 3 to 3.5 slots, when the minimum number of CSs per signalling frame was below 5 slots.

In Figure 6.2 we can appreciate that the maximum system throughput achieved was approximately 51% (for CSs-4) with an offered load about 52% of the channel capacity yielded by 25 stations. This difference of approximately 1% between theoretical and simulation results was attributed to the fact that the CSA assigned on average 3.08 contention slots per successful request (see Figure 6.3), instead of the optimum value of 2.718. Furthermore, the average Pk_{size} (registered by simulations) was 374 bytes with 8.42 slots requested per packet, which made the difference of 1%.

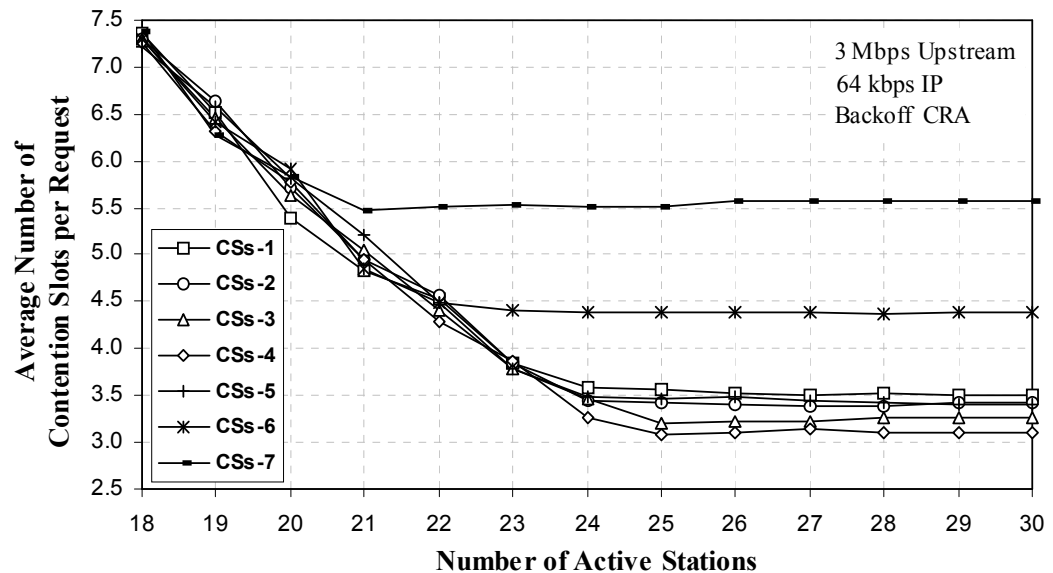


Figure 6.3 – Average No. of CSs per reservation request vs. No. of active stations.
Exponential backoff algorithm with a Simple-CSA.

6.2.2.2 Splitting tree algorithm performance using a Simple-CSA

For this algorithm, the maximum theoretical bound for the system throughput can now be obtained by using Equation 4.44, in which reservation requests are transmitted using minislots (MSs) of 21 bytes. The maximum theoretical system throughput that can be achieved with the *splitting tree algorithm* is $\approx 63\%$ of the channel capacity, as shown below.

$$S_{\max} = \frac{Pk}{Pk_{\text{slots}} + eMSs} = \frac{368.1}{8.3 * 64 + 2.718 * 21} = 62.6\% \approx 63\%$$

From simulation results shown in Figures 6.4 to 6.6, it is evident that the highest system performance is yielded with at least 1 CSs per signalling frame. This performance is achieved because a contention slot is reserved in the next signalling frame when a collision happens. This slot (which will be divided into three minislots) is then used among stations that caused the collision. By reserving an additional contention slot after a collision, apart from the *Min. No. of CSs per signalling frame*, the *Contention-Resolution-Grant Cycle* (described in Section 3.4.1) is considerably decreased, thus resulting in a reduction in access delay.

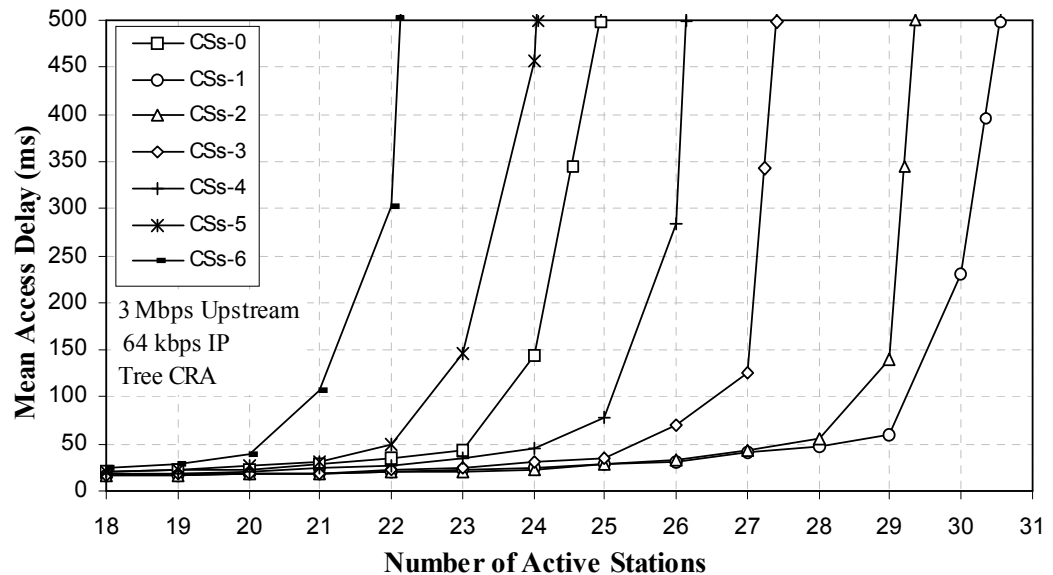


Figure 6.4 – Mean access delay vs. No. of active stations.
Splitting tree algorithm with a Simple-CSA.

Having a short value for the *Min. No. of CSs per signalling frame*, (e.g. 1 or even 2 CSs), most of the bandwidth is allocated to the reservation access region, producing an increase in the overall system performance. Larger values for this parameter (above 2 CSs) are not required, because this algorithm efficiently resolves collisions. Defining a large value for this parameter may result in many contention slots being unused, wasting bandwidth resources and therefore achieving a low system performance.

Conversely, having at least 0 CSs per signalling frame may cause (on heavy loads) those stations with new incoming packets to wait until all the previous reservation requests have been granted. Then, compete for contention access among all waiting stations. This increases considerably both the risk of collisions and the *CRGC*, resulting also in a reduced performance.

From Figure 6.5 the maximum system throughput obtained by using this algorithm was approximately 62%. The remaining bandwidth of 0.6%, (to get the 62.6% as estimated) was attributed, in this case, because the mean Pk_{size} (registered by simulations) was of 374 bytes with 8.42 slots requested per packet.

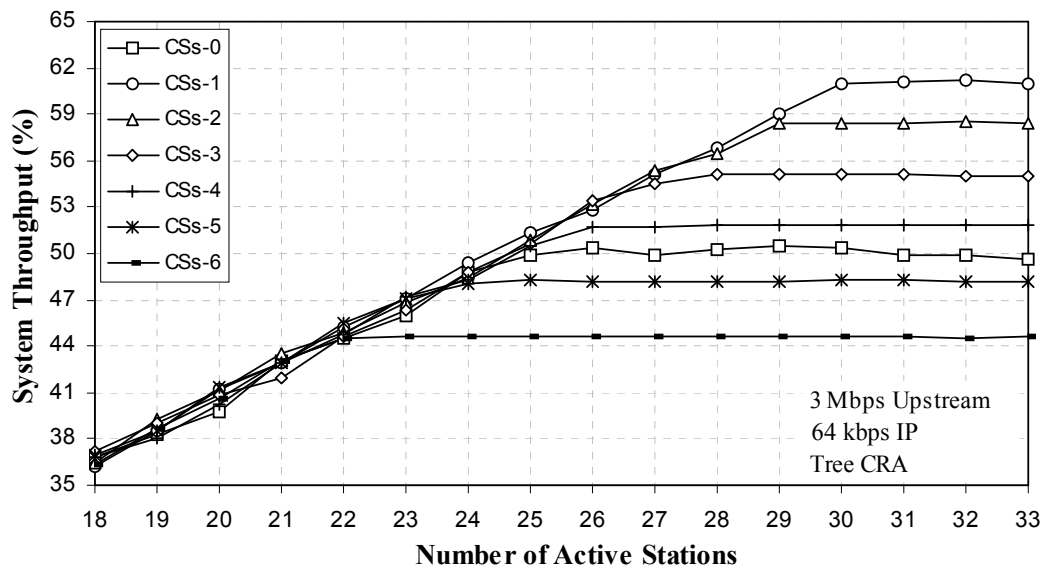


Figure 6.5 – System throughput vs. No. of active stations.
Splitting tree algorithm with a Simple-CSA.

In addition, the CSA assigned on average 1.07 CSs (or 3.21 MSs) per successful request as shown in Figure 6.6, instead of the optimum value of '2.718 MSs \approx 0.89 CSs'.

As a general observation, results presented for both CRAs showed that the optimal system performance depends on the *Min. No. of CSs per signalling frame* allocated. For the *exponential backoff algorithm* we found that the optimum value for this parameter was 4 CSs for a maximum system throughput and 3 CSs for lower access delays. For the *splitting tree algorithm* this value appeared to be of 1 CSs.

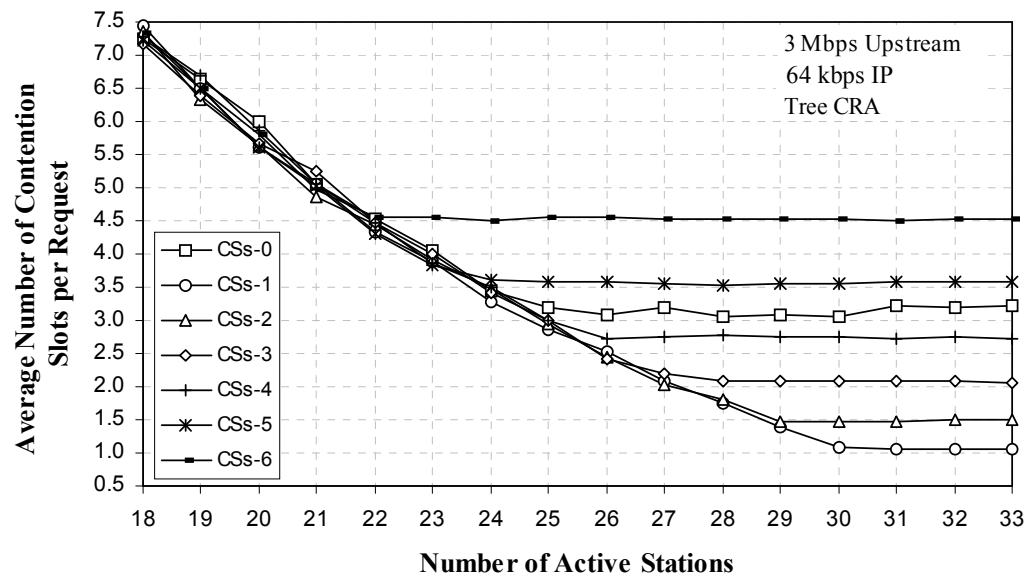


Figure 6.6 – Average No. of CSs per reservation request vs. active stations.
Splitting tree algorithm with a Simple-CSA.

However, for different traffic types (e.g. Internet at 32 kbps per station and mixed at 41.7 kbps traffic as summarised in Table 6.1), for the *exponential backoff algorithm*, the optimal system performance, was found with at least 4 and 5 CSs per signalling frame for Internet and mixed traffic, respectively. For the *splitting tree algorithm* this value (1CSs) was unchanged. In the following section we introduce two enhanced CSAs, which will optimise further the maximum system performance.

Table 6.1 - Maximum system performance for Internet and mixed traffic scenarios.

CRA	Traffic Type	Min.CSs per Sign. Frame	Maximum Throughput (%)	Mean Access Delay (ms)	CSs per Request	Offered Load (%)	Active Stations
<i>Exponential Backoff Algorithm</i>	Internet	2	47.9	236	3.6	48.4	48
	Internet	3	47.9	602	3.62	49.4	48
	Internet	4	49.3	57	3.31	49.4	48
	Internet	5	48.4	178	3.51	48.8	48
	Mixed	2	43.5	1435	3.26	45.7	34
	Mixed	3	43.8	926	3.18	45.7	34
	Mixed	4	44.5	764	3.05	46.1	34
	Mixed	5	45.23	179	2.88	45.5	34
<i>Splitting Tree Algorithm</i>	Internet	1	61.01	155	1.09	61.1	60
	Internet	2	58.06	189	1.54	58.5	57
	Internet	3	55.07	153	2.08	55.1	54
	Internet	4	51.6	257	2.8	52.2	51
	Mixed	1	57.27	282	0.93	57.9	43
	Mixed	2	55.3	98	1.27	55.5	41
	Mixed	3	52.4	331	1.63	53	39
	Mixed	4	48.99	191	2.15	49.4	37

Shaded rows represent the configuration for optimum system performance.

6.2.3 Enhanced - CSAs

In this section two enhanced CSAs are introduced. We have called these enhanced mechanisms '*Forced-CSA*' and '*Variable-CSA*'. Such mechanisms adjust dynamically the number of CSs per signalling frame according to the current traffic load, mean packet size, mean requested slots and possible collisions. These mechanisms will improve the maximum system performance for the *exponential backoff algorithm* by sending more CSs when they are needed (and not when they are available) and by reducing the average number of contention slots needed per request, from 3.08 CSs (Figure 6.3) to a value very close to the optimum ' $e(=2.718)$ '

For the *splitting tree algorithm* a more efficient CSA is not necessary, since the system performance has already been maximised by fixing the *Min. No. of CSs per signalling frame* to 1 CSs, regardless of the traffic type and when a medium size network has been considered (under 70 stations).

6.2.3.1 Forced-CSA used with the *exponential backoff algorithm*

This mechanism is based on the dynamics of the *splitting tree algorithm*. When a collision occurs, the *splitting tree algorithm* automatically allocates a CS in the next signalling frame, which is then split into 3MSs and used only between the stations involved in the collision. The *Forced-CSA*, on the other hand, allocates a flexible number of CSs in the next signalling frame. We refer to these additional slots as forced contention slots (FSs). With this new functionality, stations competing for contention access have more chance to transmit successfully, since more contention slots are allocated when they are needed, reducing considerably packet access delays.

The idea of allocating more contention slots, in addition to the unreserved slots that are also allocated to the contention-based access region, was first reported in [96] and [98]. Here the authors introduced a new contention slots allocator, referred to as '*Forced Mini-Slots CSA*' for the IEEE 802.14 protocol. The main difference between our *Forced-CSA* and the *Forced Mini-Slots CSA*, is that the latter allocates more CSs according to the maximum throughput of the Slotted Aloha System, defined in Equation 6.1.

$$\eta_{\max} = N_a \cdot p \cdot (1 - p)^{N_a - 1} \quad (6.1)$$

where p is the retransmission probability and N_a is an estimation of the number of stations competing for a contention slot.

On high traffic loads, the *Forced Mini-Slots CSA* allocates $e(=1/\eta_{\max})$ CSs for each data message to be transmitted. Conversely on low traffic loads, it allocates less than $e(=1/\eta_{\max})$ CSs. However, our mechanism (*Forced-CSA*) allocates more CSs when a collision occurs and on high traffic loads, the average number of CSs required per requested data message approaches very close to $e(=1/\eta_{\max})$.

For the *Forced Mini-Slots CSA*, the authors found that by setting the number of forced mini-slots to 2 instead of e , good results were obtained. In our analysis, we also found that by allocating 2 FSs, after a collision, an improvement in system performance was obtained. Results for the *Forced-CSA* are presented in Section 6.2.3.3.

6.2.3.2 Variable-CSA used with the *exponential backoff algorithm*

This mechanism uses a variable slot regime algorithm in which the ratio of CSs to reservation data slots (RSs) is varied from signalling frame to signalling frame, based on the current traffic load, mean packet size and the mean of requested slots. Variable slot allocators have been used since 1998 in MAC protocols for HFC and wireless access networks. They were firstly introduced in [112] and [113]. Later on, they were reported in [41] for the IEEE 802.14 protocol. The mechanism presented here is similar to [41] with slight modifications for the DVB/DAVIC protocol. The variable number of CSs (VCSs) to be added in the signalling frame is dynamically adjusted as the headend converts the number of RSs into CSs (N_{RSs}) according to Equation 6.2.

$$N_{RSs} = \left\lceil \frac{2 \cdot MAX_DATA}{2 + Req_Size} \right\rceil \quad (6.2)$$

where MAX_DATA is the maximum number of data slots that can fit in a signalling frame (= 18 – *Min. No. of CSs per signalling frame*, for a 3.088Mbps upstream channel) and *Req_Size* is the average number of RSs that can be requested at once. VCSs can be determined from Equation 6.3.

$$VCSs = \begin{cases} 0 & \text{if } MAX_REQ \geq \alpha \cdot (MAX_DATA - N_{RSs}) \\ N_{RSs} & \text{else} \end{cases} \quad (6.3)$$

where MAX_REQ is the total number of data slots requested but not yet allocated by the headend, α is a design parameter set to 2.5 as suggested in [41]. The total number of CSs to be included in the next signalling frame is then represented by following parameters: the *Min. No. of CSs per signalling frame* (as proposed in the Simple-CSA), the variable number of CSs derived from Equation 6.3 and the unused RSs converted to CSs as recommended in [96] and [98].

6.2.3.3 Performance comparison of Enhanced-CSAs

In order to provide a complete performance comparison between the Forced and Variable CSAs, three different traffic configurations will be analysed: 64 kbps IP streams, 32 kbps IP streams and 41.7 kbps mixed traffic.

A summary of the maximum system performance produced by each CSA for Internet traffic is presented in Table 6.2. A detailed comparison for mixed traffic is shown in Figures 6.7 to 6.10.

For Internet traffic at 32 kbps per station, the best system performance was achieved by the *Forced-CSA*. Three different configurations were analysed using this scheme (as indicated in Table 6.2). The highest system throughput (about 52% of the cc) and the lowest mean access delay (of 584 ms) was achieved by both allocating 2 FSs after a collision and defining at least 2 CSs per signalling frame, at approximately 53% of the channel capacity (produced by 51 stations). In addition, the average number of CSs needed per request resulted in 2.819, which is quite close to the optimum $e(=2.718)$ with a maximum deviation under 1% of the cc between simulation and theoretical results.

Table 6.2 – Maximum system performance for Internet traffic and different CSAs.

Internet Traffic	CSA	Min. CSs Per Sign. Frame	Maximum Throughput (%)	Mean Access Delay (ms)	CSs per Request	Offered Load (%)	Active Stations
32 kbps	Simple	3	46.8	3099.98	3.96344	52.3501	51
	Simple	5	48.0	2004.82	3.57891	51.3815	51
	Forced-FSs 1	3	51.1	1284.42	2.87358	53.1611	51
	Forced-FSs 2	2	51.5	584.026	2.81989	52.8379	51
	Forced-FSs 2	3	51.0	1090.09	2.91893	53.0158	51
	Variable	2	47.9	2628.13	3.68146	53.153	51
	Variable	3	47.8	2195.96	3.6845	52.9651	51
64 kbps	Simple	4	50.6	1226.38	3.07588	52.5	25
	Forced-FSs 1	3	50.7	913.49	3.01059	51.7	25
	Forced-FSs 2	2	50.6	494.56	3.01297	51.3	25
	Forced-FSs 2	3	49.8	416.05	3.15268	50.7	25
	Variable	2	50.8	777.72	2.95227	51.9	25
	Variable	3	51.3	624.79	2.87162	52.1	25

The shaded rows present the CSA with optimum system performance.

For Internet traffic at 64 kbps per station, the highest system throughput was achieved with the *Variable-CSA* and the lowest access delay with the *Forced-CSA*. For the *Variable-CSA* two configurations were used. The maximum system throughput of approximately 51% at 52% of the channel capacity (generated by 25 stations) was found by allocating at least 3 CSs per signalling frame. For this traffic type, a slight increase in the average number of CSs needed per request (with 2.871 CSs) was registered, which resulted in a small reduction in system performance of 0.2% in comparison with Internet traffic at 32 kbps.

For a better appreciation of the performance impact when using enhanced CSAs, a mixed traffic situation will be carefully analysed. For this traffic type (comprised of 32 kbps Internet and 9.7 kbps VoIP traffic), the maximum theoretical bound for the system throughput can now be estimated by taking the mean packet size transmitted (measured in bytes and in slots), as indicated in Equation 6.4.

$$S_{\max} = \frac{Pk}{Pk_{\text{slots}} + eCSs} = \frac{Pb_{IP} \cdot Pk_{IP} + Pb_{VoIP} \cdot Pk_{VoIP}}{Pb_{IP} \cdot Pk_{\text{slots_IP}} + Pb_{VoIP} \cdot Pk_{\text{slots_VoIP}} + eCSs} \quad (6.4)$$

where Pk_{IP} (=368.1) and $Pk_{\text{slots_IP}}$ (8.3) are the average Internet packet sizes measured in bytes and data slots respectively. Similarly, Pk_{VoIP} (=146) and $Pk_{\text{slots_VoIP}}$ (=4) are the VoIP packet sizes measured in bytes and data slots respectively. Finally, Pb_{IP} and Pb_{VoIP} are the probability that an IP or VoIP packet will be generated, respectively. Equations 6.5 and 6.6 give these probabilities.

$$Pb_{IP} = \frac{Dr_{IP}}{Dr_{IP} + \frac{Pk_{IP}}{Pk_{VoIP}} \cdot Dr_{VoIP}} \quad (6.5)$$

$$Pb_{VoIP} = \frac{Dr_{VoIP}}{Dr_{VoIP} + \frac{Pk_{VoIP}}{Pk_{IP}} \cdot Dr_{IP}} \quad (6.6)$$

where Dr_{IP} (= 32 kbps) and Dr_{VoIP} (= 9.7 kbps) are the mean data rates of Internet and VoIP traffic generated per station, respectively. Therefore, the maximum theoretical system throughput that can be yielded for mixed traffic is 46.4% of the channel capacity.

With reference to Figure 6.7, we can appreciate that with an offered load over 46% of the channel capacity (produced by 34 stations), the highest system throughput was ranged from ≈ 45.5 to 46% with the configurations *Forced-CSA(FSs-2, CSs-2)* and *Forced-CSA(FSs-2, CSs-3)*. The maximum deviation between simulation and theoretical results was under 1% of the *cc* with an offered load above 46%.

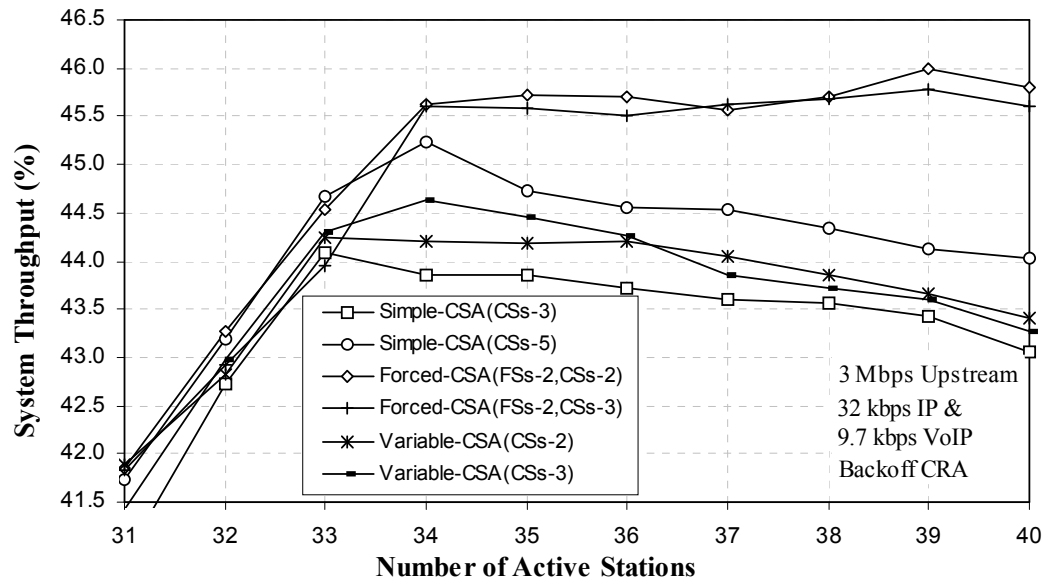


Figure 6.7 – System throughput vs. No. of active stations. Exponential backoff algorithm with varying CSA applied.

On high traffic loads, with this enhanced CSA only the contention slots needed to resolve the collisions are sent, allocating on average from 2.821 to 2.783 per request (according to the current traffic load), as shown in Figure 6.8, optimising further the bandwidth to be allocated to the reservation and contention-based access regions. In

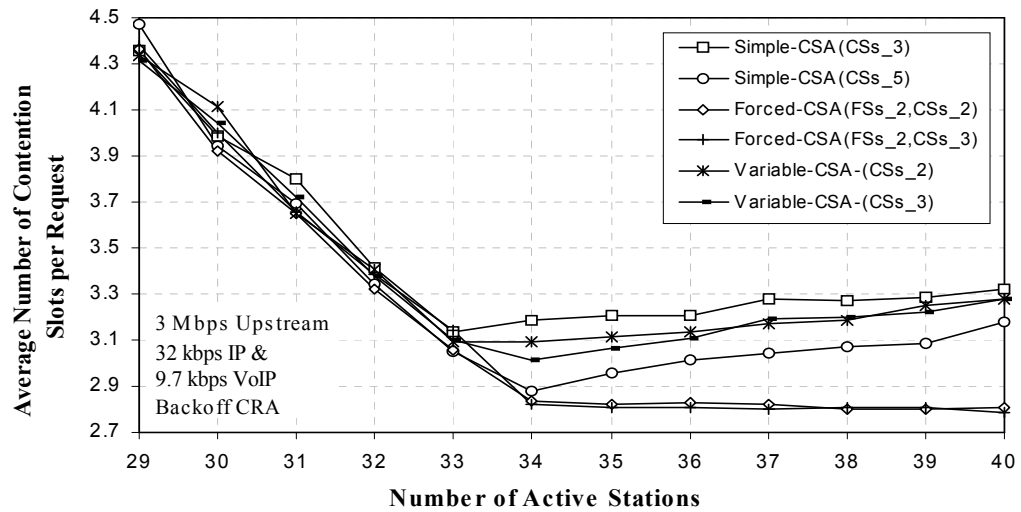


Figure 6.8 – Average No. of CSs per reservation request vs. No. of active stations. Exponential backoff algorithm with varying CSA applied.

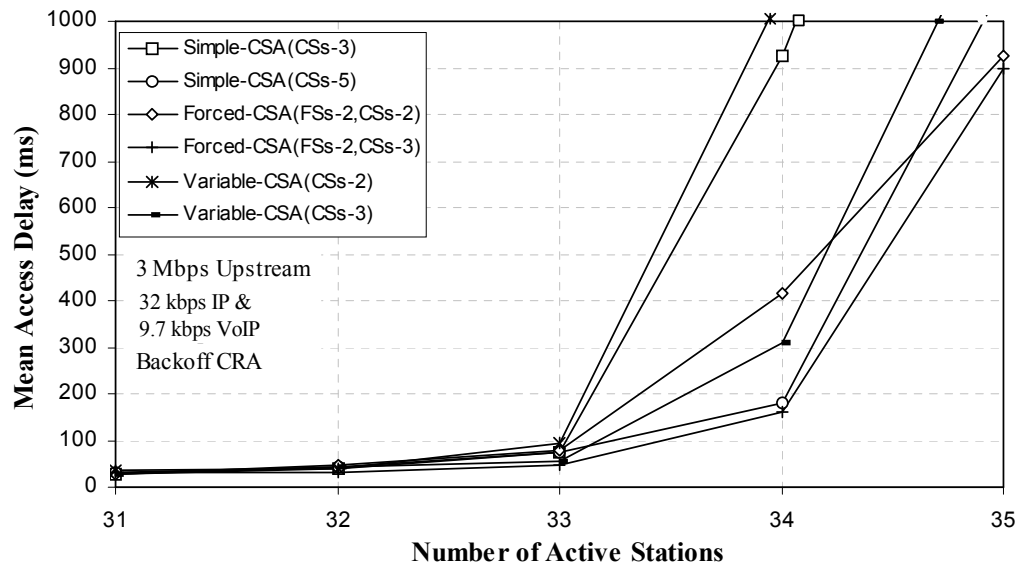


Figure 6.9 – Mean access delay vs. No. of active stations. Exponential backoff algorithm with varying CSA applied.

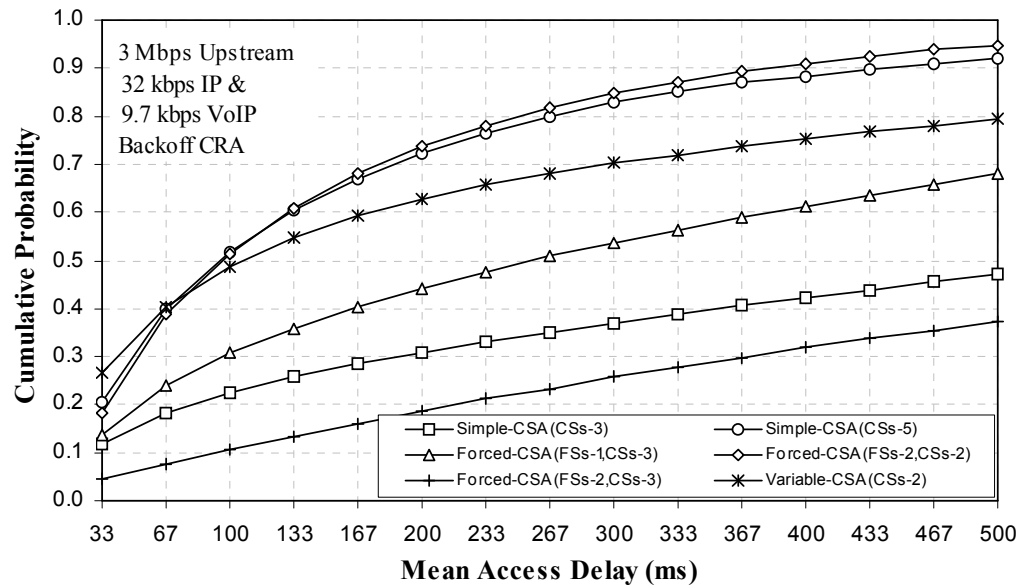


Figure 6.10 – Cumulative probability vs. No. of active stations.
Exponential backoff algorithm with varying CSA applied.
Offered load = 46% produced by 34 stations.

terms of packet delays and number of streams supported, from Figures 6.9 and 6.10, the lowest access delays were gained using the *Forced-CSA*.

For instance, with 45% of the channel capacity (produced by 33 stations), tolerable mean access delays for VoIP streams (under 50 ms) were seen only with *Forced-CSA (FSs-2, CSs-3)*, supporting up to 33 stations. With a slightly increase in offered load (e.g. 46% produced by 34 stations), only the *Simple-CSA (CSs-5)* and the *Forced-CSA (FSs-2, CSs-3)* yielded tolerable low delays for the support of IP traffic. Approximately 73% of all data packets were transmitted in less than 200 ms as illustrated in Figure 6.10. The *Simple-CSA (CSs-5)* produced relatively low packet access delays because it sent more CSs than currently needed to resolve collisions, resulting in a reduction on the average CRGC. On the other hand, the *Simple-CSA (CSs-5)* wasted many CSs, trying to minimise the CRGC and therefore a slight decrease in system throughput was obtained.

As a final remark for this section, in general the *Forced-CSA* outperforms the other two mechanisms (*Simple* and *Variable-CSA*). The *Forced-CSA* not only provided the lowest access delays, but also in most of the cases simulated provided the highest system

throughput. This makes the *Forced-CSA* the optimal slot allocator for different traffic configurations.

6.3 Dynamics of the DVB Contention Resolution Algorithms

In this section, we pay attention to the details of the *exponential backoff algorithm* and the *splitting tree algorithm* adopted by the DVB/DAVIC communications protocol.

6.3.1 Exponential backoff algorithm optimisation

For CATV networks that make use of the *exponential backoff algorithm*, the performance of such networks is also determined by the *initial* and *truncated* backoff values defined (also referred to as *backoff window*: $Bw[i-t]$). Thus, the aim of this section is to provide a performance analysis for different backoff windows and to indicate which windows offer the highest system performance. Two traffic configurations (mixed traffic and VoIP traffic) will be used.

In order to achieve optimum system performance, the *Forced-CSA* (introduced in Section 6.2.3.1) will be utilised. With the exception of the minimum and maximum backoff values, for this analysis we have used the same simulation parameters presented in Table 5.4 (Chapter 5).

a) Results for mixed traffic using the *exponential backoff algorithm*

This analysis deals with mixed traffic at 41.7 kbps per station (32 kbps Internet plus 9.7 kbps VoIP traffic). Figure 6.11 presents the cumulative probability plotted against access delay close to saturation. This performance was recorded with 34 stations, producing an offered load of 46% of the channel capacity.

From Figure 6.11, it can be seen that by defining short values for the *initial* backoff exponent (e.g. Bw[2-3] or Bw[2-5]) a poor system efficiency is obtained. Up to 33% of all packets generated were transmitted in less than 100 ms. This is because the *backlogged* NIUs are forced to transmit in any of the next 4 contention slots, thereby increasing the risk of collision with new incoming packets.

Similar inefficiencies are seen with large values for the *initial* and *truncated* backoff exponent parameters (e.g. Bw[5-7]). The consequence of this being to cause *backlogged* NIUs to wait for a relatively long period before they can transmit a request, which results in many contention slots passing without being used.

By defining intermediate values (e.g. Bw[4-6], Bw[3-4], B[3-5] and Bw[2-4]), a good system performance can be obtained. However, the backoff window that offered best system performance appeared to be Bw[4-6].

a) Results for VoIP traffic using the *exponential backoff algorithm*

This second traffic configuration emulates pure VoIP traffic where a large network size (up to 110 active stations) is considered to cover saturation points. Each station now produces 9.7 kbps CBR traffic. The point of saturation for this traffic configuration varied according to the *backoff windows* selected. Therefore, the analysis is presented in terms of mean access delays and system throughput.

We can see in Figure 6.12 that the network load becomes busy at $\approx 31\%$ of *cc* (produced by 100 active stations). At this point, all *backoff windows* offered mean access delays under 20ms. However, with a higher offered load of $\approx 33\%$ (yielded by 106 stations) the *backoff window* that still offered mean packet access delays of under 50 ms with maximum system throughput was Bw[5-7]. The *backoff window* Bw[4-8] also offered a

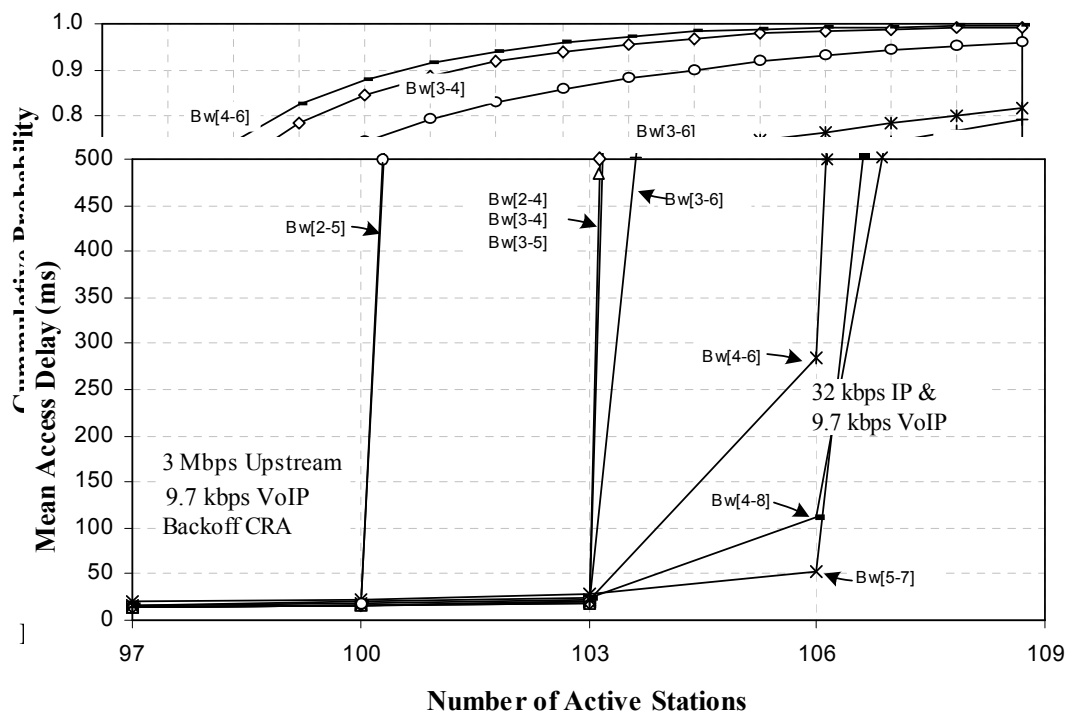


Figure 6.12 – Mean access delay vs. No. of active stations for different backoff windows and VoIP traffic.

good performance. In general, simulation showed us that when there are more than 70 active stations, a good performance is achieved if we set the *initial* backoff value in the range [4-5] and the *truncated* backoff value in the range [6-8].

From the previous analysis (for mixed traffic), the maximum system throughput supported by the network has been reduced from 45% down to 32% of the channel capacity (see Figure 6.13). This is due to the extended number of stations supported, which produce a larger number of collisions when competing for contention-based slots, and also because the mean packet size has been decreased from 368 bytes, (which is the

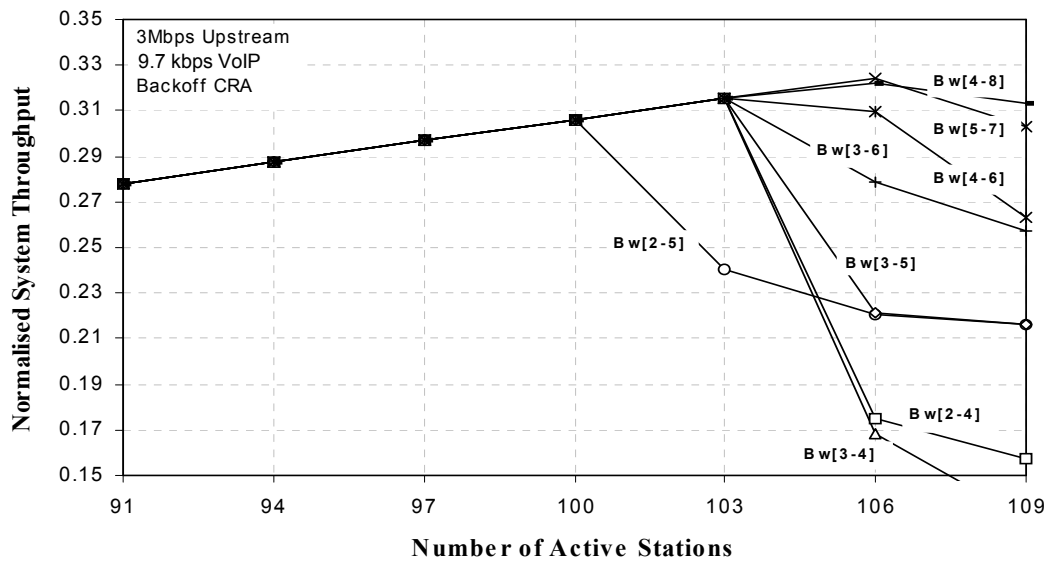


Figure 6.13 – System throughput vs. No. of active stations for different backoff windows and VoIP traffic.

average packet size used for Internet traffic, as indicated in Section 5.3.1), to 146 bytes (used for VoIP traffic), leading to a higher protocol overhead and a reduction in system throughput.

6.3.2 Splitting tree algorithm optimisation

In CATV networks, INAs and NIUs that support a *splitting tree algorithm*, achieve a better system performance by reducing the collision risk when transmitting reservation request messages at the cost of a higher processing times and complexity. This increase in performance is obtained by i) dividing one upstream contention-based slot into three independent minislots that carry shortened request messages and ii) providing two

regions of contention-based slots: the first region is normally used to resolve collisions of *backlogged* stations, where NIUs with new incoming packets are not allowed to transmit reservation requests in these contention-based slots; and the second region is used only to transmit requests of new incoming packets.

When selecting the *splitting tree algorithm*, a factor that contributes also to the performance of the network is the ‘*Entry Spreading*’ (*Es*) value, introduced in Section 3.5.3. This parameter can be either computed or fixed at the INA. For general purposes, the value of the *Es* factor will be fixed and ranged from 4 to 9 in our simulations. We use here the same traffic types as used for the previous algorithm (Internet and VoIP traffic). The simulation parameters presented in Table 5.4 were also utilised for this analysis with the exception of the *Es*, which is the variable for this part of the discussion.

a) Results for mixed traffic using the *splitting tree algorithm*

The performance impact in terms of mean access delay, when varying the *Es*, is presented in Figure 6.14. We can observe that for *Es*-8 and *Es*-6 tolerable delays under 50 ms (for VoIP) are yielded when the offered load reaches 54% of the *cc* (generated by 40 stations). By increasing the offered load by just two stations (to 42 NIUs, 57% of

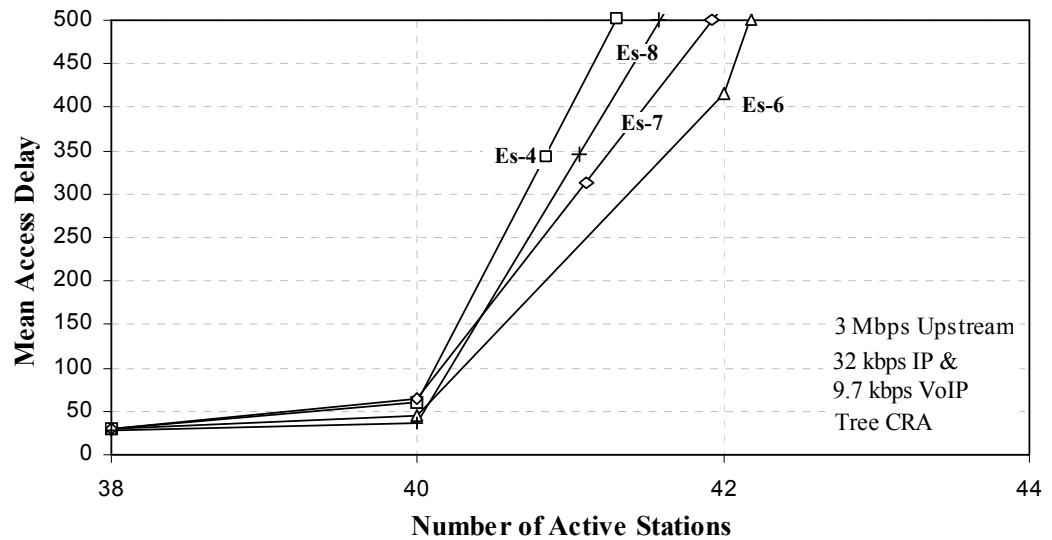


Figure 6.14 – Mean access delay vs. No of active stations for different *Entry spreading* factors and mixed traffic.

the cc), very large delays are evident. At this point, the Es value that still offers acceptable delays (for IP traffic only) under 500 ms was $Es-6$. The other two values $Es-5$ and $Es-9$, (not included in Figure 6.14), showed the same results as for $Es-7$ and $Es-8$, respectively.

In terms of system throughput, the difference of using distinct Es values was much less significant. In general, regardless of the *Entry spreading* factor, the maximum system throughput ranged between 54.5 and 55% of the cc .

b) Results for VoIP traffic using the *splitting tree algorithm*

Results from simulations for VoIP traffic (see Figure 6.15), revealed that delays of under 30 ms can be obtained for all values of the Es factor, given an offered load up to 46% of the cc (with 145 active stations). Optimum system performance is obtained with $Es-6$ and $Es-7$, achieving a maximum system throughput over 45% of the cc , as shown in Figure 6.16. In general, it was found that by defining small values for the *Entry spreading* factor (e.g.: $Es-4$), at high traffic loads (above 45% of the cc), NIUs with new arrivals are forced to transmit in one of the next three available contention-based minislots, reserved for new incoming packets with probability of $3/4 = 0.75$. This leads to a higher risk of collision when more than two NIUs are competing for request transmission.

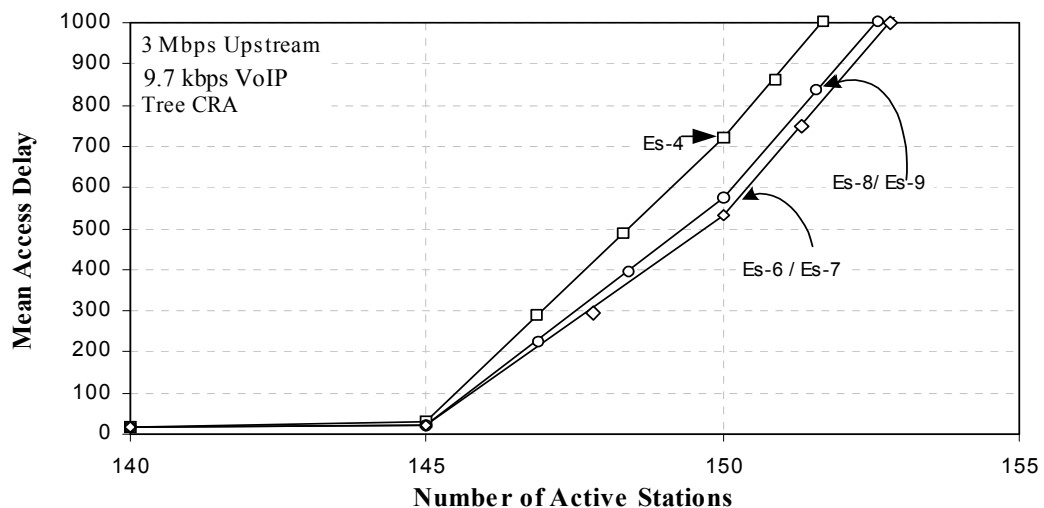


Figure 6.15 – Mean access delay vs. No. of active stations for different *Entry spreading* factors and VoIP traffic.

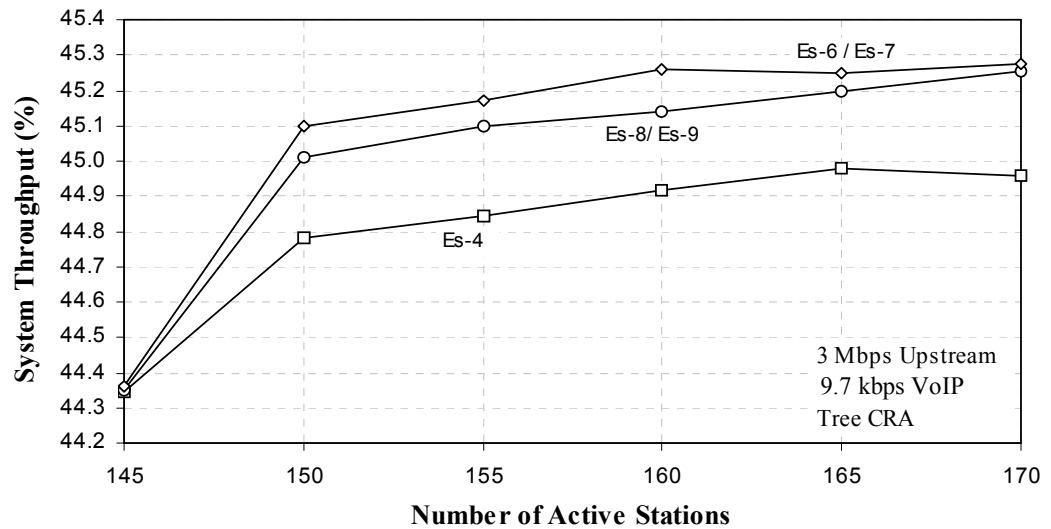


Figure 6.16 – System throughput vs. No. of active stations for different entry spreading factors and VoIP traffic.

By defining larger values for the *Entry spreading* factor, (e.g. *Es-8* and *Es-9*), the probability of transmission of requests in one of the next 3 contention minislots is $(3/8 \approx) 0.38$ and $(3/9 \approx) 0.33$ for *Es-8* and *Es-9*, respectively. This is equivalent to passing (on average) 2 or 3 contention slots before transmitting a request, which also results in a slight reduction in system performance.

Results reported in this section suggest that for optimum performance, the *Es* factor should be set to 6 for mixed traffic (or medium size networks) and 6 or 7 for VoIP traffic (or large networks).

6.3.3 Performance comparison between the exponential backoff algorithm and the splitting tree algorithm

In the previous analysis of Chapter 5, we presented an initial performance comparison between both CRAs of the DVB/DAVIC protocol. In this section we focus explicitly on the details and dynamics of each CRA in order to demonstrate why the *splitting tree algorithm* outperforms the *exponential backoff algorithm*.

Results presented in Figures 6.17 to 6.20 show a performance comparison between the *exponential backoff algorithm* and the *splitting tree algorithm*, for three different traffic configurations (32 kbps IP, 9.7 kbps VoIP and 41.7 kbps mixed traffic).

Figure 6.17 indicates that the increase in system throughput when using the *splitting tree algorithm* is 9%, 12.3% and 9.7% of the *cc* for Internet, VoIP and mixed traffic, respectively. Figure 6.17 also shows that the maximum throughput achieved varies from 32% to 58% of the *cc*.

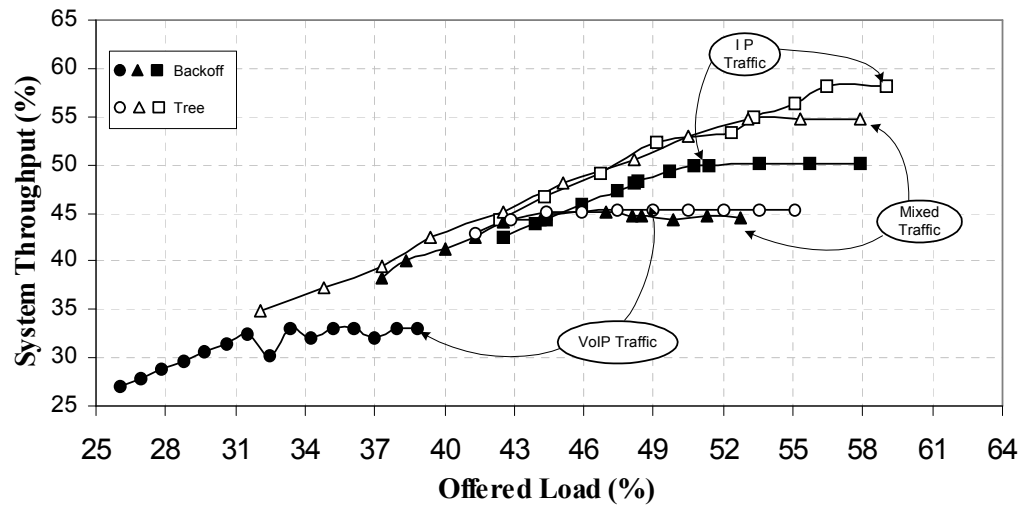


Figure 6.17 – System throughput. *Exponential backoff* vs. *splitting tree algorithm* for different traffic configurations.

From Figure 6.18, by taking the limits before the network became unstable, the increase in terms of supported streams corresponded to 8, 39 and 6 stations for Internet, VoIP and mixed traffic, respectively.

In terms of efficiency, results not shown in this analysis indicate that the maximum channel utilisation varied from 76% to 86% of the *cc* for the *exponential backoff algorithm* and from 88% to 90% of the *cc* for the *splitting tree algorithm*. The *exponential backoff algorithm* achieved a lower channel utilisation due to the considerable number of unused forced-contention slots that were allocated to resolve collisions, especially when a large network size was considered (as it was the case of VoIP traffic).

Another important comparison is that according to the *Contention-Resolution-Grant Cycle* introduced in Section 3.5.1, there are three types of delays involved in the transmission of data packets, such as waiting delay, contention delay and grant delay, as depicted in Figure 3.3. The *exponential backoff algorithm* yields a low system performance mainly because this algorithm requires more time to resolve collision

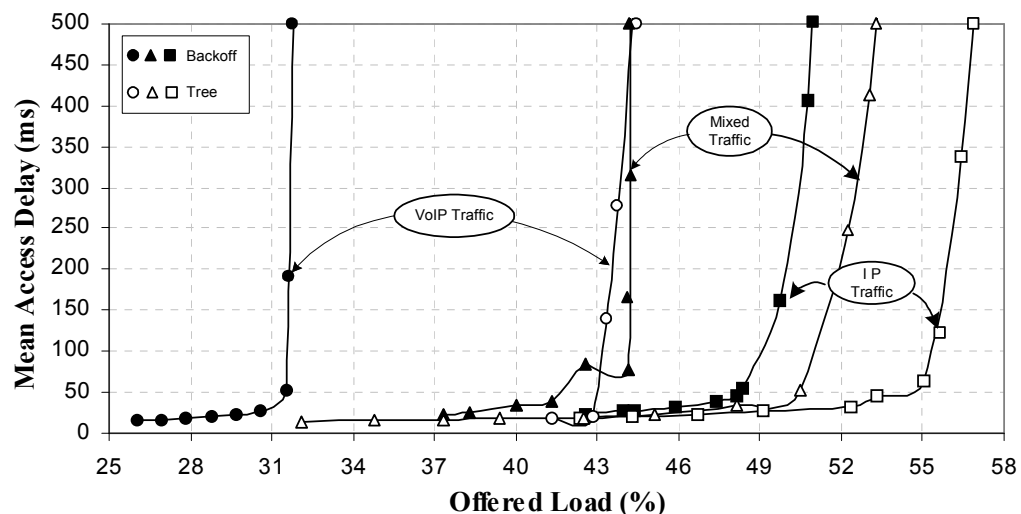


Figure 6.18 – Mean access delay. *Exponential backoff* vs. *splitting tree algorithm* for different traffic configurations.

(referred to as contention delay) than its counterpart *splitting tree algorithm*.

In order to make a direct comparison among the components of delays involved in the transmission of data packets, we can analyse the situation when only IP traffic is delivered. From Figure 6.19, on high traffic loads, the *splitting tree algorithm* requires on average 10 ms to transmit successfully a reservation request (this corresponds to the *mean contention delay* curve in Figure 6.19b), in comparison to approximately 19 ms required by the *exponential backoff algorithm* (Figure 6.19a.). This variation of 9 ms makes a difference in system performance, because having a large service time (comprised of the *mean contention delay* plus the *mean grant delay*) results in packets being queued for a longer period of time as the offered load increases further, which in turn results in large mean access delays.

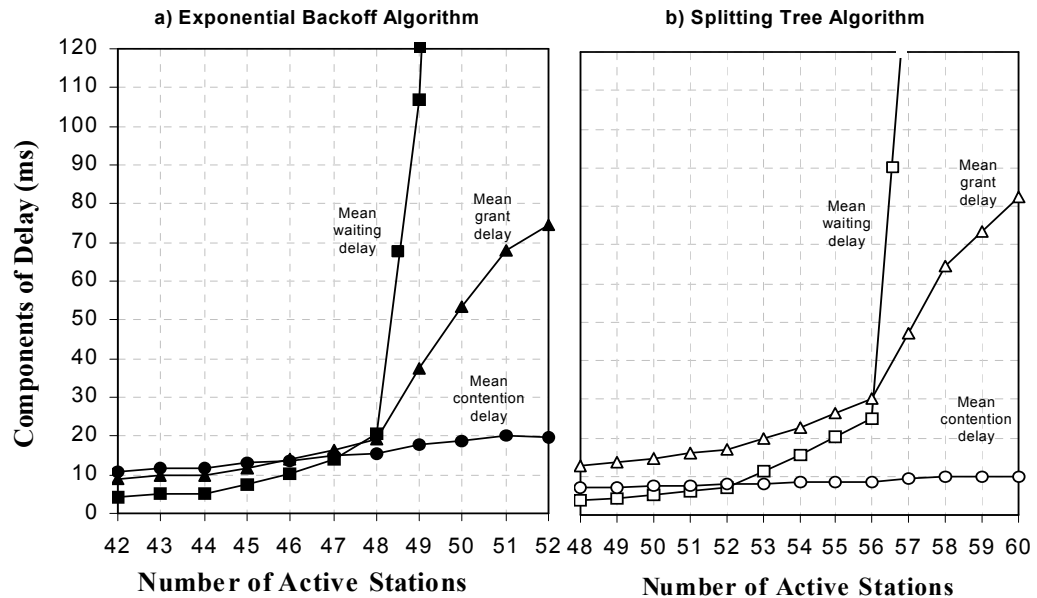


Figure 6.19 – Components of delay, *Exponential backoff* vs. *splitting tree algorithm* for IP traffic.

This effect can be appreciated in Figure 6.20, which presents in detail the collision of three NIUs. For this analysis the period begins when three stations collide and ends when the NIUs transmit their data message successfully.

In Figure 6.20 each line shows the use of the 18 slots described in each signalling frame (MCI). For the *exponential backoff algorithm*, stations 43, 25 and 8 transmit a reservation request at MCI 1, using the same contention slot. Then, the stations should wait until the headend sends back the acknowledgements of these reservation requests, which is at MCI 3. At this point, the stations detect the collision. Station 43 backs off for 10 CSs, and stations 8 and 25 transmit in CS 0 and 1, respectively.

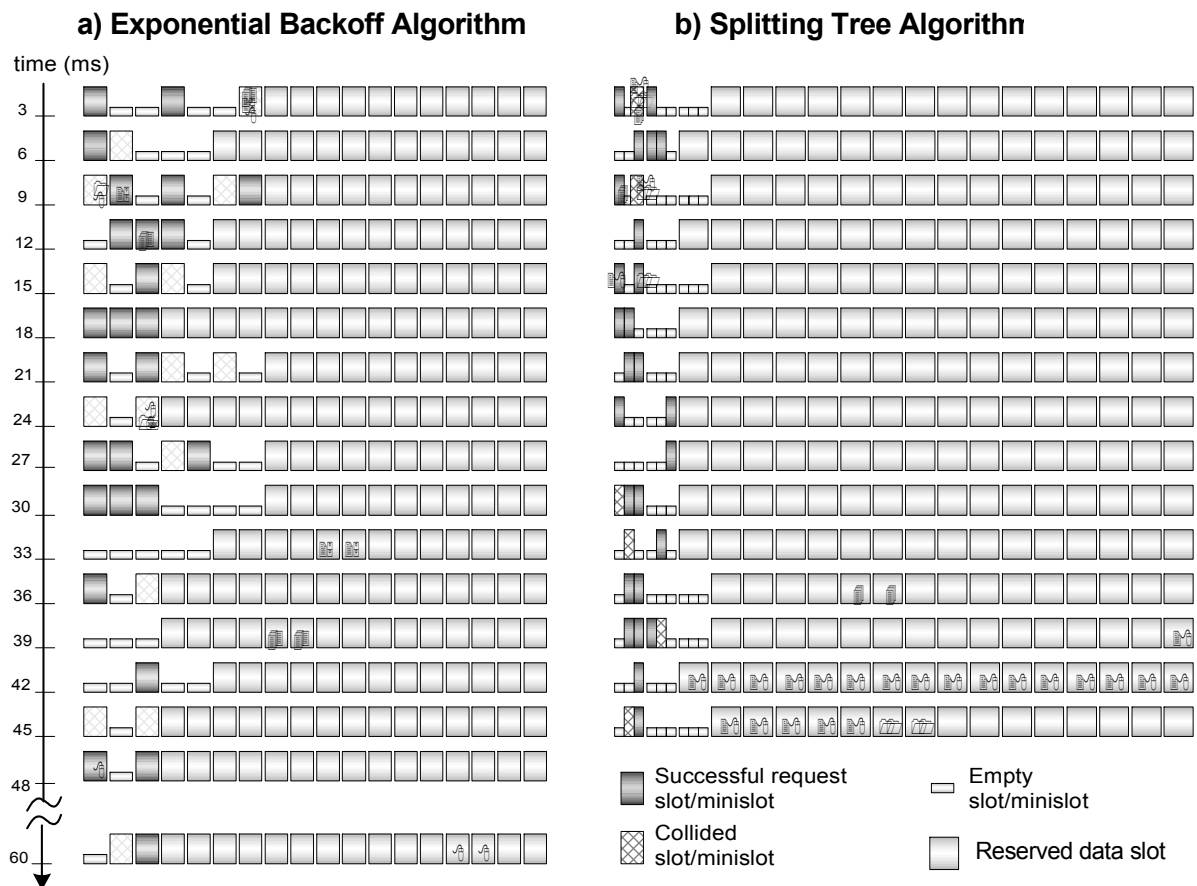


Figure 6.20 – Upstream slot usage. *Exponential backoff* vs. *splitting tree algorithm* for IP traffic with 49 stations. Sequence captured at 30 seconds of simulation.

Station 25 transmits successfully and then waits until the headend grants its request, which is at MCI 11. This is because the headend, at this point, was serving requests from other stations not shown. Station 8 collides two more times at MCI 3 (with station 1) and MCI 8 (with station 11). In MCI 16, station 8 transmits a successful request, which is granted in MCI 20. Station 43 transmits successfully at MCI 4 and its request is served in MCI 13. This algorithm required 16 signalling frames (48 ms) to resolve a collision among three stations.

Lets us now analyse the case when the *splitting tree algorithm* is utilised. At MCI 1, there are also three stations that collide (stations 28, 11 and 4). At MCI 3, the headend allocates one more CS (in the first position of the MCI), which is used only among the collided stations. So, all stations transmit again after two MCIs once the collision has been detected. At this moment, station 4 transmits a successful request using the first minislot of this additional CS (scheduled for MCI 12). Station 11 and 28 collide again using the third minislot. It is at MCI 5 when they transmit successfully and served completely at MCI 15.

Generally, the *splitting tree algorithm* is much more efficient because when a collision happens (e.g. between two or even three stations), it takes on average from 2 to 5 MCI cycles to resolve, whilst the *exponential backoff algorithm* takes from 2 to 16 (and some times longer). Another advantage of the *splitting tree algorithm* is that it requires fewer contention slots than the *exponential backoff algorithm* in order to transmit successful requests. From Figure 6.20, we can appreciate that the number of contention slots used by the *splitting tree algorithm* ranged from 2 to 3 CSs, in comparison to 3 to 7 CSs required by the *exponential backoff algorithm* in each MCI cycle.

The same performance analysis and a comparison between these two contention resolutions algorithms, for an upstream channel of 6.176 Mbps, has been reported in [86] and summarised in Figure E.1 of Appendix E.

6.4 Conclusions

In this chapter we have shown that the overall system performance of the DVB/DAVIC cable communications system can be significantly improved by the use of three novel Contention Slot Allocators (*Simple-CSA*, *Variable-CSA* and *Forced-CSA*), which allocate dynamically the number of contention slots that should be allocated in the next signalling frame, based on the current traffic load. Results presented in this chapter have pointed out that the *Forced-SCA* not only provides the highest system throughput but in most of the cases also offers the lowest packet access delays for the *exponential backoff algorithm*. Simulation results have been shown to be accurate when compared with results from theoretical analysis.

In addition, a performance comparison of the contention resolution algorithms adopted by the DVB/DAVIC protocol was also presented. The *splitting tree algorithm* takes advantage of the *exponential backoff algorithm* in the sense that feedback and allocation information allows a station, (with new incoming arrivals) to compete for contention-based slots without the risk of collision with *backlogged* stations. One more advantage of the *splitting tree algorithm* is that the use of minislots for reservation requests further decreases the risk of collisions, since one contention-based slot is divided into three minislots, increasing the probability of successful request transmissions. However, drawbacks of the *splitting tree algorithm* are higher complexity at the headend, increased processing times of the feedback and allocation information at the station, and since every contention slot should be acknowledged regardless of whether it is used or not, higher control information at the downstream channel is assigned.

Results presented by both contention resolution algorithms showed that the system performance is a trade-off between mean access delay/system throughput and the values selected for the *initial/truncated* backoff values and the *Entry Spreading* factor.

In general, results presented here showed that an increase of over 9% on system performance can be obtained by the use of a *splitting tree algorithm* when backoff values (*initial/truncated*) and the *Entry spreading* factor have been optimised, for different traffic configurations such as Internet, VoIP and mixed traffic.

Chapter 7

PERFORMANCE OPTIMISATION FOR THE SUPPORT OF TCIS AND A PERFORMANCE COMPARISON OF DVB/DAVIC AND DOCSIS

7.1 Introduction

The DVB/DAVIC protocol specification supports only a limited reservation access mechanism (referred to as '*pure reservation access*' - *PRA*) and has not yet been optimised for the delivery of isochronous streams. The functionality of the *PRA* was previously touched on under the *CRGC*, introduced in Section 3.5.1.

Some contention resolution algorithms, such as the *exponential backoff algorithm*, produce relatively high access delays for upstream transmissions. The *splitting tree algorithm*, which uses less contention slots and shortened reservation requests, rapidly resolves the contention resolution cycle, thus reducing access delays and improving system throughput. However, the *splitting tree algorithm* combined with optimised configuration parameters as discussed in Chapters 5 and 6 is not ideal for the support of the next generation timing critical interactive services.

Therefore, in this chapter some novel improvements and techniques are introduced, which will enable the DVB/DAVIC MAC protocol to provide the delay requirements optimally for the delivery of delay sensitive services. The objective is to achieve an increased performance by introducing new reservation request techniques that reduce or avoid the increased risk of collisions during congestion periods.

A second approach involves QoS with a guaranteed delivery. Although, this technique is only supported by the second version of EuroModems.

Finally, a performance comparison between the two standardised cable communications systems, DVB/DAVIC and DOCSIS, is also considered. Here, the major characteristics and the fundamental performance properties of these two leading protocols are thus evaluated.

7.2 Enhanced-Reservation-Request Mechanisms

Six improved mechanisms are now discussed. Each mechanism reduces access delays and increases overall system performance required in support of *TCIS* and also high-speed bursty traffic. All six use only the reservation access mode of the DVB/DAVIC protocol.

7.2.1 Reserved Request (RR)

The '*Reserved Request (RR)*' mechanism follows the same principle as the *pure reservation access* request. A difference is that the *RR* allows a station to request one slot more than currently needed in order to transmit a further request as depicted in Figure 7.1. This additional slot is only requested when there are other data packets in the transmission queue. Thus, the NIU can indicate its request for additional bandwidth without any collision risk between two data messages. A risk of collision exists when the transmission queue is empty and a new data message arrives.

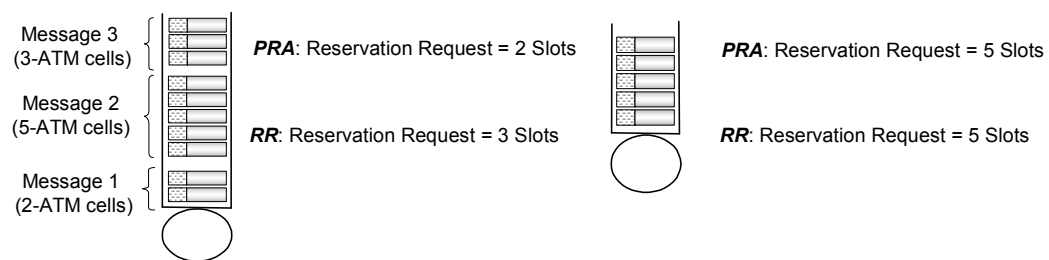


Figure 7.1 – Functionality of the *RR* mechanism for different packet sizes in the transmission queue.

7.2.2 Continuous Reserved Request (CRR)

The '*Continuous Reserved Request (CRR)*' mechanism allows a station to request one more slot than needed, regardless of the state of the transmission queue (see Figure 7.2). If the transmission queue of the NIU is empty, then the extra slot is not used. This mechanism may reduce the mean access delays, because for every two MCI cycles (6 ms), there is a reserved slot in which to place a request. A drawback of this mechanism is that unused extra slots cause a considerable waste of bandwidth during congestion periods.

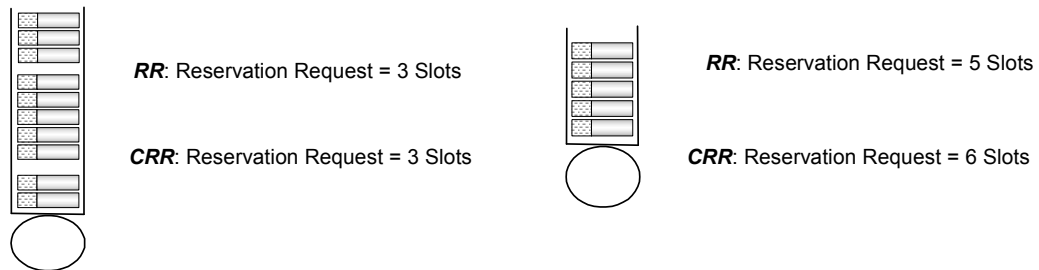


Figure 7.2 – Functionality of the *CRR* mechanism for different packet sizes in the transmission queue.

7.2.3 Enhanced Pure Reservation Access (EPRA)

A disadvantage of the *pure reservation access*, the *RR* and the *CRR* request mechanisms, is that they are all based on a per-packet transmission basis rather than a per-block transmission. In other words, a station only requests the number of slots needed to transmit the current data message (up to 256 ATM cells). As the offered load is increased, short messages tend to accumulate and then saturate the NIU buffer. Thus, long packet access delays render the system unstable. This effect is true even at lower loads but does depend on of the traffic type.

The ‘*Enhanced Pure Reservation Access (EPRA)*’ mechanism has been designed to support per-block transmissions and to allow a station to request slots of up to 256-ATM cells, so that most of the packets waiting in queue can be sent using a single reservation request. This is shown in Figure 7.3.

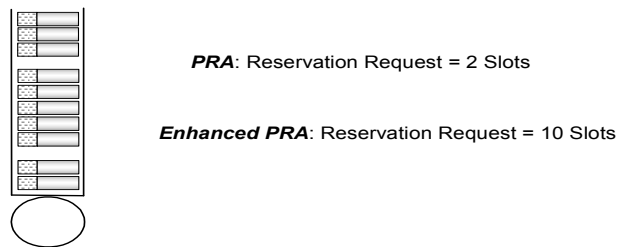


Figure 7.3 – Functionality of the *EPRA* mechanism for different packet sizes in the transmission queue.

7.2.4 Piggyback Request (PG)

The ‘*Piggyback Request (PG)*’ mechanism avoids reservation requests between data messages. When there are more data packets in the transmission queue waiting to be transmitted, two unused bits of the ATM header (in the upstream slots) are used to carry the slot request. This is illustrated in Figure 7.4.

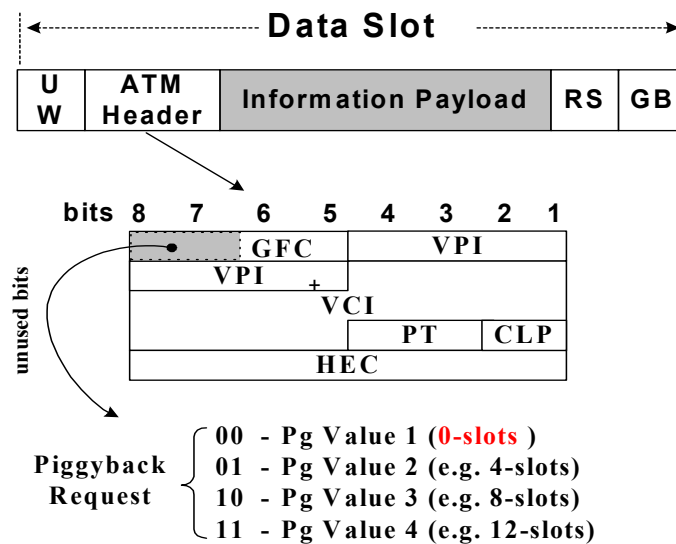


Figure 7.4 – Data slot structure with a piggyback request.

Four states are defined. These are mapped to indicate four distinct numbers of requested slots, for example 0, 4, 8 and 12, rather than the actual number of slots required [101]. Figure 7.5 shows the mechanism regime.

The *PG* mechanism (which is also supported by the DOCSIS protocol) tends to be very efficient for the delivery of bursty traffic [120].

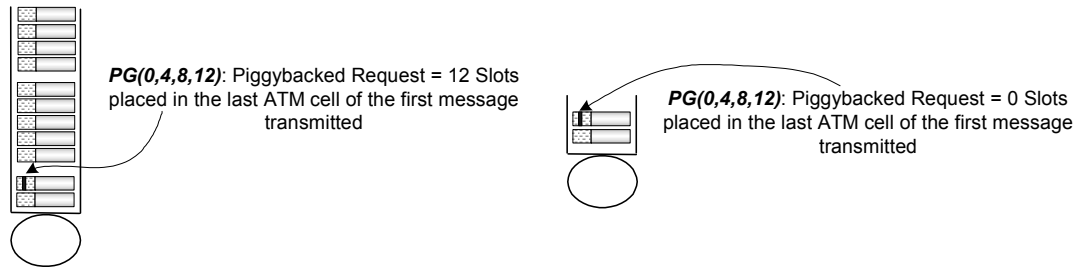


Figure 7.5 – Functionality of the *PG* mechanism for different packet sizes in the transmission queue.

7.2.5 Continuous Piggyback Request (CPG)

The '*Continuous Piggyback Request (CPG)*' mechanism allows a station to always piggyback a request and prevents the station from falling back into the contention request mode. This is shown in Figure 7.6. At least one slot is requested regardless of the state of the transmission queue. The *CPG* mechanism is efficient but not ideal for CBR traffic. This is because there are only four possible values for the piggyback request. Unused slots cause wasted bandwidth and particularly when a large number of NIUs are active.

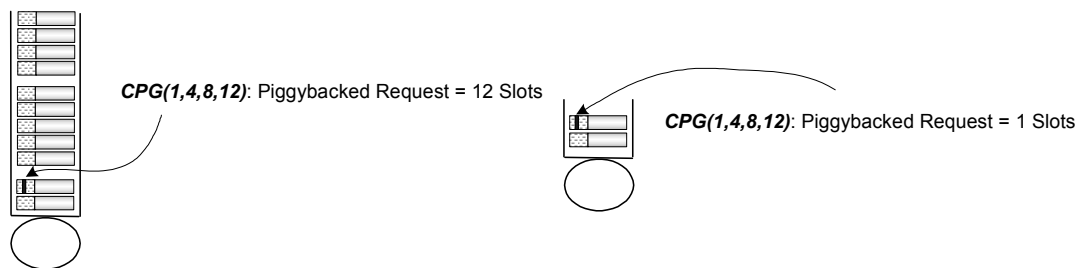


Figure 7.6 – Functionality of the *CPG* mechanism for different packet sizes in the transmission queue.

In addition, during long idle periods, the *continuous piggyback mechanism* fails due to the excessive number of slots that remain unused. A timer function stops this mechanism after a time threshold is exceeded. Time periods of up to 2.286s can be bridged without using contention slots [35].

7.2.6 Unsolicited Grant Slot (UGS)

The '*Unsolicited Grant Slot (UGS)*' mechanism provides a station with a fixed number of slots at periodic intervals, using the *reservation grant message*. The stations must use only unsolicited grant slots for upstream transmissions. The number of slots needed by a station and the periodic intervals can be negotiated during the connection set-up. This mechanism can be used only when the CBR parameters are satisfied regarding the actual data rate, packet size and packet interval. In addition, the *UGS* mechanism completely avoids the transmission of reservation requests and allocates more bandwidth to the reservation-based region, thereby reducing significantly the packet access delay and increasing the overall system throughput.

The *continuous reserved request*, *piggyback request* and *continuous piggyback request* mechanisms were first reported in [43], where a comprehensive performance analysis in terms of the cumulative probability of packet delay was presented. Due to their enhanced performance these three mechanisms have now been integrated into the latest version of the DVB/DAVIC standard 'ETSI EN 200 800' [35].

The *unsolicited grant slot*, *enhanced pure reservation access* and *reservation request* mechanisms are novel and have been developed in this project. Results derived from this research show that the *UGS* mechanism significantly outperforms the other mechanisms in terms of system performance and can be easily adopted in the ETSI EN 200 800 protocol specification.

7.2.7 Performance comparison of Enhanced-Reservation-Request Mechanisms

Two different traffic situations are presented in order to give an overview of the performance of the *enhanced-reservation-request mechanisms* in reservation-access mode. For this analysis both contention resolution algorithms were used.

The traffic types were as previously seen and consisted of VoIP traffic at 9.7 kbps and mixed traffic at 41.7 kbps. Table 7.1 shows the values of the simulation parameters that were shown to have provided optimum system performance in Chapters 5 and 6.

Table 7.1 – Optimised simulation parameters.

Simulation Parameter	Value
Upstream data rate (QPSK)	3.088 Mbps
Downstream data rate (64-QAM, In band)	42 Mbps
Buffer capacity per NIU	3000 ATM cells
Headend and NIU processing delay	2 microseconds each
Maximum contention access message length	0 ATM cells
Maximum number of active NIUs (EuroModems)	180
Maximum reservation/piggyback request length (in slots)	32 for PRA, EPRA, RR, CRR, and 256 for EPRA
Backoff windows (<i>exponential backoff algorithm</i>)	0,3,6,12 for PG and 1,3,6,12 for CPG
Minimum CSs per signalling frame (<i>exponential backoff algorithm</i>)	[4-6] for Mixed traffic and [5-7] for VoIP traffic
Contention slot allocator used (<i>exponential backoff algorithm</i>)	3 for PRA, EPRA, RR, PG and 1 for CRR, CPG, UGS
Entry spreading factor (<i>splitting tree algorithm</i>)	Forced
Minimum CSs per signalling frame (<i>splitting tree algorithm</i>)	6
Propagation delay (coax and fibre)	2 for PRA, EPRA, RR, PG and 1 for CRR, CPG, UGS
Simulation time for each run	5 μ s/Km
Distance from nearest/farthest NIU to the Headend	60s
Transmission time (cycle) of the signalling frame	10-16 Km, randomly distributed
	3 ms

A) Performance analysis for the *exponential backoff algorithm*

From simulation results, Figures 7.7 to 7.9 present a performance comparison between the *enhanced-reservation-request mechanisms* when the *exponential backoff algorithm* is used.

Results for mean access delay showed that on high traffic loads, above $\approx 33\%$ of the *cc* (105 stations) for VoIP traffic (Figure 7.7a) and 46% of the *cc* (34 stations) for mixed traffic (Figure 7.7b), all the enhanced mechanisms offered lower access delays than the *pure reservation access mode*. For both traffic types, it was found that the best mechanism is the *continuous piggyback request*. This mechanism achieved a better performance because all requests are piggybacked on the same data messages. Also, if the buffer is empty when a station receives the piggybacked slots, such reserved slots are then used to send further requests for at least one slot. Contention access is therefore avoided. Thus, more bandwidth can be allocated to the reservation access region. This results in the lowest access delays (during high traffic loads) and the highest system throughput.

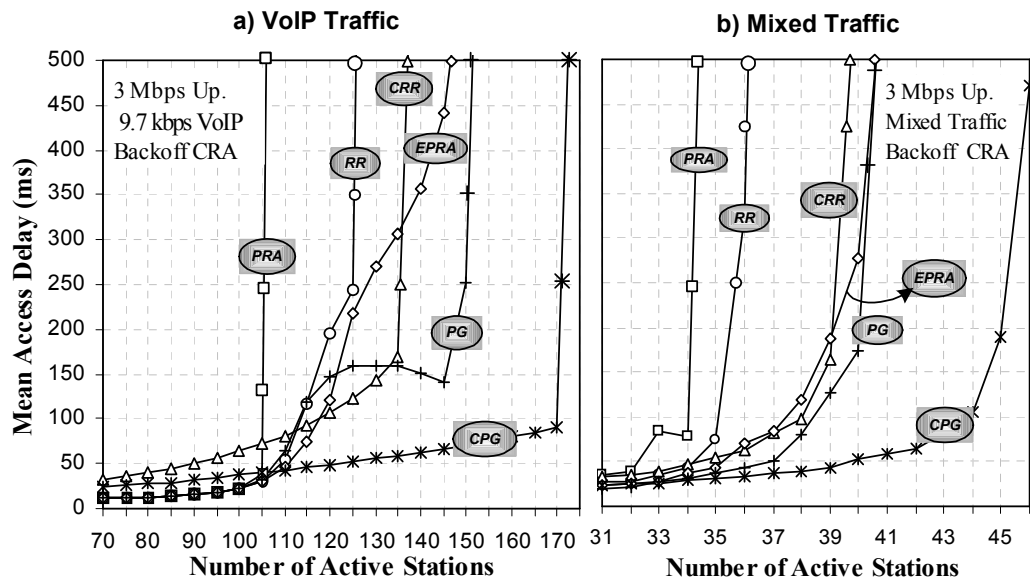


Figure 7.7 – Mean access delay vs. No. of active stations for VoIP and mixed traffic. *Exponential backoff algorithm with enhanced-reservation-request mechanisms.*

For mixed traffic on loads of under 46% of the *cc* (34 stations), the *CPG* mechanism also provided the lowest access delays. However, for the transmission of VoIP streams on loads of under 33% *cc* (105 stations), it did not offer the lowest access delays because it was seen that with a large number of active stations, the headend has to grant two types of requests. These are the requests made for one slot, which are then used to send a further reservation request and the requests made for the transmission of data messages. In simulation, it was seen that for low traffic loads, most requests are made for one slot, which delay the other requests for data messages. Thus, the NIUs have to wait longer in order to transmit their data slots.

A similar effect is also appreciated with *CRR*. However, in this case the number of requests made for one slot is larger than with the *CPG* mechanism. This is because the reservation requests are made for the transmission of the current data packet, and the *CPG* not only requests bandwidth for the current data message but also for other packets in the queue, as long as the number of slots requested does not exceed the maximum-piggybacked request threshold (set to 12 slots). Hence, mean access delays are slightly larger than those produced by the *CPG* and the other mechanisms.

In terms of number of VoIP streams supported, with the *PRA* and *PG* mechanisms only 105 connections can be upheld with mean access delays of under 50 ms. That number can be increased to 110 connections with the use of the *RR* and *EPRA* mechanisms. However, with the *CPG* mechanism the network capacity can be extended up to 124 connections.

Results for mixed traffic indicate that the number of VoIP streams supported in the presence of Internet traffic is 32 connections with the *PRA* mechanism. Simulations have also shown that this number can be extended by up to 8 connections using the *enhanced-reservation-request mechanisms*.

In terms of throughput, Figure 7.8 shows the maximum system throughput achieved by each mechanism. The *PRA* and *RR* achieve under $\approx 38\%$ *cc* for VoIP traffic (Figure 7.8a) and under 47% of the *cc* for mixed traffic (Figure 7.8b). This is due to continuous contention access. The maximum achievable throughput is produced by the *CPG* mechanisms, which can be up to 52% and 61% of the channel capacity for VoIP and mixed traffic, respectively.

All of the enhanced mechanisms achieve a higher system throughput than the *PRA* mechanism, lowering collision risk and reducing (or even avoiding) contention access transmissions, thereby allocating more bandwidth to the reservation access region.

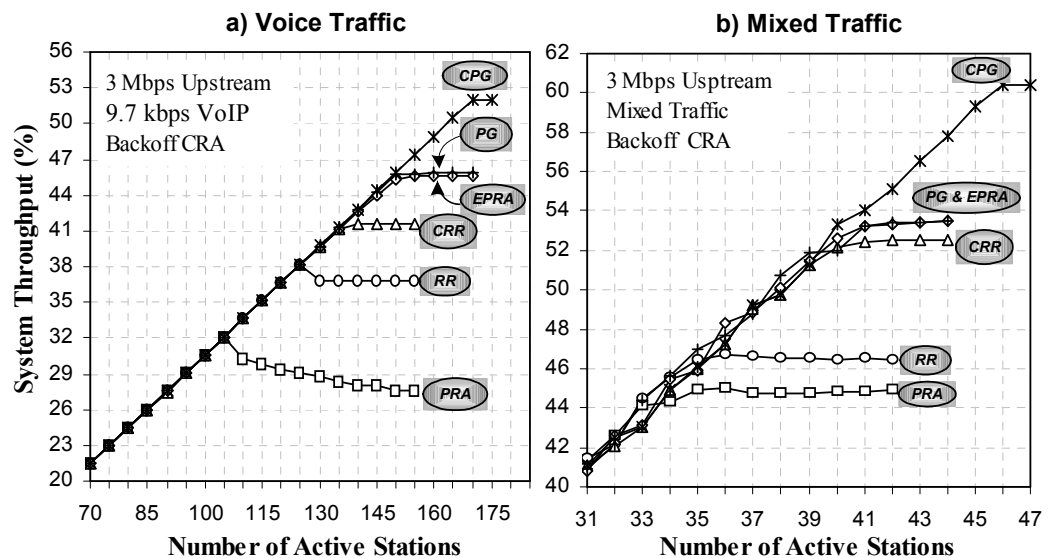


Figure 7.8 – System throughput vs. No. of active stations for VoIP and mixed traffic. Exponential backoff algorithm with enhanced-reservation-request mechanisms.

In Figure 7.9, we can appreciate how the amount of bandwidth wasted by collisions is decreased considerably by the *enhanced-reservation-request mechanisms* as the offered traffic load becomes more intense. The continuous request mechanisms, (*CRR* and *CPG*) were designed to avoid contention access entirely. The *RR* and *PG* follow the same dynamics as the *CRR* and *CPG* mechanisms respectively, with the exception that when there are no packets in the buffer, stations fall back to contention access.

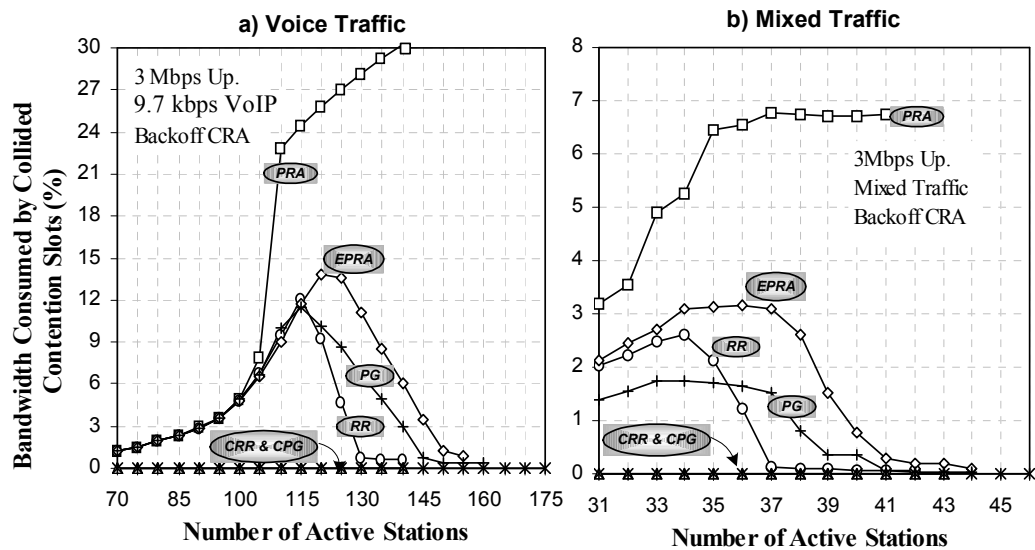


Figure 7.9 – Bandwidth consumed by collisions vs. No. of active stations for VoIP and mixed traffic. *Exponential backoff algorithm with enhanced-reservation-request mechanisms.*

B) Performance analysis for the *splitting tree algorithm*

Simulations with the *splitting tree algorithm*, Figure 7.10a indicates that for pure VoIP traffic optimum system performance may be obtained using the *unsolicited grant mechanism*. With this mechanism the network capacity is considerably increased to the highest figure for this research at ≈ 170 connections. This equates to approximately 53% of the available channel capacity and produces mean access delays below 7 ms. Hence, this mechanism has been shown to match the requirements of CBR. The other mechanisms, with the exception of *CPG*, support the same number of connections equal to ≈ 145 .

Results for mixed traffic are shown in Figure 7.10b. It should be noted that discussion of the *UGS* mechanism is absent here since in a mixed traffic situation the constant bit rate requirements cannot be satisfied using this technique. From the figure we can also appreciate that the *enhanced-reservation-request mechanisms* are more effective with high traffic loads. For example, for traffic loads above 55% of the *cc*, the maximum number of Internet connections that can be supported was ≈ 45 with the *CPG* mechanism, compared with ≈ 41 with the *PRA* mechanism. For delay sensitive applications the number of connections supported resulted in ≈ 40 , regardless of the mechanism utilised.

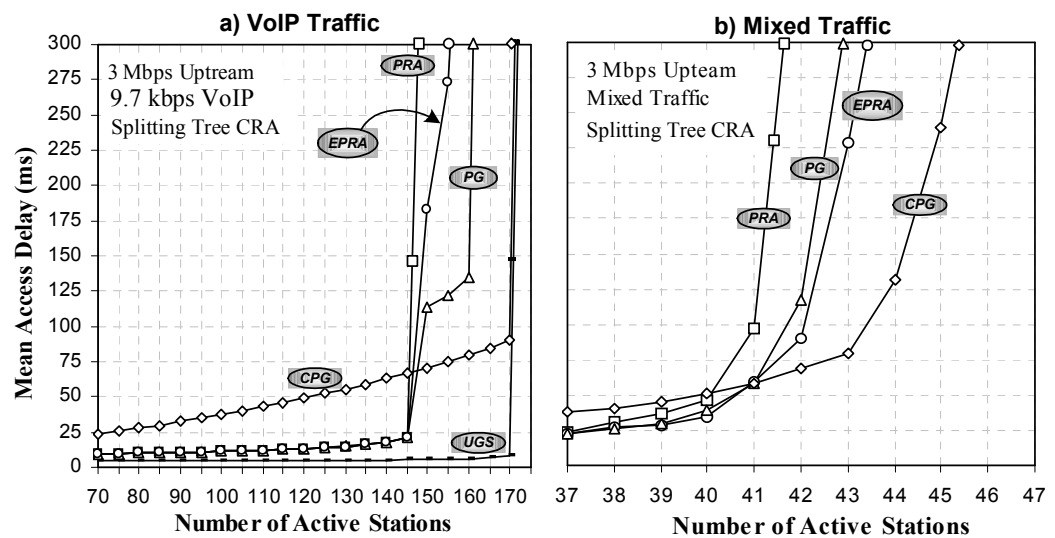


Figure 7.10 – Bandwidth consumed by collisions vs. No. of active stations for VoIP and mixed traffic. *Splitting tree algorithm* with *enhanced-reservation-request mechanisms*.

In general, the *UGS* and also the *CPG* (for small networks) are quite efficient and best suited for the delivery of VoIP streams. For Internet traffic, it was found that the *CPG*, *PG* and *EPRA* mechanisms, combined with the use of the *splitting tree algorithm*, are more appropriate for the transmission of Internet and bursty traffic.

With all of the *enhanced-reservation-request mechanisms*, it has been demonstrated that the number of VoIP streams and Internet connections can be significantly extended, using only the reservation access mode of the DVB/DAVIC communications protocol. This access scheme is the basic mode for the first version of EuroModems [36]).

A drawback of using the reservation access mode for the provision of delay sensitive applications is that bandwidth cannot be guaranteed. This disadvantage is because all traffic transmitted using this mode is treated with a ‘best-effort’ service and in some situations, the delivery of *TCIS* requires a special treatment for an improved service. Therefore, in order to provide guaranteed bandwidth for *TCIS*, the use of QoS is crucial and is addressed in the following section.

7.3 Quality of Service (QoS)

This section outlines some issues related to QoS and how a cable network compliant to the DVD/DAVIC protocol specification can make use of QoS characteristics for the provision of guaranteed bandwidth. Here a comprehensive performance analysis for QoS is presented for the support of *TCIS*.

In the simplest sense according to [82], QoS is the ability of a network element (e.g. an application, host or router) to have some level of assurance that service requirements can be satisfied, providing a consistent predictable data delivery service.

The Internet Protocol provides what is called a ‘*best-effort*’ service, making no guarantees about when data will arrive, or how much traffic can be delivered. Timing critical interactive applications, including voice, audio and video streaming, demand high data throughput with low-latency in two-way communications.

In order to provide service guarantees, a level of quantifiably reliable service may need to be supplemented with the ability to differentiate traffic and enable different service levels. Two types of QoS are proposed. They are complementary and designed for use in combination in different network contexts. These schemes are known as ‘prioritisation’ and ‘reservation’. The analysis presented in the following two subsections examines to what extent performance can be improved upon when prioritisation and reservation mechanisms are used.

7.3.1 Prioritisation in QoS

One approach to QoS is to use the Type of Service (ToS)-based relative priorities of the IPv4 header (or the 4-bit Priority field of the IPv6 header), which indicates in a simple way the relative delay and drop sensitivity of a packet, as depicted in Figure 7.11. This method gives *TCIS* streams higher priority than data packets, but does not provide a guarantee of bandwidth or latency.

Although traffic prioritisation is not part of the DVB/DAVIC specification, a faster transmission for the delivery of *TCIS* streams can be provided by mapping the *ToS* field with 2-levels of priority at the DVB/DAVIC MAC layer, as explained below.

The ToS indicates the desired usage of the packet. The field itself contains a 3-bit precedence indicator for the priority of the packet and 3 flags (D, T and R) to show whether delay, throughput or reliability are relevant for the transmission. Most routers in the Internet are as yet set to ignore the ToS field.

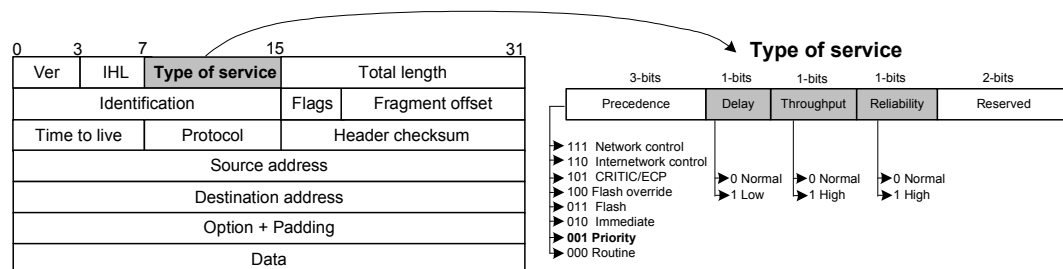


Figure 7.11 – IP header structure.

For DVB/DAVIC cable networks, this field can be used to provide a faster delivery of *TCIS* streams by assigning a higher priority to such streams. In this way, a packet that arrives at the NIU can be classified according to its *ToS* requirement and placed in a queue with two or more levels of prioritisation.

In this section, a mixed traffic situation with 41.7 kbps (as described in Section 5.3.3) was used in order to provide a performance analysis when prioritisation for QoS is considered. For this analysis only the *splitting tree algorithm* was used because of its superior performance over the *exponential backoff algorithm*. Results presented here are for the *pure reservation access* and the *piggyback* mechanisms.

From Figure 7.12a, it can be seen that packet access delays produced by the *PRA* mechanism when prioritisation is not supported, were almost the same for both traffic types. The maximum number of stations supported was 40 for VoIP traffic (with mean access delays of under 50ms) and 42 for Internet traffic (with mean access delays of under 500 ms). This is to be expected, since both traffic types were treated equally.

However, from Figure 7.12b when prioritisation was supported, on high traffic loads or congestion periods (above 44 stations or 59% of the *cc*), it was seen that packet access delays for VoIP traffic produced by the *PRA* and the *PG* mechanisms remained at

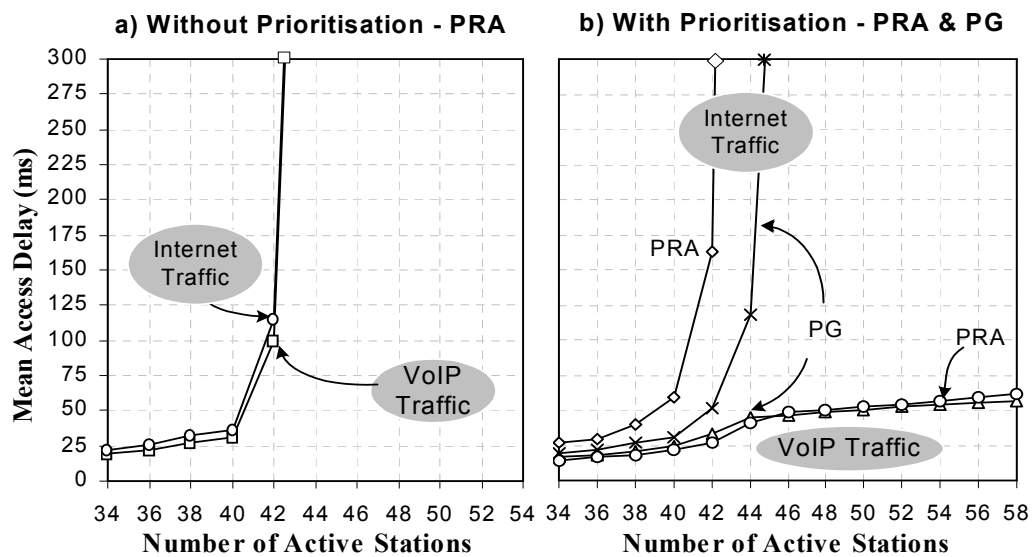


Figure 7.12 – Mean access delay vs. No. of active stations for VoIP and Internet traffic. *Splitting tree algorithm* with and without prioritisation.

approximately 50 ms, which is suitable for the support of audio streams with heavy traffic loads. Delays for Internet traffic were lower with the *PG* mechanism due to the continuous piggyback requests on high traffic loads, which avoids contention access.

In terms of system throughput, when prioritisation is not used (see Figure 7.13a), we can observe that the throughput for both traffic types increases almost linearly with respect to the NIUs population up to the point of saturation (reported at 57% of the *cc*, with 42 stations). After this, VoIP and Internet packets start accumulating in the NIUs queues. Conversely, with prioritisation (Figure 7.13b), it can be seen that both reservation request mechanisms transmit all VoIP streams in the first place and then use remaining bandwidth for Internet packets. As explained in Section 7.2.7, the *PG* mechanism achieves a higher system throughput than the *PRA* because of the reduced number of collisions in periods of congestion.

With very high traffic periods (above 73% of the *cc* or 54 stations), there is a slight difference for VoIP traffic. With the *PG* mechanism after transmitting the VoIP streams some piggybacks requests are made for Internet traffic. These requests at the headend delay subsequent piggyback requests made for VoIP traffic. This is because the headend grants all requests using a *best-effort* service. This problem can be solved by replacing

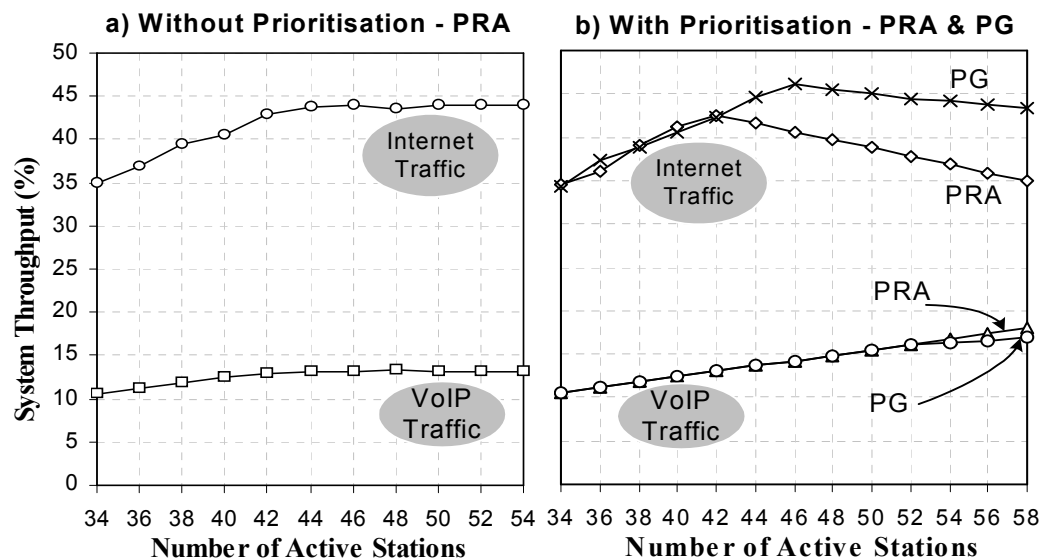


Figure 7.13 – System throughput vs. No. of active stations for VoIP and Internet traffic. *Splitting tree algorithm* with and without prioritisation.

the scheduling mechanism at the headend with one that supports at least two levels of piggyback or even reservation requests and also gives higher priority to VoIP traffic or *TCIS* applications in general.

Although this prioritisation mechanism offers an improved performance for *TCIS* applications, a more efficient mechanism is examined in the following section.

7.3.2 Reservation (Fixed-rate access) in QoS

The second approach for QoS with a guaranteed service, uses a fixed-rate access connection. In such connections data is sent in slots assigned at a fixed-rate based access region in the upstream channel, as described in Section 3.5.4, Figure 3.6. These slots are uniquely assigned to a connection by the INA. The number of slots needed by a connection and the periodic intervals are negotiated during the connection setup. The complete set of procedures is illustrated in Figure 7.14.

When an NIU requires a new fixed-rate connection or needs to change fixed-rate parameters, a **Resource Request** message is first sent to the INA by the NIU, including the new parameters. Example parameters are *new cyclic assignment needed*, *requested bandwidth*, *distance between slots* and *connection identifier (CID)*. The INA answers such requests by sending a **Connect Message** to the NIU, indicating whether the *new cyclic assignment* is granted and if so, a new set of fixed-rate parameters are provided such as *frame length*, *fixedrate start*, *fixedrate distance*, *fixedrate end* and *CID*.

Connection-Setup

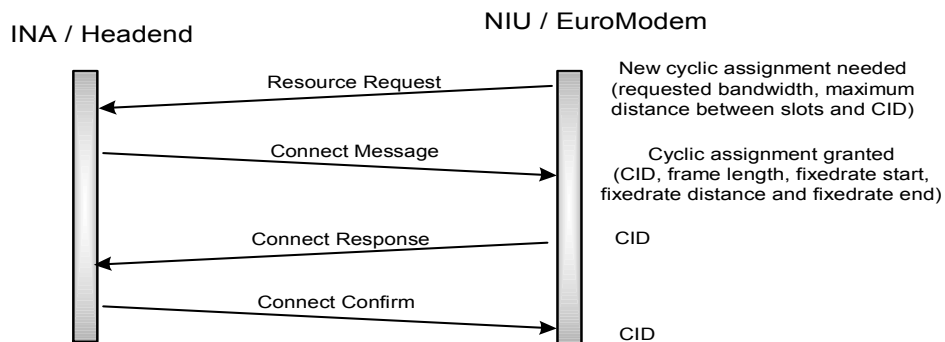


Figure 7.14 – Connection-setup for a Fixed-rate access connection.

The *frame length* parameter indicates the number of successive slots to use in the fixed rate access region associated with each fixed rate slot assignment. The *fixedrate start* parameter indicates the starting slot within the fixed rate access region that is assigned to the NIU. The NIU may use the next *frame length* slots of the fixed rate access regions. The *fixedrate distance* parameter represents the distance between additional slots assigned to the NIU and the *fixedrate end* parameter indicates the last slot that may be used for fixed rate access.

Subsequently, the NIU accepts the connection by sending a **Connect Response** message (indicating its CID) to the INA, which in turn answers this message by sending a **Connect Confirmation** message to the NIU.

The analysis to follow will address the optimal system performance of the DVB/DAVIC cable communications protocol when QoS (with a guaranteed service) is supported. Two traffic types were analysed: mixed traffic and VoIP traffic. For this analysis the optimised simulation parameters presented in Table 7.1 were used with the following exceptions:

- The upstream data rate of 3.088 Mbps was changed for an upstream channel with a capacity of 6.176 Mbps. This change was considered necessary because the results presented in the previous sections and also in the previous two chapters were based on a 3.088 Mbps upstream channel. The DVB/DAVIC protocol is also capable of supporting a 6.176 Mbps in the upstream direction. Therefore, in order to approach the maximum system performance in terms of the maximum number of active stations supported, an upstream channel with a capacity of 6.176 Mbps was used.
- The number of signalling frames transmitted in the 3 ms period was changed to 4, instead of 1. This is because stations can achieve a faster interaction (as examined in Section 5.4.6) with the headend, if more than one signalling frame is transmitted within the 3 ms period, which results more convenient for the support of *TCIS*.
- The minimum number of CSs allocated per signalling frame was changed to one, instead of two. In Section 6.2.2.2, it was proved that the maximum system

performance is yielded with at least one CS per signalling frame when the *splitting tree algorithm* is used.

- The maximum number of active stations analysed was up to 680 instead of 180 in order to cover the point when saturation is experienced and very low-data rate streams are delivered.

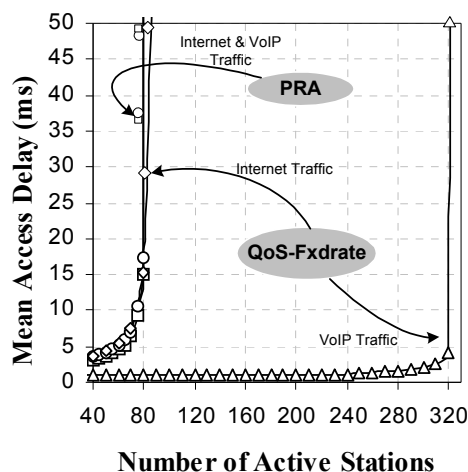
A) Performance analysis for mixed traffic

In this section, a performance comparison is provided when the *PRA* and QoS (with a fixed-rate connection) mechanisms are used for the transmission of mixed traffic. The simulations were carried out using only the *splitting tree algorithm* and the mixed traffic consisted of 9.7 kbps of VoIP traffic and 32 kbps of Internet traffic, as described in Section 5.3.3.

It can be seen in Figure 7.15a that the delay characteristics for both traffic types were similar for the *PRA* mechanism, which supports approximately 80 stations. On the other hand, with QoS not only an extended number of VoIP connections can be supported (up to 320), but also the mean access delay is reduced and ranged between ≈ 1 or 2 ms before the system becomes unstable.

This reduction in access was because the number of MCI frames transmitted in the 3ms period was increased to 4 and this helped the NIUs to achieve a faster interaction with the INA, and also because VoIP traffic had precedence over IP traffic at the headend.

a) Mixed Traffic: Mean Access Delay



b) Mixed Traffic: System Throughput

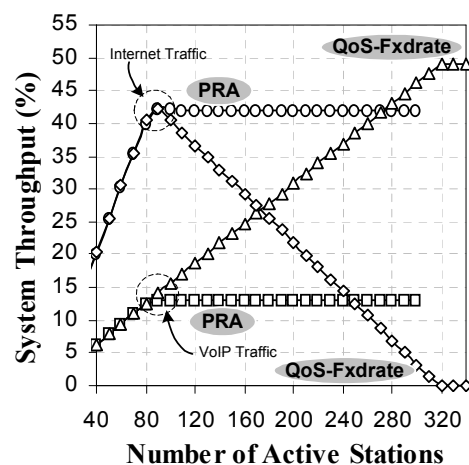


Figure 7.15 – Mean access delay and system throughput vs. No. of active stations. *Splitting tree algorithm* with mixed traffic.

From Figure 7.15b, the maximum sustainable system throughput was $\approx 56\%$ of the cc for the *PRA* mechanism. For the mechanism provided by QoS, the maximum system was also $\approx 56\%$. Internet traffic was choked off because in every MCI frame, the headend schedules first the bandwidth for VoIP traffic and then in the remaining slots IP traffic. This becomes more evident when the maximum system throughput is achieved (at 56% of the cc , produced by 80 stations).

B) Performance analysis for VoIP traffic

In this section the performance is focused on the case in which a fixed-rate connection is used for the delivery of VoIP streams. Here two different audio codecs are used for the analysis, G.711 [56] and G.723.1 [57]. So far, most of the VoIP streams examined have been based on the codec G.723.1, which encapsulates 4 VoIP frames of 20 bytes every 120 ms. In order to approach the maximum system performance that the DVB/DAVIC protocol can achieve, we have used four possible configurations for each codec. In addition, to increase the number of streams supported, header suppression (which is an advanced functionality in the latest version of this protocol [35]) is also considered in this analysis.

Codec G.711 was considered here to stress the CATV network and also because EuroModems (class B) are more likely to use this codec for quality voice calls. G.711 is the mandatory codec according to the ITU-T H.323 conferencing standard [59], which uses Pulse Code Modulation (PCM) to produce a data rate of 64 kbps. This audio codec creates and encapsulates a VoIP frame of 80 bytes every 10 ms. In order to reduce the protocol overhead involved, a frame size of 30 ms (240 bytes) can also be used, according to the *Packet Cable Audio/Video Codecs Specification* [80].

From Table 7.2 (second and fourth column) we can see that without header suppression (HS) only 55% (for a frame size of 10ms) and 78% (for a frame size of 30ms) is voice data. According to [47], there is interest in the cable modem communities in removing the requirement for LLC encapsulation of voice streams at the MAC layer, which would increase the fraction of each voice data in each packet to 63% and 83%, respectively.

Further, header suppression involving the RTP, UDP and IP headers would result in even more significant bandwidth savings. In the simulations it was assumed that the RTP, UDP and IP headers can all be suppressed, which is reasonable if the INA maintains additional state information on all active voice connections [47]. Thus, the efficiency calculated from Table 7.2 (third and fifth column) is considerably improved to 91% and 97% for frame sizes of 10 and 30 ms, respectively.

Similarly for G.723.1, four configurations were also analysed as indicated in the last four columns of Table 7.2. The frame sizes for this codec were of 30 and 120 ms.

Table 7.2 – VoIP codecs: G.711 and G.723.1

	G.711 - 64 kbps (ISDN)				G.723.1 - 5.3 kbps (Internet)			
	without HS	with HS	Without	with HS	without HS	with HS	without HS	with HS
Frame Size [ms]	10	10	30	30	30	30	120	120
Voice Frame [bytes]	80	80	240	240	20	20	80	80
RTP [bytes]	12	0	12	0	12	0	12	0
UDP [bytes]	8	0	8	0	8	0	8	0
IP [bytes]	20	0	20	0	20	0	20	0
LLC [bytes]	3	3	3	3	3	3	3	3
SNAP [Bytes]	5	5	5	5	5	5	5	5
Ethernet MAC [bytes]	18	0	18	0	18	0	18	0
Voice Packet size [bytes]	146	88	306	248	86	28	146	88
Net Data rate [kbps]	116.8	70.4	81.6	66.1	22.9	7.5	9.7	5.9

a) G.711 Performance

Results presented in Figure 7.16a revealed that the maximum number of streams supported with a frame size of 10 ms is about 27 stations, which achieves a maximum system throughput at approximately 50% of the channel capacity (Figure 7.16b). Increasing up to three the number of frames per audio packet not only results in a gain of about 10% on system throughput, but also in an increased number of VoIP streams supported (up to 45 stations).

The maximum system capacity is yielded when header suppression is considered, achieving up to 53 VoIP connections, regardless of the frame size. However, the maximum system throughput of about 60% of the cc is only produced with a frame size of 10 ms. The other frame size achieves a reduced capacity at approximately 56% of the cc . This is because with a frame size of 30 ms, the VoIP packet size to be transmitted is of 248 bytes that requires an additional (8-byte) AAL5 header plus 32 bytes of padding stuff. This extra padding is used to get an entire AAL5-PDU multiple of 48 bytes, which results in a waste of bandwidth and degrades the system throughput as indicated in Figure 7.16b.

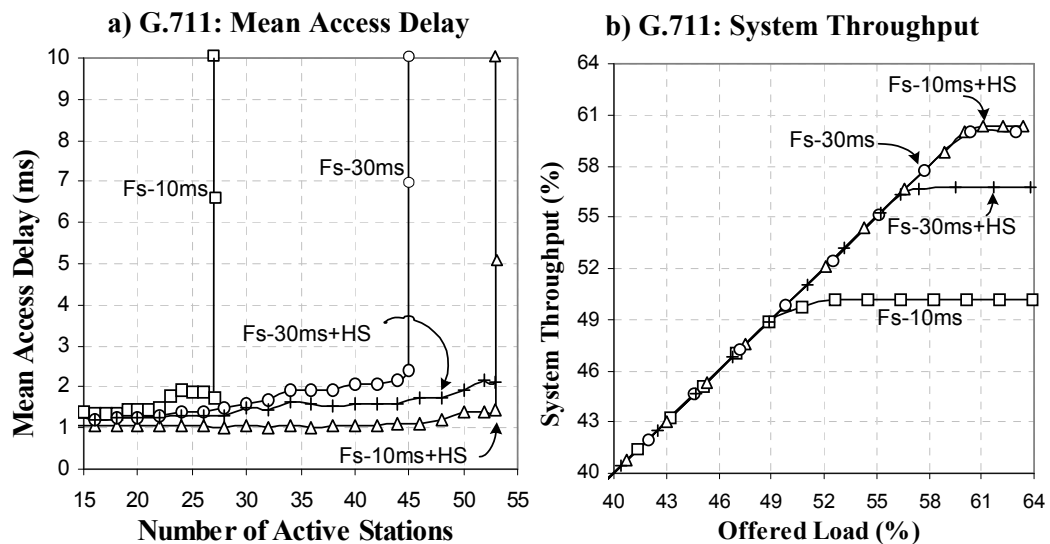


Figure 7.16 – Mean access delay and system throughput.
Splitting tree algorithm and VoIP traffic (G.711 at 64 kbps).

b) G.723.1 Performance

Results for the codec G.723.1 (presented in Figure 7.17a) show that by using a frame size of 30 ms, up to 160 stations can be sustained at 59% of the *cc*. This number can be extended to 320 VoIP connections with header suppression, but a reduction in terms of system throughput is obtained, achieving about 39% of the channel capacity. This loss (of about 20% of the *cc*) is caused by the 12 bytes padding used to encapsulate the 28-byte audio packet to fit into one (48-byte) AAL5 PDU (as indicated above). The other three Voice packet configurations do not suffer this large reduction, because they fit properly into a fixed number of ALL5-PDUs.

With codec G.723.1, optimum system performance is obtained when header suppression is considered and a frame size of 120 ms is used. The network capacity increases to the highest figure for this research supporting up to approximately 640 VoIP connections and producing a mean access delay below 2 ms before congestion is experienced (Figure 7.17a). This equates to 60% of the available channel capacity (Figure 7.17b).

The results presented in this subsection for both audio codecs can be validated by using Equations 7.1 and 7.2, for the maximum system throughput sustainable and the maximum number of streams supported, respectively.

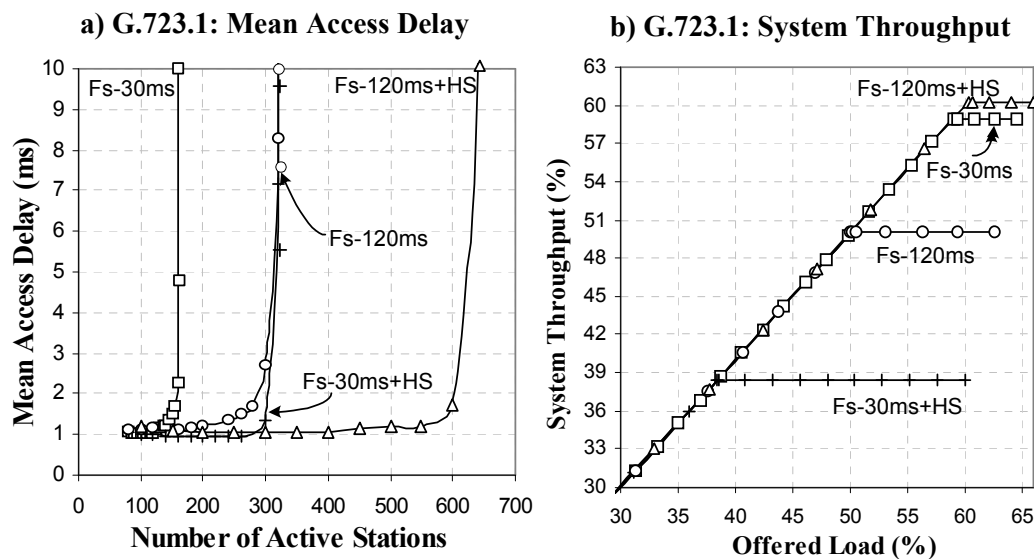


Figure 7.17 – Mean access delay and system throughput.
Splitting tree algorithm and VoIP traffic (G.723.1 at 5.3 kbps).

$$S_{\max} = CC \cdot \left(\frac{Pk_{size}}{Pk_{slots}} \right) \cdot \left(\frac{RS}{RS + CS} \right) \quad (7.1)$$

$$N_{\max} = \frac{S_{\max}}{Pk_{size} \cdot \lambda} \quad (7.2)$$

where RS and CS corresponds to the number of slots to be allocated to the reservation and contention access respectively. For instance, for the codec G.723.1 with a frame size of 30 ms (20 bytes of sampling), without header suppression the audio packet size to be transmitted becomes of 86 bytes (see Table 7.2, fifth column). Then, from Equations 7.1 and 7.2,

$$S_{\max} = 6176000 \cdot \left(\frac{86 \cdot 8}{2 \cdot 64 \cdot 8} \right) \cdot \left(\frac{8}{8+1} \right) = 3688 \text{ kbps} = 59.7\% \text{ of the } cc$$

$$N_{\max} = \frac{S_{\max}}{N_r} = \frac{S_{\max}}{P_s \cdot \lambda} = \frac{3688000}{(86 \cdot 8) \cdot (1/0.03)} \approx 161$$

From simulations using this codec, a maximum system throughput of 59% of the *cc* and a maximum number of streams supported of 160 was obtained. Thus, theoretical results were seen to be similar. In general, for all four configurations presented for each codec a maximum deviation between simulation results and theoretical results was found to be less than 1.5%. This deviation can be attributed mainly to collisions and protocol overheads.

7.4 Comparison: DVB/DAVIC vs. DOCSIS

So far only a performance characterisation and optimisation for the European cable communication system ‘DVB/DAVIC’, under various configurations and different traffic situations has been introduced. However, as stated in Section 2.5, the EuroDOCSIS protocol is also a serious alternative for the European market.

This section continues with a comparison between both standardised cable communication protocols: DVB/DAVIC vs. DOCSIS. The comparison focuses on performance issues and technical characteristics at the MAC and PHY layers.

In Barton [8], a comparison of the main characteristics of each protocol was provided from an American perspective. Here, an overview and status of each standard is provided. Furthermore, a technical comparison of the major characteristics, strengths, advantages and disadvantages of each standard are also described.

A less extensive comparison but with a European perspective is reported in [55]. In this paper the authors presented an overview of each protocol followed by a brief comparison of the major technical characteristics of each standard.

An initial performance comparison carried out in this research can be found in [84]. This comparison is based on the DVB/DAVIC, DOCSIS and the (withdrawn) IEEE 802.14 standard. The comparisons presented in this section focus on services, MAC layer, PHY layer technical characteristics, and performance comparisons in terms of volumetric data (at the physical layer), efficiency, mean access delays and maximum system throughput.

7.4.1 Technical comparisons

7.4.1.1 MAC layer comparisons

At the DVB/DAVIC MAC layer, four access modes are provided. The first mode (contention access) is based on a contention access mechanism, which allows users to send information at any time with the risk of collisions. The second and third modes (fixed-rate and reservation access) are contentionless, in which the INA either provides a predefined amount of slots to a specific NIU, or a given bit rate requested by an NIU until the INA stops the connection on NIU's demand. These access modes are dynamically shared between time slots, which allows the NIUs to know when contention, reservation or fixed-rate transmission is or is not allowed. This is to avoid a collision for the two contention-less based access modes. The fourth mode is called ranging access, in which slots are used to measure and adjust the time delay and the power level for upstream transmission.

Data transmission in the DOCSIS MAC layer is based on a request/grant scheme. At regular intervals, the CMTS provides timing, ranging, registration, transmission and contention resolution information for every CM on the network. In the upstream, each

minislot can be used either for contention, contention/data, initial maintenance or station maintenance. A CM with data to send issues a request, using contention minislots. In the case of a collision, this protocol applies the *exponential backoff algorithm* and the *initial* and *truncated* backoff values of this algorithm are indicated in each signalling frame called as MAP. This frame also describes the usage of the upstream bandwidth.

7.4.1.2 PHY layer comparison

The basic characteristics at the physical layer are summarised in Table 7.3. In this table the downstream and upstream spectrum allocations and the slight variations in the frequency ranges for the DVB/DAVIC and DOCSIS standards are shown. The downstream channels support the legacy analogue broadcast television (frequency range are \approx 80-450 MHz) and multiples of 1-6 MHz (for DOCSIS) or 1-8 MHz channels (for DVB/DAVIC and EuroDOCSIS) in the 450-860 MHz region for transmission of digital data. The upstream channels are also divided into 1-6 MHz but poor SNR limits the data capacity to 1-10 Mbps per channel for DOCSIS and 1-6 Mbps for DVB/DAVIC. This is a low data rate compared to the downstream channel where 28-40 Mbps and 42-52 Mbps are available for DOCSIS and DVB/DAVIC, respectively.

In terms of modulation schemes, most manufacturers have implemented 64 and 256-QAM for the downstream and QPSK for the upstream channels. DOCSIS also supports a 16-QAM modulation in the upstream direction.

For upstream transmission at the physical layer, DOCSIS MAC PDUs (which are composed of Ethernet packets) are segmented into PHY codewords, which are of

Table 7.3 – Frequency allocation and modulation characteristics.

	Feature	DVB/DAVIC	DOCSIS
Upstream	Modulation	QPSK, 16-QAM under development	QPSK, 16-QAM
	Frequency	5 – 65 MHz	DOCSIS: 5–42MHz, EuroDOCSIS: 5-65MHz
	Spacing	200 kHz, 1 MHz, 2 MHz, 4 MHz	200, 400, 800 kHz, 1.6, 3.2MHz
	Data rates	Mandatory: 3.088 Mbps, Optional: 256 kbps, 1.54 kbps, 6.176 Mbps	320 kbps, 640 kbps, 1.28 Mbps, 2.56 Mbps, 5.12 Mbps, 10.24 kbps
Downstream	Modulation	IB: QPSK, OOB: 16, 32, 64, 256-QAM	64-QAM or 256-QAM
	Frequency	OOB: 70-130 MHz / 300-862 MHz,	80 – 860MHz
	Spacing	IB: 300-862 MHz OOB: 1 MHz, 2 MHz,	DOCSIS: 6MHz, EuroDOCSIS: 8 MHz
	Data rates	IB: 7/8 MHz 52 Mbps (256-QAM), 42 Mbps (64-QAM)	42 Mbps (256-QAM), 30 Mbps (64-QAM)

variable size (ranging from 16 to 253 bytes). These codewords contain FEC parity, also of variable size (from 0 to 10 bytes). Both a preamble and guard time fields of variable length are then added at the beginning and at the end of all the codewords for synchronization purposes between the CM and the CMTS. Such codewords are transmitted as a continuous series of minislots. The size of the minislot is set by the CMTS during initialisation but can be varied. The length of a minislot is a multiple of 6.25 μ s (i.e. 8, 16, 32 bytes etc.).

In the upstream, DVB/DAVIC uses a 64-byte slot format for the transmission of data at the physical layer. The slot format consists of a Unique Word which provides a burst mode acquisition method, a payload area that contains a single ATM cell and a Reed-Solomon parity field [34], which provides 3-bytes of FEC over the payload area and a Guard Band field for synchronization. The basic characteristics of these two protocols are outlined in Table 7.4.

Table 7.4 – Technical characteristics.

Feature	DVB/DAVIC	DOCSIS
Services	Internet access, high speed interactive Set-top-box, VoIP, SNMP	Internet access, low speed interactive Set-top-box, VoIP, SNMP
Upstream Packet Format	64-byte Slot based on a ATM cell transport with IP adaptation layer translation	Variable Length (based on 64-1500 bytes Ethernet packets), Native IP with QoS
QoS Granularity	53 bytes + 11 bytes PHY overhead	8-16 bytes
QoS Services	BE plus ATM derived class of services: CBR, ABR.	BE, CIR and prioritisation in DOCSIS 1.0 and RTP, nrt-RTP, UGS added in DOCSIS 1.1
Collision Resolution	<i>Exponential backoff</i> and <i>Splitting tree algorism</i>	<i>Exponential backoff algorithm</i>
ATM Support	Mandatory	Optional
Access Modes	Ranging, contention, reservation and fixed	Contention and reservation
Security	RSA/DES Encryption and clone detection	RSA/DES Encryption and clone detection
Commercial Deployment	DVB/DAVIC EM: Q3-99, Q3-00.	DOCSIS 1.0: Q2-99; DOCSIS 1.1 Q4-99

7.4.2 Performance comparisons

The performance comparison to follow addresses some of the fundamental properties and scalability of the upstream channel for both protocols. The analysis focuses on the performance comparison for the upstream channel, which is the limiting factor on CATV networks (as stated in Section 1.2.2), and is critical in the delivery of services to individual subscribers on demand. Key issues of the analyses address the following performance comparisons: protocol efficiency and volumetric data at the physical layer,

maximum throughput achieved per cable modem, effects of varying the packet size in isochronous streams and maximum system capacity in terms of active cable modems.

For the DOCSIS protocol, the CSF (v.13) based on the OPNET simulation package was used [77]. The simulation parameters considered for the performance analysis are summarised in Table 7.5. In both protocols a scheduler at the headend with a simple FIFO service was used. For the DOCSIS protocol, priority service was not considered, since the DVB/DAVIC standard does not support traffic prioritisation. Therefore, all streams generated are assumed to have the same access priority.

In addition, the default upstream data rate of 3.088 Mbps for the DVB/DAVIC protocol was selected. The corresponding data rate for the DOCSIS protocol was set at 2.056 Mbps. The concatenation and piggyback features of the DOCSIS protocol were not considered for this analysis, since such characteristics are not supported by the DVB/DAVIC protocol.

Table 7.5 – Simulation parameters.

Parameter	DOCSIS	DVB/DAVIC
Upstream channel capacity (QPSK)	2.56 Mbps	3.088 Mbps
Number of cable modems	Up to 350	Up to 350
Minislot/slot size in bytes	16 (minislot)	64 (slot), 21 (minislot)
Max. No. of minislots/slots in MAP/MCI	Max. 4096	Fixed 36
Min. No. of contention slots in MAP/MCI	16	[2-3]
Contention resolution algorithm	<i>Exponential backoff algorithm</i>	<i>Splitting tree algorithm</i>
Max. number of IEs in MAP	240	-----
FEC-T bytes	3-SD, 5-LD*	3
Codeword length	75-SD, 245-LD*	53 (ATM cells)
Last codeword shortened	True	-----
Guard time size	2 bytes (variable)	1 byte (Fixed)
Distance from nearest/farthest modem to the headend	10-16Km, rnd. distributed	10-16Km. rnd. distributed
Simulation time	60 seconds	60 seconds

*SD = Short data codeword, LD = Large data codeword

A) Protocol efficiency and volumetric data comparisons

This analysis addresses a comparison of the volume of data (transferred upstream by a user data request transmission) and protocol efficiency for the DVB/DAVIC and the DOCSIS standards.

For the upstream volumetric data (Figure 7.18a), the DVB stepped curve is caused by the encapsulation of PDU messages into ATM cells (of 53 bytes), which are then transmitted throughout the upstream channel as 64-byte frame slots. Therefore, a user message of 1450 bytes (without protocol overhead) causes the DVB protocol to request the following data: $(1450\text{-payload} + 20\text{-TCP} + 20\text{-IP} + 3\text{-LLC} + 18\text{-MAC} + 8\text{AAL5}) / 48$ AAL5 PDUS $\approx 32\text{-slots} = 2048$ bytes. This results in 71 % upstream efficiency as indicated in Figure 7.18b. Here, every upstream slot transmitted includes 6 bytes of FEC information to correct up to 3 bytes (T=3) over each ATM cell. Figure 7.19 presents the upstream PDU structures of this protocol.

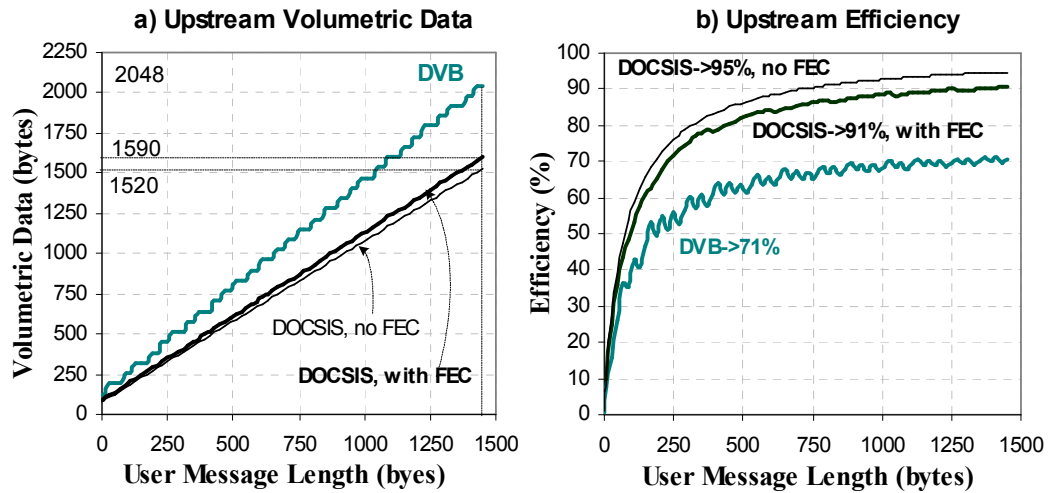


Figure 7.18 – Upstream volumetric data and efficiency.

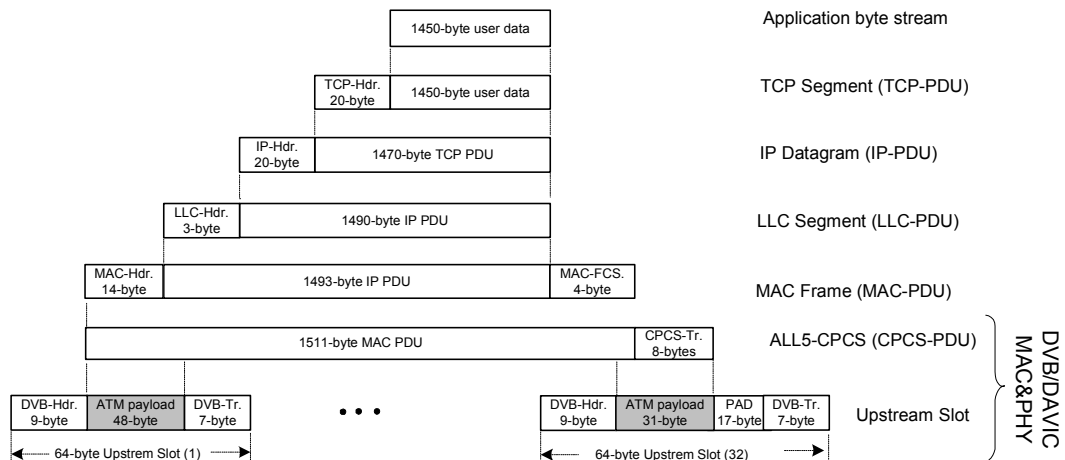


Figure 7.19 – DVB/DAVIC upstream PDU structures.

Similarly, the DOCSIS stepped curve is caused by the encapsulation of PDU messages into codewords (of ≈ 220 bytes), which are then transmitted as a continuous series of minislots. Figure 7.20 shows the PDU structures for this protocol. For instance, for the same user data of 1450 bytes, the DOCSIS protocol requests $(1450\text{-payload} + 20\text{-TCP} + 20\text{-IP} + 3\text{-LLC} + 18\text{-MAC} + 6\text{-DOCSIS MAC}) / 245 \text{ byte-codeword} = 6$ large codewords + 1 short codeword ≈ 1590 bytes (or ≈ 100 minislots), for an upstream efficiency of 91%. Here, every codeword includes 10 bytes of FEC information to correct up to 5 bytes over the MAC PDU. Without FEC the volumetric data at the PHY layer becomes of 1520 bytes (≈ 95 minislots) for an increased efficiency up to 95%.

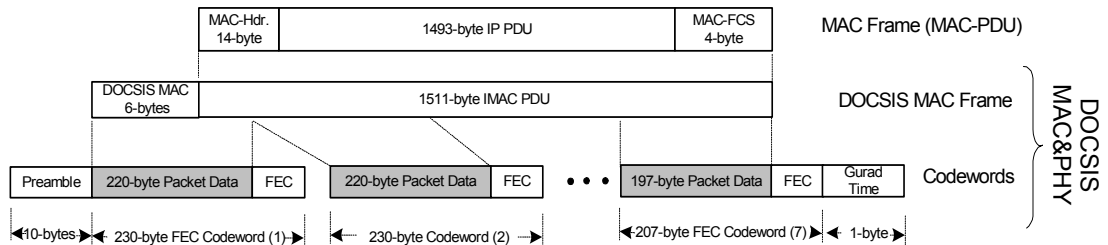


Figure 7.20 – DOCSIS upstream PDU structures.

B) Maxim throughput achieved per cable modem

In order to obtain the maximum throughput per cable modem that each protocol can achieve, a single node (cable modem) network was considered with typical Ethernet packet sizes ranging from 64 to 1518 bytes. A traffic generator within the node was set to produce the maximum number of packets that it could receive from the CPE interface, (as introduced in Section 5.4.7.1), which was a 10 Mbps constant stream with variable number of packet per second depending on the packet size.

All references to packet size in this analysis (and also in the following analyses) will refer to the size of the packet as it enters the system from the CPE interface. This would be the packet size that the cable modem submits for delivery over the cable network, which does not include CATV MAC and PHY overheads. However, the CPE MAC and possible layer 2 and layer 3 overheads are included.

The maximum throughput and upstream channel utilisation versus the packet size generated is shown in Figure 7.21. In terms of throughput (Figure 7.21a), the DOCSIS protocol is superior to the DVB/DAVIC protocol. For the maximum Ethernet packet size (1518 bytes), the maximum sustainable throughput per DOCSIS cable modem was ≈ 1.6 Mbps, which corresponds to about 62% of the upstream channel capacity, in comparison to ≈ 1 Mbps (32% of the cc) achieved by each DVB/DAVIC cable modem. In terms of utilisation (Figure 7.21b), the difference was 19% of the link capacity in favour of the DOCSIS protocol.

However, for small packet sizes under 256 bytes, results reveal that both communication protocols are highly inefficient. The maximum throughput does not exceed 25% of the link capacity even though the offered load was a constant 10 Mbps stream. The worst case scenario is released for the 64-byte packet size, where the throughput was as low as 2% of the cc (≈ 56 kbps) for DVB/DAVIC cable modems, in comparison to 7% of the nominal channel capacity (≈ 181 kbps) provided by the DOCSIS cable modems.

A closer analysis of the difference between throughput and utilisation, simulation results showed that an increased protocol overhead is involved in the DVB/DAVIC protocol that reduce considerably the protocol efficiency, as examined in the previous scenario.

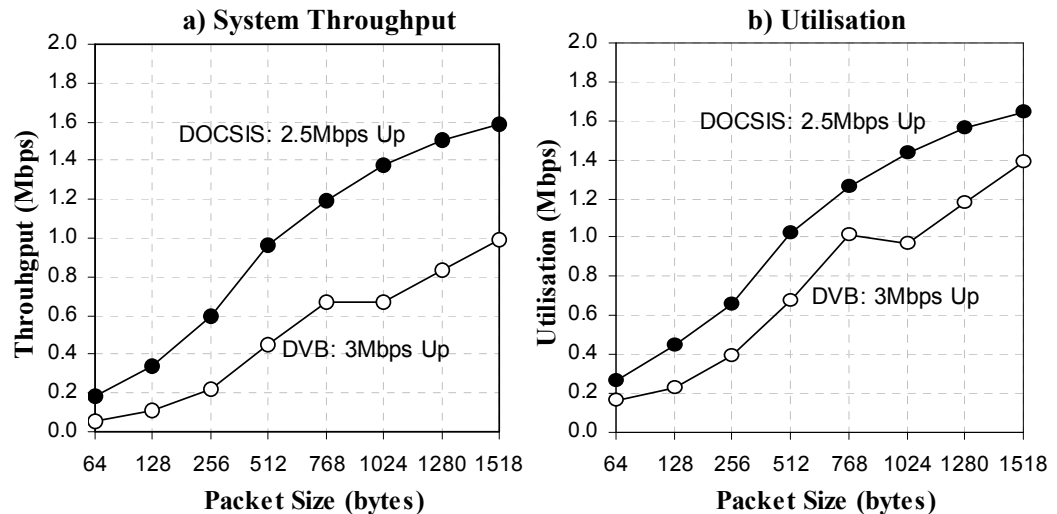


Figure 7.21– Maximum channel capacity for DVB/DAVIC and DOCSIS for one node.

For instance, for packet sizes of 1518 bytes, this difference resulted in approximately 2% and 13% of the cc for the DOCSIS and DVB/DAVIC protocols, respectively.

C) Maximum network capacity

In this analysis, we now address the maximum scalability of the network in terms of mean access delays and system throughput/utilisation for an increased network population. The number of stations was increased to 340 to move beyond saturation points. The traffic load generated by each station was a single-Ethernet packet of 1518 bytes with an exponentially distributed inter-arrival rate of 1 packet per second. This is equivalent to 12 kbps streams.

Results presented in Figure 7.22a tell us that both protocols can support the same number of stations, (at least in this traffic situation) with approximately 160 streams supported, after which delays become too large for the chosen application. It is apparent that for medium network traffic loads, DVB/DAVIC cable modems provide slightly lower delays than the DOCSIS protocol, due to the faster interaction with the headend. DVB/DAVIC always sends an MCI frame every 3 ms and DOCSIS uses a variable scheme, where the number of minislots described per MAP can be from a few minislots

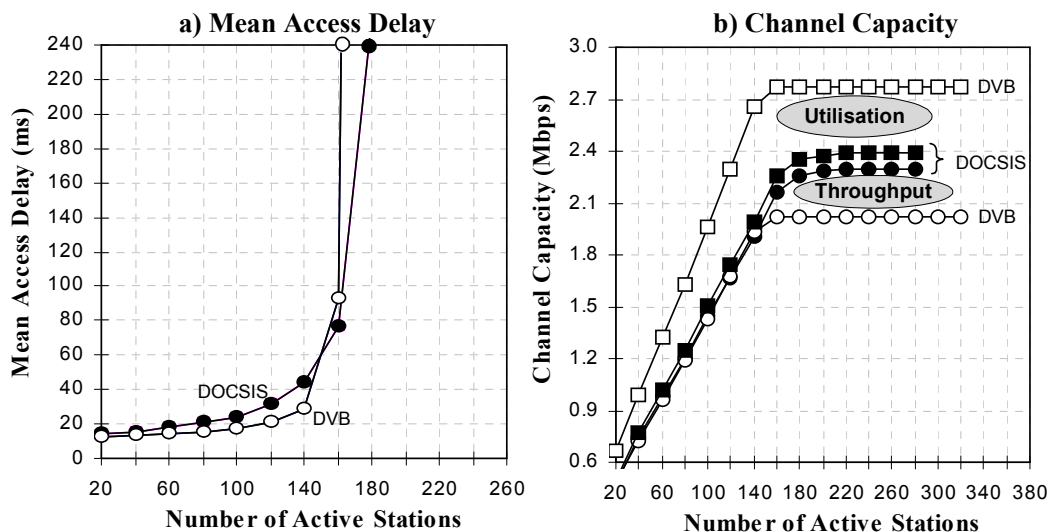


Figure 7.22 –Mean access delay and maximum channel capacity, for DVB/DAVIC and DOCSIS, 12.4 kbps streams with packet size = 1518 bytes.

up to 4096. This results in MAP being transmitted from ≈ 1 ms when only contention minislots are described up to $(4096 \text{minislot} \cdot 128 \text{bits} / 2560000 \approx) 200$ ms on high traffic loads.

Figure 7.22b shows an increased system performance for DOCSIS over the DVB/DAVIC protocol in terms of system throughput and utilisation, even though a reduced upstream channel capacity was used for the DOCSIS protocol. For this analysis, the maximum system throughput resulted in $\approx 65\%$ of the channel capacity for the DVB/DAVIC protocol, in comparison to 90% of the cc achieved by the DOCSIS protocol.

In terms of utilisation, the DVB/DAVIC protocol uses 89% of the channel capacity in comparison to 93% utilised by the DOCSIS protocol. Both protocols cannot achieve 100% of the upstream channel capacity due to slots assigned to the contention access region that remained unused. Reducing the minimum number of contention slots per MAP or signalling frame may result in an increase of upstream channel utilisation, but there may be also some drawbacks, such as an increased number of collisions in periods of congestion and therefore an increased mean access delay.

D) Effects of packet size in isochronous streams

Here, an analysis of the performance impact in terms of mean access delay is presented. The system throughput provided by each protocol when the packet size is ranged from 64 to 1518 bytes for different isochronous streams is shown. For this analysis, streams from 16 kbps to 128 kbps, suitable for low-rate timing critical interactive services such as compressed/ uncompressed voice, audio and low quality video were considered.

Figure 7.23 shows the performance comparison when isochronous streams at 32 kbps were used for small, medium and large packet sizes.

From the results for a 64-byte packet size, in Figure 7.23a it can be seen that the maximum number of streams that the DVB/DAVIC protocol can support is approximately 24, with access delays under 20 ms. The DOCSIS protocol supports about 8 streams more than the number achieved by DVB/DAVIC. Results for the other two packet sizes (256 and 1024 bytes) show that both protocols support the same number of streams for delay sensitive isochronous streams (under 20m), which corresponds to 54 and 57 streams for 256 and 1024-byte packet sizes, respectively. For some isochronous streams that support larger access delays over 20 ms, e.g. video, audio, the number of streams sustainable resulted in a large figure for the DOCSIS protocol.

In terms of system throughput, (see Figure 7.23b) the minimum Ethernet packet size yields a maximum system throughput of 1.4Mbps (55% of the cc) for the DOCSIS protocol and 1 Mbps (32% of the cc) for the DVB/DAVIC standard, which indicates that the DOCSIS protocol is capable of using approximately 23% of the upstream channel capacity for the delivery of more user data. For packet sizes of 256 and 1024 bytes, this difference resulted in 25% and 28% in favour of the DOCSIS protocol.

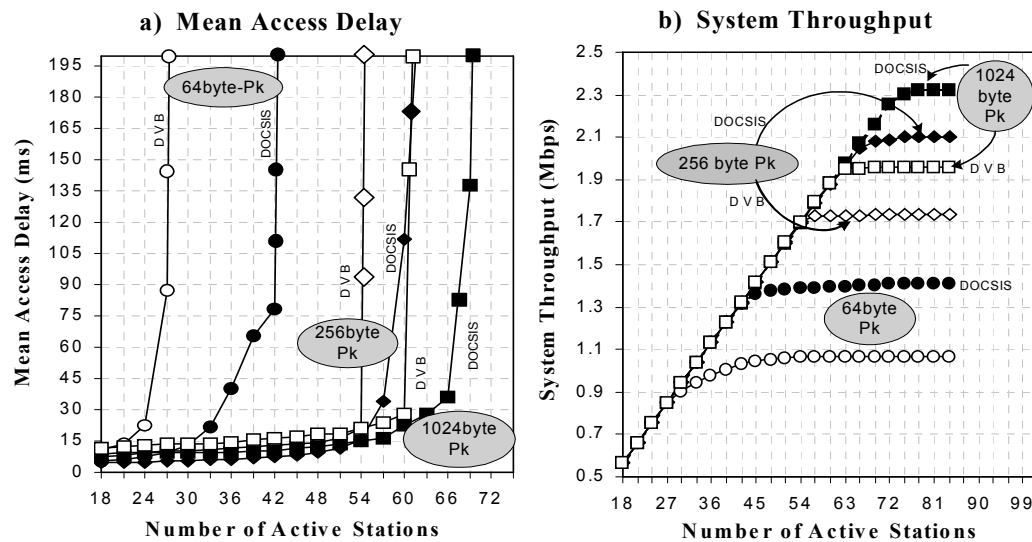


Figure 7.23 - Mean access delay and system throughput, for DVB/DAVIC and DOCSIS with 32 kbps isochronous streams and different packet sizes.

Results for the complete range of packet sizes and isochronous streams are summarised in Table 7.6. These results are displayed in terms of maximum number of streams supported for both protocols. For packet sizes from 64 to 512 bytes, access delays under 20 ms were considered. Meanwhile for larger packet sizes, (1024 and 1518 bytes) access delays under 100 ms were used, since it is more likely that larger delays are accepted for the support of isochronous streams when large packet sizes are utilised.

In general, for most of the packet sizes and isochronous streams evaluated, DOCSIS outperformed the DVB/DAVIC protocol. There were just few cases where the DVB/DAVIC supported an increased number of streams, as it was the case of low rate isochronous streams of 16 kbps and 32 kbps for the packet size of 128-bytes.

For higher data rate isochronous streams and small packet sizes, there were three instances where the DVB/DAVIC protocol failed to provide low transmission delays (under 20 ms). This is because according to the *CRGC* of the DVB/DAVIC protocol (depicted in Figure 3.3) the minimum packet delay is approximately of the order of 9 ms, and the inter-arrival time for the three cases indicated in Table 7.6, were either 8 ms or 4 ms. This resulted in infinite queue built-up at the transmitter side causing the streams to fail. In order to fix this problem, *enhanced reservation request mechanisms* are required, as introduced in Section 7.2 and [87]. Another possible solution is to decrease the transmission cycle of the DVB/DAVIC signalling frame, as analysed in Section 5.4.6.

Table 7.6 – Number of streams supported for DVB/DAVIC and DOCSIS.

Packet		Isochronous Streams							
Size (bytes)	Access Delay(ms)	16 kbps		32 kbps		64 kbps		128 kbps	
		DVB	DOCSIS	DVB	DOCSIS	DVB	DOCSIS	DVB	DOCSIS
64	≤20	62	62	24	32	No supported	20	No supported	6
128	≤20	96	87	48	43	19	23	No supported	12
256	≤20	108	110	54	54	27	27	12	14
512	≤20	112	125	59	62	31	31	15	15
1024	≤100	122	143	61	70	31	35	15	18
1518	≤100	125	144	63	71	31	36	15	18

As a final remark, results in terms of maximum number of streams supported for a 6.176 and 5.12 Mbps upstream channel can be found in [85] and [121] for the DVB/DAVIC and the DOCSIS protocol, respectively.

7.5 Conclusions

In this chapter several novel improvements were implemented that would give the DVB/DAVIC MAC protocol a superior performance and increased efficiency for the support of the next generation of communications services.

The first improvement was through the use of six enhanced-reservation-request mechanisms. As these mechanisms become more complex, they efficiently reduce and in some cases avoid the collision risk of reservation request transmissions, increasing markedly the system performance by reducing the access delay and achieving a higher system throughput. Results presented have shown that the basic reservation access mechanism of the DVB/DAVIC standard can be improved upon significantly by adopting these enhanced mechanisms. The *unsolicited grant slot* mechanism (when the CBR parameters are satisfied with regard to the actual data rate, packet size and packet interval) and the continuous piggyback mechanism are the most appropriate for the support of timing critical interactive services. However, the use of minislots with or without piggybacking requests, and the *enhanced pure reservation access* mechanisms are best suited for Internet traffic or bursty traffic. The performance increase provided by these mechanisms can only be obtained when the traffic characteristics are carefully analysed. The choice of a poor mechanism may reduce the service quality considerably.

The second approach for increased performance was through the use of Quality of Service. Here, the use of traffic prioritisation was found to have a significant effect on system performance. By giving higher transmission priority to isochronous streams than to data packets, low packet access delays can be sustained (in medium and in some cases in high traffic loads) achieving an increased number of delay-sensitive streams. The major drawback of this prioritisation technique is that it does not provide any guarantee of bandwidth availability or latency. However, in the reservation technique, through the use of a fixed-rate connection, a reduced packet access delay for the

delivery of isochronous streams, in the range of 1 and 2 ms could be guaranteed by the DVB/DAVIC protocol before large periods of congestion are experienced.

Combined with header suppression, bandwidth efficiency is increased to a large extent, achieving a much higher figure regarding the maximum number of streams sustainable. For instance, for VoIP streams (with codec G.723.1) it was found that by using header suppression and a large frame size of 120 ms, the maximum number of streams supported could be increased up to approximately 640 connections, in comparison to 160 when these features are not considered.

In this research a minimum packet access delay of about 1ms (for VoIP streams) was possible. However, with advanced synchronisation techniques this delay can be further reduced (under 1ms), for the delivery not only of VoIP packets but *TCIS* in general.

Finally, a comparison between the DVB/DAVIC and the DOCSIS protocol was also addressed in this chapter in order to evaluate the major properties provided by each standard at the MAC and PHY layers, which may help Internet Service Providers or Cable Network Operators to make the right choice when performance issues for the support of timing critical interactive applications, and high-speed Internet traffic are considered.

The DVB and the DOCSIS organisations are each producing their own cable modem specification. At the PHY layer these specifications are similar. However, at the MAC layer the solutions provided by these groups have little in common. In general, in most of the analyses, the DVB/DAVIC protocol achieved a reduced performance compared with the DOCSIS protocol. This was mainly because the DVB/DAVIC protocol encapsulates every datagram into ATM-AAL5 cells and suffers an overhead penalty for Segmentation and Reassembly (SAR) in an attempt to provide a faster transmission for QoS over the ATM protocol, while the DOCSIS uses a scheme that favours the delivery of variable length Internet protocol packets rather than ATM transfer, in an attempt to keep cost and complexity of cable modems down.

Chapter 8

FINAL CONCLUSIONS

8.1 Introduction

The work presented in this dissertation has addressed the issues of digital communications over the European cable communication protocol (DVB/DAVIC, ETS 200 800). The focus of this work has been the performance evaluation and optimisation of the upstream channel, which is more complex to analyse when compared with the downstream channel. This is because transmission on the downstream channel is handled exclusively by the headend (the Interactive Network Adaptor), simplifying operation. In contrast, the upstream channel is a shared access medium, which uses random (contention), reservation and fixed access techniques. The interval distribution of the start time of these access modes is dynamic and controlled by the headend. The boundaries of contention and reservation access are broadcast periodically in the downstream channel. The boundary of fixed rate access is assigned to a station at the beginning of connection. When collisions occur, a contention resolution algorithm is used to resolve them. This protocol uses two different CRAs, which are the *exponential backoff algorithm* and the *splitting tree algorithm*.

The series of analyses presented in this dissertation have concentrated on the effectiveness of the access modes and the CRAs defined in the DVB/DAVIC protocol specification with the key performance issues for access and data transmission in the upstream channel. In addition, several improvements have also been introduced to extend this protocol in support of high-speed data transmissions and in particular timing critical interactive services.

8.2 General discussions

Chapter 1 presented an overview of the main characteristics of CATV networks. Here it was highlighted that the bandwidth available in the upstream direction, makes CATV networks limited for the support of an increased number of users, particularly when transmitting interactive services with high capacity. The chapter went on to give an overview of the major measurements used in the field of performance analysis of communication protocols, and provided a comprehensive review of the current research in relation to stability of contention resolution algorithms and modelling of CATV protocols relevant to this work. An important point here is that all contributions in research of performance analysis of CATV protocols have been made in the last six years.

This chapter also discussed the contributions and novelty of this research. This project was started about two months after the first version of the DVB/DAVIC protocol came out. With the exception of the work carried out in [100], most of the research found in the literature focused on performance evaluation, improvements and optimisations for the IEEE 802.14 and DOCSIS MAC protocols.

In *Chapter 2* the relevant theory was presented for this work in relation to CATV networks, which included a description of traditional and modern cabling infrastructures, their differences and similarities. Here, the changes that have to be made to upgrade CATV networks were pointed out for the support of bi-directional digital communications.

In addition, background material was also covered in this chapter that included an overview of *cable modem technology*, possible cable modem configurations (internal, external or STB), cable modem providers and identified the requirements for the next generation of communications services.

This chapter also outlined the main characteristics of the DOCSIS, IEEE 802.14 and DVB/DAVIC and presented an overview of alternative technologies for high-speed digital access, including xDSL, and FTTx infrastructures.

In **Chapter 3** a comprehensive description of the DVB/DAVIC protocol and the DVB project was addressed. Here the main characteristics as well as the architecture, MAC operation, and a detailed description of the *exponential backoff algorithm* and the *splitting tree algorithm* adopted by the DVB/DAVIC protocol specification were described.

Chapter 4 presented the simulation model that was implemented for the performance evaluation of the DVB/DAVIC protocol. In summary, this model was based on the Common Simulation Framework, initially developed by MIL 3 and CableLabs. Such a framework was implemented in the OPNET Simulation Package and is used for the modelling of the DOCSIS protocol. We have used the basic network topology of this framework and programmed the functionality of the MAC and PHY layers of the DVB/DAVIC protocol (including the INA and NIUs), replacing the functionality of the MAC and PHY layers of the DOCSIS protocol, respectively.

In addition, new features for further optimisations were also incorporated in the simulation model. The major issues were the incorporation of three contention slot allocators, six enhanced reservation requests mechanisms, and a prioritised scheduler algorithm at the station's premises that were used in subsequent analysis.

In this chapter, an analytical model was also formulated based on single node scenario to validate the result that could be obtained through the simulation model. A multi-node analytical model was beyond the scope of this research, due to the complexity of the DVB/DAVIC protocol and its hybrid access method, which uses both random access with *exponential backoff* or *splitting tree algorithm* and elements of TDMA renders its theoretical analysis very complex. We have found that studies of contention resolution algorithms are either based on simulation for accurate results or examine the stability of the algorithms with several simplifying assumptions, such as an infinite number of stations.

The simplified analytical model implemented for the DVB/DAVIC protocol was based on an M/G/1 queuing system and addressed one of the basic performance properties of computer communications protocols, which relates to the mean packet access delays and maximum sustainable throughput of a station.

We have seen that results obtainable using the simulation model were in good agreement with theoretical results, with a maximum deviation of 3% for mean access delays and 2% for throughput. In this analysis it was found that the *scheduler-look ahead* is the major delay element in the transmission cycle, which dramatically reduces the throughput.

In *chapter 5*, before a performance optimisation or the incorporation of new mechanisms was approached, we first presented a rigorous performance evaluation of the fundamental properties of the DVB/DAVIC protocol, based on a 3 Mbps upstream channel.

The first issue addressed was the protocol efficiency and performance characteristics in terms of global offered load. Simulation results revealed that for the *exponential backoff algorithm*, the protocol can sustain linear throughput increase and bounded delays for offered loads up to 61% of the *cc*. The maximum system throughput was even higher and reached $\approx 65\%$. However, in order to achieve this throughput, the offered load needs to be higher than 70% of the *cc* at which point frame delay becomes unbounded. This throughput is higher when compared to other pure random access mechanisms, such as Aloha (37%), and CSMA/CD (60%), [117]. The main advantage is that the throughput maintains maximum values even under heavy traffic loads and does not degrade as rapidly as these other random access protocols.

A similar performance characteristic with linear throughput and delay increase was confirmed in the following scenario studied (*capacity in terms of active stations*), where an increased node population was considered. Here, the maximum sustainable throughput was also 61% of the *cc*, with bounded access delays under 40 ms. Results for the *splitting tree algorithm* revealed that this algorithm could achieve a higher system performance than the *exponential backoff algorithm*, achieving a maximum system throughput of up to $\approx 65\%$ of the *cc*.

In addition, for these two analyses (*offered load scalability* and *capacity in terms of active stations*) the maximum channel utilisation achieved by the protocol was $\approx 90\%$ of the channel capacity. The difference between maximum system throughput and utilisation is attributed to the extensive protocol overhead involved, generated by the

ATM protocol and the DVB PHY layer, reservation request transmissions, collisions and retransmissions. The remaining bandwidth of $\approx 10\%$ was accounted for unused contention slots, due to the random nature of CRAs.

The third analysis studied the delay and system throughput characteristics when transmitting a determined number of cells in the '*contention-based access region*', using the *exponential backoff algorithm* and a mixed traffic configuration of 32 kbps IP traffic and 12 kbps VoIP streams per station. In this analysis, it was found that optimum system performance is achieved when stations are allowed to transmit messages of 1-ATM cell in the contention access region. Here low delays under 25 ms were seen with an offered load up to 35% of the *cc*. Higher offered loads resulted in unbounded delays. In general, it was seen that when larger messages (e.g. 2-6 ATM cells) are transmitted using contention access, the risk of collision is increased slightly, resulting in a reduction in system throughput. This loss in throughput can be of 2% when messages up to 6-ATM cells are transmitted using contention access or up to 5% when all messages are transmitted using only reservation access.

Results for effects of *maximum request size* suggested that this parameter should be set as large as possible if the upstream channel is only used for Internet traffic. For instance, a value of 22 or 32 ATM cells was found to provide optimum system performance for this traffic configuration. However, if the upstream channel supports the transmission of both traffic types (IP and VoIP), a higher interaction for VoIP streams can be obtained if the *maximum request size* is set as short as possible (e.g. 6 ATM cells). Furthermore, when the performance of both CRAs was compared, it was seen that the *splitting tree algorithm* could support up to 10 stations more than the number supported with the *exponential backoff algorithm*. This is equivalent to a performance increase of up to 17% of the channel capacity.

Results for the analysis of *buffer capacity* revealed that by using small buffer sizes, for example of 50 or 100 ATM cells, only a small fraction, below $\approx 1\%$ of the channel capacity, is held in the station's buffers on high traffic loads (above 53% of the *cc*), compared to over 10% of the *cc* when a large buffer capacity is used (e.g. 1000 or 3000 ATM cells). Here, it was found that a small buffer capacity resulted in a better

performance for the support of VoIP traffic, since lower access delays are obtained with a short buffer capacity than with a large buffer size. However, a drawback is that the smaller the buffer capacity the higher the number of discarded packets, which may result in a degradation of service quality. For this analysis the number of discarded packets resulted in ≈ 3 packets per second when a buffer capacity of 50 ATM cells was considered. This number was of ≈ 1 and 0 for buffer capacities of 1000 and 3000 ATM cells, respectively.

In general, from this analysis of *buffer capacity*, it was found that there is a natural trade-off between giving sessions free access to the network and keeping delay at a level low enough so that interactive applications (e.g. VoIP, audio and video) are supported and retransmissions or other inefficiencies do not degrade the network performance.

From the analysis of the *effects of increasing the number of signalling frames* it was found that by transmitting two signalling frames in the 3 ms period, a slight decrease in access delay can be obtained. This reduction resulted in approximately 4 and 5 ms for the *exponential backoff algorithm* and the *splitting tree algorithm*, respectively. However, in terms of system throughput, for the *exponential backoff algorithm* there was a decrease in throughput of $\approx 6\%$ of the *cc*. This reduction was because the number of contention slots allocated in each signalling frame remained the same, reducing the bandwidth for data transmissions in every 3 ms cycle, in an attempt to resolve faster collisions.

The last analysis of this chapter addressed the *effects of varying the packet size in isochronous streams*. Here, it was shown that the major factors affecting the system performance were seen to be the length of the packet being transmitted for delivery. In the analysis of a *single node scenario*, it was demonstrated that regardless of the offered load, a station cannot achieve throughput higher than 32% of the maximum channel capacity in a 3.088 Mbps upstream channel. Even worse, this figure can be as low as 1.8% of the *cc* when delivering minimum Ethernet packets, which is attributed to the *scheduler-look ahead* delay.

In the *multiple node scenario* analysis, the DVB/DAVIC protocol proved again very inefficient due to the excessive number of reservation requests, collisions, retransmissions and DVB MAC and PHY protocol overheads.

Table 8.1 presents the maximum system performance and bandwidth characterisations for the six different packet sizes considered.

In this scenario the maximum system throughput achieved was $\approx 34.5\%$ of the *cc* when packet sizes of 64 bytes were considered. Increasing the packet size can significantly improve the performance with maximum system throughput up to $\approx 65\%$ of the *cc*.

Results for channel utilisation were higher than for throughput results with a deviation of 54% to 24% for minimum and maximum size Ethernet packets, respectively. The major source of performance inefficiency was the extensive protocol overhead involved at the MAC (ATM encapsulation) and PHY (FEC) layers, which can be as high as 35% and 23% of the upstream channel capacity for 64-byte and 1518-byte packets, respectively. A second significant factor degrading system performance is the bandwidth consumed for the transmission of successful requests, which accounted for $\approx 12\%$ when minimum size Ethernet packets are transmitted. Furthermore, collisions also contribute to system inefficiency with up to 8% of the *cc*.

Chapter 6 introduced three novel contention slots allocators (*Simple-CSA*, *Variable-CSA* and *Fixed-CSA*). These techniques were studied because the DVB/DAVIC protocol specification did not define any mechanism for the allocation of contention

Table 8.1 – Summary of maximum system performance and bandwidth characterisation.

Bandwidth Characterisation (%)	Packet Size (bytes)					
	64	128	256	512	1024	1518
Maximum Throughput	34.5	51.3	56.2	62.9	63.4	64.6
Maximum Utilisation	89.1	90.1	90.5	89.8	88.9	88.5
Deviation	54.6	38.8	34.3	26.9	25.5	23.9
Supported Streams	25.0	48.0	54.0	60.0	60.0	63.0
Request Collided	8.3	4.1	1.3	0.4	0.1	0.1
Request Successful	11.8	8.5	4.6	2.6	1.3	0.9
MAC&PHY overhead	34.5	26.2	28.4	23.9	24.1	23.0
Bandwidth unused	10.9	9.9	9.5	10.2	11.1	11.5

slots into the signalling frames. Here it is most likely that unscheduled slots are assigned to contention access).

In an analysis for the IEEE 802 protocol (reported in [94] and [96]) it was pointed out that the performance of multi-access reservation protocols heavily depends on the overall structure and the capacity assigned to the reservation and contention access regions. Therefore, in this chapter we have demonstrated that the performance of the DVB/DAVIC can be improved if these techniques are used.

The first strategy studied was the *Simple-CSA*. Here the *minimum number of contention slots per signalling frame* was ranged from 1 to 7 using the *exponential backoff algorithm* and from 0 to 6 using the *splitting tree algorithm*. When the *exponential backoff algorithm* was used it was found that by allocating at least 3 or 4 CSs per signalling frame, not only lower access delays were yielded but also an increase in system throughput was achieved. Larger values (e.g. 5, 6 and 7 CSs) caused a waste of bandwidth because more CSs are allocated than the number of CSs required to resolve collisions. Similarly, a low system performance is also obtained when short values (e.g. 1 and 2 CSs) are considered. Here the system inefficiency is attributed to an increased number of collisions, due to a reduction of slots for contention access. Results for the *splitting tree algorithm* revealed that optimal system performance is achieved by allocating at least 1 CS per signalling frame. The increase in performance is achieved because an extra slot is reserved in the next signalling frame when a collision happens. This extra slot (is split into three minislots that carry shortened reservation requests) and is used only by the stations that caused the collision, so that new arrivals do not compete with *backlogged* stations for contention access. The increase in system throughput is achieved because by allocating at least 1 CSs in each signalling frame, more slots can be scheduled for data transmissions.

When the performance of the *Simple-CSA*, *Forced-CSA* and *Variable-CSA* was compared, it was found that the *Forced-CSA* outperformed the other two mechanisms. Table 8.2 presents a summary of the maximum system performance achieved by each CSA for three different traffic configurations. The *Forced-CSA* not only provided the lowest access delays, but also (in most of the scenarios) the highest system throughput for the *exponential backoff algorithm*, as appreciated in Table 8.2. The increase in performance was because this mechanism allocates more contention slots when they are needed to resolve collisions. By allocating two additional contentions slots in the following signalling frame, after a collision has been detected, the probability of a new collision among *backlogged* stations and new arrivals is decreased. This in turn, results in a reduced *Contention-Resolution-Grant Cycle* that cannot be obtained with the other strategies.

This chapter also presented a performance optimisation and a comparison between the *exponential backoff algorithm* and the *splitting tree algorithm*. The first analysis studied

Table 8.2- Summary of maximum system performance for different CSAs.

Traffic Type	CSA	Min. CSs per sign. frame	Maximum Throughput (%)	Mean Access Delay (ms)	CSs per Request	Active Stations
Internet 32 kbps	Simple	3	46.8	3099.98	3.96344	51
	Simple	5	48.0	2004.82	3.57891	51
	Forced-FSs 1	3	51.1	1284.42	2.87358	51
	Forced-FSs 2	2	51.5	584.026	2.81989	51
	Forced-FSs 2	3	51.0	1090.09	2.91893	51
	Variable	2	47.9	2628.13	3.68146	51
	Variable	3	47.8	2195.96	3.6845	51
Internet 64 kbps	Simple	4	50.6	1226.38	3.07588	25
	Forced-FSs 1	3	50.7	913.49	3.01059	25
	Forced-FSs 2	2	50.6	494.56	3.01297	25
	Forced-FSs 2	3	49.8	416.05	3.15268	25
	Variable	2	50.8	777.72	2.95227	25
	Variable	3	51.3	624.79	2.87162	25
Mixed 41.7 kbps	Simple	3	43.9	925.94	3.18	34
	Simple	5	45.23	178.62	2.88	34
	Forced-FSs 2	2	45.6	417.15	2.84	34
	Forced-FSs 2	3	45.6	162.18	2.82	34
	Variable	2	44.2	1061.55	3.09	34
	Variable	3	44.6	309.13	3.01	34

The shaded rows present the CSA with optimum system performance.

focused on the optimisation of the *initial* and *truncated* backoff bounds (or *backoff window* $Bw[i-t]$) for the *exponential backoff algorithm*. Results for mixed traffic at 41.7 kbps per station revealed that an increase in system performance is achieved with $Bw[4-6]$. With this *backoff window*, up to 75% of all data packets are transmitted in less than 100 ms.

A good system performance can also be obtained with $Bw[3-4]$, $Bw[2-4]$, $Bw[3-6]$, and $Bw[3-5]$. From this analysis it was seen that defining short values for the *initial* backoff exponent, such as $Bw[2-3]$, results in a poor system performance. This is because *backlogged* stations are forced to transmit in the next $2^i = 4$ contention slots, which increases considerably the risk of collision with new incoming packets. A similar performance degradation is obtained with a large *backoff window* (e.g. $Bw[5-7]$). The consequence of this is because *backlogged* stations are now forced to wait a relatively long period of time before they can compete again for contention access. With $Bw[2-3]$ and $Bw[5-7]$ only up to $\approx 33\%$ of all packets generated are transmitted in less than 100 ms.

We also found the *backoff window* that provided optimal performance when only VoIP traffic (9.7 kbps) is transmitted. They were $Bw[5-7]$ and $Bw[4-8]$. If we compare the range of the optimum *backoff window* for mixed traffic and VoIP streams, we can see that larger *backoff windows* are needed when only VoIP traffic is transmitted. This is to be expected since the data rate of VoIP stream is lower than the data rate for mixed traffic, thus the number of streams (or stations) can be increased. This in turn increases the probability of collisions, and indicates that larger *backoff windows* should be considered to reduce this probability. An important point here is that the maximum system throughput is reduced from $\approx 45\%$ when mixed traffic is transmitted to $\approx 32\%$ when only VoIP is delivered. This reduction is attributed to the increased number of streams supported, which requires more bandwidth to resolve collisions, and the higher protocol overhead involved when delivering VoIP traffic.

Similarly, the second performance optimisation focused on the *Entry spreading* (Es) factor of the *splitting tree algorithm*. In this analysis it was found that there is not much difference from selecting different Es values. Simulations results suggested that for

optimum system performance the E_s factor should be set to 6 for mixed traffic (or medium size networks) and 6 or 7 for VoIP traffic (or large networks). In terms of system throughput, the difference of using distinct E_s values was much less significant. It was found that the maximum system throughput ranged between $\approx 54.5\%$ to 55% for mixed traffic and from 44.8% to 45.2% of the cc for VoIP traffic.

The last analysis presented in Chapter 6 approached a performance comparison between both CSAs. In this analysis it was concluded that the *splitting tree algorithm* outperforms the *exponential backoff algorithm*. This is because the former uses feedback and allocation information that allows a station, with new incoming arrivals, to compete for contention slots without the risk of collision with *backlogged* stations. The most important advantage is that the use of minislots for reservation requests further decreases the risk of collisions, since one contention-based slot is divided into three minislots, increasing the probability of successful request transmissions. The major drawbacks of this algorithm are higher complexity at the headend, increased processing times of the feedback and allocation information at the station, and since every contention slot should be acknowledged regardless of whether it is used or not, higher control information at the downstream channel is assigned. In general, simulation results showed that an increase over 9% on system performance, could be achieved with the *splitting tree algorithm* when the backoff values (*initial/truncated*) and the *Entry spreading* factor have been optimised, for different traffic configurations (such as: Internet, VoIP and mixed traffic).

In *Chapter 7* several novel improvements were implemented. They enable the DVB/DAVIC MAC protocol to provide the delay requirements and an increased efficiency for the support of timing critical interactive services and high-speed data transmissions. The first analysis focused on the performance increase achieved by the use of six enhanced-reservation-request mechanisms (*Reserved Request*, *Continuous Reserved Request*, *Enhanced-Pure Reservation Access*, *Piggyback Request*, *Continuous Piggyback Request* and *Unsolicited Grant Slot*).

The introduction of these techniques was because the DVB/DAVIC protocol uses a limited reservation access mechanism (*PRA*) by default for the transmission of

reservation requests that was not optimised for the delivery of isochronous streams (timing critical interactive services), and this is evident from the increased risk of collision with reservation requests among stations and the poor system performance as seen in previous analysis. In general, it was demonstrated that by adopting these enhanced mechanisms not only lower access delays are possible but also the system throughput is considerably improved. The highest system performance is produced with the *UGS* mechanism. Results for VoIP traffic revealed that the number of streams supported could be greatly increased from 105 to approximately 170 connections, achieving also relatively low access delays in the order of 4-7 ms. This appeared to be the highest performance increase for this research, using only the contention and the reservation access modes of the DVB/DAVIC protocol. The *CPG* mechanisms also offered a good system performance. With this technique, the number of supported VoIP streams can be increased up to ≈ 125 .

From this analysis, it was highlighted that the use of minislots with or without piggybacking requests, and the *enhanced pure reservation-access* mechanism are best suited for Internet traffic or bursty traffic.

In general, the six enhanced mechanisms achieve an increased performance over the default *PRA*, because as they become more complex, they efficiently reduce and in some cases avoid the collision risk of reservation request transmissions and therefore the *CRGC* is minimised. The increase in system throughput is achieved because as collisions are avoided, these mechanisms dynamically allocate more bandwidth for data transmissions. An important point here is that the performance increase provided by these mechanisms can only be obtained when the traffic characteristics are carefully analysed. The choice of a poor mechanism may reduce the service quality considerably.

The performance of the piggyback mechanisms and the *CRR* mechanisms is also reported in [101], where a comprehensive analysis in terms of the cumulative probability of packet delay was presented. Due to their enhanced performance, these three mechanisms have already been integrated into the latest version of the DVB/DAVIC standard 'ETSI EN 200 800' [35].

A second approach for increased performance was through the use of Quality of Service. We have found that the main drawback of using the reservation access mode for the provision of delay sensitive applications is that bandwidth cannot be guaranteed. This is because all traffic transmitted using this mode is treated with a ‘best-effort’ service and in some situations the delivery of *TCIS* requires an especial treatment for an improved service.

Therefore, in order to provide a guaranteed service for *TCIS* the use of QoS is fundamental. Two types of QoS were studied. They are known as prioritisation and reservation.

Although traffic prioritisation is not part of the set of functionalities of the DVB/DAVIC protocol specification, we have demonstrated that a faster transmission for the delivery of *TCIS* streams can be achieved by mapping the *ToS* field of the IP protocol, with at least two 2-levels of priority at the DVB/DAVIC MAC layer. This will give a faster delay treatment to *TCIS* stream. Results from a mixed traffic configuration (at 41.7 kbps) revealed that after the saturation point, the prioritised mechanism still offers tolerable delays (about 50 ms) for *TCIS* streams. Here, the number of VoIP streams was increased from ≈ 40 with the default *PRA* up to ≈ 50 connections with the prioritised mechanism. The major drawback of this technique is that it does not provide any guarantee of bandwidth availability or latency with high traffic loads.

On the other hand, the reservation technique uses the fixed access mode of the DVB/DAVIC protocol to provide a guaranteed service. Here the number of slots needed by a station and the periodic intervals is negotiated during the connection setup. With this technique, we have seen that reduced packet access delays for the delivery of isochronous streams, in the range of 1 and 2 ms could be guaranteed by the DVB/DAVIC protocol, before large periods of congestion are experienced. It was also found that for the same traffic configuration (at 41.7 kbps), on high traffic loads, the service of IP traffic gets starved at the headend and the number of VoIP streams could be significantly increased up to 320. In order to avoid the starving service for IP traffic, the scheduler algorithm at the headend could be slightly modified to control the

bandwidth to be assigned to each access mode (e.g. 40% of the cc for the fixed-rate access and 60% for the reservation and the contention access mode).

The best performance of the reservation technique was achieved when only *TCSI* streams were transmitted. Here we studied the performance of two codecs for VoIP traffic (G.711 and G.723.1). Codec G.711 was considered to stress the CATV network and mainly because EuroModems (class B) are more likely to use it for high quality voice calls. In addition, header suppression, which is an advanced functionality in the latest version of this protocol [35], was also considered in this analysis. In general, it was demonstrated that the use of header suppression, combined with a larger voice frame (30 ms for G.711 and 120 ms for G.723.1), bandwidth efficiency is considerably increased to a large extent, achieving a much higher figure regarding the maximum number of streams sustainable. In addition, in this research a minimum packet access delay about 1ms (for VoIP streams) was possible. However, with advanced synchronisation techniques this delay can be further reduced (under 1ms), for the delivery of *TCIS* streams in general.

For example, results for codec G.711 revealed that the number of streams supported could be augmented up to 53 connections when header suppression is considered and three audio frames are encapsulated in each audio packet, compared to 27 connections achieved when only voice packets consist of one voice frame and header suppression is avoided. This corresponds to an increase of $\approx 96\%$. Similarly, for codec for G.723.1, this increase, in terms of streams supported, could be up to 400%. In order to achieve these remarkable figures, we have assumed that the RTP, UDP, IP and MAC headers can all be suppressed, which is reasonable if the INA maintains additional state information on all active voice connections in accordance to [47].

Finally, the last analysis provided in this dissertation addressed a comparison between the DVB/DAVIC and the DOCSIS protocol. We have evaluated some of the fundamental properties provided by each standard at the MAC and PHY layers. In general, results revealed that in most of the analysis, the DVB/DAVIC protocol achieved a reduced performance compared with the DOCSIS protocol. This was mainly because the DVB/DAVIC protocol encapsulates every datagram into ATM-AAL5 cells

and suffers an overhead penalty for Segmentation and Reassembly (SAR) in an attempt to provide a faster transmission for QoS over the ATM protocol, while the DOCSIS uses a scheme that favours the delivery of variable length Internet protocol packets rather than ATM transfer.

8.3 Application of results to the Industry

The results presented in this dissertation can be used in implementation for the European cable modem protocol (DVB/DAVIC) by cable network operators and manufactures. For example, results for mean access delay, throughput and utilisation (discussed in Chapter 5) can act as a guideline as to how subscribers should be allocated in the upstream in order to achieve a desired performance and service level balance.

In addition, by using the given performance levels for different loading scenarios, an appropriate charging scheme can be selected. Analysis on capital investment may be based on results presented herein by deriving how the network could be segmented in order to provide services with specific performance requirements. In general, recommendations for operators and service providers on how to achieve optimum system performance are as follows.

The effects of the '*Maximum Contention Access Message Length*' (Section 5.4.3) revealed that an increase in the overall system performance could be up to 3% by setting the this parameter to 1. This parameter can be modified by the operator using the device manager at the headend. The '*Maximum Reservation Access Message Length*' can also be modified by the service operator. A recommendation is to set this parameter to 32 slots if the subscriber service is only for Internet traffic. If the service is also for voice over IP, (with a best effort service) this parameter should be reduced to about 6 slots, but there is a drawback, reducing this parameter increases the risk of collisions of reservation request and this results in a loss in system performance of about 6% in an attempt to provide short delays for the delivery of VoIP streams (as indicated in Figure 5.12), on high periods of congestion.

The optimisation of the number of contention slots carried out in Sections 6.2.2.1 and 6.2.2.2 for the *exponential backoff algorithm* and the *splitting tree algorithm*, respectively, showed that a simple contention slots allocator can increase performance by up to $\approx 10\%$ (Figure 6.2 and Figure 6.5). This mechanism can be directly implemented by manufacturers by slightly modifying the DVB/DAVIC MAC protocol.

The only parameter that is needed for the operation of the Simple-CSA is the ‘*Min. No. of CSs per signalling frame*’, which was not defined by the DVB/DAVIC standard. This parameter can be added to the set of operational parameters of the INA and then modified via the device manager, either by the manufacturer or by the operator. Results presented in Section 6.2.2.1 can be used as a guidance of how to set the correct number of contention slots (for optimum performance).

Similarly, the Forced-CSA can also be implemented by the manufacturer by simple adding the parameter *forced contention slots* (FSc), which represents the number of slots added to the signalling frame after a collision occurred. This scheme was the best contention slot allocator, because it dynamically adjusts the number of contention slots needed to resolve collisions according to the traffic load, and achieves an increase in the overall system performance by up to 3% more than the figure reported by the Simple-CSA. We recommend setting this parameter to 2 for optimum system performance.

The optimisation of the two contention resolution algorithms used by the DVB/DAVIC protocol (approached in Section 6.3) does not require the intervention of the manufacturer. For the *exponential backoff algorithm* we recommend to the operator to use the backoff windows of Bw[4-6] and Bw[4-8], which performs well for Internet and VoIP traffic. For the *splitting tree algorithm* the *Entry spreading* factor should be set to 6 for best system performance.

From results presented in Section 6.3.3, we recommend that cable network operators buy EuroModems that support the *splitting tree algorithm*. In this Section 6.3.3 it was shown that this algorithm always performs better than the exponential backoff algorithm, regardless of the traffic load and number of active stations. Although EuroModems with this functionality tend to be more expensive than EuroModems supporting only the *exponential backoff algorithm*, this extra cost is compensated for by

the increase in system performance, which can be of up to $\approx 12\%$ of the maximum channel capacity.

There are however modifications proposed herein, namely modifications at the INA scheduler and the NIU for the support of enhanced reservation request mechanisms. The *piggyback mechanism* and the *continuous piggyback* have already been integrated in the third version of the DVB/DAVIC protocol [35] due to their remarkably performance over the default reservation mechanism (defined in the second version). We strongly recommend to cable network operators to buy EuroModems supporting these piggyback mechanisms, here it was proved that the increase in system throughput could be up to 25% of the maximum channel capacity for upstream channels supporting Internet and voice traffic (see Section 7.2.7, Figure 7.8).

The *unsolicited grant slots (UGS)* mechanism, however can achieve a better system performance than the piggyback mechanisms in terms of access delays for the support of CBR traffic, here access delays below 7 ms can be obtained on high traffic loads. The implementation of this mechanism requires slight modifications to the INA scheduler and NIU MAC protocol. Here the '*INA_Capabilities*' field of the *Default Configuration Message* and the '*NIU_Capabilities*' field of the *Sign-on Response Message*, contain a reserved field (of 5 bits for the) for future implementations, which can be used for the support of the UGS mechanism. This functionality requires only one bit, leaving the other four bits for future implementation. This reserved field is of one bit in the last version of the DVB/DAVIC standard [35], which can also be used for this functionality.

With the *UGS* mechanism, as stated in Section 7.2.6, the number of slots needed by a station and the periodic intervals can be negotiated during the connection set-up through the use of the '*Resource Request Message*'. This structure is the same as the one used for the establishment of fixed-rate connections, with the exception that instead of assigning slots to a NIU in the fixed-rate region of the signalling frame, these slots are allocated in the reserved access region of the signalling frame, using the '*Reservation Grant Message*'.

8.4 Future work

In this dissertation we have provided a rigorous performance analysis and introduced novel techniques that increase the network efficiency of the upstream channel. However, there are many interesting issues requiring further research.

The latest version of the DVB/DAVIC protocol specification (v.3) [35] has been recently released. New features such as header suppression, piggyback mechanisms, connection priority and the performance over a 12 Mbps upstream channel need to be researched.

In this dissertation we have seen that the use of header suppression provides the highest network efficiency. In the latest version, a different approach is used for the suppression of RTP sessions, which consists of the combination of RTP/UDP/IP headers. Instead of maintaining additional state information on all active connections at the headend, so that these headers and also the MAC header can be all suppressed (as suggested in [47]), only the fixed fields of the RTP/UDP/IP headers are suppressed in [35], as illustrated in Figure 8.1. This means that only up to 27 bytes (out of 12-bytes RTP + 8-bytes UDP + 20-bytes IP = 40 bytes) can be suppressed. An interesting issue here would be to analyse the change in system performance when this strategy is used for the delivery of *TCIS*.

The *continuous piggyback request* and the *piggyback request* mechanisms have already been included as part of the functionality of the DVB/DAVIC MAC protocol.

The performance comparison of the *enhanced-reservation request mechanisms* provided in this dissertation considered fixed piggyback values (e.g. 0, 4, 8 and 12 for *PG* and 1, 4, 8 and 12 for *CPG*). According to the new protocol specification these values (with the exception of the first one, which is always fixed) could be up to 255. An additional analysis would be to provide a performance optimisation of the piggyback values for these two strategies using different traffic configurations (e.g. Internet traffic, bursty traffic, VoIP, Video conferencing, multimedia traffic, etc.).

The analysis of connection priorities would allow the DVB/DAVIC protocol to prioritise traffic transmissions. A connection with low priority can be reprovisioned in order to accommodate the requirements of connections with high priority. The new protocol specification defines up to 255 levels of priorities as indicated in Table 8.3. This functionality would require an improved scheduler algorithm at the INA, which gives preference to connections with higher priority.

Table 8.3 – Priorities for connections.

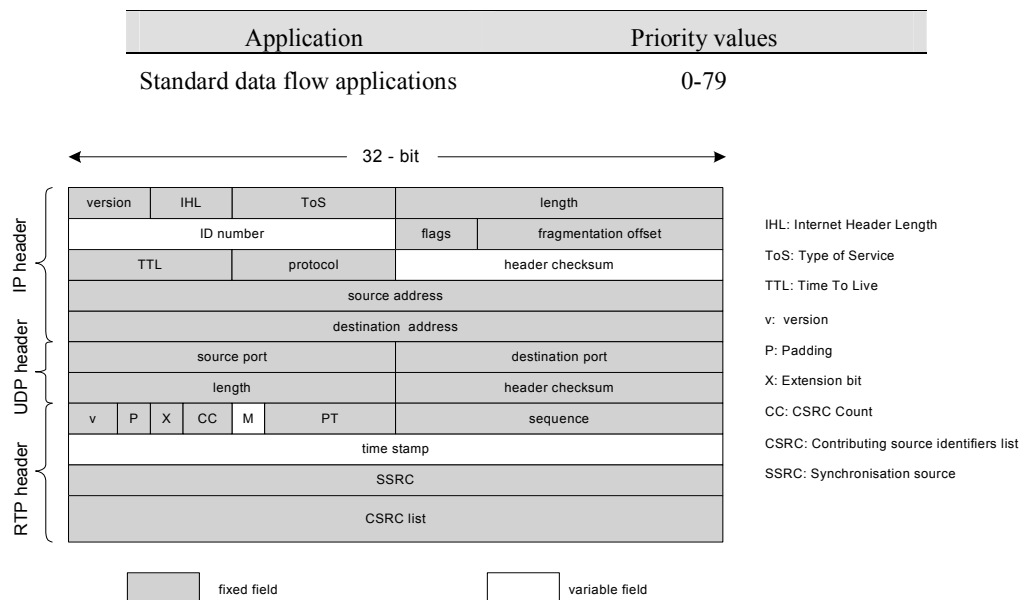


Figure 8.1 – Fixed fields for RTP/UDP/IP headers.

Applications with QoS	80-200
High priority applications	201-255

In this research, most of the analysis focused on 3 Mbps and 6 Mbps upstream channels. An interesting issue is to analyse the performance when applications that demand higher bandwidth requirements are delivered using a 12 Mbps channel.

Another important topic that needs to be addressed as future work is quality of service in the downstream direction. The mapping of up to eight upstream channels onto a single downstream channel raises the issue of congestion in the downstream too. For the transmission of traffic that does not originate in the CATV network, for example traffic coming from either the Internet, PSTN or other CATV network via the router/ATM switch attached to the INA, an efficient scheduler will be required in order to prioritise and police traffic as well as maintain maximum link utilisation in the downstream channel.

Finally, the work presented in this dissertation is highly relevant to emerging fixed, mobile and satellite architectures [20]. Fixed wireless architectures have the same access topology as CATV networks. Figure 8.2 presents three different configurations of CATV networks with satellite, terrestrial or HFC architectures. The spectrum is divided into different unidirectional channels for the upstream and downstream direction. The major difference is the physical interface. In wireless architectures, the signals propagate through the air while in CATV networks either coax or optic fibre links or a combination of both is used for transmissions. The physical interface would be significantly different but the MAC would be identical.

CATV networks have been considered as the ideal backbone network architecture for Personal Communications Service (PCS), as illustrated in Figure 8.2c, [50]. The broadcast nature of CATV networks significantly simplifies the hand-off mechanism of mobile stations thus reducing cost and complexity at both the base station and the telecom switch.

The major drawback of using CATV networks for satellite communications, as seen in Figure 8.2a, is the long propagation link delays, which can result in serious performance implications.

This is because very large propagation delays (in the order of hundreds of milliseconds) will result in equally large *scheduler look-ahead* delays. The propagation delay to the headend and back in such systems will be twice the roundtrip delay to the satellite, which in the case of Geo-stationary Earth Orbit (GEO) satellites placed at about 22.3 miles above the Earth would be of ≈ 500 ms [9]. Therefore the explicit *CRGC* cycle will result in large intervals of network idle time due to the number of contention slots the scheduler will have to subscribe.

For delay sensitive applications, such as voice conversations, the propagation delay over the two-way path between conversants influences the perceived quality of service. This problem can be ameliorated with LEO satellites placed at $\approx 500 - 1000$ miles above the Earth. This relatively short distance reduces transmissions delays significantly to ≈ 50 ms [9]. The drawback is that a large number of satellites, arrayed on a constellation of multiple satellites in a polar orbit is required to maintain path continuity.

The long propagation delays of satellite systems undoubtedly will reduce the system efficiency of a CATV MAC protocol. An interesting issue requiring further research is to analyse to what extent long propagation delay can degrade the quality of service of

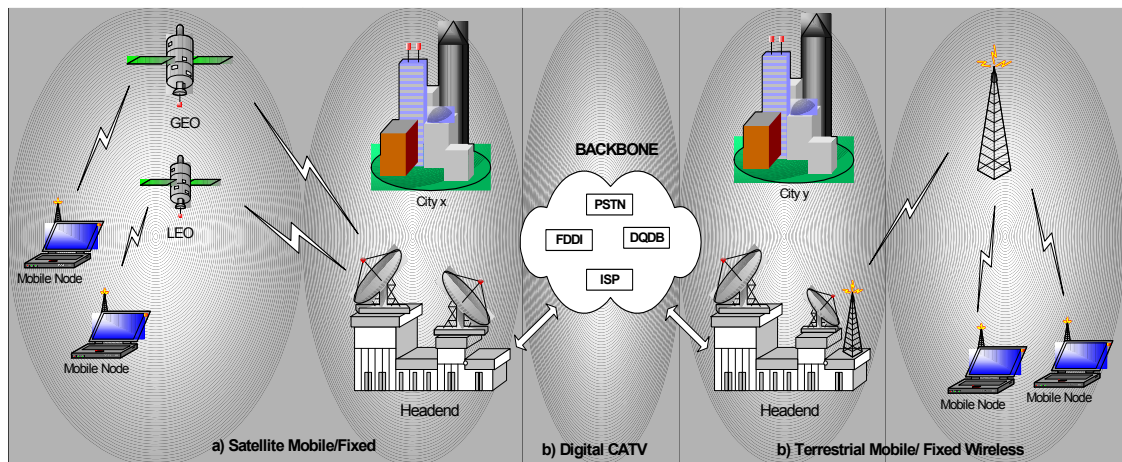


Figure 8.2 – Similarities of CATV with fixed/mobile terrestrial and satellite wireless architecture.

TCIS streams over the DVB/DAVIC protocol. The direct consequence of having long propagation delays is that the contention resolution algorithm interval will increase simply because each station will have to assume longer time-out periods for the reception of a grant. In the case where a collision does not occur this will result in the CRA to be initiated with a delay equal to two times the round trip. If the traffic pattern of a station is of such a nature (e.g. sporadic packet generation), this results in the use of contention access for the transmission of reservation requests, and performance can be degraded significantly.

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APPENDIX A: GLOSSARY

AAL	ATM Adaptation Layer
ABR	Available Bit Rate
ADSL	Asymmetrical Digital Subscriber Loop
ATM	Asynchronous Transfer Mode
BC	Broadcast Channel
BE	Best Effort service
BIM	Broadcast Interface Module
BNA	Broadcast Network Adaptor
BONeS	Block Oriented Networks Simulator
CAP	Carrierless Amplitude/Phase modulation
CATV	Community Antenna Television
CC or <i>cc</i>	Channel Capacity
CBR	Constant Bit Rate
CIR	Committed Information Rate
CM	Cable Modem
CMTS	Cable Modem Termination System
CPE	Customer Premise Equipment
CPG	Continuous Piggyback Request
CPR	Centralised Priority Reservation
CRGC	Contention-Resolution-Grant Cycle
CRR	Continuous Reservation Request
CSA	Contention Slot Allocator
CSRC	Contributing source identifier list
CSF	Common Simulation Framework
DAVIC	Digital Audio Video Council
DDIC	DVB/DAVIC Interoperability Consortium
DES	Data Encryption Standard
DHCP	Dynamic Host Configuration Protocol
DMT	Discrete Multi-Tone
DOCSIS	Data Over Cable System Interface Specification
DSL	Digital Subscriber Loop

DVB	Digital Video Broadcast
ECCA	European Cable Communications Association
ECL	EuroCableLabs
EM	Euro Modem
EN	European Norm
EPRA	Enhanced Pure Reservation-Access
ES	European Standard
ETS	European Telecommunication Standard
ETSI	European Telecommunication Standard Institute
FA-TDMA	Frequency Agile – Time Division Multiple Access Division
FEC	Forward Error Correction
FIFO	First In First Out
FSM	Finite State Machine
FTP	File Transfer Protocol
FTTC	Fibre To The Curb
FTTH	Fibre To The Home
GB	Guard Band
HDSL	High data rate Digital Subscriber Line
HE	Headend
HFC	Hybrid Fibre Coaxial
HS	Header Suppression
IB	In Band
IC	Interaction Channels
IEEE	Institute of Electrical and Electronic Engineers
IETF	Internet Engineering Task Force
IHL	Internet Header Length
IIM	Interactive Interface Module
INA	Interactive Network Adaptor
IP	Internet Protocol
ISDN	Integrated Systems Digital Network
ISP	Internet Service Provider
ITU	International Telecommunications Union

kbps	Kilobits per second
LAN	Local Area Network
LLC	Logical Link Control
MAC	Media Access Control
MAN	Metropolitan Area Network
MAP	MAC Management Access
MCI	MAC Control Information
MCNS	Multimedia Cable Network Systems
MPEG	Motion Pictures Experts Group
MPEG-TS	Motion Pictures Experts Group Transport Stream
MSOs	Multiple Service Operators
MVA	Mean Value Analysis
NAT	Network Address Translator
NIC	Network Interface Card
NIU	Network Interface Unit
NTT	Nippon Telegraph and Telephone
NVOD	Near Video On Demand
ONU	Optical Network Unit
OOB	Out Off Band
OPNET	Optimised Network Engineering Tool
OSI	Open System Interconnection
PCI	Peripheral Component Interconnect
PCS	Personal Communications Service
PCM	Pulse Code Modulation
PDF	Probability Density Function
PDU	Protocol Data Unit
PG	Piggyback Request
PHY	Physical Interface
PON	Passive Optical Network
POTS	Plain Old Telephone System
PPP	Point to Point Protocol
PRA	Pure Reservation Access
QAM	Quadrature Amplitude Modulation

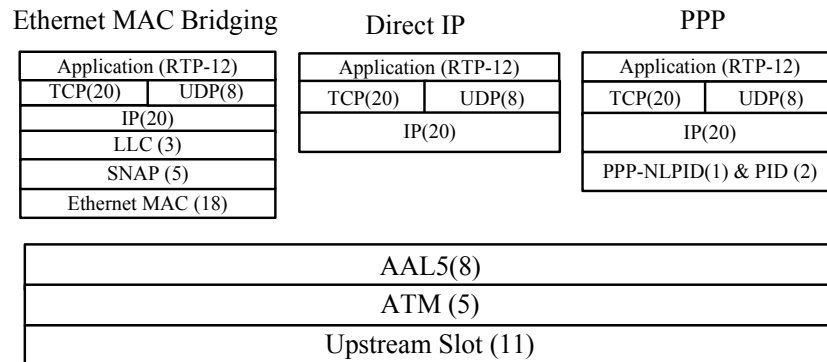
QoS	Quality of Service
QPSK	Quaternary Phase Shift Keying
RF	Radio Frequency
RFC	Request For Comments
RR	Reservation Request
RS	Reed Solomon
RSA	Rivest-Shamir-Adleman
RTP	Real Time Protocol
rtPS	real-time Polling Services
SA	Scheduling Advance
SAR	Segmentation and ReAssembly sublayer
S-CDMA	Synchronous Code Division Multiple Access
SCF	Slot Configuration Field
SCFQ	Self Clock Fair Queuing
SCTE	Society of Cable Telecommunications Engineers
SDSL	Single Line Digital Subscriber Line
SF	Start Field
SL-ESP	Signalling -Link Extended Super Frame
SNAP	SubNetwork Attachment Point
SNMP	Simple Network Management Protocol
SSRC	Synchronisation Source
STB	Set Top Box
SuperPON	Super Passive Optical Networks
TCIS	Timing Critical Interactive Service
TCP	Transport Control Protocol
Telcos	Telephone Companies
TDM	Time Division Multiplexing
TDMA	Time Division Multiple Access
ToS	Type of Service
TTL	Time To Live
TS	Transport Stream
UDP	User Datagram Protocol
UGS	Unsolicited Grant Slot

USB	Universal Serial Bus
UW	Unique Word
VDSL	Very high data rate Digital Subscriber Line
VoD	Video On Demand
VoIP	Voice over IP
VSF	Vestigial Side Band
WAN	Wide Area Network
WWW	World Wide Web

APPENDIX B: DVB/DAVIC PROTOCOL STACK AND PACKET FORMATS

B.1 Protocol stack

In accordance with [34], INA and NIU support three different solutions for the upstream channel: Ethernet MAC bridging, Direct IP and Point to Point (PPP) and as illustrated in Figure B.1. These three solutions allow compatible and interoperable implementations for transmitting IP datagrams, Ethernet MAC frames and PPP frames over ATM AAL5 [49] and DVB Multiprotocol Encapsulation [33]. The Direct IP solution is mandatory for both INA and NIU. However, the other two solutions are optional.



RTP: Real-time Transport Protocol	ATM: Asynchronous Transfer Mode
UDP: User Datagram Protocol	AAL5: ATM Adaptation Layer 5
TCP: Transfer Control Protocol	PID: Protocol Identifier
IP: Internet Protocol	NLPID: Network Layer PID
SNAP: SubNetwork Attachment Point	MAC: Media Access Control

Note: Numbers in parenthesis represent the protocol overhead per data unit

Figure B.1 – Protocol stack.

B.2 Packet formats

In the upstream channel, a 64-byte slot format is used for the transmission of user data or MAC messages, as shown in Figure B.2a. A Unique Word (UW) of 4-bytes provides a burst mode acquisition method for synchronisation purposes. The payload area is of 53-bytes and contains a single ATM cell for user data or MAC control transmissions. Then, a Reed-Solomon (RS) parity field of 6-bytes provides 3-bytes of RS protection over the payload area. The Guard Band of 1-byte provides spacing between adjacent packets.

In the case where minislots are used, one contention-based upstream slot is divided into three independent minislots of 21-bytes long as illustrated in Figure B.2b. Each minislot consists of 4-byte UW, a single byte Start Field (SF), a 16-byte payload and a single byte Guard Band (GB), and can only be used to carry only a shortened MAC reservation request message when the *splitting tree algorithm* (introduced in Section 3.4.3) is selected.

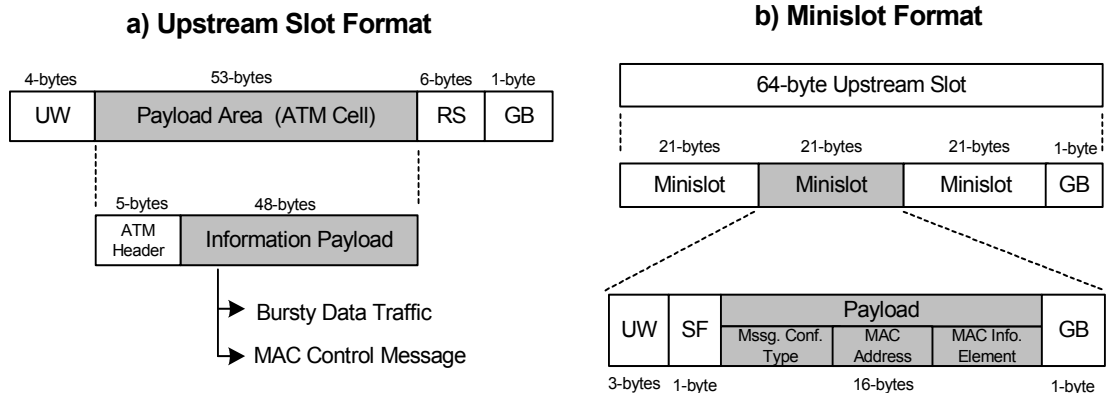


Figure B.2 – Upstream slot and minislot frame structure of the DVB/DAVIC protocol.

The Protocol Data Unit (PDU) structures for the upstream direction are depicted in Figure B.3. In this channel, ATM Adaptation Layer (AAL5) is used to encapsulate MAC PDUs (Ethernet MAC Frames, IP Datagrams or point-to-point -PPP packets) in ATM cells. At the physical layer, these ATM cells are further encapsulated using the (64-byte) slot format (presented in Figure B.2a).

In the downstream direction two signalling methods are used: in-band (IB) and out-of-band (OOB).

- IB:** In the IB signalling method the downstream channel is embedded in the broadcast channel and is oriented for the EuroModem solution. This method is used to transmit high data rates. Up to 52 Mbps can be transmitted in one 8-MHz downstream channel with a 256-Quadrature Amplitude Modulation (QAM). The transmission of data packets (MAC PDUs) is based on the DVB specification for data broadcasting- ETSI EN 301 192 [33], in which Motion Pictures Experts Group (MPEG-2) Transport Stream (TS) frames are used to encapsulate the MAC PDUs.

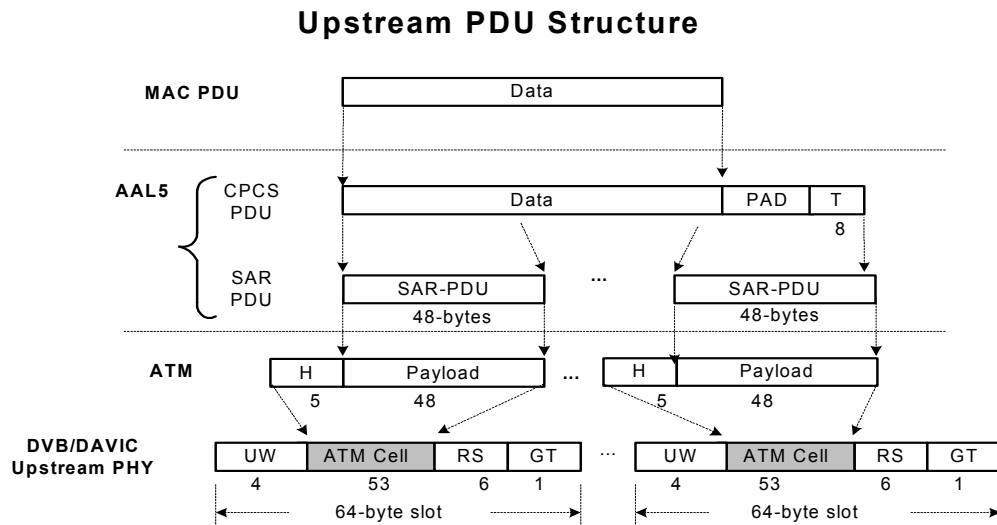


Figure B.3 – Upstream PDU Structure of the DVB/DAVIC protocol.

The transmission of MAC control messages (or simply MAC messages) is also based on MPEG-2 frames as illustrated in Figure B.4. Up to three MAC control messages can be encapsulated into one MPEG-2 TS frame. Most of the MAC messages are 40-bytes in length and the other 68-bytes of the MPEG-2 TS frame is comprised of a 4-byte MPEG header, a 60-byte MAC Control Information (MCI) field which contains the signalling parameter for the control of its associated upstream channels, and 4 bytes reserved for future implementations. No AAL5 layer is defined for MPEG-2 TS packets cells. At the physical layer, a 16-byte RS parity field is added to each MPEG-2 TS frames to form an RS-coded packet of 204 bytes in length.

- **OOB**: For downstream OOB, the maximum slot transmission rate is 3.088 Mbps with Quaternary Phase Shift Keying (QPSK) modulation and a 2-MHz channel bandwidth. A higher transmission data rate is not necessary since the OOB downstream channel is mainly used to transmit control messages. AAL5 adaptation is used to encapsulate data information and MAC control messages in ATM cells. This signalling method is mainly oriented for the Set Top Box solution.

OOB uses a Signalling Link Extended Super frame (SL-ESF) framing structure based on ATM cells. Ten ATM cells are mapped into 24 sub-frames with additional signalling (MCI) and error correction information. The description of the packet format structures for this method is beyond the scope of this research, since we focus our research for the EuroModem solution, with a downstream IB signalling mode. For a full description of the downstream (and also the upstream) packet formats the readers are referred to [34].

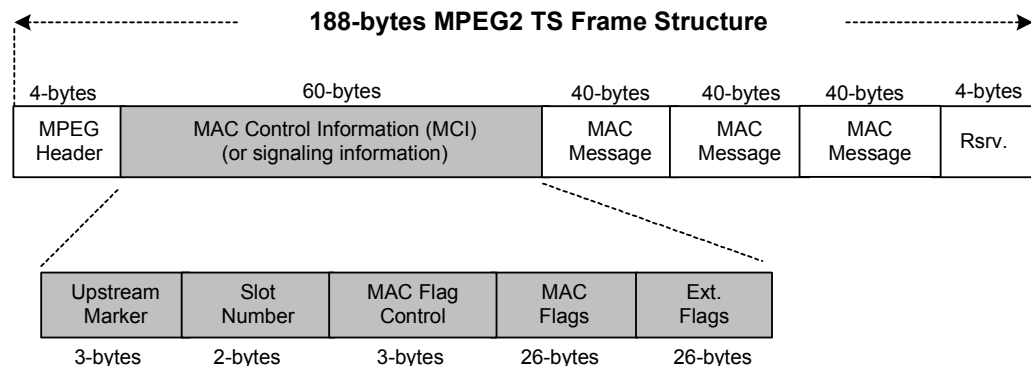


Figure B.4 – Downstream frame structure.

APPENDIX C: CATV PENETRATION PER COUNTRY

Table C. 1 – Cable TV penetration in Europe, North America and Asia. Main sources from ECCA annual meeting 2001 [7] and European Union [30].

Country	Population (millions)	Households (millions)	Homes passed by cable (millions)	(%)	Cable Subscribers (millions)	(%)
Austria	8	3	1.7	56.7	1.2	70.6
Belgium	10	3.9	3.9	100	3.85	98.7
Czech Rep.	10.3	3.6	1.955	54.3	0.765	39.1
Denmark	5.3	1.7	1.7	100	1.375	80.9
Estonia	1.5	0.576	0.246	42.7	0.167	67.9
E Finland	5.181	2.28	1.4	61.4	0.902	64.4
U France	58.5	23.8	8.4	35.3	3.041	36.2
R Germany	82	33	26	78.8	22	84.6
O Greece	10.5	3.65	0.1	2.7	0.011	11
P Hungary	10.1	3.8	2.45	64.5	1.7	69.4
E Ireland	3.7	1.2	0.600	50	0.470	78.3
Italy	57.3	20.4	0.963	4.7	0.073	7.6
Luxembourg	0.429	0.185	0.135	73	0.121	90
Netherlands	15.8	6.5	6.3	96.9	6.12	97.1
Norway	4.45	1.8	0.9	50	0.789	87.7
Poland	38.8	13.3	6.5	48.9	5	76.9
Portugal	9.96	4.16	2.18	52.4	0.925	42.4
Romania	22.5	8.2	6.5	79.3	2.8	43.1
Slovenia	2.35	0.67	0.45	67.2	0.243	54
Spain	40	11.8	2	16.9	0.25	12.5
Sweden	9	4.1	2.7	65.8	2.7	100
Switzerland	7.166	3.162	2.6	82.2	2.6	100
UK	59.7	25	13.14	52.6	3.43	26.1
USA	276	105.7	102.5	97	65	63
Other Canada	31	11.5	10.9	95	8.4	77
China	1265	333	112.5	33.8	90	80
Taiwan	23	5.67	-	-	4.5	80
Japan	129	47	46.6	99	9.5	20.3

Shaded rows represent the countries with the highest cable penetration in Europe.

APPENDIX D: VERIFICATION TEST FOR THE EXPONENTIAL BACKOFF ALGORITHM

Number of active stations: 30, Data rate: 32 Kbps, IP + 9.7 Kbps, VoIP, Data taken from simulation time at 59.0 to 60.0 seconds

MCI/Slot	Data Slot																		Arrival
	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	18	
1																			26
2	20	2		5	5	28	28	28	24	24	24	24	13	13	13	30	30	21	
3	29,33,3	3		30	30	30	30	30	30	30	30	30	30	30	30	30	30	21	
4				30	30	30	30	30	30	30	30	30	30	30	30	30	30	21	
5				25	23,19	22	22	22	22	22	22	22	22	22	22	22	22	15	
6	24			22	22	22	22	22	22	22	22	22	22	22	22	22	22	15	
7	22	19		27	4	4	12	12	12	10	10	10	10	10	20	20	20	20,14	
8				26,15	20	2	2	2	3	3	3	3	25	25	24	24	24	1	
9				2				5	18	22	22	22	22	19	19	19	19	27	
10				23	15			26				21	6					18,27	
11				24	2	2	5	5	5	5	5	5	5	5	5	5	5	11,11	
12				10				5	5	5	5	5	18	18	18	23	23	16,3	
13				23	23	15	15	26	26	26	26	21	21	21	6	6	6	29,5	
14				23,18				15,5	26	14	6	24	24	10	10			16,7	
15				1	13						10	8						12,27,7	
16	23			18	15	9,5						26	26	14	14	14	14	30,5,4	
17				1	1	1	1	13	13	13	13	13	13	13	13	13	13	18,27	
18				13	13	13	13	13	13	13	13	13	13	13	10	10	8	29,21,7	
19	9			25				16	16	16	16	23	23	18	15	15	15	26,19,17,16	
20	27	11						23,14				29						14,26	
21				9	9	9	9	25	25	25	25	25	25	25	25	25	25	8	
22				25	25	25	25	25	25	25	25	25	25	25	25	25	25	17	
23				25,14	23	30,4	16	27	27	11	11	29	29	29	19	19	19	14,26	
24				5	11	9	9	9	9	9	9	9	9	9	9	9	9	8	
25				30				9	9	9	9	9	23	23	23	23	23	17	
26				30				23	23	23	23	23	23	23	23	23	23	17	
27	17			23	5	5	5	5	11	11	11	11	25	25	4	4	4	1	
28				14				25	11	7			8,1					20,21,1	
29				8	1			14	14	25	25	25	11	11	7	7	7	13	
30													8	8	8	1	1	30,6,28,1	
31																		29,21,7	
32																		14,26	
33																		8	
34																		17	
35																		17	
36																		17	
37																		19,7	
38																		22	
39																		22	
40																		26,18,15	
41																		12,18,20	
42	7,2	20,11	28															21	
43																		21	
44																		21	
45																		21	
46																		21	
47	22,2	20	26	7	7	7	7	7	7	7	7	7	7	7	7	7	7	9	
48																		9	

APPENDIX E: RESULTS FOR A 6 MBPS UPSTREAM CHANNEL

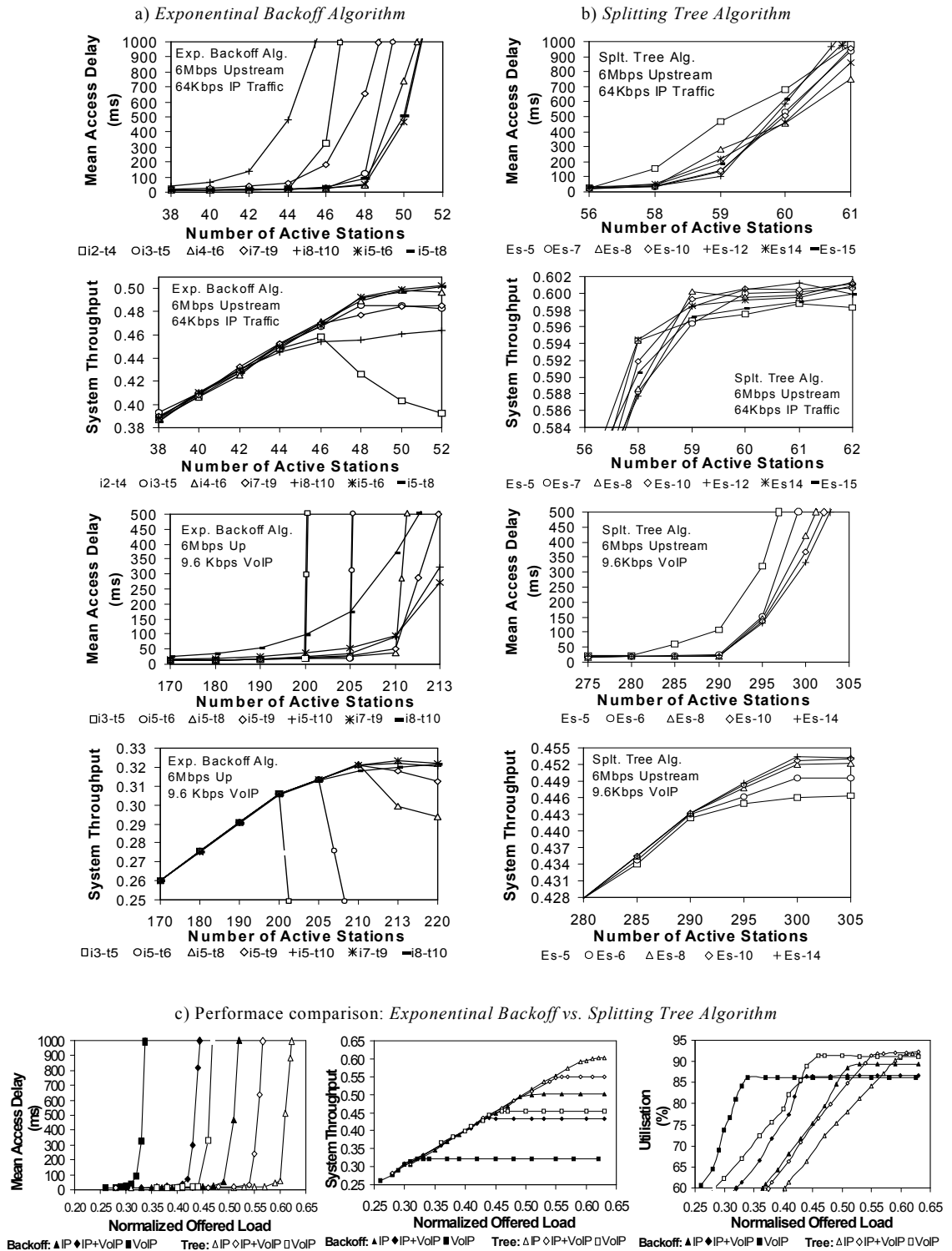
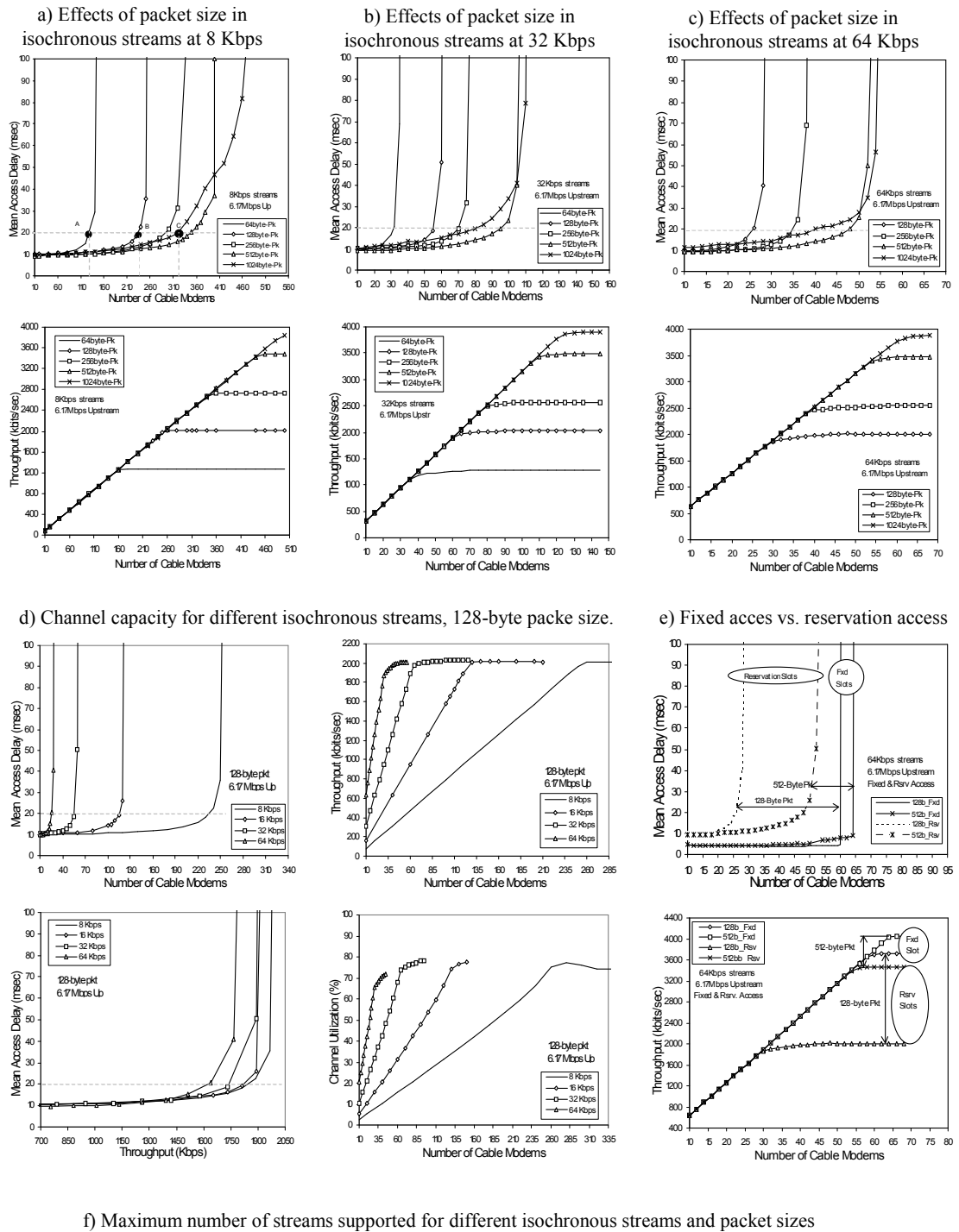


Figure E. 1 – Performance optimisation and comparison of the *exponential backoff algorithm* and the *splitting tree algorithm*, for different traffic patterns and 6 Mbps Up. Channel.






Streams	8 Kbps		16 Kbps		32 Kbps		64 Kbps		128 Kbps	
	Rsv	Fxd	Rsv	Fxd	Rsv	Fxd	Rsv	Fxd	Rsv	Fxd
64 byte	130	352	67	176	32	88	----	44	----	----
128 byte	230	469	116	234	56	117	26	58	----	29
256 byte	300	>500	144	234	70	117	35	58	17	29
512 byte	360	>500	194	256	95	128	48	64	24	32
1024 byte	325	>500	155	256	85	128	42	64	21	32

Figure E. 2 – Performance analysis of isochronous streams for different packet sizes, using the *exponential backoff algorithm* and a 6 Mbps upstream channel.










APPENDIX F: GUIDE TO CD-ROM

There is one CDs attached at the back cover of the thesis. The structure of the files contained is as follows:

F.1 DVB simulation model





-  DVB_Process_Models
-  DVB_Network_Models
-  OPNET_Compatibility Process

F.2 Publications








-  CTS_DVB_QoS_Thesis_Paper(4)
-  IBC_DVB99_PERFORMANCE EVALUATION_DVB_DOCSIS_Thesis_Paper(7)
-  IBC_Tv99_DOCSIS_DVB_Comparisons_Thesis_Paper(9)
-  IBC2001&CTE2002_DVB_Optimisation_of_CRA_Thesis_Paper(11)&(2)
-  ICT2000_DVB_DOC_IEEE_Comparisons_Thesis_Paper(10)
-  IEEE_BT_DVB_Performance_Evaluation_Thesis_Paper(1)
-  IEEE_BT_DVB_Performance_Ischr_Strms_Thesis_Paper(1)_Abstract
-  IEEE_Goblecom2001_DVB_Opmimisation_IschrStrms_Thesis_Paper(12)
-  CTE_DVB_Opmimisation_CSA_Thesis_Paper(13)

F.4 Specifications

Docsis

-  DOCSIS_MAC_Model_Thesis_Ref[77]
-  DOCSIS1_0_RF_Interface_Spec_Thesis_Ref[21]
-  DOCSIS1_1_RF_Interface_Spec_Thesis_Ref[22]
-  DOCSIS1_2_RF_Interface_Spec_Thesis_Ref[23]


Etsi_dvb





-  DVB_CODING_MOD_(ETS_300_429)_Thesis_Ref[31]
-  DVB_DATA_BROADC_(EN_301_192)_Thesis_Ref[33]
-  DVB_DAVIC1_1_(ETSI_300_800)_Thesis_Ref[32]
-  DVB_DAVIC1_2_(ETSI_ES_200_800)_Thesis_Ref[34]
-  DVB_DAVIC1_3_(ETSI_200_800)_Thesis_Ref[35]
-  DVB_S_(EN_300_421)_Thesis_Ref[29]
-  ETSI_TIPHON_QOS_(TR_101_329)_Thesis_Ref[117]

EuroDOCSIS

-  EuroDOCSIS_Overview_Thesis_Ref[2]

EuroModem

-  EuroModem_Thesis_Ref[36]

 **IEEE802_14** IEEE_802_14_DRAF3_REV3_Thesis_Ref[54] IEEE_802_14_FATDMA_SCDMA_Thesis_Ref[54] **ITU_H323** ITU-T Rec. H.323_Thesis_Ref[59] **PacketCable** PacketCable_Spec_Thesis_Ref[80] **F.5 Thesis documents** **F.6 Tools** Acrobat-Reader-4.0 Ghost-View22

The *DVB Simulation model* directory contains the OPNET model used for the performance analysis and optimisations of the DVB/DAVIC protocol. The *Publications* directory contains the papers authored in the course of this work in PDF format. Papers that have been submitted for publication only contain the abstract. The *Specifications* directory contains the specifications for CATV data protocols presented in Chapters 1, 2 and 3, which are available in the public domain. Other copyrighted documents such as ITU-T documents are not included.

The *Thesis Documents* directory contains: an electronic version of this document in postscript and pdf format and the excel file used for the theoretical analysis of the results presented in Section 4.5. Finally, the *Tools* directory contains Acrobat Reader 5.0 and Ghostscript, for reading and printing Acrobat and postscript files under Windows 95/98TM and NT.