Design of a Medium Access Protocol and Scheduling Algorithm for Multimedia Traffic over a DVB-RCS Satellite Link using a Cross-Layer Approach

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EXAMINER’S COPY

To my Dad and Mom

Preface

As the candidate's Supervisor I agree/do not agree to the submission of this thesis

Supervisor's Signature______________________________

The research work performed in this thesis was performed by Jared Wilmans, under the supervision of Prof. Fambirai Takawira, at the University of KwaZulu Natal's School of Electrical, Electronic and Computer Engineering, in the Research Centre for Radio Access and Rural Technologies. This work was partially sponsored by Telkom.

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Abstract

Satellite networks provide an alternative to terrestrial networks where cost and lack of infrastructure are driving parameters. For a satellite network to be cost effective one needs to be able to increase the efficiency of the network: this is accomplished by focusing on the parameters that affect the performance of the system and improving on them where possible. The factors affecting the network performance include the capacity, the propagation delay, the protocol used, and the channel error rate, among others. There are various ways to implement a satellite network depending on the satellite orbit, the architecture used, the access technique used, the radio interfaces used, etc.

This thesis work describes the chosen satellite standard, Digital Video Broadcasting – Return Channel via Satellite (DVB-RCS) and the associated Medium Access Control (MAC) protocols.

Two protocols were designed and investigated under ideal channel conditions, these being the Combined Free/Demand Assigned Multiple Access with Piggy Backing – Packet Dropping (CF/DAMA-PB-PD) protocol; and the Combined Free/Demand Assigned Multiple Access with Piggy Backing – Prioritised Earliest Deadline First (CF/DAMA-PB-PEDF) protocol, both derived from the Combined Free/Demand Assigned Multiple Access with Piggy Backing (CF/DAMA-PB) protocol.

The multimedia traffic models for voice, video and web classes are described, validated through simulations and presented; these provide the heterogeneous

traffic required for evaluating the performance of the satellite system implemented and the designed protocols. Under the multimedia traffic, CF/DAMA-PB-PD was shown to excel in average packet delay reduction while reducing the overall system throughput. The CF/DAMA-PB-PEDF does not contribute to an improvement over the CF/DAMA-PB-PD protocol.

The effects of a non-ideal channel on the CF/DAMA-PB-PD protocol was investigated and presented along with the design of three MAC protocols that take the channel characteristics into account to improve on the system performance. The cross-layer interactions, more specifically the interaction between the physical and data–link layers, were used, investigated and presented. The channel state information in terms of signal-to-noise ratio (SNR) was used to improve the system performance.

The five protocols evaluated under non-ideal channel conditions were the CF/DAMA-PB, CF/DAMA-PB-PD, CF/DAMA-PB-BSNRF, CF/DAMA-PB-DD and the CF/DAMA-PB-BSNRF+DD. The best overall performance, both in average packet delay while maintaining good QoS levels and throughput was shown to be that of the CF/DAMA-PB-DD protocol.

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List of Acronyms

ACK	Acknowledgement
ACM	Adaptive Coding and Modulation
ADSL	Asymmetric Digital Subscriber Line
AMD	Advanced Micro Devices
APSK	Amplitude Phase-Shift Keying
ATM	Asynchronous Transfer Mode
AVBDC	Absolute Volume Based Dynamic Capacity
BCH	Bose Ray-Chauduri Hocquenghem
BER	Bit Error Rate
BoD	Bandwidth on Demand
BSM	Broadband Satellite Multimedia
BSNRF	Best Signal-to-Noise Ratio First
BSS	Broadcasting Satellite Service
BTP	Burst Time Plan
CDMA	Code Division Multiple Access
CF/DAMA	Combined Free/Demand Assigned Multiple Access
CF/DAMA-PB	CF/DAMA with Piggy Backing
CF/DAMA-PB-PD	CF/DAMA-PB – Packet Dropping
CF/DAMA-PB-PEDF	CF/DAMA-PB – Prioritised Earliest Deadline First
CF/DAMA-PB-BSNRF	CF/DAMA-PB – Best Signal to Noise Ratio First
CF/DAMA-PB-DD	CF/DAMA-PB – Dynamic Deadlines
CF/DAMA-PB-BSNRF+DD	CF/DAMA-PB – Best Signal to Noise Ratio First and Dynamic Deadlines
CMF	Control and Monitoring Functions
CRA	Constant Rate Assignment
CRC	Cyclic Redundancy Check
CRSC	Circular Recursive Systematic Convolutional
CSC	Common Signalling Channel
DBS	Direct Broadcast Satellites

D-MAP	Discrete Time Markovian Arrival Process
DVB	Digital Video Broadcasting
DVB-RCS	Digital Video Broadcasting – Return Channel via Satellite
DVB-RCS NG	Digital Video Broadcasting – Return Channel via Satellite Next Generation
ETSI	European Telecommunications Standards Institute
EUL	Enhanced Uplink
FAQ	Free Assignment Queue
FCA	Free Capacity Assignment
FCFS	First Come, First Serve
FDMA	Frequency Division Multiple Access
FIFO	First In, First Out
FSS	Fixed Satellite Service
FTP	File Transfer Protocol
GEO	Geostationary Earth Orbit
GMR	GEO – Mobile Radio
GMSK	Gaussian Minimum Shift Keying
GPS	Global Positioning System
GUI	Graphical User Interface
HEO	High Elliptical Orbit
HS	Heavy Shadowing
HSDPA	High-Speed Downlink Packet Access
HSUPA	High-Speed Uplink Packet Access
ICO	Intermediate Circular Orbit
IMT-2000	International Mobile Telecommunication - 2000
IP	Internet Protocol
IPoS	Internet Protocol over Satellite
ISDN	Integrated Services Digital Network
ISL	Inter-Satellite Link
ISN	Interactive Satellite Network
ITU	International Telecommunication Union
LDPC	Low-Density Parity-Check
LEO	Low Earth Orbit
LOS	Line Of Sight
LS	Light Shadowing
MAC	Media Access Control
MCPC	Multiple Carrier per Channel
MEO	Medium Earth Orbit
MF-TDMA	Multiple Frequency – Time Division Multiple Access
MPE	Multiprotocol Encapsulation
MPEG	Moving Picture Experts Group
MPEG2-TS	Moving Picture Experts Group (version 2) – Transport Stream
MSS	Mobile Satellite Service
NCC	Network Control Centre
OBP	On-Board Processor

OSI	Open System Interconnection
PER	Packet Error Rate
QoS	Quality of Service
QPSK	Quadrature Phase-Shift Keying
RBDC	Rate Based Dynamic Capacity
REQ	Request
RRM	Radio Resource Management
RRQ	Reservation Request Queue
RS	Reed-Solomon
RSM-A	Regenerative Satellite Mesh – Air Interface
RTD	Round Trip Delay
SCPC	Single Carrier per Channel
S-DOCSIS	Satellite – Data over Cable Service Interface Specification
SMAC	Satellite Medium Access Control
S-UMTS	Satellite – Universal Mobile Telecommunications System
SRMA	Scheduled-Retransmission Multiaccess
SRMA/FF	Scheduled-Retransmission Multiaccess – Fixed Frame
SRMA/DF	Scheduled-Retransmission Multiaccess – Dynamic Frame
TBTP	Terminal Burst Time Plan
TD-CDMA	Time Division – Code Division Multiple Access
TDMA	Time Division Multiple Access
TD-SCDMA	Time Division – Synchronous Code Division Multiple Access
TG	Traffic Gateway
UDP	User Datagram Protocol
UMTS	Universal Mobile Telecommunications Systems
VBDC	Volume Based Dynamic Capacity
VCM	Variable Coding and Modulation
VoIP	VoIP
W-CDMA	Wideband – Code Division Multiple Access

List of Symbols

a	Exponent of the auto-covariance
α	Geometric probability
A	Video mini-source bit rate factor
$\hat{A}_v$	Peak video bit rate
b	Coefficient of exponential packet error rate
B_s	Voice source degree of burstiness
B_v	Video source degree of burstiness
B_w	Web source degree of burstiness
β	Geometric probability
c	Signal-to-noise ratio coefficient
D	Delay
$E[x]$	Expected value of the random variable *x*
f_{ci}	Carrier frequencies (i = 1, 2,...,M)
G	Offered Load
I	Input load
k	Minimum length of a we datagram
l_d	Random pareto length of a web datagram
M	Number of video mini-sources
m	Maximum length of a web datagram

m_{w_d}	The mean number of datagrams
$m_{w_{ia}}$	The mean inter-arrival time
m_{w_r}	The mean reading state time
N_{atm}	Number of ATM cells
N_{mpeg}	Number of MPEG packets
PD	Packets Dropped
p	Mean ON time
P_{ij}	Probability of moving to state *j* from state *i*
$\boldsymbol{P}$	State transition probability matrix
PER	Packet error rate
q	Mean OFF time
q_w	Factor used to modulate burstiness of web source
R_s	Voice peak bit-rate
S	Throughput
SNR	Signal to Noise Ration in decibels
$\overline{SNR}_i$	Mean Signal to Noise Ratio of state i
T_s	Slot time
σ^2	Variance
σ_{state}	Channel state standard deviation
U	Utilization
μ_{ON}	Mean exponential ON time
μ_{OFF}	Mean exponential OFF time
μ_v	Video mean bit rate
μ_s	Voice mean bit rate
μ_w	Web mean bit rate
v	Shape of a Pareto distribution
ω	The exponential ON rate

W_i	Probability of being in state i
$\boldsymbol{W}$	State probability matrix
ζ	The exponential OFF rate
ψ_v	Video activity factor
ψ_s	Voice activity factor
ψ_w	Web activity factor

1 Introduction

In this age of global communications explosion and the huge benefits accruing from this, it is very important to create a communication system that is reliable, efficient, and cost effective and reaches all corners of the globe. Satellite communication networks have an integral part to play in achieving these goals.

1.1 Satellite Systems

Without satellite communications many a technology would not have existed as we know them today, such as Global Positioning Systems (GPS), satellite television broadcasting, satellite space observation, weather services, military applications, humanitarian efforts after disasters, etc.

There are still many areas of the globe that remain in the dark when it comes to global communication infrastructure. And many rural and underdeveloped areas do not have access to high speed communication systems. The cost of such terrestrial infrastructure has posed an enormous challenge to these areas, both financially and in terms of the skills required to implement and support such systems.

Satellite communication networks are an attractive means to overcome such challenges. They are able to provide multimedia communication services to areas without, and form a more complete global coverage than terrestrial systems currently provide.

1.1.1 Satellite Orbit Types

There are four main types of satellite orbits depicted in Figure 1, these being the Low Earth Orbit (LEO), the Medium Earth Orbit (MEO), the Geostationary

Earth Orbit (GEO) and the High Elliptical Orbit (HEO). Each of the orbital methods has their pros and cons.

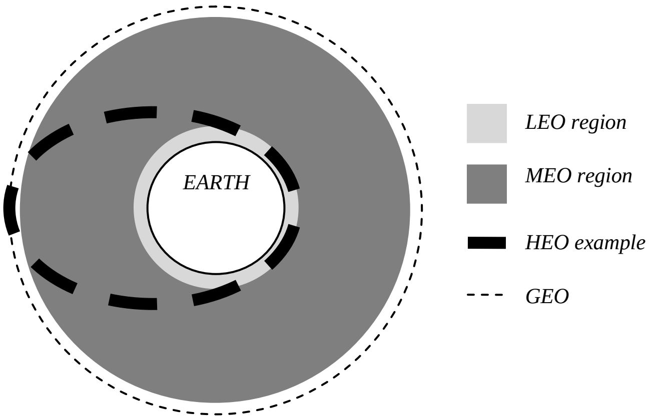

Figure 1: Various satellite orbits

LEO

The LEO satellites orbital range is between 300 km and 2,000 km from the surface of the earth, below the Van Allen radiation belts. The period, i.e. the time taken to circle the Earth, of a LEO satellite is of the order of 100 minutes, which translates into a satellite passing overhead every 15 minutes, due to the Earth having 1440 minutes in a revolution. With a satellite having such a short visibility period, many satellites are required to create a network that ensures continuous visibility which leads to more regular connection handover between satellites. This creates a challenge in terms of the cost of deployment as well as a greater complexity in maintaining the system coverage.

MEO

The MEO satellite orbit, also referred to as the Intermediate Circular Orbit (ICO), has an orbital range above the LEO and below the GEO. It is placed between the two Van Allen belts. This orbit has, due to its altitude and speed, greater periods of visibility (2-4 hrs). With a rotation period ranging between 5 hrs and 12 hrs, the number of satellites required to maintain global coverage is reduced, compared to that of the LEO. The number of connection handovers between satellites is less, but not eliminated, adding to the complexity of the system compared to that of GEO satellites. The increase in altitude results in a greater path loss and a longer round trip delay (RTD). The MEO can be either circular or elliptical in shape.

GEO

The GEO satellite has a geosynchronous orbit around the Earth, directly above the equator. The orbit is positioned at an altitude of 35,780 km and has a period of 24 hrs. This maintains a fairly constant coverage footprint on the Earth's surface with one satellite, i.e. it remains stationary with respect to a fixed point on the Earth's surface. This reduces the complexity of the system in terms of not requiring connection handover. A minimum of 3 GEO satellites are required to obtain global coverage. It has relatively large path losses due to the large RTD of 270 ms.

HEO

HEO satellites have an elliptical orbit characterised by an extremely high-altitude apogee and relatively low-altitude perigee. The apogee has a long visibility reaching up to 12 hrs, whereas the perigee has a short and quick phase. This orbit is used mainly at the poles for communications purposes, as GEO tends to not cover these areas adequately. A large number of these satellites are used for Russia, due to its high latitude position.

The satellite orbit type that this thesis will be focusing on is the GEO. This orbit does not require complex hand-over techniques and is capable of providing high bandwidth to areas lacking in terrestrial infrastructure without a huge deployment of multiple satellites. This satellite type is particularly well suited for rural area connectivity.

1.1.2 Satellite Architectures

The architecture chosen affects the propagation delay experienced when transmitting data and/or requesting system capacity to transmit data. There are two general satellite architectures, the bent-pipe and the regenerative satellite architecture.

Bent-Pipe

The bent-pipe architecture (Figure 2) has three main components; the Earth station with a gateway, the satellite repeater and the user Earth terminals. The satellite does not do any data manipulation or processing; it simply receives a signal, amplifies and retransmits it.

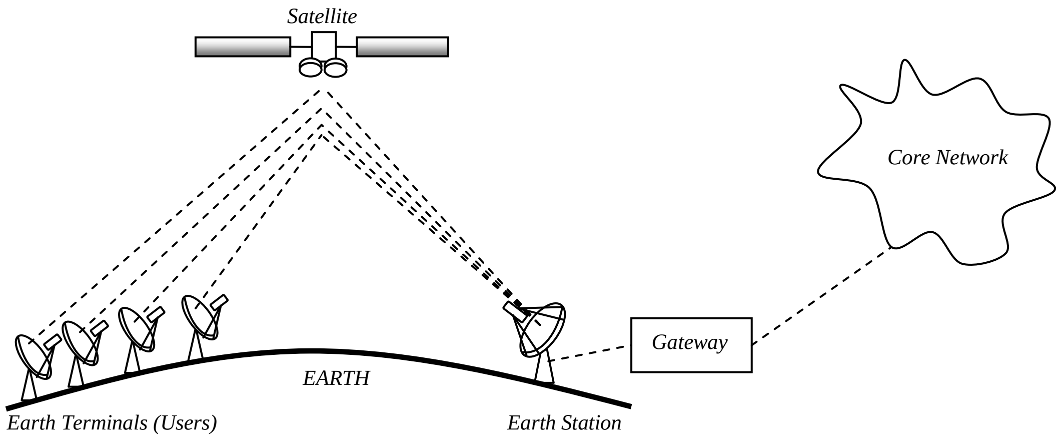

Figure 2: Bent-pipe satellite architecture.

Regenerative

The regenerative architecture differs to the bent-pipe architecture as it has an added component, namely the ability to process data onboard the satellite, Onboard Processing (OBP). It allows the satellite to demodulate and decode the received signal and retransmit after signal error recovery is done. Employment of Inter-Satellite Links (ISLs) is made possible. This allows for the most efficient methods of mesh connectivity establishment.

There are three classifications of the onboard processors; the regenerative with onboard switching (Figure 4) the regenerative without onboard switching (Figure 3) and the regenerative hybrid, i.e. the hybrid has both the transparent repeater of a bent-pipe as well as the onboard processing capabilities.

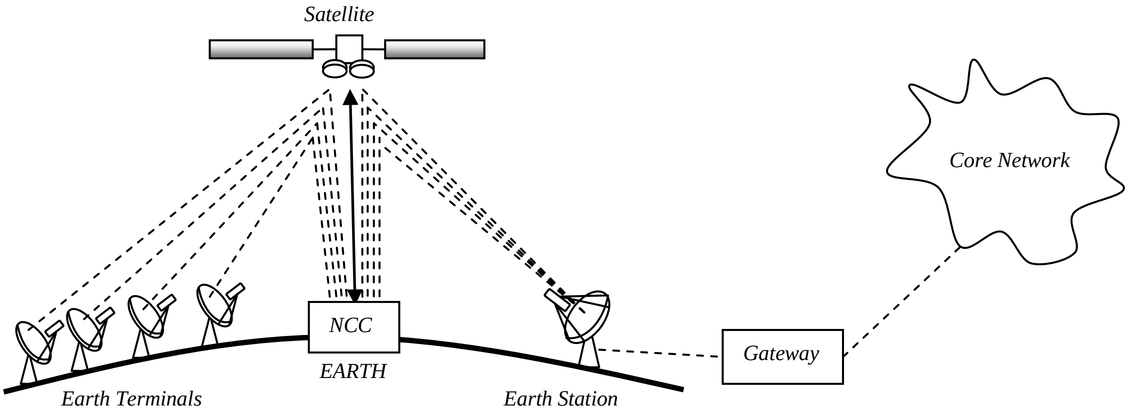

Figure 3: Regenerative satellite architecture (Without onboard switching).

The Network Control Centre (NCC), seen in Figure 3 above, provides control and monitoring functionality as well as timing for the network.

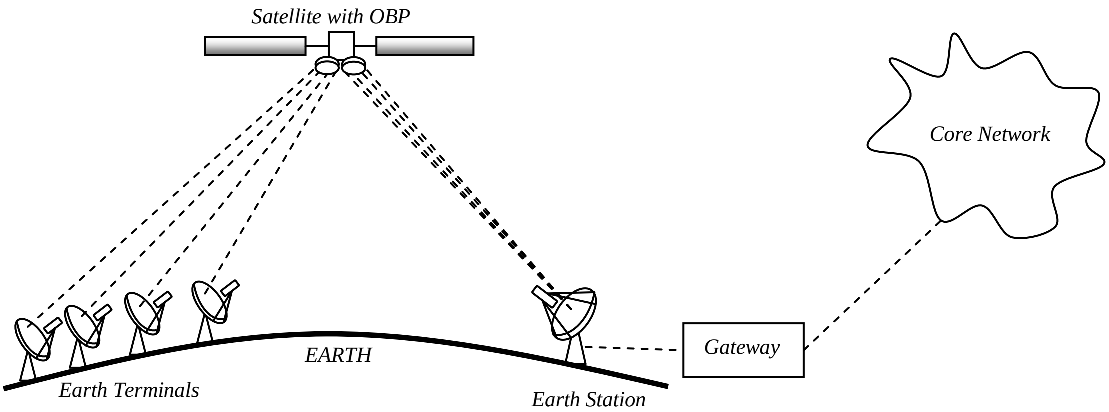

Figure 4: Regenerative satellite architecture (With onboard switching).

The satellite architecture that this work was interested in is the regenerative with OBP capabilities. The OBP will have similar functionality as the NCC. The satellite will have the MAC algorithm and make decisions of allocating the return link capacity requests.

1.1.3 Satellite Networks

Satellite networks can be used in many different situations; in point-to-point communication, in point-to-multipoint communications and in multipoint-to-multipoint communications. Satellite networks are able to form a bridge virtually anywhere within a network if required. They can be used within a core network or to gain access to a core network, where access is limited or nonexistent.

This is all made possible through the correct standardisation of network implementation. There are various aspects that form a network; from the users of the network right through to the servers where content is stored and accessed by the users.

The European Telecommunications Standards Institute has put together new network architecture for a Broadband Satellite Multimedia (BSM) network. The network is divided into five domains [13]:

- User Domain
- Access Domain
- Distribution Network
- Core Network
- Content Domain

The User Domain is the group of end users that utilise the network services. The Access Domain is the method used to connect to the services of the network (e.g. ISDN, ADSL, Satellite, etc). The Distribution Network is the intermediate network that connects the core network to the access network. The Core network is the transport network that interconnects and defines the networking in a geographical region. And the final network, the Content Domain, is where the services are that the end user has access to through the network and the domains.

A standardized protocol stack (Figure 5) is defined, which separates the satellite dependant layers from the satellite independent and the external layers. The dependent and independent layers have an interface specification, the Satellite Independent – Service Access Point (SI-SAP).

The SI-SAP is further divided into three SAPs, [14]:

- SI-U-SAP
- SI-C-SAP

- SI-M-SAP

The SI-U-SAP is the User SAP, the SI-C-SAP is the Control SAP and the SI-M-SAP is the Management SAP.

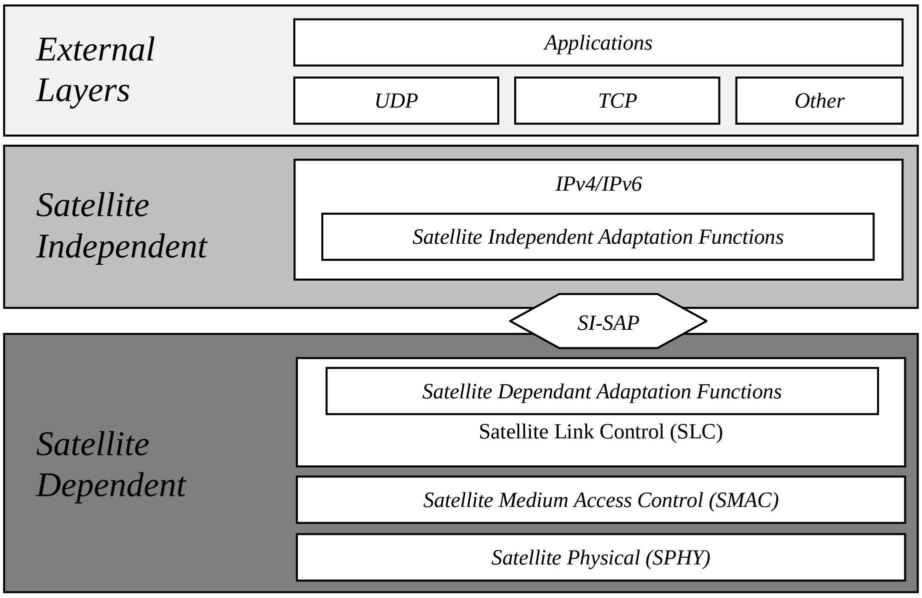

Figure 5: Standardized Protocol Stack with SI-SAP [2], [3], [4].

1.1.4 Multiple Access Techniques

In general, multiple access techniques are the methods that are used to share a common resource amongst several devices. Examples of resources that are shared in communications are the medium for the communication, such as copper cable, space, fibre optics, etc. The medium access technique is the way in which the resource is made useful, examples of medium access techniques used in communications are; Time Division Multiple Access (TDMA), Frequency Division Multiple Access (FDMA), Code Division Multiple Access and a combination, or mix, of these techniques. In satellite communications, it is the ability to have multiple Earth stations all sharing the common air interface to communicate, or simultaneously connect, with the satellite station in space in order to transmit traffic to single/multiple destinations. The techniques ensure that each Earth station has a means to utilise the capacity offered by the satellites transponders. Figure 6 depicts the basic access techniques in terms of their throughput characteristic.

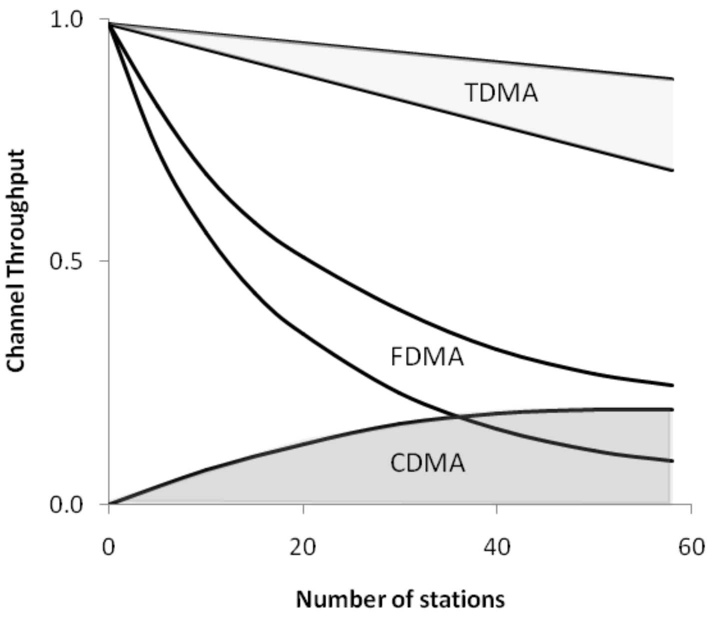

Figure 6: Throughput of the basic access techniques [17].

FDMA

FDMA is a multiplexing technique used in radio communications. It is an analogue transmission technique where frequency is used to share the channel. The range of frequencies, or bandwidth, is usually subdivided into even portions. Each portion has a carrier frequency to identify the range and guard bands to protect the band from interference caused by adjacent frequencies. Each user is assigned its own carrier frequency. This technique is generally used where the number of users sharing the channel is very small, as it has a limited range of frequencies depending on the system and its assigned radio spectrum. FDMA is rather simple to implement as it does not require synchronization between users but tends to be wasteful in that if a user does not need to transmit data, the frequency remains unused, although it is possible to divide the frequency ranges proportional to the needs of the users. FDMA is depicted in Figure 7.

Figure 7: FDMA structure.

TDMA

In TDMA, depicted in Figure 8, a single frequency is shared amongst multiple users. This method divides the signal into time slots. Each time slot is assigned to a user. The time slots are usually organised into time frames. This method tries to ensure that all users get a time slot to utilise the channel capacity. A greater number of users leads to a decrease in the time slot length for a set time frame. The result is a non-continuous transmission per user. Between each time slot there is a guard period, similar to the guard band in FDMA, which ensures no interference from adjacent slots as a result of the system losing synchronization between sender and receiver. This method has been widely used in satellite systems.

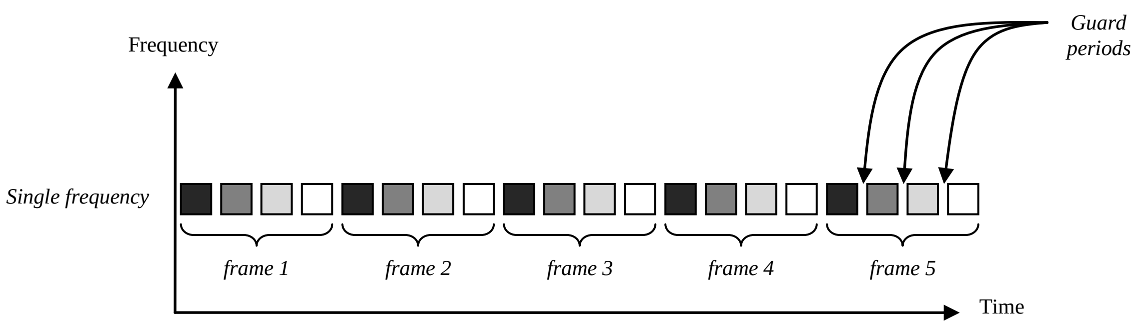

Figure 8: TDMA structure.

CDMA

CDMA uses spread-spectrum and a coding scheme to multiplex multiple users over a shared channel. Each transmitter is assigned a code. The code is used to modulate a user's signal, allowing the receiver to determine what message was sent and from whom it was sent. An analogy is often used to describe how CDMA works. Suppose there are many people in a room all trying to communicate with each other, if all are speaking at the same time and in the same language it creates confusion. If each pair that wishes to communicate a message with one another decides to use a unique language, this gives each pair the ability to convey a message concurrently without much confusion at all. Some technologies make use of this concept in communications (e.g. 3G, GPS, etc).

Mixed

The mixed/hybrid method makes use of a combination of the above mentioned methods, such as Multi Frequency – Time Division Multiple Access (MF-TDMA) [1], Time Division – Code Division Multiple Access (TD-CDMA), etc. The focus in this thesis is the MF-TDMA.

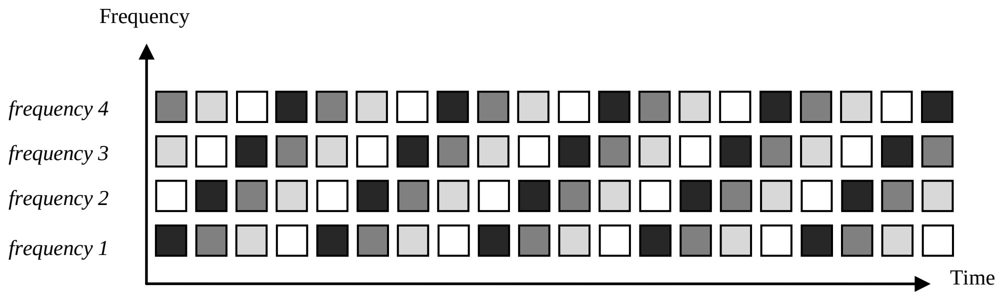

Figure 9: MF-TDMA Structure (Basic).

MF-TDMA has both FDMA and TDMA; this is implemented by applying TDMA to the FDMA frequency bands. Thus allowing multiplexing of both time, in terms of slots defined by frames, and frequency, in terms of frequency bands defined by carrier frequencies.

1.1.5 Frequency Bands and their Uses

Frequency bands fall into various categories; the ones of interest are those that are related to satellite communications. The type of service usually determines the frequency bands to be used.

Table 1: Frequency bands used for satellite communications [2].

Band Name	*Frequency Range*
L	*1 – 2 GHz*
S	*2 – 4 GHz*
C	*4 – 8 GHz*
X	*8 – 12 GHz*
Ku	*12 – 18 GHz*
K	*18 – 26 GHz*
Ka	*26 – 40 GHz*
V	*40 – 75 GHz*

Fixed Satellite Service (FSS) makes use of the most common of these frequencies, the C (6 GHz uplink and 4 GHz downlink) and Ku (14 GHz uplink and 11GHz downlink) bands. The X, Ka, and V bands are also used for these services. FSS uses fixed terrestrial terminals and tend to be broadband. Both GEO and non-GEO satellite systems are used for FSS. Examples of these satellites are the Hughes Galaxy, GE American, Loral Skynet, etc.

Direct Broadcast Satellites (DBS) or Broadcasting Satellite Service (BSS) uses the S and Ku bands. These services are used in direct-to-home video/audio commercial services, although other uses are also implemented. Examples of these satellites are the DirecTV, Echostar, etc.

Mobile Satellite Service (MSS) uses the L and Ka bands. These services were initially designed for voice and low speed data to mobile units. Examples of these satellites are the Inmarsat, ACES, etc.

1.1.6 Performance Parameters

In communications networks one has various parameters which affect the performance of the system. For a system to be improved in any way one has to evaluate these parameters and determine what can and cannot be varied. In satellite systems, the following parameters, affecting the performance, have been identified, [5]:

- Capacity
- Propagation Delay
- Number of bits per frame
- Medium access protocol
- Offered load
- Number of stations
- Error rate

The capacity of the system is a fixed parameter defined by the channel and system hardware. It is a given parameter which generally defines the upper limit of the system. The propagation delay is, in the case of non-GEO satellites, varying within a given range according to the orbit of the satellite or, in the case of GEO satellites, it is a fixed parameter with insignificantly small variations. In both non-GEO and GEO cases the values are predefined by the altitude and orbit of the satellite. The number of bits per frame is a fixed value according to the capacity of the system and the multiple access technique used. The first three parameters are the system defined performance parameters. The next parameter, the medium access protocol, is the main design problem in this thesis. The MAC protocol is where the design choices are needed to be made in order to increase the performance of the system. The next two parameters, the offered load and the number of stations, are generally seen as the independent variables of the system. The performance of the system is determined as a function of these two parameters. The offered load is the traffic that is generated by the stations in the network. The final parameter, the error rate, is due to channel effects and results in packet loss.

1.2 Radio Interfaces

The radio interface specifies the common boundary in a radio network. The boundary is defined by various characteristics, such as the functional, the physical interconnections, the signal, etc. There are numerous standards which define the radio interface. The standards of importance to this work are those of satellite communications. Some existing radio interface satellite standards are:

- Digital Video Broadcasting – Satellite (DVB-S), [6].
- Digital Video Broadcasting – Return Channel via Satellite (DVB-RCS), [1], [6], [7].
- Digital Video Broadcasting – Satellite –Second Generation (DVB-S2), [1], [6]-[8].
- Satellite – Universal Mobile Telecommunications System (S-UMTS), [9]-[12].
- Regenerative Satellite Mesh – Air Interface (RSM-A).
- GEO – Mobile Radio (GMR).
- Satellite – Data over Cable Service Interface Specification (S-DOCSIS).
- Internet Protocol over Satellite (IPoS).

A brief overview of the first four is presented, while chapter 2 will describe DVB-RCS in more detail.

1.2.1 S-UMTS

The Universal Mobile Telecommunications Systems (UMTS) technology fall under the International Mobile Telecommunications – 2000 (IMT-2000) family of technologies. The IMT-2000 family is better known as 3G or 3rd Generation and was defined by the International Telecommunication Union (ITU). UMTS is also known as W-CDMA, TD-CDMA or TD-SCDMA. UMTS helps to achieve a high spectral efficiency and data rates that are capable of supporting download speeds of up to 14Mbits/s, through High-Speed Downlink Packet Access (HSDPA), and upload speeds up to 11.5Mbits/s, using High-Speed Uplink Packet Access (HSUPA) – also known as Enhanced Uplink (EUL).

The Satellite UMTS (S-UMTS) standard adds to UMTS the support of satellite communication. ETSI made a contribution to the S-UMTS standard by ensuring that it was possible to interwork the S-UMTS with UMTS terrestrial networks. S-UMTS helped in achieving a far greater coverage of mobile handset connectivity through satellite communications.

For further insight into the S-UMTS standard and technical details see references [9]-[12], [16], ETSI Technical Reports; TR 101 865, TR 101 866, TR 102 058 and TR 102 277.

1.2.2 DVB-S

The ETSI Digital Video Broadcasting set of standards define the interface for video broadcasting of digital television using various technologies (i.e. terrestrial, cable, satellite, etc.). These set of standards define the physical layer, as well as the data link layer of the distribution system. They transport data in Moving Picture Experts Group Transport Streams (MPEG-TS). DVB defines the transmission of voice, video and data.

The Digital Video Broadcasting – Satellite (DVB-S) standard is the original DVB standard which defines the satellite interface of DVB. DVB-S defines the Forward Error Correction (FEC) coding and the modulation used for satellite television. The standard uses Moving Picture Experts Group version 2 as its packet structure and defines the physical link characteristics as well as the frame structure implemented. It is used in both Multiple Carrier per Channel (MCPC) and Single Carrier per Channel (SCPC) system implementations.

1.2.3 DVB-RCS

The Digital Video Broadcasting – Return Channel via Satellite (DVB-RCS) standard is an open standard with highly efficient bandwidth management that helps in being a cost effective solution compared to that of the alternatives.

The DVB-RCS standard defines the interactive channel for video broadcasting having the capability to be used not only for voice and video but also for data transmission. Data transmission is accomplished by means of Multiprotocol

Encapsulation (MPE). DVB-RCS supplies an alternative to that of its terrestrial equivalent systems and reaches, in many situations, far larger areas than the current terrestrial network does.

The encoding method used is that of Phase-Shift Keying (QPSK and GMSK) and the multiple access technique used is MF-TDMA.

Work has begun on the second version of the DVB-RCS standard, also known as DVB-RCS Next Generation (DVB-RCS NG).

1.2.4 DVB-S2

The Digital Video Broadcasting – Satellite version 2 (DVB-S2) standard is built on that of the DVB-S standard and one of its aims were to add the ability to efficiently broadcast High-Definition Television (HDTV) using satellites. It has been claimed in [8] that the standard can achieve up to 30% performance gain over that of DVB-S.

The system is capable of transmitting one or more MPEG2 or MPEG4 audio-video streams. It makes use of QPSK, 8PSK 16APSK or 32APSK modulation with concatenated encoding. It can employ Variable Coding and Modulation (VCM) and Adaptive Coding and Modulation (ACM). For the FEC it makes use of Bose Ray-Chauduri Hocquenghem (BCH) and Low-Density Parity-Check (LDPC) coding.

The DVB-S2 standard is defined in ETSI EN 302 307.

1.3 MAC Schemes

MAC is an essential part of a communication network; it ensures that a network is able to share a channel amongst multiple users/nodes according to a set of rules.

MAC protocols can be classified into categories as depicted in Figure 10. The highest of these categories is the user category. One generally has two user types, the passive user and the active user. The passive set of protocols gives the control to a central controller that decides who may access the channel at any one time, i.e. a user may access the channel only once it has been polled. The active set of protocols gives the user the ability to try and obtain access to the channel without having to wait for the controller to poll them. The active user set is further subdivided into contention free protocols and contention oriented protocols. Contention free protocols ensure that the users/nodes do not access the same channel capacity simultaneously, as this would cause a conflict and require some sort of conflict resolution scheme. On the other hand, contention oriented protocols allow for conflicts and define a conflict resolution scheme to resolve the conflict. Contention free protocols are further subdivided into Fixed Assignment, Demand Assignment, Free Assignment and the hybrid sets. Contention oriented protocols are further subdivided into the Random Access, Reservation and hybrid sets of protocols.

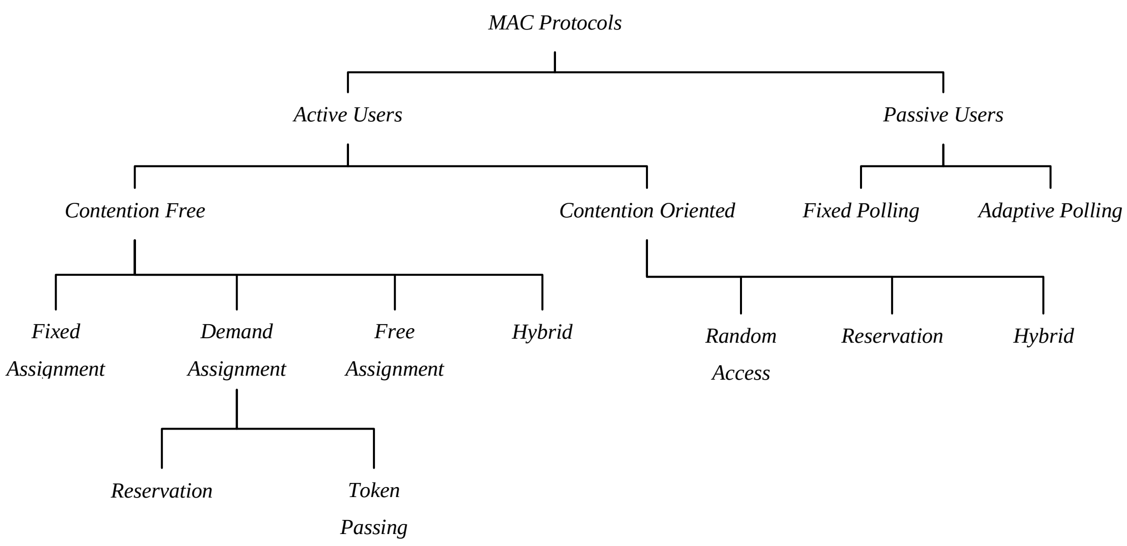

Figure 10: MAC protocol classification [18] [19].

1.3.1 Fixed Assignment

Fixed assignment is a set of protocols that decide before hand how and when the user/node may access the channel. This assignment is a static assignment and does not depend on the users/nodes activity. Examples of these protocols are that of the TDMA, FDMA and CDMA access techniques. In TDMA, the channel is divided into time slots, one or more per user, and the assignment is fixed, i.e. it cannot be changed according to the user's activity. In FDMA, the channel is divided into frequency ranges, each being assigned to one user/node, and the assignment is also fixed. Similarly, CDMA is also a fixed assignment protocol, the benefit of this over that of TDMA and FDMA is that it is a conflict free scheme that allows for overlapping of transmission in both frequency and time domains. The limitation is that CDMA has a very low upper bound for throughput as opposed to that of TDMA and FDMA, see Figure 6. Refer to pp. 8.

These protocols are found to be very effective in satellite communication when one has a finitely small number of users with a stable and predictable traffic load. Fixed assignment also has the benefit of being able to guarantee the throughput and Quality of Service (QoS) that the network may offer due to each user having exclusive rights to a portion of the total satellite capacity.

1.3.2 Demand Assignment (BoD)

Demand assignment, also referred to as Bandwidth on Demand (BoD), is a set of MAC protocols that allow the users/nodes to utilise channel capacity according to their traffic requirements. In the case where the traffic is unpredictable and each user's requirement unknown, it is required that one assign capacity dynamically according to the needs of the users/nodes. Demand assignment is theoretically capable of ensuring high throughput and hence high channel utilisation of the system. One can subdivide this set of protocols into reservation and token based protocols, in reservation, the user requests capacity according to its requirements; this can be explicit or implicit reservation. In explicit reservation, a single slot is assigned to each user as required. In implicit reservation users compete for the reservation slots, this reduces the number of requests made and hence request slots required per frame. In token passing a user can use the system capacity as needed whilst holding the token and once done, or time has expired, pass the token on to the next logically ordered user in the network.

There are three general phases in demand assignment:

- The first phase in which the connected stations make their requests
- The second phase where the satellite or NCC allocates requests according to an algorithm
- The final phase is transmitting the data in the allocated slots

Demand assignment can also be divided into two categories according to the types of requests made. The fixed rate and the variable rate demand assignment.

In the fixed rate demand assignment, the user connects to the network and makes a request for channel capacity. If the request is granted, the user is allocated a fixed channel capacity for the period of its connection, at which point it then disconnects and makes its allocated capacity available to other users.

In the case where a user remains connected irrespective of whether it has traffic or not, the variable rate demand assignment is more desirable. The variable rate demand assignment allows the user to request capacity when required and then allocates the capacity once off according to a predetermined algorithm (e.g. First Come, First Serve – FCFS also referred to as First in, First Out - FIFO). This method is favourable for all traffic types even if they have varying requirements over time.

1.3.3 Free Assignment

Free assignment is similar to that of fixed assignment, as it assigns capacity to users without the user requesting that capacity. The difference between the fixed and free assignment strategies is that the fixed assignment is a guaranteed assignment that ensures the user is allocated capacity irrespective of whether they require it or not, whereas in the free assignment strategy the free capacity is not a guaranteed capacity but rather extra capacity that is not being utilised. The controller allocates the free capacity according to an algorithm (e.g. round robin).

This is beneficial to those users who, at the time of being allocated the free capacity, have extra traffic to make use of the allocation. Free assignment has the benefit of decreasing the end-to-end delay of a user but has the limiting factor dictated by the number of users that the controller shares the free capacity amongst. As the number of users increases, the less probability there is of receiving a free assigned capacity and hence this assignment has less effect on the overall system performance. The main benefit of this system is a probable increase in channel utilisation.

1.3.4 Random Access

Random access is different to the other strategies, as it gives the control almost completely to the user. The user does not make requests for channel capacity, it rather tries to access the channel in hope that it does not conflict with any other user accessing the channel. Random access is relatively simple to implement and takes into account varying demand but, due to collisions, can be quite wasteful of channel capacity. It cannot guarantee a QoS and therefore tends to be a bad choice for real time applications. Random access is a contention based protocol that allows for the uncoordinated accessing of channel capacity. The Aloha set of protocols are good examples of random access, i.e. Pure Aloha, Slotted aloha, etc.

1.3.5 Hybrids

The hybrid set of protocols tries to make use of a combination of the above mentioned protocol sets in the hope of obtaining the benefits of the protocols used while reducing the limitations of the protocols.

This set of protocols has proven to be of great interest in research and has achieved far more efficient systems than the protocol sets alone.

It is important to realise that no single strategy used will be able to provide both a reduction in end-to-end delay as well as an increase in throughput without combining two or more of these strategies. For this reason the focus in this thesis was on a hybrid MAC protocol.

1.4 Scope of Thesis

This thesis covered performance evaluations and simulations of several Medium Access Control Protocols, each having been derived from the CF/DAMA-PB protocol by Le-Ngoc et al in [21]-[23]. The system was based on a satellite standard for video broadcasting, DVB-RCS.

System evaluation considers the average packet delay performance of each protocol, trying to reduce the delay was one aim. The throughput of the system was evaluated, as well as the individual throughput of each traffic class contributing to the system throughput. The percentage of dropped packets was also evaluated in systems affected by this, i.e. where deadlines for traffic delays were implemented and channel loss was taken into account.

The work made use of three traffic models, video, voice and web, based on Markovian Processes and used these traffic models as the offered load to the system in order to evaluate the system.

Two channel models were looked at, the Good-Bad channel model and the 3 state channel model. The 3 state channel model was used to create a more realistic channel environment.

Evaluation of six different protocols was undertaken, some under both ideal channel conditions as well as more realistic channel conditions. The protocols evaluated were:

- Combined Free/Demand Assigned Multiple Access with Piggy Backing (CF/DAMA-PB)
- Combined Free/Demand Assigned Multiple Access with Piggy Backing – Packet Dropping (CF/DAMA-PB-PD)
- Combined Free/Demand Assigned Multiple Access with Piggy Backing – Prioritised Earliest Deadline First (CF/DAMA-PB-PEDF)
- Combined Free/Demand Assigned Multiple Access with Piggy Backing – Best Signal to Noise Ratio First (CF/DAMA-PB-BSNRF)

- Combined Free/Demand Assigned Multiple Access with Piggy Backing – Dynamic Deadlines (CF/DAMA-PB-DD)
- Combined Free/Demand Assigned Multiple Access with Piggy Backing – Best Signal to Noise Ratio First with Dynamic Deadlines (CF/DAMA-PB-BSNRF+DD)

1.5 Original Contributions

The first contribution made in this thesis is the adaptation of the CF/DAMA-PB protocol to the DVB-RCS system. The adaptation to an MF-TDMA based air interface was needed as a result of DVB-RCS being based on the MF-TDMA air interface while the original CF/DAMA-PB protocol was based on TDMA.

Cross-layer design was implemented by taking into account the interaction between the MAC scheme and the information that can be obtained from the physical layer. This was used in further system improvement in either average packet delay or system throughput.

The idea of dynamic deadlines according to the SNR states of the terminals, RCSTs, was taken advantage of in order to reduce packet dropping while still benefiting from the average packet delay that traffic time deadlines offer.

The following protocols were proposed and their performance evaluated in this work:

- CF/DAMA-PB-PD
- CF/DAMA-PB-PEDF
- CF/DAMA-PB-BSNRF
- CF/DAMA-PB-DD
- CF/DAMA-PB-BSNRF+DD

These protocols are evaluated under multimedia, non homogeneous, traffic.

1.6 Summary

This chapter reveals the many ways in which a satellite network may be classified and setup. Satellites are characterised by their architecture, orbit, network type, medium access technique, frequency considerations, performance parameters, air interface standardization and the various MAC protocol classifications.

Each method or technique was designed for particular system requirements and hence were defined by the requirements. These requirements could vary from user QoS, system utilization, cost of deployment of network, legal concerns in terms of standards that needed to be adhered to, etc.

The orbit used decides on the cost of the network and the complexity it offers. This also defines the traffic types that may be possible in that network as a result of the delay differences between orbits.

There are various classifications of the MAC protocols, these being Fixed Assignment, BoD, Free Assignment, and Random Access among others.

The main conclusion that can be drawn from this chapter is there are many ways to implement a satellite network, but the design problem comes in deciding what method is best for the task at hand and whether costs may have to be considered and to what extent.

2 DVB RCS

2.1 Introduction

This chapter is concerned with the DVB-RCS standard and its relative specifications as depicted in Figure 11.

The DVB-RCS standard specification defines the physical characteristics and the medium access characteristics needed to implement an Interactive Satellite Network (ISN). The need for the DVB-RCS standard arose when the terrestrial return channel did not supply a high enough rate for the growing demand of multimedia communications. This was due to a lack of terrestrial infrastructure or a terrestrial infrastructure without the necessary capabilities. An interactive satellite return link proved to be able to manage the demand of new multimedia traffic while not requiring a huge outlay of costly terrestrial networks.

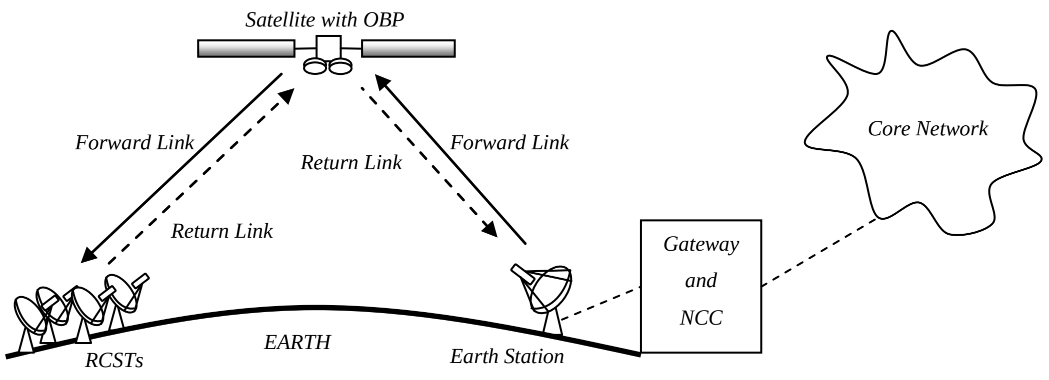

Figure 11: DVB-S/DVB-RCS system architecture example.

Examples of the frequency bands used in DVB-RCS Terminals (RCSTs) are, [1]:

Table 2: DVB-RCS satellite reception and transmission band examples.

Reception Bands	*Transmission Bands*
10.70 GHz-11.70 GHz	*14.00 GHz-14.25 GHz*
11.70 GHz-12.50 GHz	*27.50 GHz-29.50 GHz*
12.50 GHz-12.75 GHz	*29.50 GHz-30.00 GHz*
17.70 GHz-19.70 GHz	
19.70 GHz-20.20 GHz	
21.40 GHz-22.00 GHz	

The reception bands are those from the FSS and BSS allocations, while the transmission bands are from the FSS allocation.

This chapter is organised as follows: Section 2.2 gives a description of the reference models used in the DVB-RCS standard; Section 2.3 briefly describes the forward link of the DVB standard; Section 2.4 gives the details of the DVB-RCS return link; Section 2.5 describes the multiple access techniques used in the DVB-RCS standard and how data is encapsulated using the Multiprotocol Encapsulation (MPE); and in Section 2.6 a conclusion is given.

2.2 DVB Reference Models

The DVB-RCS has the ability to make use of regenerative satellite architectures with onboard switching, without onboard switching and bent-pipe or a combination of these.

The functional requirements of the OBP are, [1]:

- The ability to receive all RCST control data and signals.
- The ability to receive all NCC control data and signals.
- The ability to extract from the correct format (DVB-S or DVB-S2) and then route the data, meant for the downlink, to the appropriate output(s).
- The ability to generate/extract the appropriate control data intended for the NCC and appropriately route to the output.

- The ability to correctly format the downlink streams with all necessary signalling messages.

The reference models include the Protocol Stack, the System Model, the Satellite Interactive Network and Dynamic Connectivity.

2.2.1 Protocol Stack

The DVB-RCS standard implements a simplified version of the Open Systems Interconnection (OSI) stack for interactive services as depicted in Figure 12. This protocol stack defines two broad categories of protocols, these being the network dependent and the network independent sets of protocols. This ensures that the satellite implementation does not limit the integration of the system into a network. A sub-layer of the network dependent category consists of the MAC protocols, packet structure, etc. This involves the organisation of data in terms of packets and how the physical interface, the air interface, is shared amongst multiple users and their multiple data streams to ensure efficient use of the system capacity. The other sub-layer of the network dependent category consists of the protocols which are related with the physical layer operations, these ensure that the signal is transmitted in a secure, safe, and reliable manner by making use of synchronization, modulation techniques, channel coding techniques, etc. The network independent layers involves the protocols similar to those in the OSI stack from the Network Layer and up, these establish connections, ensure correct presentation of data, etc.

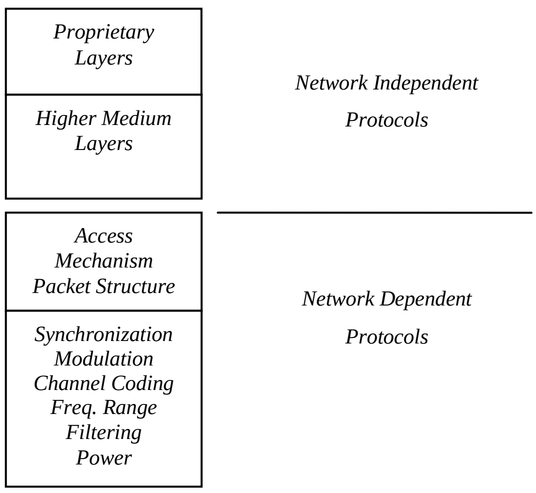

Figure 12: Layer structure for generic system reference model.

2.2.2 System Model

In the system model of the DVB-RCS interactive services, there are two channels that are established:

- Broadcast channel
- Interactive channel

The broadcast channel is a unidirectional channel set up between the service provider and the end users. It may include the Forward Interaction Channel. The interactive channel consists of a bidirectional channel between the service provider and the user or between users. It is made up of a Forward Interaction path and a Return Interaction path. The Forward Interaction path is used to deliver information from the service provider to a user or to many users. It may be implemented in the forward broadcast channel or simply not used depending on the requirements of the system. The Return Interaction channel is the link from the user to the service provider and is mainly used for requesting data or services from the service provider.

2.2.3 Reference Model of the Satellite Interactive Network

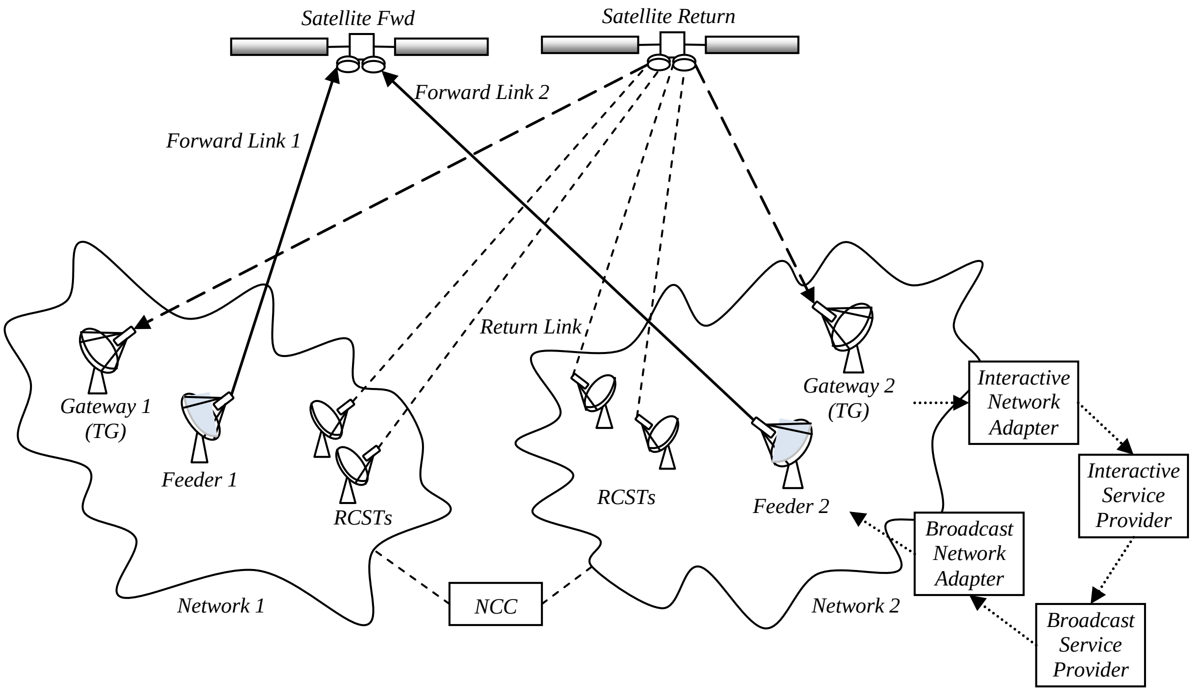

Figure 13: Reference Model for the Satellite Interactive Network.

The network model consists of three functional blocks depicted in Figure 13, these are:

- The Network Control Centre (NCC).
- The Traffic Gateway (TG).
- The Feeder.

The NCC provides the Control and Monitoring Functionality (CMF) of the interactive channels, as well as producing the timing and control signalling that the Feeder stations transmit. The Traffic Gateway is the gateway that receives all the RCST requests and connects them to the external service providers and networks. The Feeder is responsible for transmitting the forward broadcasting path, which may be a DVB-S or DVB-S2 channel.

2.2.4 Dynamic Connectivity

DVB-RCS has taken dynamic connectivity into consideration. Dynamic connectivity enables the system or system administrator to establish, modify and release a link with a hub or RCST. The system would require such ability depending on the traffic events of the links and to ensure proper management and control of the system. Dynamic connectivity is used to build additional reference models that exceed the reference model.

2.3 Forward Link

The forward link can be the DVB-S, the original broadcasting standard used in DVB-RCS interactive channels, or more recently the DVB-S2 (ETSI EN 302 307) broadcasting standard. The DVB-S2 and DVB-RCS standards were only combined and finalised in 2009.

The forward link is used to respond to the requests made by the user via the RCS interactive channel. It provides the users with their requested data and services. The DVB-S and DVB-S2 satellite broadcasting standards are further described in ETSI's standard documentation EN 300 421, [6], and EN 302 307, [8], respectively.

2.4 Return Link

The return link is defined in the ETSI EN 301 790. The generic digital signal processing of the RCST's transmission, which is performed at the RCST transmitter side, is divided into the following functional blocks depicted in Figure 14, [7]:

- Burst formatting.
- Energy dispersal.
- Channel coding.
- Modulation.
- Synchronization.

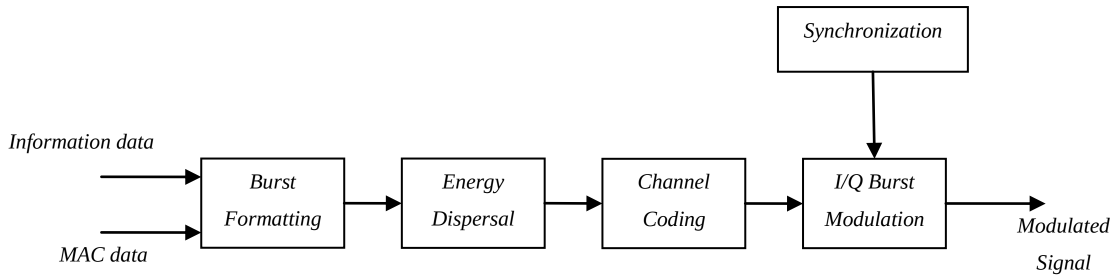

Figure 14: Block diagram of the RCST return link baseband signal processing, [7].

2.4.1 RCST Synchronization

Synchronization is a crucial feature in TDMA based systems such as DVB-RCS, as synchronization affects the throughput efficiency of the system. DVB-RCS has added a few measures to ensure that the system is correctly synchronized.

DVB-RCS uses the information in the Network Clock Reference (NCR) and the signalling in the DVB/MPEG2-TS private sections to implement timing control. The forward link signalling contains the NCR information to synchronize the RCSTs with the NCC carrier and this achieves a normalised carrier frequency accuracy of better than 10^{-8}. Bursts are sent according to a Burst Time Plan (BTP) which is synchronised using the signalling information from the forward link.

2.4.2 Burst Format

There are four types of bursts, [7]:

- Traffic.
- Acquisition.
- Synchronization.
- Common Signalling Channel.

The traffic bursts are used to transmit data from the user to a gateway or a user. There are two types of traffic modes defined in the DVB-RCS standard; the Asynchronous Transfer Mode (ATM) and the MPEG2-TS, optional, mode.

ATM Traffic Burst

The ATM traffic burst consists of N_{atm} 53 byte ATM cells as its payload, an optional prefix and a preamble as depicted in Figure 15.

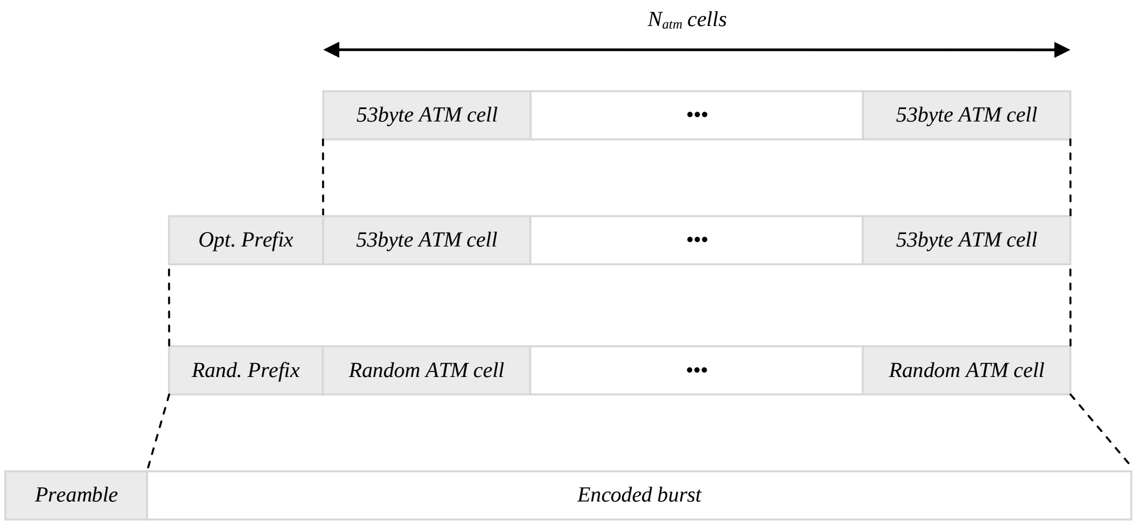

Figure 15: Composition of an ATM traffic burst, [7].

Optional MPEG2-TS Traffic Burst

The MPEG2-TS traffic burst contains several channel coding blocks. The basic containers of this traffic burst is the MPEG2-TS packets, 188 bytes in length and N_{mpeg} concatenated packets make up the bursts depicted in Figure 16.

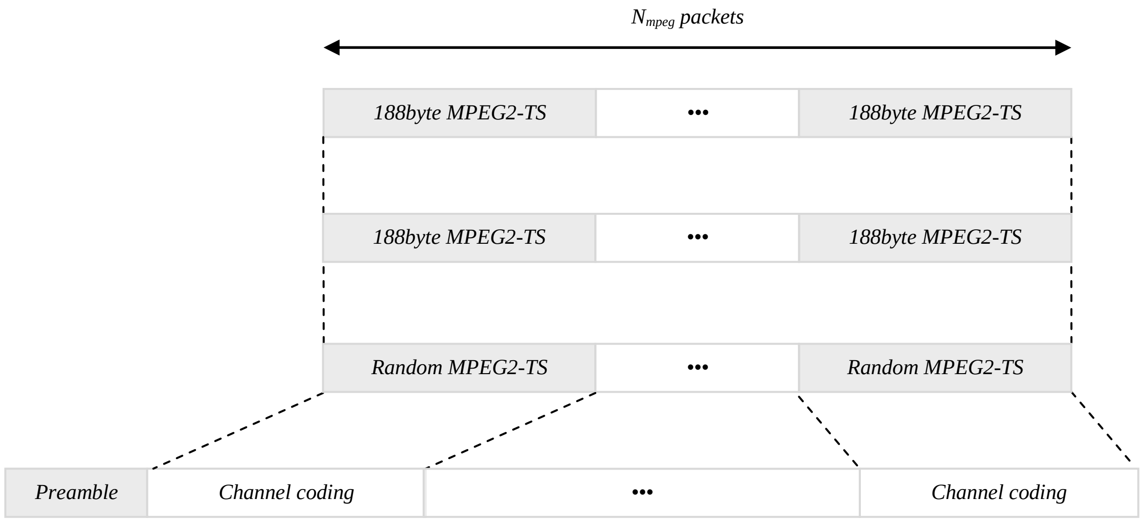

Figure 16: Composition of an MPEG-2TS traffic burst, [7].

The synchronization and acquisition bursts are defined to ensure the traffic bursts are accurately positioned. The synchronization burst ensures the burst is synchronized and is also used to send control signals to the system. The acquisition burst ensures that an RCST is able to synchronize itself with the system prior to its operation.

The Common Signalling Channel (CSC) burst is used to initiate the login procedure and return link synchronization of a RCST. It is also used to indicate the terminal beam location of mobile devices while they are not logged in.

2.4.3 Channel Coding

DVB-RCS uses both Turbo coding, as well as concatenated coding schemes. The concatenated coding schemes used are by-passable Reed-Solomon (RS) code for the outer code and by-passable non-systematic convolution code for the inner code. A by-passable Cyclic Redundancy Check (CRC) may be concatenated on the CSC and synchronization bursts for error detection purposes.

The CRC used is that of the CRC-16 with the CRC polynomial being $x^{16}+x^{15}+x^{2}+1$. For further information on the RS and convolution coding implementation see ETSI EN 300 421, [6]. The Turbo encoder makes use of the double binary Circular Recursive Systematic Convolutional (CRSC) code, [7].

2.4.4 Modulation

QPSK is used to modulate the signal. Baseband shaping is also used except where spreading spectrum is concerned in the mobile implementation. Each of the bursts is surrounded by guard times to accommodate for RCST power switch-off transients and any system timing errors that may occur. For the MPEG traffic burst the guard time has been defined to be less than half of an MPEG2-TS packet length.

2.5 Multiple Access

The DVB-RCS makes use of the DAMA schemes to efficiently manage the allocation of resources, one such possible scheme is the basis of this work, namely the Combine Free/Demand Assigned Multiple Access (CF/DAMA). The RCST is responsible for determining the capacity needed and then requesting the capacity from the NCC. The NCC is then responsible for allocating the requests and informing the RCSTs of the allocation via a Terminal Burst Time Plan (TBTP) over the forward channel. The RCSTs examine the TBTP and then only use the slots allocated to them if required.

2.5.1 Fixed MF-TDMA

Fixed Multiple Frequency – Time Division Multiple Access (MF-TDMA) is the default option used in the DVB-RCS standard. The width of each frequency (f_{ci}) and the length of each TDMA slot (s_i) are fixed throughout the execution of the system. This is depicted in Figure 17.

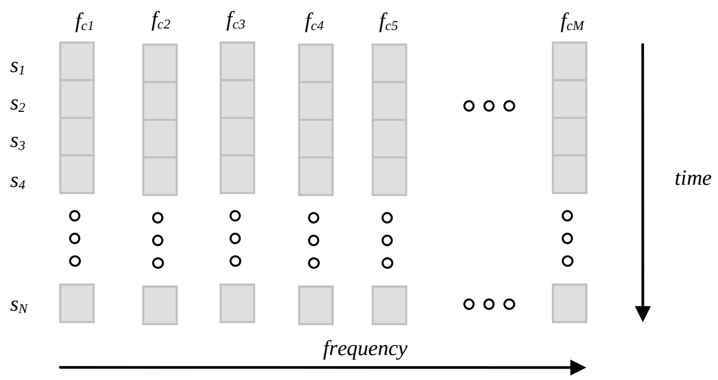

Figure 17: MF-TDMA structure.

The fixed MF-TDMA access technique is the technique considered in this work.

2.5.2 Dynamic MF-TDMA

Dynamic MF-TDMA is an option in the DVB-RCS standard. This allows the MF-TDMA frame structure to vary in terms of the bandwidth of the carrier frequencies (f_{ci}), as well as the length of each slot (s_i) over time.

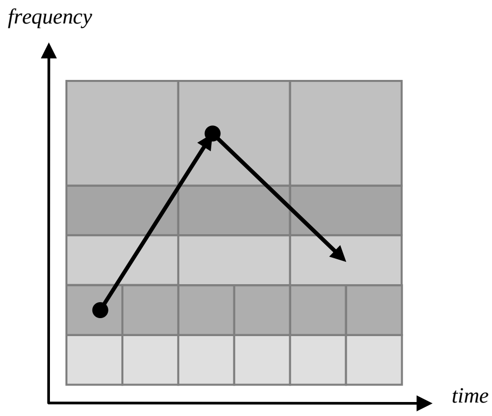

Figure 18: Dynamic-slot MF-TDMA (Optional)

The Dynamic Slot MF-TDMA, as depicted in Figure 18 above, creates a more flexible channel by increasing or decreasing the number of slots and hence the sizes of each slot as per system requirements. The slots need to be managed in a logical manner, multiples of a minimum slot size, to ensure synchronization and efficient usage of the frame time, which is preferably fixed. The ability to change a slot size helps in varying channel conditions and traffic packet sizes but adds a significant level of complexity to the system. If the implementation is not optimized, this method can significantly degrade system performance. Similar to the way in which slots are varied, the size of the frequency bands may also be varied in a dynamic MF-TDMA environment.

2.5.3 Resource Organisation of MF-TDMA

The frequency and time domain is, at the highest level, divided into super-frames, each having a super-frame_id assigned to it. These super-frames, characterized by superframe_counter numbers, are further sub-divided into frames, once again each having their own frame_number or frame_id associated with it. The frame is further subdivided into the time slots, each having a timeslot_number assigned to them. Each level is able to vary in time and in frequency or have a fixed time and frequency depending on which implementation is used. This is illustrated in Figure 19 below.

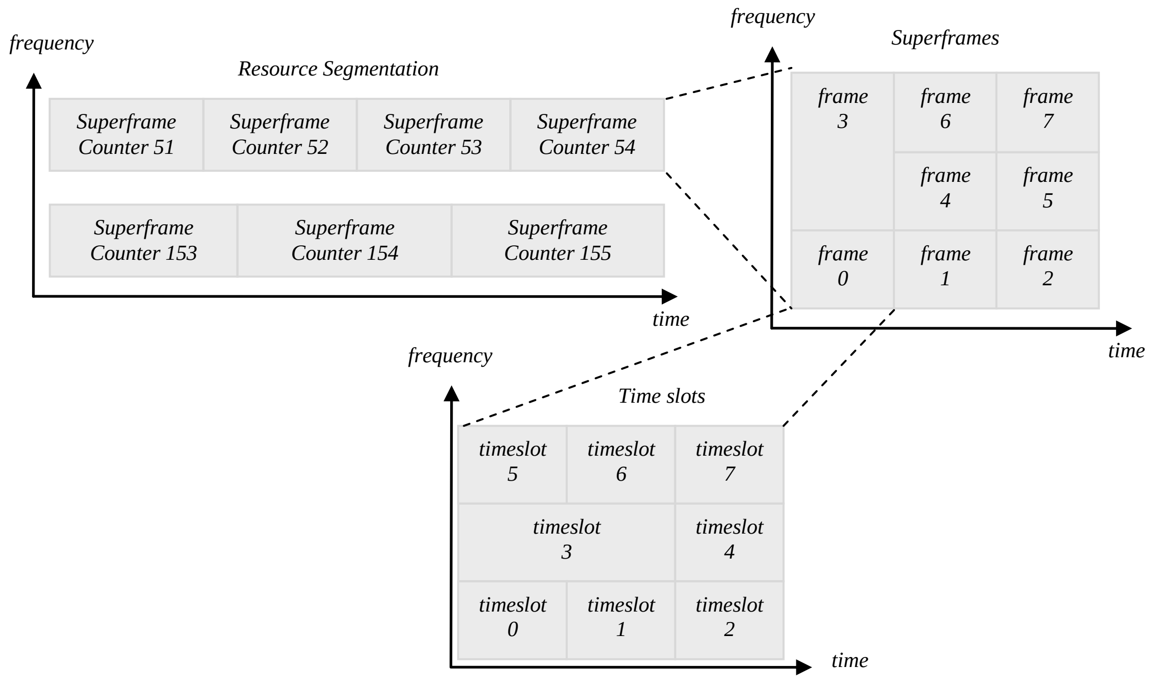

Figure 19: MF-TDMA resource organization, [2], [7].

2.5.4 Capacity Request Categories

DVB-RCS takes into account that traffic has varying priority and hence has defined the following five capacity categories:

- Continuous Rate Assignment (CRA)
- Rate Based Dynamic Capacity (RBDC)
- Volume Based Dynamic Capacity (VBDC)
- Absolute Volume Based Dynamic Capacity (AVBDC)
- Free Capacity Assignment (FCA)

The CRA category is a fixed assignment technique, whilst the RBDC, VBDC and the AVBDC are variable assignment categories. The FCA is an assignment category which assigns the unused capacity of the system, the “bonus” capacity assignment. The categories have a priority order; CRA, being the highest, RBDC, VBDC and AVBDC have equal priority and finally FCA being the least priority category.

CRA

The CRA is a static rate capacity assigned in full at the start of each super-frame and agreed upon at network setup. The setup phase requires the negotiation between the RCST and the NCC. The fixed assignment of slots per super-frame is released once the RCST sends a release message to the NCC when it no longer requires the capacity. This kind of assignment is usually used for subscription based services. It is used for QoS guarantees, as it is easy to ensure that the user has the agreed upon capacity. VoIP is an example of a service that would benefit from this category of capacity assignment.

RBDC

This capacity assignment category is a dynamic resource assignment method that is based on the traffic rate. It is dynamically requested by the RCST and each request made overrides the previous request. The rate requested is governed by an upper limit defined by the system. If no updated request is received by the NCC within a set period, the rate for the RCST in question is then terminated and reset to zero, this period is set to two super-frames as default.

VBDC

This category is assigned according to the traffic volume of the RCSTs and is also requested by the RCSTs. The request made is that of the number of slots needed over several super-frames. Each successive VBDC request adds to the previous request. Due to the nature of this category allowing for the requested slots to be over several super-frames, this category is not suited for jitter intolerant traffic types that require real time guarantees. Therefore a good use of this category would be for FTP based communications.

AVBDC

This category is similar to the VBDC in most aspects, but instead of accumulating the requests, the most recent request replaces the previous requests as it is an absolute value. This category is suitable to support the traffic classes that VBDC supports.

FCA

The FCA category is the only category, of the five, that is an automatic capacity assignment. This is assigned in a volume based manner, i.e. number of slots, to RCSTs. The RCST does not request this capacity but rather is allocated the capacity as an extra capacity that would otherwise remain unused. This gives the RCST the ability to use extra slot assignments if required. The assignment of this unrequested capacity is based on either the performance optimization of TCP/IP to reduce timeouts or in equity fashion, i.e. ensuring each RCST obtains a fair share of this capacity one could apply a round-robin assignment or something similar.

DVB-RCS has a round trip delay determined by the time to request and receive an acknowledgement of the request. This can be one hop, two hops or multiple hops, depending on whether the satellite is bent-pipe, has an OBP with NCC, or where the NCC is located relevant to the RCST. In general the timing between the RCST and NCC is as depicted in Figure 20.

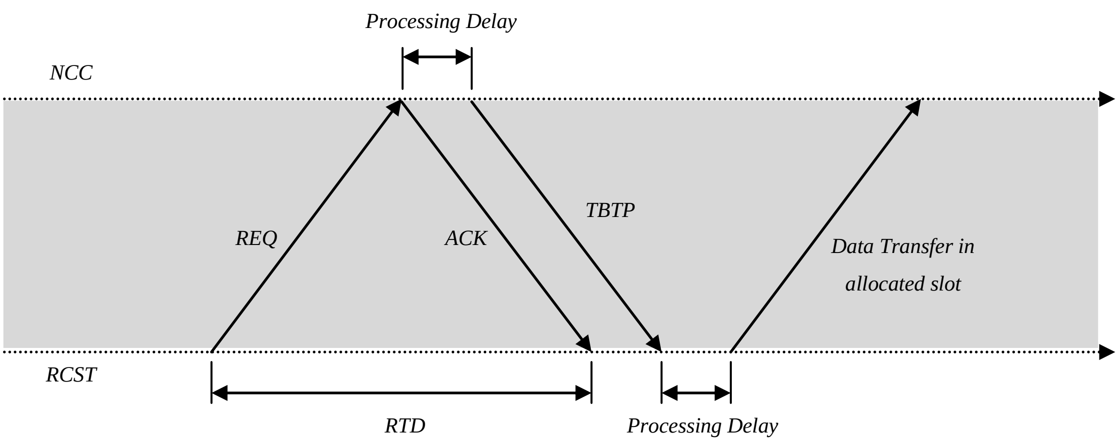

Figure 20: Time delays in requesting and allocating resources to terminals.

The method in which an RCST receives, channelizes, queues and makes the necessary capacity requests is depicted in Figure 21.

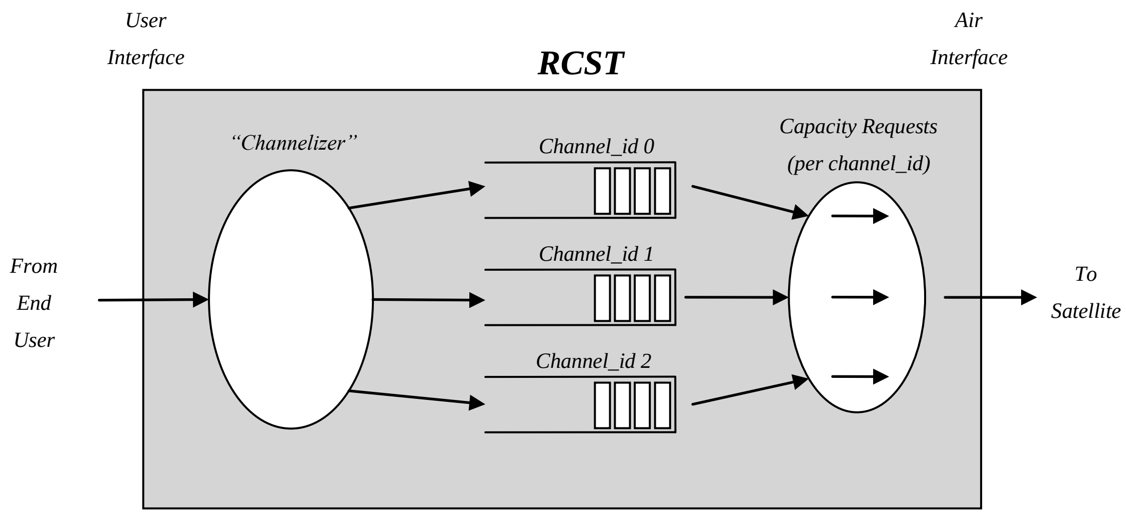

Figure 21: Functional Representation of Channel_id Usage, [1].

2.5.5 Multiprotocol Encapsulation

Multiprotocol encapsulation (MPE) is a protocol that defines the encapsulation of packet oriented protocols (i.e. IP, UDP, etc.) onto a MPEG transport stream (TS). The protocol specifications were defined by DVB and included in the ETSI EN 301 192 publication. The MPE of a transport stream is performed at the RCST for return link data transfers.

The MPE for IP traffic is depicted in Figure 22. An IP packet is checked for its size, if the size of the IP packet exceeds the MPEG2-TS section size of 4080 bytes, then it is fragmented into numbered sections. The sections are then each fragmented into MPEG2 packets if their lengths exceed the MPEG2 packet payload size of 183 bytes (184 if the pointer field is not required). An MPE header is attached to each section with a CRC for error detection purposes. The MPEG2 packets have the MPEG2-TS header attached to them and then each packet is concatenated to form a traffic burst or transport stream.

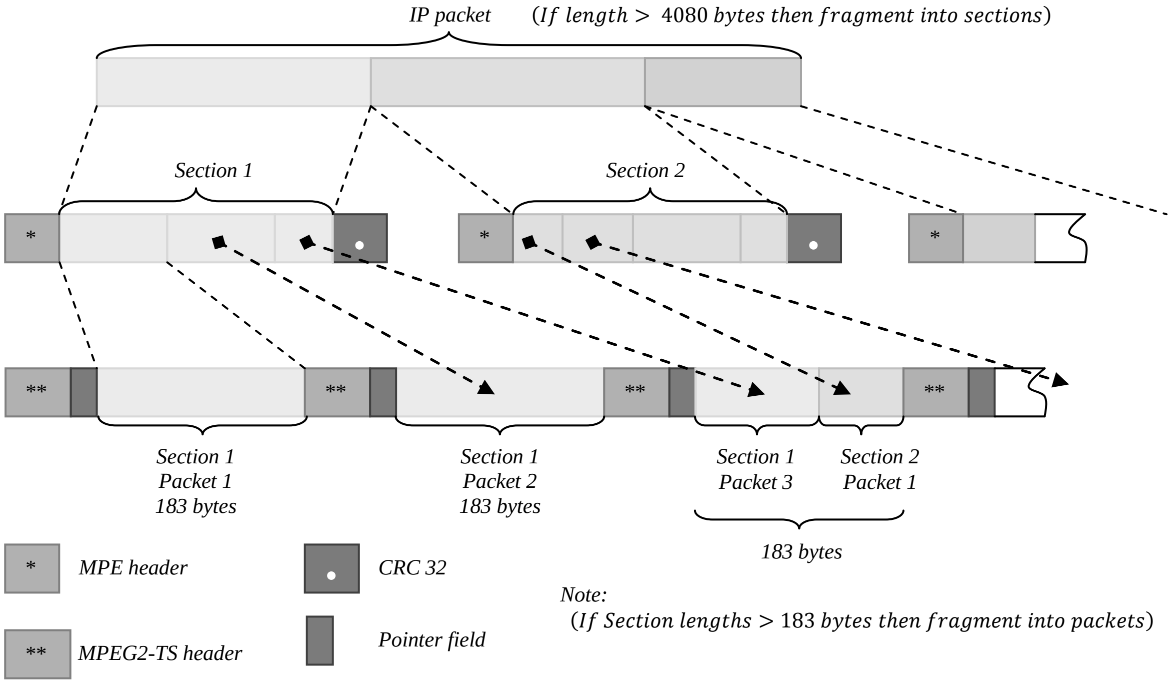

Figure 22: MPE encapsulation for IP traffic.

2.6 Conclusions

This chapter discussed the details of the DVB-RCS standard affecting the MAC algorithms. The ability to have an architecture that supports OBP and hence reduce the round trip delay of the system is beneficial to the overall packet delay performance, especially in the time sensitive traffic classes such as voice and video.

The TDMA-based MF-TDMA multiple access technique allows for the adaptation of the CF/DAMA-PB protocol, used in this work, to that of a GEO satellite system.

The coding, Turbo coding and concatenated coding, and modulation, QPSK, offer a better level of accuracy in terms of successful packet transmission than that of a non-coded system as claimed in EN 302 307 [8]. This helps in reducing the effects of the channel losses.

The MPE was also taken into account, which allows for several differing types of traffic to be encapsulated, using MPE, and transmitted in the system without having to take the different traffic types into account in the MAC scheme on an individual basis which would increase system complexity significantly.

3 Enhanced MAC Schemes for DVB-RCS

3.1 Introduction

This chapter introduces two new MAC protocols for DVB-RCS and explains how they function.

In Section 3.2 a survey of some of the existing MAC protocols is presented placing greater focus on the CF/DAMA-PB protocol from which the proposed protocols are derived. Section 3.3 gives a detailed description of the proposed CF/DAMA-PB-PD MAC protocol, and Section 3.4 presents a detailed description of the CF/DAMA-PB-PEDF MAC protocol. The simulation implementation and structuring is briefly discussed in Section 3.5. Section 3.6 gives the three traffic models used to create the multimedia traffic sources required for simulation. In Section 3.7 the various performance parameters used in comparing the protocols are discussed. Section 3.8 details the simulation parameters used in obtaining the results of Section 3.9. The chapter conclusion is given in Section 3.10.

3.2 Satellite MAC protocols

Satellite Medium Access Control (SMAC) protocols that are capable of supporting multimedia traffic were of interest in this work. MAC schemes that were designed for terrestrial wireless networks have very limited applicability to the satellite systems as their propagation delays are substantially smaller than that of satellite communication networks, especially the GEO satellites, which was the focus of this work. The various schemes that will be mentioned will not be discussed in detail in terms of their performance parameters and analysis as there are many varying satellite network parameters and scenarios. There are four broad categories of SMAC protocols [24]:

- Combined Random Access and DAMA schemes
- Combined Fixed Assignment and DAMA schemes
- Combined Free Assignment and DAMA schemes
- Predictive schemes

Some of the MAC protocols mentioned in the literature are, [24]:

- Announced Retransmission Random Access (ARRA), [25]
- Generalised Retransmission Announcement Protocol (GRAP), [26]
- Scheduled Retransmission Multiple Access (SRMA), [27]
- Controlled Multi-access Protocol, [28], [29]
- Combined Random/Reservation Multiple Access (CRRMA), [30], [31]
- Transmit Before Assignment using Collision Requests (TBACR), [32], [33]
- Integrated Access Scheme, [34]
- Movable Boundary Random/Demand Assignment Multiple Access (MB/R-DAMA), [35]
- Random-Reservation Adaptive Assignment (RRAA), [36]-[38]
- Dynamic Random-Reservation Adaptive Assignment (D-RRAA), [39]
- Interleaved Frame Flush-Out (IFFO), [40]
- Packet Reservation Multiple Access with Hindering States (PRMA-HS), [41]-[45]
- Combined Free/Demand Assignment Multiple Access (CF/DAMA), [21]-[23]
- Combined Fixed/Demand Assignment Multiple Access, [46], [47]
- Response Initiated Multiple Access (RIMA), [48]

- Predictive Demand Assignment Multiple Access (PDAMA), [49], [50]

This section gives a summary of some of the above protocols.

3.2.1 ARRA

In ARRA [25], each message packet has a control mini-slot (using at most 2% of the data frame) to announce the next frame slot position the terminal intends to use for retransmission if the packet collides. All users monitor and keep a list of the message packet collisions to determine which slot to use in the following frame. Packets generated in the current frame are held until the terminal can determine the permitted set of slots for the following frame. The retransmissions of the collided packets are transmitted first, then, at random, the new packets are transmitted. If there are no slots available for the newly transmitted packets, then there is a common mini-slot pool to make the retransmission announcements. This protocol ensures that there are no collisions between the retransmitted and newly generated packets. This protocol ensures the simplicity of a random access protocol while at the same time ensuring no conflicts. Random access protocols still lack in their ability to fully utilise the system capacity while reducing the average packet delay. In GEO-satellite communications, the large RTD would not be suited to a random access protocol.

GRAP takes this protocol a bit further and adds a NCC to broadcast the available slots for new packets on a frame by frame basis.

3.2.2 SRMA

In this TDMA Aloha based protocol [27], [56], the return link is divided into reserved and random access sub-frames. The new packets arriving at the random access sub-frame are transmitted in the following slot. The packets that arrive at the reservation sub-frame are transmitted in the next random access sub-frame slot. Mini-slots associated with the random access data slots are used, at random, to announce retransmissions. These announcements are reservation requests that are only allocated in the case of a collision. Both slots reservation information and ACKs are produced and sent to the users, hence if a user fails to receive an ACK for a packet the SRMA scheme starts to look for an allocated reservation slot. In the case that the reservation slot also had a collision, the user will wait a random time before retransmitting the packet on a random access slot.

Below, in Figure 23, the SRMA Fixed Frame (SRMA/FF) protocol flowchart is presented; another variation is the SRMA Dynamic Frame (SRMA/DF) protocol.

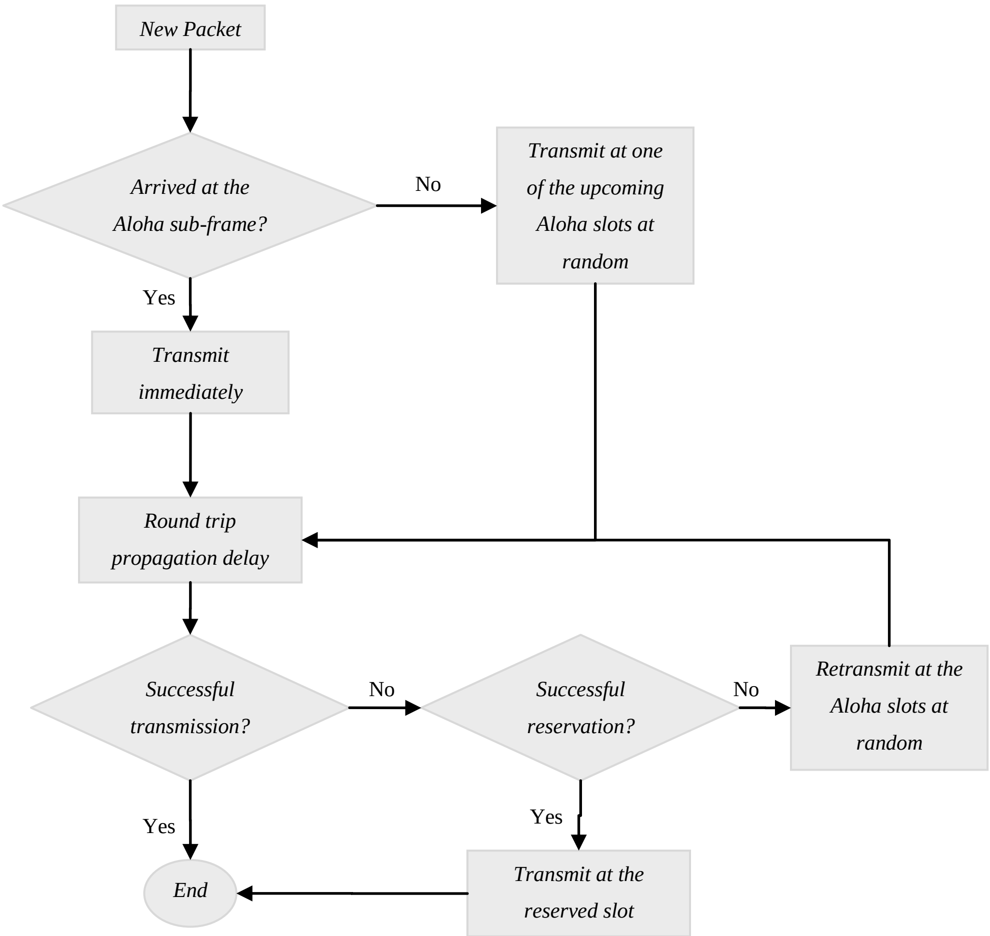

Figure 23: SRMA/FF protocol algorithm, [56].

The main disadvantage of this protocol is the low throughput characteristics as is seen in [56] on the performance of the SRMA protocol. The maximum throughput achieved at an offered load of 1.0 was a normalised throughput of 0.65for SRMA/FF as illustrated in [56].

3.2.3 PRMA

In this protocol [41]-[45], the return link is divided into frames which are further subdivided into equal data slots without any overhead capacity for capacity reservation requests. The length of the slot and frame are chosen to be suitable for the traffic type, i.e. voice. Initially, terminals randomly transmit their packet in an unreserved slot. If a collision occurs, then all collided packets will be retransmitted at random in a future unreserved slot according to a permission probability. If a terminal obtains a successful transmission the slot is reserved for that terminal in successive frames. To release the slot reservation the terminal leaves the slot empty.

3.2.4 PDAMA

This set of protocols, [49] and [50], tries to estimate the future resource requirements of each terminal by monitoring the traffic. This scheme tries to improve on the CF/DAMA scheme by predictive allocation of resources. Free assigned slots are assigned to terminals more likely to require them instead of using a round-robin strategy.

The disadvantage of this kind of protocol is that the nature of traffic that enters a network does not always arrive at the expected point and hence a prediction adds an extra level of complexity to the system in order to more accurately predict the system requirements. Many of these protocol schemes tend to work well with a specific arrival process instead of across the board, thus not being desirable under unpredictable traffic loads or heterogeneous traffic scenarios.

3.2.5 CF/DAMA-PB

The Combined Free/Demand Assignment Multiple Access protocols (CF/DAMA) were proposed by Le-Gnoc and J. I. Mohammed, [21], [22]. The Combined Free/Demand Assignment Multiple Access – Piggy-Backed (CF/DAMA) was a refinement in the way requests were made in the CF/DAMA set of protocols, [23].

The CF/DAMA protocol algorithm will be explained in detail since it forms the basis of the proposed protocols in this work.

CF/DAMA essentially combines the benefits of both the Free Assignment technique and the Demand Assignment technique. The protocol ensures a short delay at the low to medium end of traffic space while maintaining a high channel throughput offered by that of the DAMA technique. CF/DAMA-PB falls into the contention-free category of MAC protocols.

Piggy-Backed request strategy enables the system to add a request to a packets header so as to reduce the overhead required if one used a whole packet for requesting system capacity.

There are three different request schemes that Le-Ngoc described for the CF/DAMA protocol:

- CF/DAMA with Fixed Assignment (CF/DAMA-FA)
- CF/DAMA with Random Access (CF/DAMA-RA)
- CF/DAMA with Piggy-Backed requests (CF/DAMA-PB)

All the schemes have the same scheduling algorithm with differing ways for a terminal to make a request.

The CF/DAMA-PB protocol has a centralised scheduling algorithm and a terminal requesting algorithm at each terminal. The scheduling algorithm can be situated at the satellite or an NCC on the earth. The implementation used is that of the scheduling algorithm at the satellite, thus reducing the number of hops to be made in order for a request to be allocated. In this case a minimum of one satellite hop is required to request channel capacity and have the request acknowledged.

The scheduling algorithm is depicted in Figure 24. The scheduler has two main tables or queues, the Reservation Request Queue (RRQ) and the Free Assignment Queue (FAQ). The FAQ is populated with the addresses of each of the terminals connected to the satellite. The RRQ is used to hold the requests made by the earth terminals or RCSTs. The RRQ holds the source address of the RCST that made the request, as well as the number of slots requested by the

RCST. Each received request is placed into the RRQ in a FIFO fashion and served in a FIFO fashion. The scheduler allocates the slots of the TDMA on a frame-by-frame basis and then sends the assignment to the RCST using a packet on the downlink frame. The number of slots in a TDMA frame is equal to the number of RCSTs, effectively giving each a turn to initially request resource capacity. In the MF-TDMA version of this work, the number of slots shared among all the frames in a frame time period is equal to the number of RCSTs. In the case of the DVB-RCS, the Return Link is MF-TDMA and thus requires refinement of the TDMA structure that CF/DAMA protocols are developed for to that of a MF-TDMA structure.

The scheduler first queries the RRQ to determine whether there are any requests made. If the RRQ is not empty then the scheduler assigns a slot in a frame to the RCST request at the head of the queue and subtracts one from the number of requested slots of the RCST in question. This is repeated, if the RCST request at the head of the RRQ becomes zero, the RCST is removed from the Head of the queue and the corresponding FAQ entry is moved to the tail end of the FAQ. The slot allocation is repeated until the frame is full, at which point it waits to allocate the following frames slots, or if the RRQ is empty then the slots are assigned to the RCST at the head of the FAQ. The RCST then will be removed from the head of the FAQ and placed at the tail end of the queue. The FAQ is serviced in a round-robin fashion ensuring that the free assignment is fairly allocated to the RCSTs.

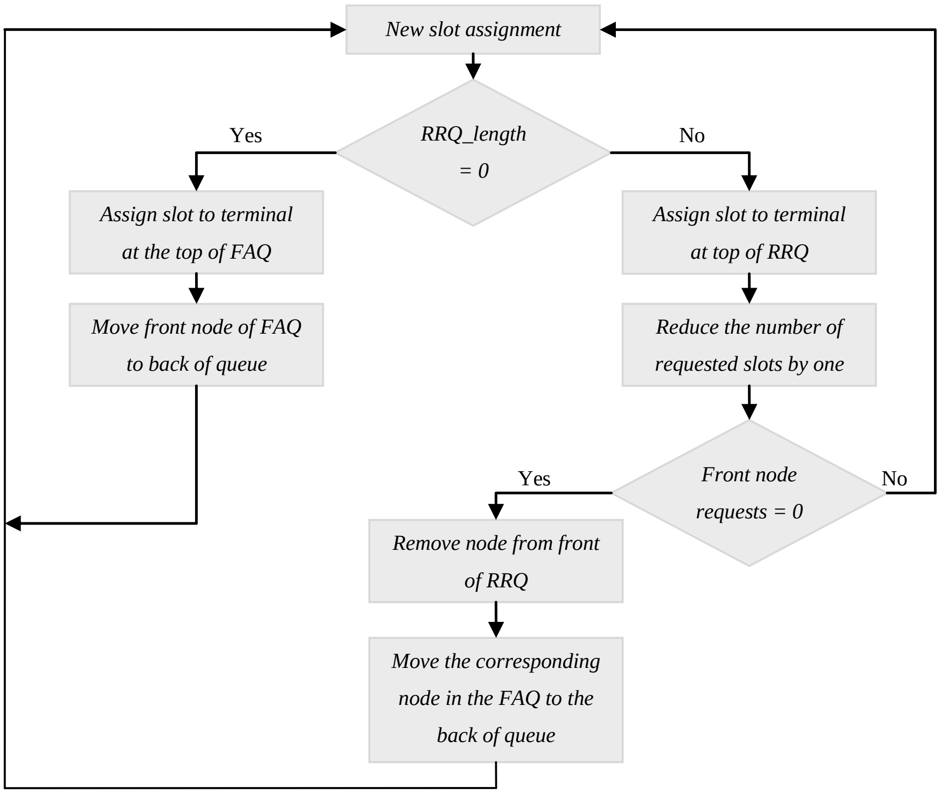

Figure 24: CFDAMA-PB scheduling algorithm, [20].

The piggy-backed request strategy used in CF/DAMA-PB-PD is depicted in Figure 25. The request is piggy-backed on the data packets being sent. The RCST may only make a request on a packet in a slot allocated to it. The format used to add a request to a packet in a TDMA frame is depicted below.

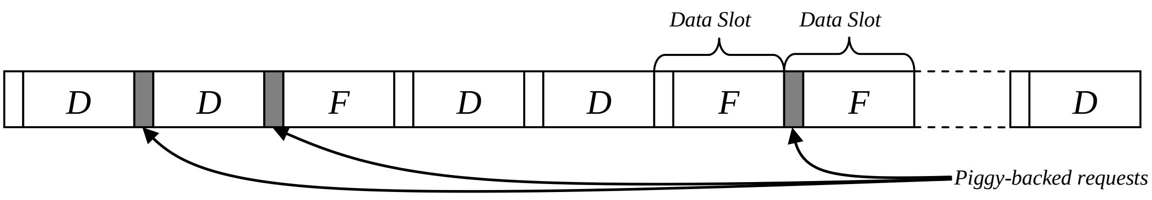

Figure 25: Return link piggy-backed frame format in TDMA.

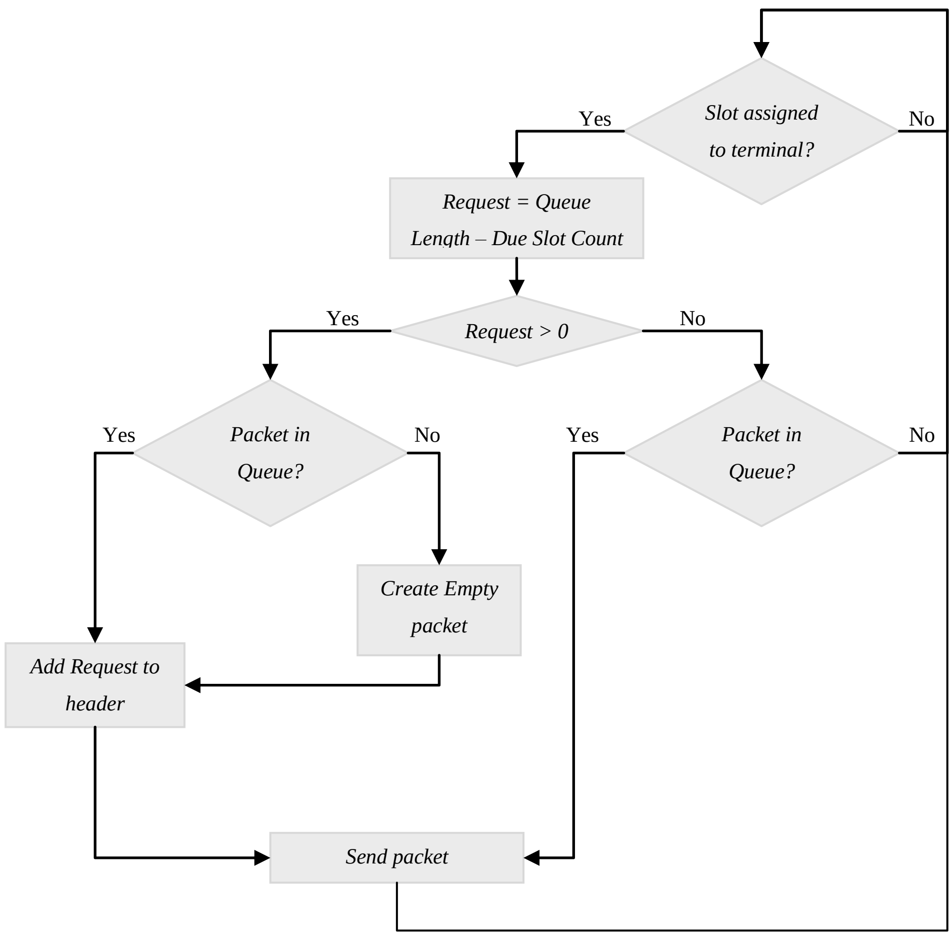

Figure 26: CF/DAMA-PB terminal algorithm.

The terminal algorithm for the RCSTs is illustrated in Figure 26 and discussed below.

The RCST waits for a slot assignment. Once a slot has been assigned to the RCST, either by free assignment or demand assignment, the RCST then calculates if it needs to request any additional slots for queued packets. If a request is necessary the RCST adds the request to the packet at the head of its traffic queue and sends the packet. If there are no packets in the queue and the RCST has a request and a slot assigned to it, then the RCST creates an empty packet, adds the request to that packet and sends the packet in the allocated slot. If the RCST does not have any additional requests to make, but has packets queued in the traffic queue, the packet at the head of the queue is sent in the allocated slot. On the instance of not receiving a slot for multiple frames, the request is accumulated until the RCST receives a slot allocation to send the request. This ensures that a request is made once per frame at maximum per RCST.

3.3 CF/DAMA-PB-PD

This section proposes a Combined Free/Demand Assigned Multiple Access with Piggy Backing and Packet Dropping (CF/DAMA-PB-PD) protocol which is derived from the CF/DAMA-PB derived protocol.

The CF/DAMA-PB-PD protocol has a centralised scheduling algorithm and a terminal requesting algorithm at each terminal. The scheduling algorithm can be situated at the satellite or an NCC on the earth. The implementation used is that of the scheduling algorithm at the satellite, thus reducing the number of hops to be made in order for a request to be allocated. In this case a minimum of one satellite hop is required to request channel capacity and have the request acknowledged.

The scheduling algorithm is depicted in Figure 27. The scheduler has two main tables or queues, the Reservation Request Queue (RRQ) and the Free Assignment Queue (FAQ). The FAQ is populated with the addresses of each of the terminals connected to the satellite. The RRQ is used to hold the requests made by the earth terminals or RCSTs. The RRQ holds the source address of the RCST that made the request, as well as the number of slots requested by the RCST. Each received request is placed into the RRQ in a FIFO fashion and served in a FIFO fashion. The scheduler allocates the slots of the TDMA on a frame-by-frame basis and then sends the assignment to the RCST using a packet on the downlink frame. The number of slots in a TDMA frame is equal to the number of RCSTs, effectively giving each a turn to initially request resource capacity. In the MF-TDMA version of this work, the number of slots shared among all the frames in a frame time period is equal to the number of RCSTs. In the case of the DVB-RCS, the Return Link is MF-TDMA and thus requires refinement of the TDMA structure that CF/DAMA protocols are developed for to that of a MF-TDMA structure.

The scheduler first queries the RRQ to determine whether there are any requests made. If the RRQ is not empty then the scheduler assigns a slot in a frame to the RCST request at the head of the queue and subtracts one from the number of requested slots of the RCST in question. This is repeated, if the RCST request at the head of the RRQ becomes zero, the RCST is removed from the head of the queue and the corresponding FAQ entry is moved to the tail end of the FAQ. The slot allocation is repeated until the frame is full, at which point it waits to allocate the following frames slots, or if the RRQ is empty then the slots are assigned to the RCST at the head of the FAQ. The RCST then will be removed from the head of the FAQ and placed at the tail end of the queue. The FAQ is serviced in a round-robin fashion ensuring that the free assignment is fairly allocated to the RCSTs.

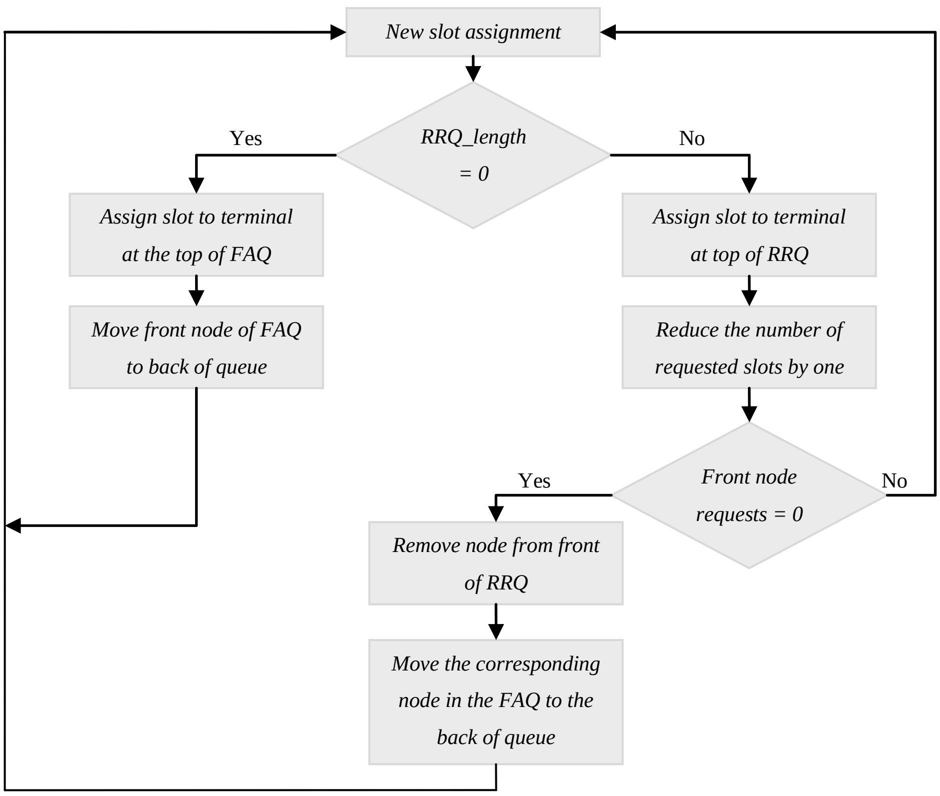

Figure 27: CFDAMA-PB-PD scheduling algorithm.

The piggy-backed request strategy used in CF/DAMA-PB-PD is the same as that of CF/DAMA-PB.

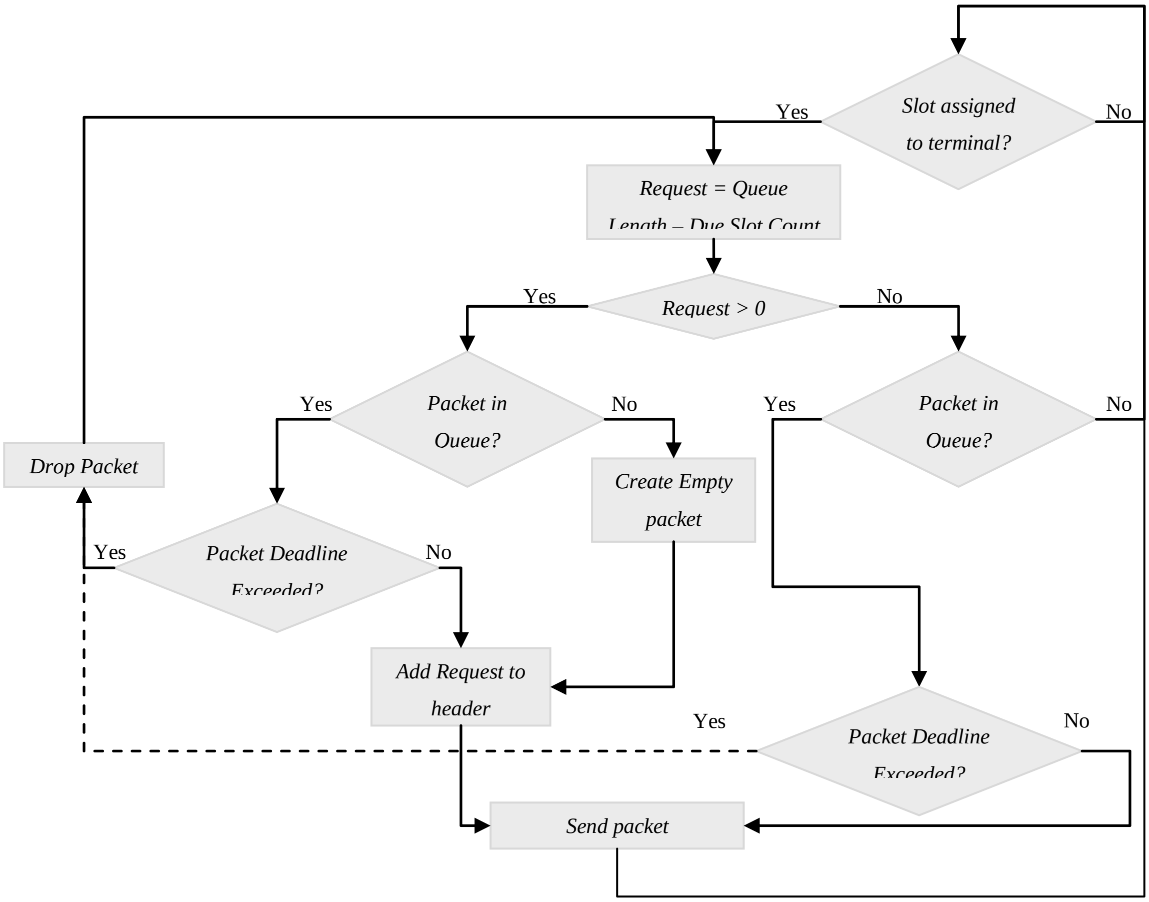

Figure 28: CF/DAMA-PB-PD terminal algorithm.

The terminal algorithm for the RCSTs is illustrated in Figure 28 and discussed below.

The RCST waits for a slot assignment. Once a slot has been assigned to the RCST, either by free assignment or demand assignment, the RCST then calculates if it needs to request any additional slots for queued packets. If a request is necessary the RCST determines if the packet at the head of the queue has exceeded its deadline. If the deadline has been exceeded for the packet in question, this packet could be a voice, video or web (virtual dropping) packet, the packet is dropped and one returns to calculating if a request needs to be made. If the deadline was not exceeded, the RCST adds the request to the packet at the head of its traffic queue and sends the packet. If there are no packets in the queue and the RCST has a request and a slot assigned to it, then the RCST creates an empty packet, adds the request to that packet and sends the packet in the allocated slot. If the RCST does not have any additional requests to make, but has packets queued in the traffic queue, the packet at the head of the queue is checked if it has exceeded the deadline and if not it is sent in the allocated slot. On the instance of not receiving a slot for multiple frames, the request is accumulated until the RCST receives a slot allocation to send the request. This ensures that a request is made once per frame at maximum per RCST.

With so many different types of traffic services being implemented in a telecommunications network, it is essential to be able to ensure a QoS for the traffic types. The aim of this protocol is to reduce the end-to-end delay of the time critical traffic classes, these being the voice and video traffic classes. In order to accomplish this, one is able to drop the late packets, prior to sending, that would have adversely increased the traffic class delay.

Traffic, such as voice and video, have a certain degree of tolerance when it comes to losing packets, as human beings are able to cope with a slight degradation of visual and vocal information. In data sensitive applications such as websites and FTP oriented connections, such as backing up of sensitive data, the accuracy of the data is far more important than that of the speed of the data. The existence of the various traffic classes and their requirements brings about a design issue that can be used to improve overall system performance.

The three traffic classes, being voice, video and web traffic, each have an associated deadline, T_{dvoice}, T_{dvideo} and T_{dweb} respectively. These deadlines are dictated by the traffic services that are being implemented in the network. Some of the types of traffic are [2],

- Conversational

- Interactive
- Streaming

A conversational traffic type requires stringent delay parameters and is less concerned with QoS. It is also concerned with preserving the temporal relation of information. The Interactive traffic type requires a low end-to-end delay, less stringent than that of the conversational class, as it is a request response interaction whereby a user requests something in a network and expects a response within a reasonable time period. The interactive class also demands a higher degree of accuracy, i.e. a low BER, in the information communicated. The final type, the streaming type, is far more concerned with the preservation of the temporal relation of the information than it is with the end-to-end delay. The end-to-end delay may be accounted for in a streaming type by using a sufficiently large buffer at the user end.

Examples of the conversational type of traffic are conversational voice such as VoIP, video calling such as that offered by Skype, Interactive gaming such as the Battle.net service, etc. Examples of the interactive type of traffic are voice messaging and dictation, Web browsing, etc. Examples of the streaming type of traffic are audio streaming such as radio, video streaming such as YouTube and still image transmission. The following table summarises traffic requirements in terms of end-to-end delay and their tolerance to information loss,

Table 3: Traffic types and classes with associated delay and loss tolerance.

Traffic Type	*Class*	*End-to-End Delay*		*Loss Tolerance*
		Preferred	*Limit*	
Conversational	*Voice*	*<150 ms*	*<400 ms*	*Medium*
	Video	*<150 ms*	*<400 ms*	*Medium*
	Web	*<250 ms*		*Zero*
Interactive	*Voice*	*<(1-2) s*		*Medium*
	Video	*<(1-2) s*		*Medium*
	Web	*< 4 s*		*Zero*
Streaming	*Voice*	*<10 s*		*Low*
	Video	*< 10 s*		*Low*
	Web	*< 10 s*		*Zero*

According to the system performance limitations, various types of traffic may or may not be suitable for satellite communications. The GEO satellite has a lower end-to-end delay dictated by its large propagation delay of around 540 ms for a RTD. With this RTD, conversational, real-time, services are not well suited to a GEO satellite system due to the limit of 400ms shown in Table 3.

The CF/DAMA-PB-PD protocol makes use of the CF/DAMA protocol in scheduling the traffic and then employs packet dropping at the RCSTs by means of deadlines that are set to a certain number of frame delays on top of the RTD. This method helps to ensure an upper limit of end-to-end delay for voice and video while the web traffic takes advantage of the "extra" requests, due to packets being dropped and not requiring their slots, made and hence reduces its delay while increasing system utilization in the process.

A light packet loss of up to 3% is considered acceptable in this work, as in [57]. With a chosen acceptable PER of less than 3%, the following delays were chosen (see Table 4):

Table 4: Traffic classes and deadlines.

Traffic class	*RTD*	*Frames*	*Deadline*
Voice	*540 ms*	*2*	$T_{dvoice} = 0.587s$
Video	*540 ms*	*1*	$T_{dvideo} = 0.5635\ s$
Web	*540 ms*	*43*	$T_{dweb} = 1.5505\ s$

Note that the deadline set for the web traffic is a virtual deadline, i.e. if a packet overshoots this deadline it is recorded but the packet is not dropped.

3.4 CF/DAMA-PB-PEDF

The proposed Combined Free/Demand Assigned Multiple Access with Piggy Backing and Prioritised Earliest Deadline First (CF/DAMA-PB-PEDF) protocol is a CF/DAMA-PB derived protocol.

The scheduling algorithm is depicted in Figure 29. The scheduler has two main tables or queues, the Reservation Request Queue (RRQ) and the Free Assignment Queue (FAQ). The FAQ is populated with the addresses of each of the terminals connected to the satellite. The RRQ is used to hold the requests made by the earth terminals or RCSTs. The RRQ holds the source address of the RCST that made the request, as well as the number of slots requested by the RCST. Each received request is placed into the RRQ in a FIFO fashion and served in a FIFO fashion. The scheduler allocates the slots of the TDMA on a frame-by-frame basis and then sends the assignment to the RCST using a packet on the downlink frame. The number of slots in a TDMA frame is equal to the number of RCSTs, effectively giving each a turn to initially request resource capacity. In the MF-TDMA version of this work, the number of slots shared among all the frames in a frame time period is equal to the number of RCSTs. In the case of the DVB-RCS, the Return Link is MF-TDMA and thus requires refinement of the TDMA structure that CF/DAMA protocols are developed for to that of a MF-TDMA structure.

The scheduler first queries the RRQ to determine whether there are any requests made. If the RRQ is not empty then the scheduler assigns a slot in a frame to the RCST request at the head of the queue and subtracts one from the number of requested slots of the RCST in question. This is repeated, if the RCST request at the head of the RRQ becomes zero, the RCST is removed from the head of the queue and the corresponding FAQ entry is moved to the tail end of the FAQ. The slot allocation is repeated until the frame is full, at which point it waits to allocate the following frames slots, or if the RRQ is empty then the slots are assigned to the RCST at the head of the FAQ. The RCST will then be removed from the head of the FAQ and placed at the tail end of the queue. The FAQ is serviced in a round-robin fashion ensuring that the free assignment is fairly allocated to the RCSTs.

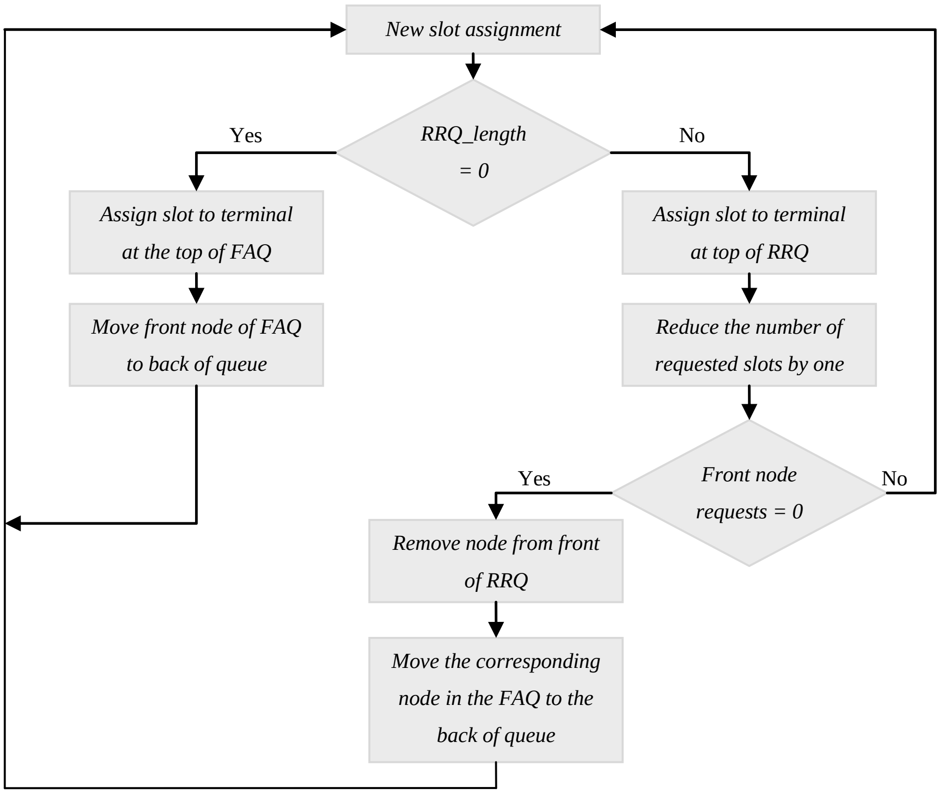

Figure 29: CFDAMA-PB-PEDF scheduling algorithm, [20].

The piggy-backed request strategy used in CF/DAMA-PB-PD is the same as in CF/DAMA-PB.

The traffic classes require certain deadlines in order to meet the QoS a user expects of the traffic types. The Prioritized – Earliest Deadline First (P-EDF) tries to accomplish a means of ensuring the deadlines are met with minimal packet dropping required.

In an Earliest Deadline First (EDF) scheduler, priorities are assigned to packets waiting in the queue to be transmitted. The priority of a packet in the queue is determined by $P_k[n]$, which is according to the current Terminal Time Interval (TTI), or in this case frame period, and the terminal k. $P_k[n]$ is defined as the ratio of the transmission delay of the oldest packet of terminal k, $d_k[n]$, and the deadline for the packet according to its class, $T_{deadline}$,

$$P_k[n] = \frac{d_k[n]}{T_{deadline}} \ (k = 1,2,3 \dots, N) \qquad (3.1)$$

where N denotes the number of terminals.

The EDF scheduler does not prioritise video traffic over that of the web traffic, hence the web traffic could deteriorate the performance of the video and voice traffic as the web traffic becomes significantly large compared to the other traffic loads. As the delay requirements of voice and video are more stringent than web a class differentiation in priorities is required.

The Prioritized-EDF ensures that the voice and video packets are given a higher priority than that of the web traffic without suffocating the web traffic. The following equation is used to determine the priority of a web packet,

$$P_k[n] = min\left\{0.9, \frac{d_k[n]}{T_{deadline}}\right\} \; (k = 1,2,3 \ldots, N) \qquad (3.2)$$

This ensures that very urgent video and voice packets are assigned a higher priority as the web packet priority cannot exceed 0.9 while the voice and video traffic can exceed 0.9.

The packets with the highest priority after calculating their priorities are transmitted first. Thus a FIFO queue is rearranged according to infinite priority possibilities instead. It has been shown that if it is possible to achieve no missed deadlines for a system, the EDF schedulers' will be able to ensure that no deadlines are missed.

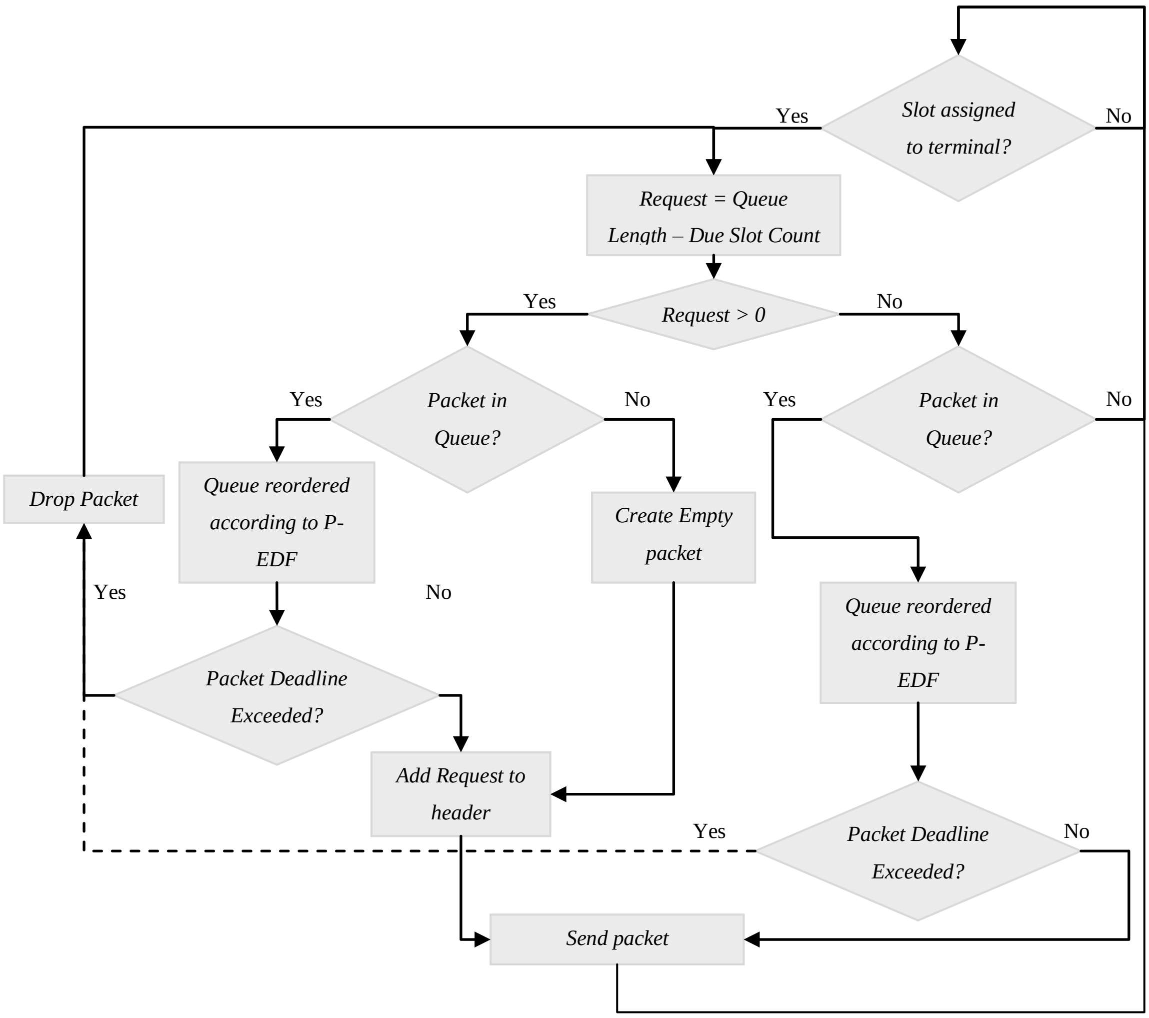

Figure 30: CF/DAMA-PB-PEDF terminal algorithm.

The terminal algorithm for the RCSTs is illustrated in Figure 30 and discussed below.

The RCST waits for a slot assignment. Once a slot has been assigned to the RCST, either by free assignment or demand assignment, the RCST then calculates if it needs to request any additional slots for queued packets. If a request is necessary the RCST reorders the queue according to P-EDF and determines if the packet at the head of the queue has exceeded its deadline. If the deadline has been exceeded for the packet in question, this packet could be a voice, video or web (virtual deadline) packet, the packet is dropped and one returns to calculating if a request needs to be made. If the deadline was not exceeded, the RCST adds the request to the packet at the head of its traffic queue and sends the packet. If there are no packets in the queue and the RCST has a request and a slot assigned to it, then the RCST creates an empty packet, adds the request to that packet and sends the packet in the allocated slot. If the RCST does not have any additional requests to make, but has packets queued in the traffic queue, the queue is reordered according to P-EDF and then the packet at the head of the queue is checked if it has exceeded the deadline, if not it is sent in the allocated slot. On the instance of not receiving a slot for multiple frames, the request is accumulated until the RCST receives a slot allocation to send the request. This ensures that a request is made once per frame at maximum per RCST.

The CF/DAMA-PB-PEDF Protocol is thus a CF/DAMA-PB-PD scheduler with a P-EDF ordered queue at the RCSTs with the deadlines for packet dropping determined according to the classes as in CF/DAMA-PB-PD. The aim of this protocol is to try and improve the packet loss by making a prioritized choice of transmission while still ensuring the same upper limit as in the CF/DAMA-PB-PD protocol.

3.5 Simulation Structure

While the simulations performed focused mainly on the end-to-end delay of packets and the system utilization in terms of the throughput of the system, packet dropping was also examined to ensure that it does not exceed acceptable levels. The system incorporated CF/DAMA-PB-PD and the CF/DAMA-PB-PEDF MAC protocols in a DVB-RCS GEO-satellite system. Certain adjustments were added to the protocols to ensure that the MAC protocol was suited to a DVB-RCS system, the main one being defined by the MF-TDMA air interface that DVB-RCS makes use of instead of the TDMA air interface that CF/DAMA has been originally designed for.

The return-link capacity of the GEO satellite was the focus of the research, which is defined by the DVB-RCS standard as described in chapter 2. The channel was assumed to be ideal, i.e. there are no channel losses, in this chapter.

The next chapter defines the channel model and adds a more realistic channel environment to the system.

3.5.1 Simulator Coding

The simulation was programmed using C++. The decision to use the C++ language was due to the speed capabilities of the C language and the object-oriented design capabilities offered by C++. The statistical functions required to implement the traffic models and channel model where obtained or coded using the C++ TR1 (The standard was defined by the C++ Standards Committee – Technical Report 1) extension as a basis.

Microsoft Visual C++ Microsoft Foundation Classes (MFC) was used in creating the Graphical User Interface (GUI) of the simulator for ease of simulation scenario adjustment.

The simulations were all run on an AMD Turion(tm) 64 X2 Mobile Technology TL-50 processor in a Microsoft Windows Vista Home Premium (32-bit) environment.

3.5.2 Simulation Flow

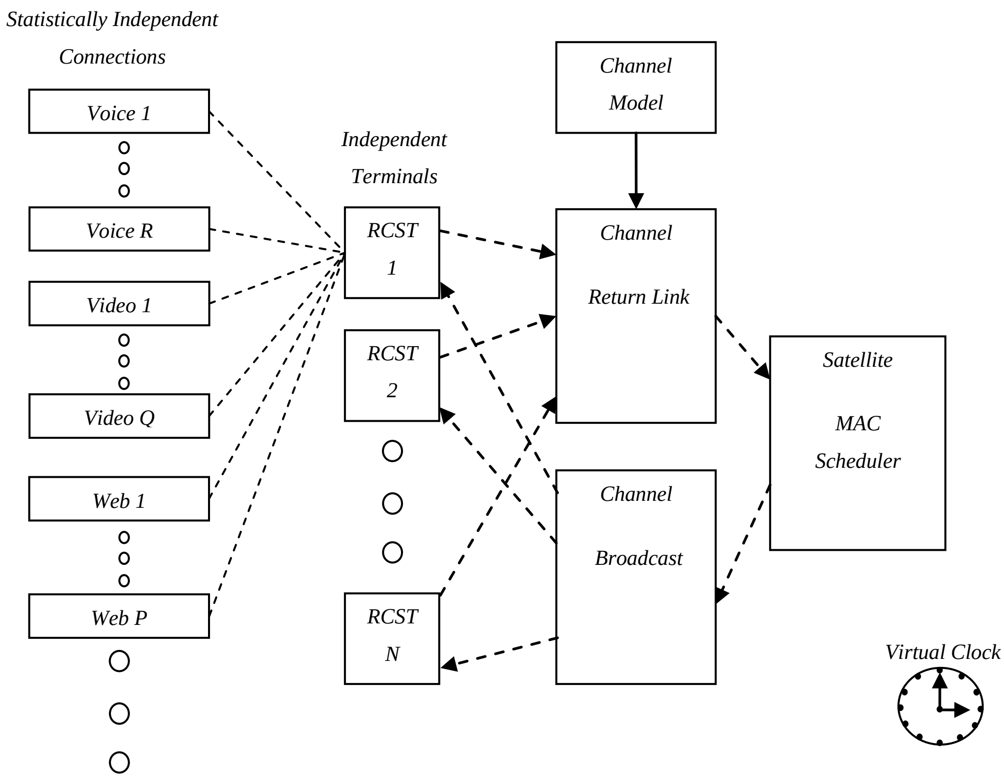

Figure 31: Simulation block diagram.

The simulation has a GEO satellite with OBP, a return link and forward broadcast link, N independent RCS terminals. Each RCST has the ability to have multiple voice, video and web connections generating traffic statistically independently from one another. The simulator has a virtual clock to ensure correct timing of simulations in terms of the channel timing, frame time, slot time, packet timing, bit-rate, etc. The dashed connecting lines represent the traffic flow of the system as shown in Figure 31.

Each RCST has the simplified OSI layered structure implemented, allowing for future additional features to be added to the system and to allow for cross-layer interaction explored in Chapter 4.

3.5.3 Simulator GUI

The GUI has four main groupings; the simulation run time, to view the progress of the simulation, the system parameters, the traffic input and the simulation results, as depicted in Figure 32.

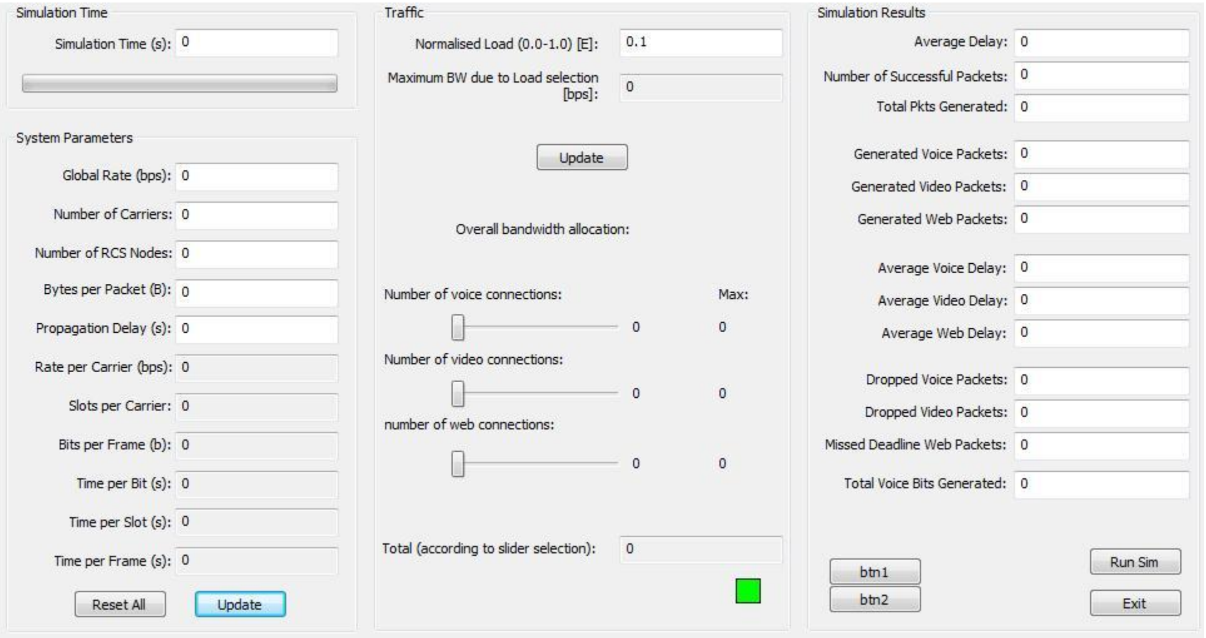

Figure 32: Simulator GUI.

3.6 Traffic Models

This section discusses the various traffic models used to implement the multimedia data sources. The traffic sources were used to evaluate the performance of the RRM schemes. Voice, video and web data sources were used to implement the multimedia traffic required. Each source was made statistically independent from each other during simulations. The traffic models were then validated by comparing the simulation components with the theoretical expectations. The statistical distributions that make up the traffic models were verified by comparing the simulated outputs of the distributions with the mathematical expected results.

Traffic sources have a characteristic measure called the degree of burstiness: this is the measure used to describe sources that have long silent or OFF periods and sudden bursts of activity. It describes the measure of maximum ideal multiplexing gain that is achievable in a system capacity equal to the peak rate of the source.

The traffic models used were standard traffic models used to evaluate radio resource management schemes as described by Andreadis and Giambene [53].

3.6.1 Video Traffic

The video traffic model implemented can be used for both conversational and streaming traffic classes. A real-time video traffic source may be seen as a combination of M independent mini-sources, each of the M mini-sources alternating between an ON and an OFF state. While a mini-source is in its ON state, a constant bit-rate of A bits/s is generated while in the OFF state the mini-source generates no bits.

In the MF-TDMA air interface technique, time evolves in discrete slots (T_s) and frames, each slot being equivalent in length to one MPEG2 packet. The time spent in the ON state, the sojourn time, is represented in terms of T_s and is geometrically distributed with a probability of,

$$\alpha = \frac{1}{p} \tag{3.3}$$

Where p is the mean time spent in the ON state in terms of T_s. The sojourn time spent in the OFF state is geometrically distributed with a probability of,

$$\beta = \frac{1}{q} \tag{3.4}$$

Where q is the mean time spent in the OFF state in terms of T_s. Each mini-source has an activity factor of,

$$\psi_v = \frac{p}{p+q} \tag{3.5}$$

To ensure that one does not have sudden variations in traffic, only a single mini-source may change state during a slot time. This video source is modelled using a discrete-time Markovian Arrival Process (D-MAP) depicted in Figure 33.

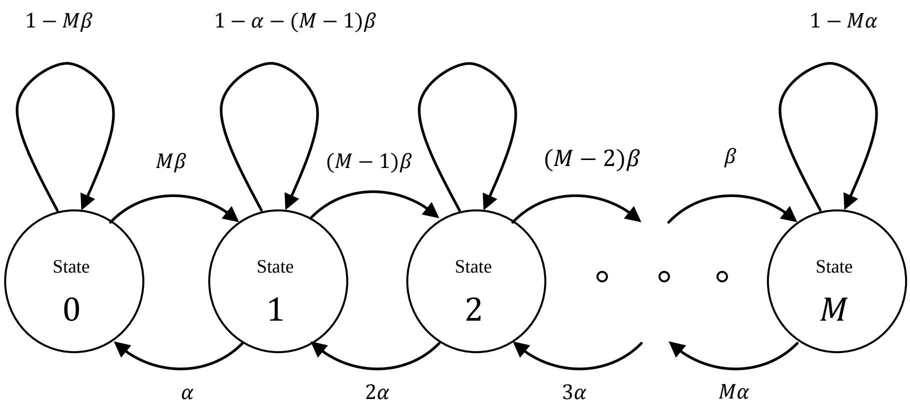

Figure 33: Modulating process for the packet generation of a video source (D-MAP).

The mean ON and OFF state sojourn times and the bit-rate A are calculated using the following formulae [53],

$$A = \frac{\mu_v}{M} + \frac{{\sigma_v}^2}{\mu_v}\left[\frac{bit}{s}\right] \tag{3.6}$$

$$q = \frac{1}{aT_s}\left(1 + \frac{M{\sigma_v}^2}{{\mu_v}^2}\right)[s] \tag{3.7}$$

$$p = \frac{1}{aT_s}\left(1 + \frac{{\mu_v}^2}{M{\sigma_v}^2}\right)[s] \tag{3.8}$$

$$Prob\{state = i\} = \binom{M}{i} {\psi_v}^i (1 - \psi_v)^{M-i} \quad i = 0, \dots, M \tag{3.9}$$

In the formulae, μ_v represents the mean bit-rate of the wanted/modelled video traffic, ${\sigma_v}^2$ is the variance of the traffic sources bit-rate and a is the coefficient of auto-covariance of the sources bit-rate. The probability of being in a state is shown to be binomial.

The parameters chosen are the mean bit-rate μ_v = 256 kbits/s and the standard deviation σ = 128 kbits/s, M = 10 mini-sources and a = 3.9 s^{-1}resulting in,

$$A = \frac{256000}{10} + \frac{128000^2}{256000} = 89600\ bits/s$$

$$q = \frac{1}{3.9 * 0.0235}\left(1 + \frac{10 * 128000^2}{256000^2}\right) = 38.189\ s$$

$$p = \frac{1}{3.9 * 0.0235}\left(1 + \frac{256000^2}{10 * 128000^2}\right) = 15.276\ s$$

And the probabilities follow,

$$\alpha = \frac{1}{15.276} = 0.065462162$$

$$\beta = \frac{1}{38.189} = 0.02618555$$

With an activity factor of,

$$\psi_v = \frac{15.276}{15.276 + 38.189} = 0.285719629$$

Therefore one achieves a mean bit-rate of,

$$\mu_v = M\psi_v A\ bits/s \tag{3.10}$$
$$\mu_v = 10 * 0.285719629 * 89600 = 256005\ bits/s$$

The peak bit-rate is,

$$\hat{A}_v = MA\ bits/s \tag{3.11}$$
$$\hat{A}_v = 896000\ bits/s$$

And finally the degree of burstiness is,

$$B_v = \frac{1}{\psi_v} = \frac{1}{0.286} = 3.497 \tag{3.12}$$

Video Traffic Model Validation

The following illustrate the accuracy of the model components by comparing the simulated results with the theoretically expected results. The geometric distributions that determine the mean sojourn times spent in each state *i* are shown to be geometrically distributed as expected in Figure 34 and Figure 35.

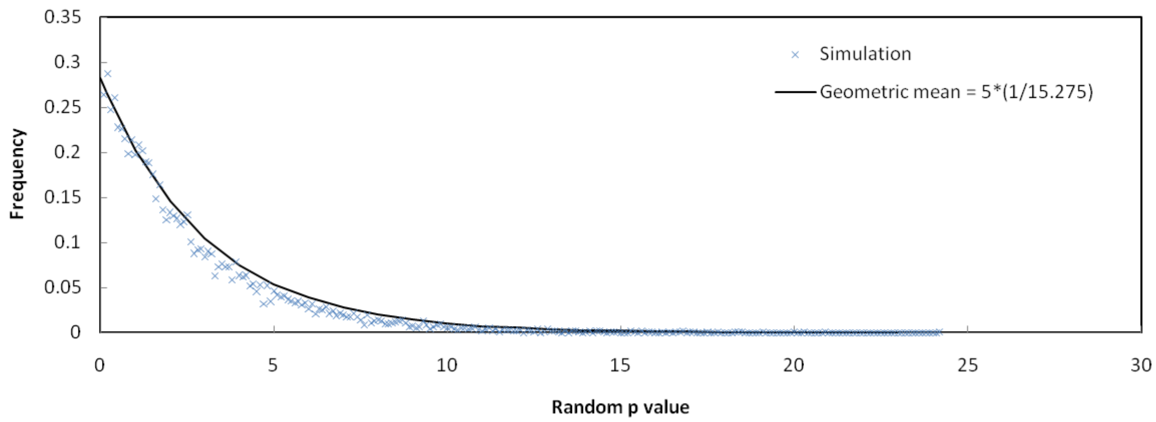

Figure 34: State *i* = 5, p (OFF) geometric distribution (100,000 data points).

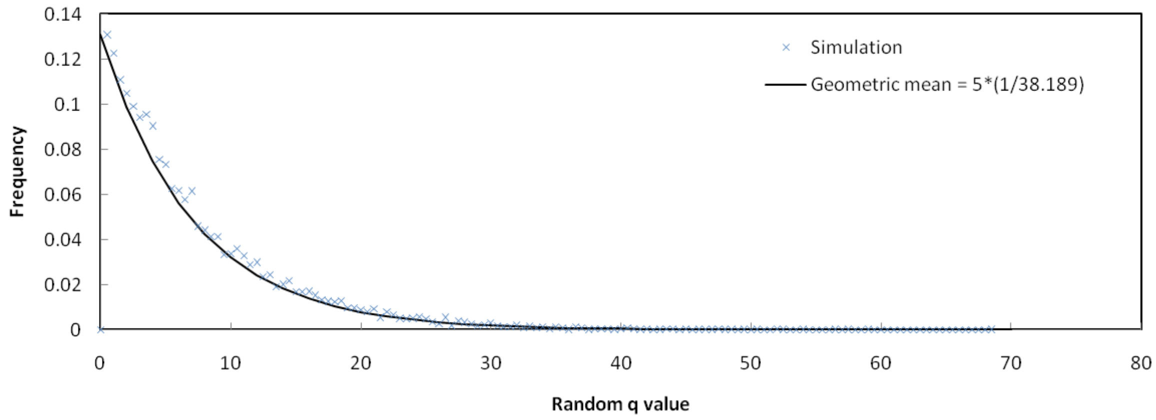

Figure 35: State *i* = 5, q (ON) geometric distribution (100,000 data points).

Figure 36 depicts the histogram in terms of the frequency a bit-rate is achieved, the mean being close to the expected mean of 256 kbits/s.

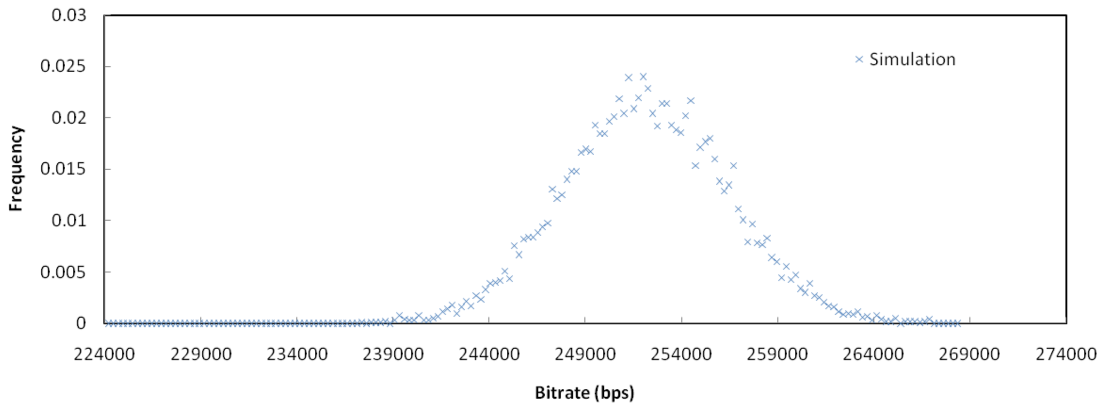

Figure 36: Video Traffic Histogram (10000 dataset).

The parameters of the simulated and theoretical results are compared in Table 5 and found to be within acceptable ranges of the expected mean bit-rate, the mean resulting bit-rate is the peak bit rate multiplied by the activity factor.

Table 5: Video traffic model parameter validation.

Parameter	*Theory*	*Simulation*
$\alpha \times i$	$0.0654 \times i$	$0.0654 \times i$
$\beta \times i$	$0.0261 \times i$	$0.0261 \times i$
ψ_v	0.285	0.285
Mean bit-rate	*256000bps*	*252000.6bps*

3.6.2 Voice Traffic

A voice conversation can vary between two states, i.e. an ON state and an OFF state, as depicted in Figure 37.

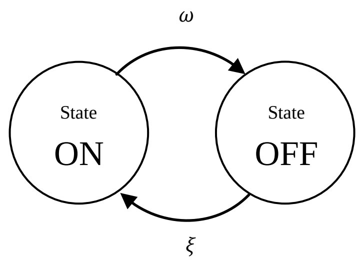

Figure 37: ON-OFF traffic model for a voice source.

While a voice source is in the ON state, it generates a constant bit-rate of R_s bits/s. While a voice source is in the OFF state, no bits are generated. The time spent in an ON or OFF state, the sojourn time, is exponentially distributed with the ON state having a mean sojourn time of,

$$\mu_{ON} = \frac{1}{\omega} = 1s \tag{3.13}$$

And the Off state having a mean sojourn time of,

$$\mu_{OFF} = \frac{1}{\zeta} = 1.35s \tag{3.14}$$

The voice source will then have an activity factor,

$$\psi_s = \frac{\frac{1}{\omega}}{\frac{1}{\omega} + \frac{1}{\zeta}} = 0.425 \tag{3.15}$$

The voice source degree of burstiness follows,

$$B_s = \frac{R_s}{R_s \psi_s} = \frac{1}{\psi_s} = 2.352 \tag{3.16}$$

The mean bit-rate for the chosen 64 kbit/s voice source is,

$$\mu_s = \psi_s R_s = 0.425 \times 64000 = 27200\ bits/s \tag{3.17}$$

Voice Traffic Model Validation

The following graphs illustrate the accuracy of the model components by comparing the simulated results with the theoretically expected results. The exponential distributions that determine the mean sojourn times spent in each state are shown to be exponentially distributed as expected in Figure 38 and Figure 39.

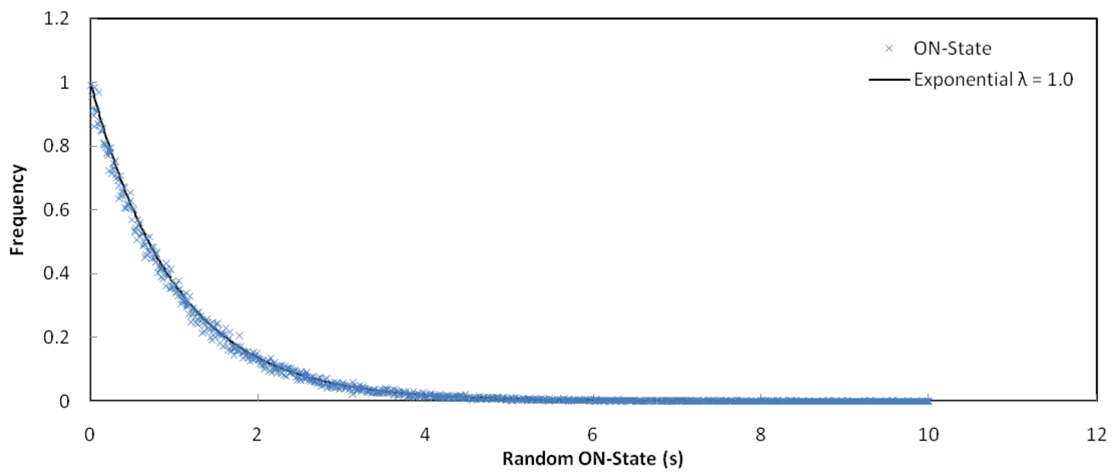

Figure 38: On-State distribution (100,000 data points).

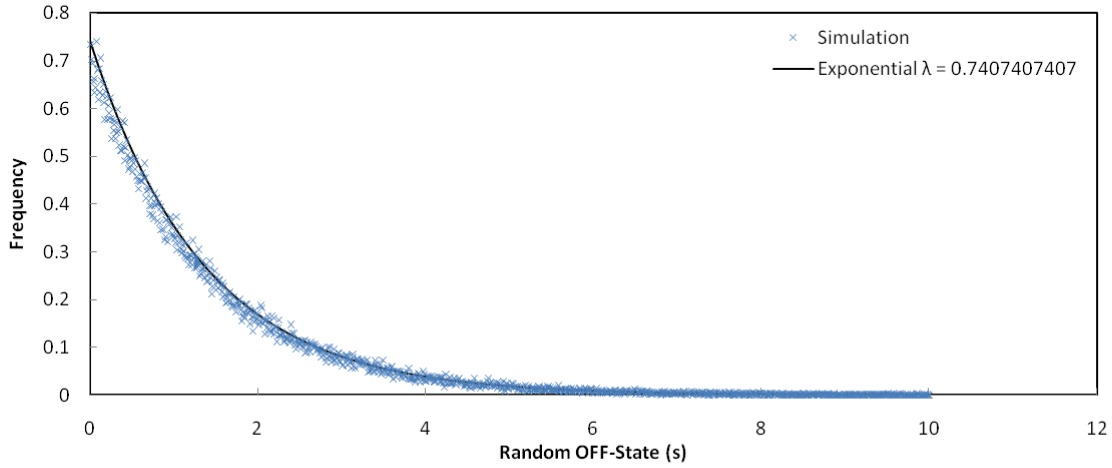

Figure 39: Off-State distribution (100,000 data points).

Figure 40 shows the voice average bit-rate over time; the voice is oscillating around the expected average bit-rate.

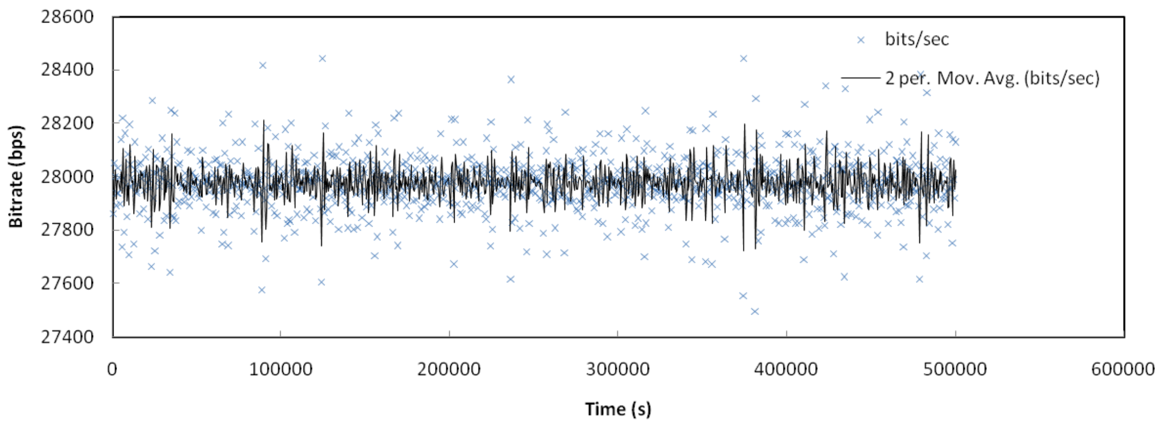

Figure 40: Voice Traffic model bit rate over time.

Figure 41 depicts the packet output of the voice source, clearly showing the ON and OFF sojourn times over time. This graph can also be used to calculate the average bit-rate of the voice source by summing the packets of a set time and multiplying by the number of bytes per packet and the number of bits per byte and dividing this by the time in seconds, achieving the bit-rate of the source.

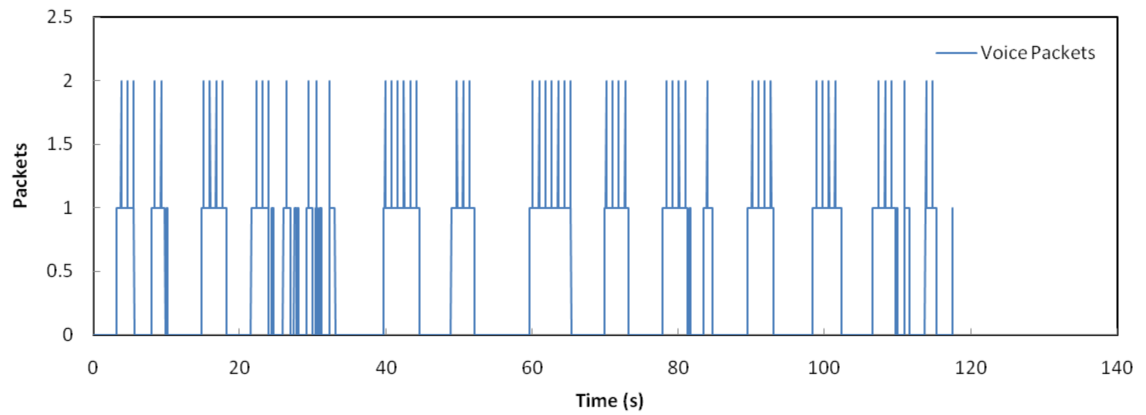

Figure 41: Voice Traffic model packet generation.

Figure 42 depicts the overall model output as a histogram of the frequency that a certain bit-rate is produced. This graph was obtained from a 10000 dataset each representing a period of 500 seconds, thus equating to a 5000000 second simulation of the voice traffic output.

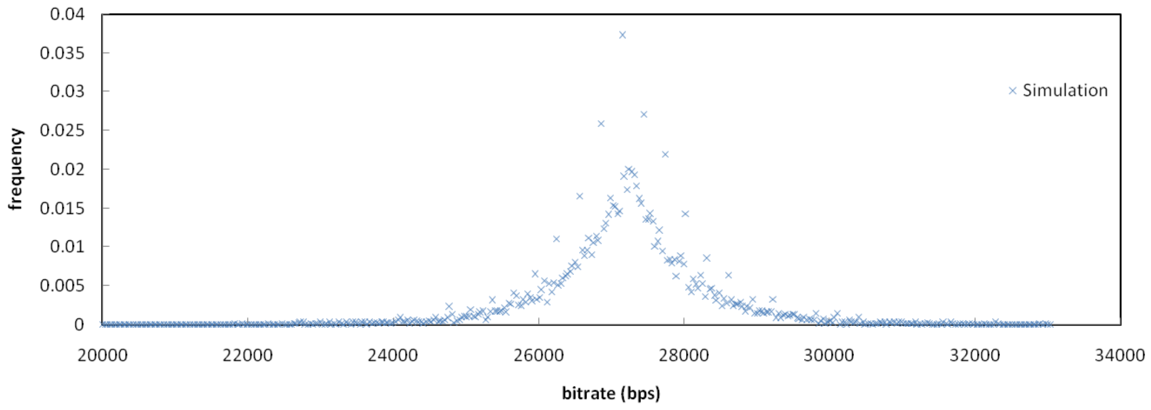

Figure 42: Voice Traffic Histogram (10000 dataset).

The parameters of the simulated and theoretical results are compared in Table 6 and found to be very accurate, the mean resulting bit-rate is the peak bit rate multiplied by the activity factor.

Table 6: Voice traffic model parameter validation.

Parameter	*Theory*	*Simulation*
$1/\omega$	*1.0 s*	*1.0 s*
$1/\zeta$	*1.35 s*	*1.35 s*
R_s	*64000 bps*	*64000 bps*
Mean bit-rate	*27200*	*27233.37*

In conclusion, the simulation of the implemented traffic model is seen to output the expected voice traffic and thus was used in creating a multimedia traffic source along with the web and video traffic sources.

3.6.3 Web Traffic

To create a web traffic source, a two state model is used as described by Andreadis and Giambene [53]. The model consists of a Reading state and a Packet Call state as depicted in Figure 43. The source alternates between the two states. During the Packet Call state web datagrams are generated, the number of datagrams has a mean of [53],

$$m_{w_d} = 25 \tag{3.18}$$

The number of datagrams has a geometrical distribution. The inter-arrival time between datagrams has an exponential distribution with a mean of,

$$m_{w_{ia}} = \frac{0.5}{q_w}\ s\ (q_w = 1, 2, 3 \ldots) \tag{3.19}$$

$$m_{w_{ia}} = \frac{0.5}{1} = 0.5\ s$$

The Reading state has a sojourn time that is exponentially distributed with a mean of [53],

$$m_{w_r} = 12\ s \tag{3.20}$$

Each of the datagrams has a random length generated by means of a Pareto distribution, the output of the Pareto distribution, *x*, is floored,

$$l_d = \lfloor x \rfloor\ bytes \tag{3.21}$$

The Pareto distribution is defined by the shape and mode, the shape being v and the mode k. The output is also truncated, m, to ensure that the largest length of the random variable x does not exceed that of 9000 bytes. The mode defines the minimum length that a datagram can have of 81.5 bytes. The probability density function that defines the random variable x is,

$$pdf(x) = \frac{vk^v}{x^{v+1}}[u(x-k) - u(x-m)] + \left(\frac{k}{m}\right)^v \delta(x - m) \tag{3.22}$$

Where $u(\bullet)$ is the unitary step function and $\delta(\bullet)$ is the Dirac Delta function.

One is able to obtain the mean value of x as follows,

$$E[x] = \frac{vk - m\left(\frac{k}{m}\right)^v}{v-1} \tag{3.23}$$

Thus, with v = 1.1, k = 81.5 bytes, m = 9000 bytes,

$$E[x] = \frac{1.1 \times 81.5 - 9000\left(\frac{81.5}{9000}\right)^{1.1}}{1.1 - 1} = 3098.76\ bits/s$$

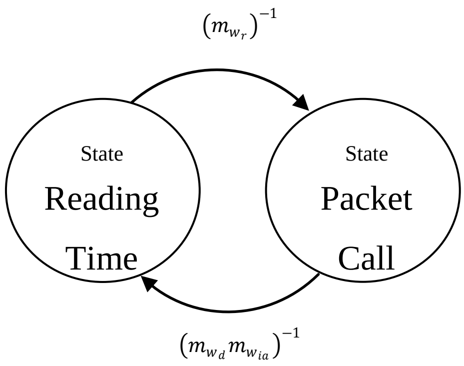

Figure 43: Modulating process for a web source.

With an activity factor defined by,

$$\psi_w = \frac{m_{w_d} m_{w_{ia}}}{m_{w_d} m_{w_{ia}} + m_{w_r}} \tag{3.24}$$

$$\psi_w = \frac{25 \times 0.5}{25 \times 0.5 + 12} = 0.5102$$

And the mean arrival rate of datagrams defined by,

$$\lambda_w = \frac{\psi_w}{m_{w_{ia}}} = \frac{0.5102}{0.5} = 1.020408163 \qquad (3.25)$$

The mean bit rate is,

$$\begin{aligned}\mu_w &= \lambda_w E[x] = 1.020408163 \times 3098.76 \\ &= 3162\ bits/s\end{aligned} \qquad (3.26)$$

The degree of web burstiness is largely determined by the value of q_w, the chosen value of $q_w = 1$ gives the burstiness,

$$B_w = \frac{1}{\psi_w} = \frac{1}{0.5102} = 1.96 \qquad (3.27)$$

Due to the C++ TR1 extension not having a Pareto distribution included in it, the following equation was used to create the Pareto distribution from a uniform distribution of the C++ TR1 extension,

$$Pareto = mode \times (1 - uniform)^{\frac{-1}{shape}} \qquad (3.28)$$

Where *uniform* is a uniform random variable between 0 and 1.

Web Traffic Model Validation

The following graphs illustrate the accuracy of the model components by comparing the simulated results with the theoretically expected results. Each statistical distribution has been simulated and the histogram graphed with 10000 simulated random variables. The exponential distributions that determine the mean sojourn times spent in the inter-arrival times and the Reading state are shown to be exponentially distributed as expected in Figure 44 and Figure 46 respectively.

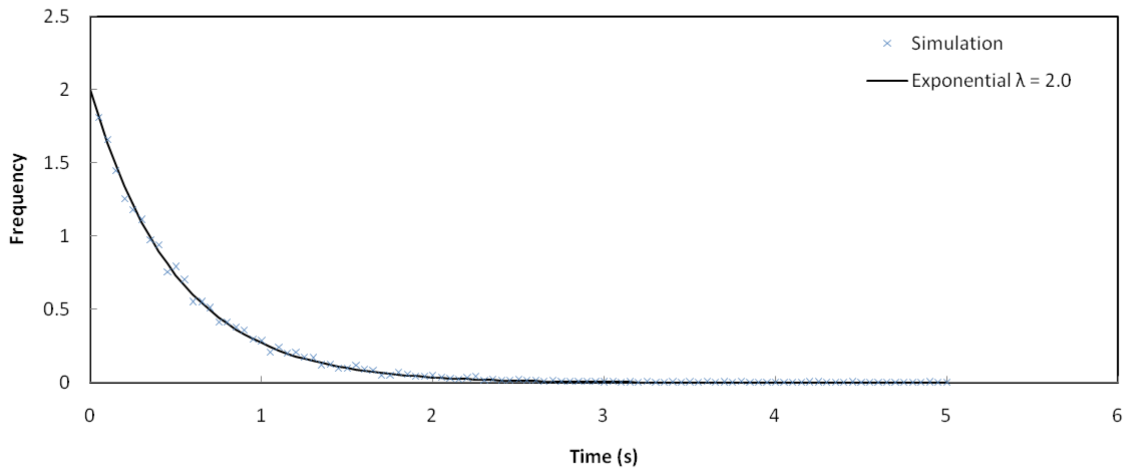

Figure 44: Inter Arrival time distribution (100,000 data points).

The pareto distributed datagram length was simulated and as depicted in Figure 45 shown to be a pareto distribution with the shape and mode as expected. Having no lengths less than 81.5 bytes and decreasing frequency of increasing lengths from 81.5 bytes.

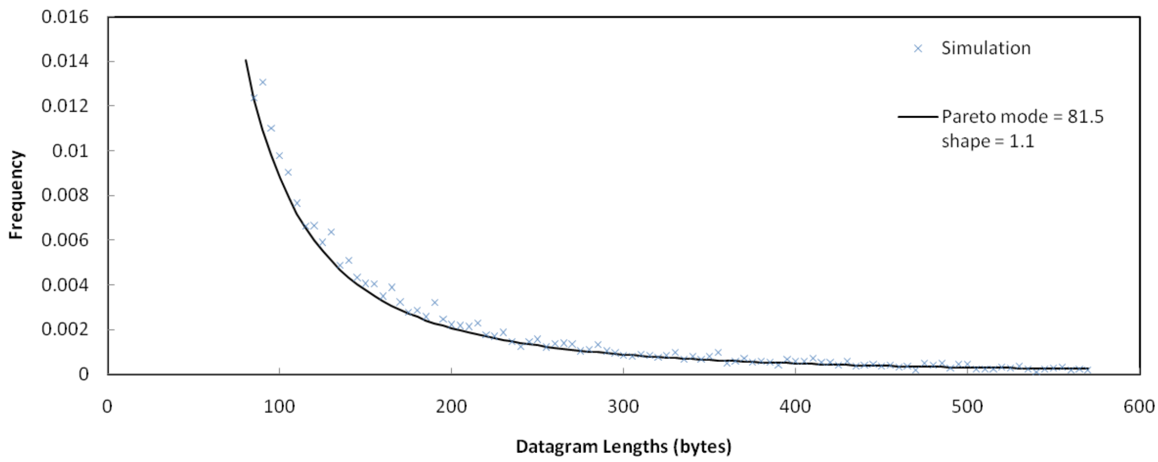

Figure 45: Datagram length distribution (100,000 data points).

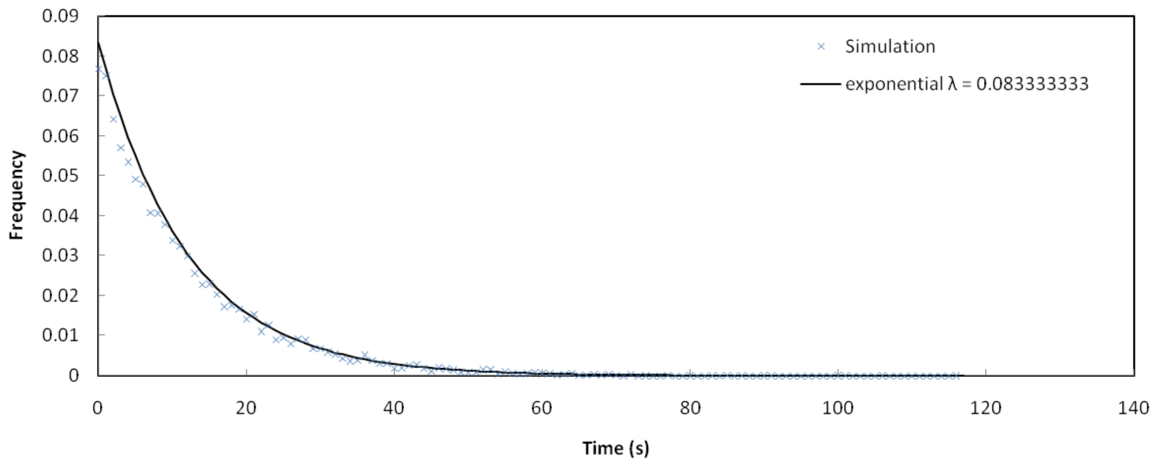

Figure 46: Reading time distribution (100,000 data points).

In Figure 47 the simulated results follow the geometric distribution closely and have the mean number of datagrams of 25 as expected. Figure 48 depicts the simulated average bit-rate of a web source to be around the expected mean bit-rate and Figure 49 illustrates the web traffic model implemented, showing the reading times as large periods of no datagrams and the packet call periods having multiple datagrams with inter arrival periods separating them. Table 7 compares the theoretical results with the simulation results and once again the mean bit-rate closely matches the expected theoretical result with a slight deviation.

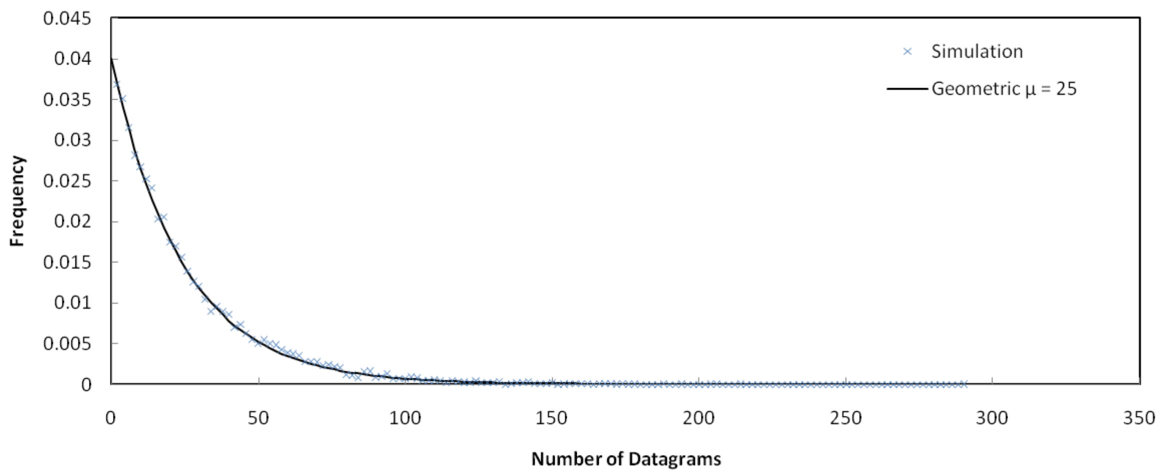

Figure 47: Datagram calls distribution (100,000 data points).

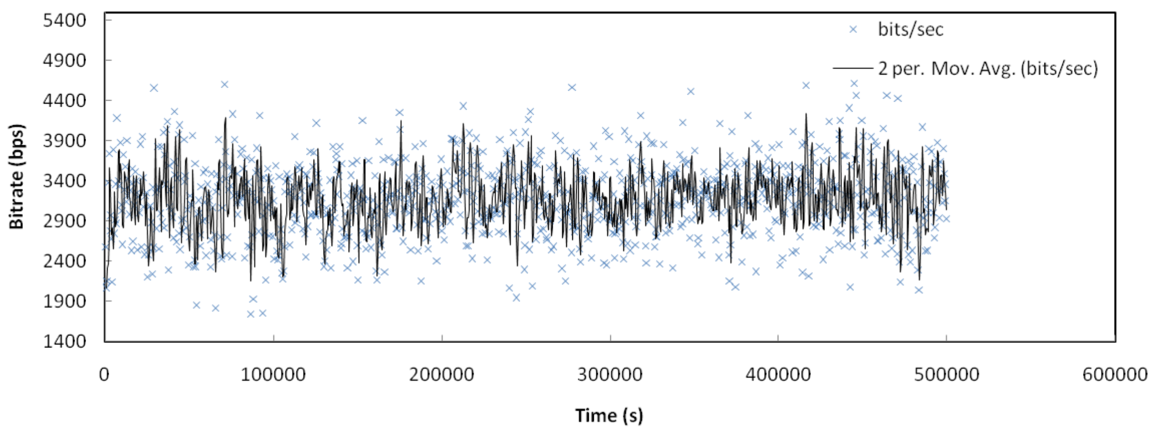

Figure 48: Web Traffic model bit rate over time.

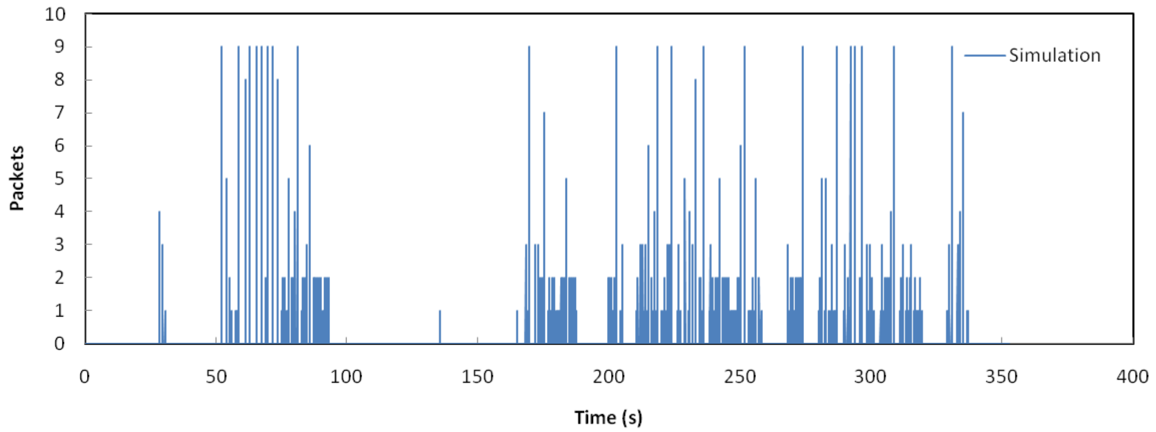

Figure 49: Web Traffic model packet generation (B_w = 1).

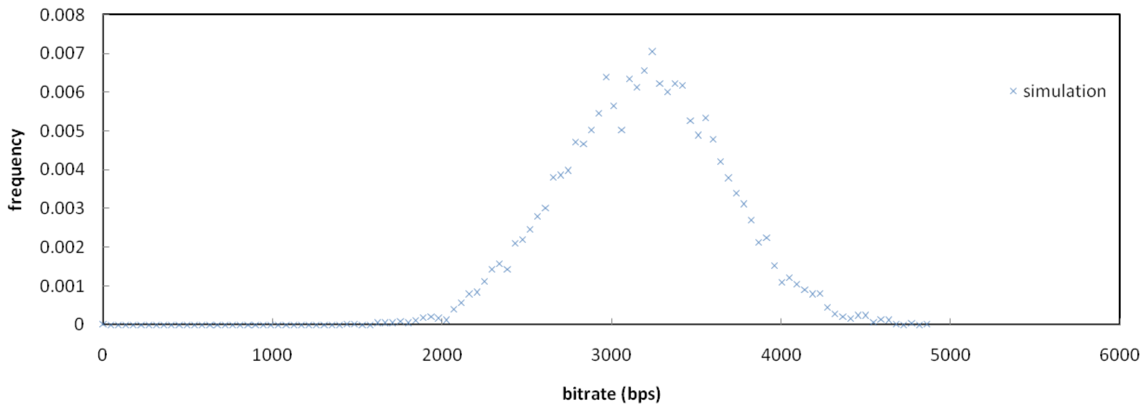

Figure 50: Web Traffic Histogram (10000 dataset).

The web traffic model histogram is depicted in Figure 50 it consists of 10000 values of packet output during a 500 s period converted to the bit-rate and displayed as a histogram illustrating the mean bit-rate of the web traffic model implemented over a period of 5000000 s.

Table 7: Web traffic model parameter validation.

Parameter	*Theory*	*Simulation*
m_{w_d}	*25*	*25*
$m_{w_{ia}}$	*0.5*	*0.5*
m_{w_r}	*12*	*12*
v	*1.1*	*1.1*
k	*81.5 bytes*	*81.5 bytes*
m	*9000 bytes*	*9000 bytes*
Mean bit-rate	*3162*	*3078*

3.7 Performance Measures

The performance measures commonly used to analyse a packet broadcast satellite network and the effects of the MAC algorithms used are:

- Delay (*D*)
- Throughput (*S*)
- Utilization (*U*)
- Packet Dropping (*PD*)

These performance measures are common to both satellite and non-satellite networks alike and are used in network performance analysis.

The performance measures are commonly displayed against:

- Input Load (*I*)
- Offered Load (*G*)

3.7.1 Delay

The delay performance measure D, is the time taken from the point in time the packet is ready to be transmitted to the point in time that the packet is successfully transmitted. This transmission delay is represented as the average delay of packets over a set time interval. The average delay is the measure one wishes to reduce as far as possible and has a lower bound dictated by the propagation delay of the GEO satellite network.

3.7.2 Throughput

The throughput performance measure is the total rate of data being transmitted between stations. This is also referred to as the carried load. Throughput is commonly normalized, i.e. it is represented as a fraction of the total capacity between stations in consideration.

3.7.3 Input Load

The input load, *I*, is the total rate of data that is generated by the stations connected to the system. This is commonly represented in the normalized form, i.e. as a fraction of the system capacity.

$$Normalised\ Load = \frac{Traffic\ generated}{Channel\ Capacity} \tag{3.29}$$

3.7.4 Offered Load

The offered load, *G*, is the total rate of data that is presented to the system for transmission. This is commonly represented in the normalized form, i.e. as a fraction of the system capacity.

3.7.5 Utilization

The utilization, *U*, is the fraction of the total capacity being used. The ideal channel utilization is depicted in Figure 51.

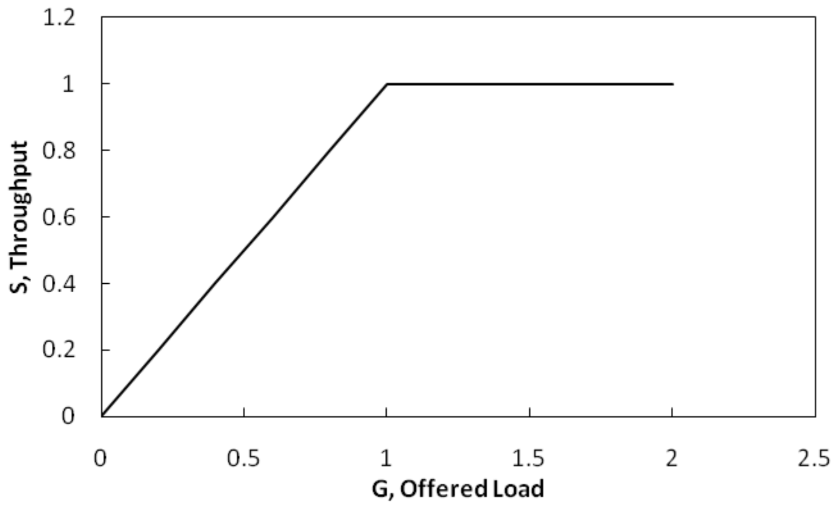

Figure 51: Ideal channel utilization.

In Figure 51 throughput is a dependent variable and offered load is the dependent variable.

3.8 Simulation Parameters

The simulations consists of 1000 s runs for each load value, a resulting virtual time of between 10000 s and 17000 s simulation runs per range of throughput, average delay and packet dropping graphs. The GEO satellite DVB-RCS system consisted of a global rate of 8 Mbits/s and a carrier rate of 2 Mbits/s each. The total number of slots is equivalent to the number of RCST nodes, i.e. 128 slots and nodes. This equates to 32 slots per carrier, 4 carriers, imposing a limit of 32 slots being allocated to a node per frame period of 23.5 ms. A channel propagation delay of 270 ms due to the distance between an RCST and the GEO satellite. Each MPEG2 packet contains 188 bytes in total and 183 of these are for the payload.

Table 8: System parameters.

Parameter	*Value*
Simulation time	*1000 s*
Global rate	*8192000 bits/s*
Number of carriers	*4*
Number of RCSTs	*128*
Packet format	*MPEG2*
Packet size	*188 bytes*
Payload size	*183 bytes*

Rate per carrier	*2048000 bits/s*
Slots per carrier	*32*
Propagation delay	*270 ms*
Frame time	*23.5 ms*

The traffic used as input to the system is that of video, voice and web combinations, making up a multimedia traffic source. The scenarios used dictated the ratios of each traffic type that make up a multimedia traffic. Due to traffic ratios being network dependent, as well as user dependent, 5 traffic scenarios were chosen to give a broad overview of possible traffic scenarios and how each traffic type affects the system. It would not be realistic to choose one traffic scenario according to a specific network or internetwork traffic ratio as this will change over time. Increasing ratios of video and data to that of voice has been the trend. The 1st, 2nd and 3rd traffic scenarios evaluate a non-heterogeneous traffic scenario to observe the general performance of a system when a single traffic class is used. The fourth scenario allows for the evaluation of the system with even ratios of the three traffic classes. This does not necessarily occur in reality, thus a more close estimate of reality, being dominated by the video class, is also examined in scenario 5.

Table 9: Simulation traffic scenarios.

Scenario	*Video*	*Voice*	*Web*
1	*1*	*0*	*0*
2	*0*	*1*	*0*
3	*0*	*0*	*1*
4	*1/3*	*1/3*	*1/3*
5	*2/3*	*1/6*	*1/6*

3.9 Results

The results presented and discussed in this chapter are those of the CF/DAMA-PB protocol, the CF/DAMA-PB-PD protocol and the CF/DAMA-PB-PEDF protocol. The scenario used will be clearly mentioned and the resulting graphs examined.

The average delay measured is the time delay from the point at which a packet is created to the time a request is allocated and the RCST places the packet onto the channel to be sent to the DVB-RCS GEO satellite minus 270ms link delay.

The normalized load describes the offered traffic as a fraction of the total system capacity of 8 Mbits/s.

The throughput is the fraction of successfully transmitted packets to the total system capacity of 8 Mbits/s.

The following simulations are presented:

- CF/DAMA-PB
- CF/DAMA-PB-PD Protocol
- CF/DAMA-PB-PEDF Protocol

3.9.1 CF/DAMA-PB Protocol

The traffic scenarios that have been investigated in the CF/DAMA-PB simulations are scenario 1, scenario 2, scenario 3, scenario 4 and scenario 5. The various traffic scenarios have been chosen to either be an equal portioning of the traffic classes, as in scenario 4, a more realistic portioning, as in scenario 5, or simply a pure video, voice, or web traffic class, as in scenario 1, 2 and 3. The CF/DAMA-PB protocol is presented in detail to ensure clarity in the performance of the proposed protocols and their enhancements when compared to the CF/DAMA-PB scheme.

Scenario 1

In scenario 1, a video traffic set of sources was used without voice and web traffic classes; this was to investigate the system performance when presented with a purely video traffic load.

The average packet delay of the video source is presented in Figure 52. The graph depicts a low average delay close to the RTD of 270 ms for loads below 0.7. This good performance at lower loads was due firstly to the fact that a minimum of one RTD was required to request and allocate the resource to the

RCSTs but also is characteristic of the CFDAMA-PB protocol where at low loads the free assignment of resources is quite high and often allowing the user to gain more allocated slots than required. Below 0.7, 22 or less video connections were sharing the total system capacity, each having a theoretical bit-rate of 256 kbits/s, requiring just over 5.5 Mbit/s data rate out of the 8 Mbit/s system capacity. The system average delay begins to increase at higher loads as a result of RCSTs competing for resource allocation, but also mainly due to the fact that less free assignment is made, giving RCSTs less opportunity to request capacity if this has not been assigned through demand assignment. The average delay becomes almost exponentially large once above a load of 0.9 indicating that the system performance degrades tremendously.

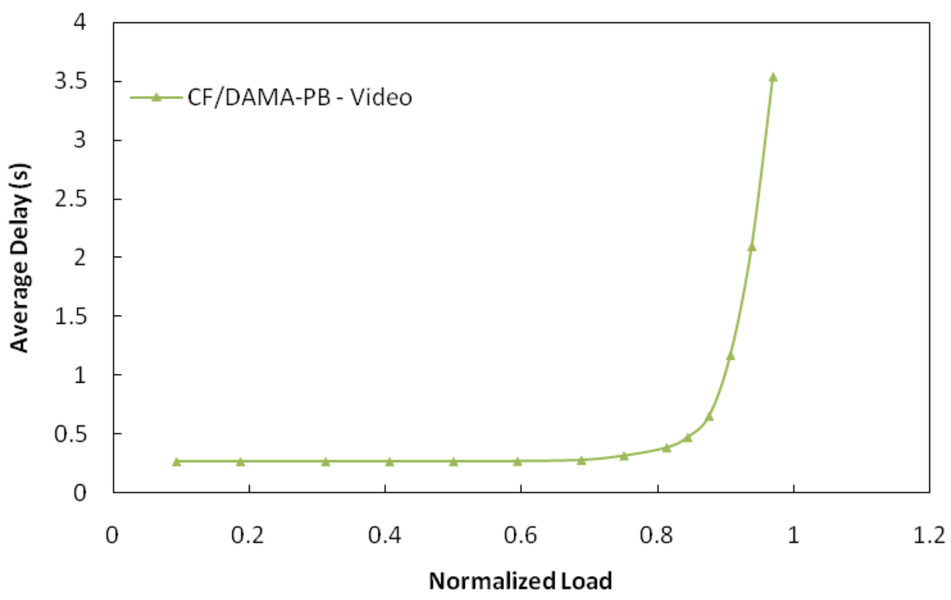

Figure 52: Average Video Delay vs. Normalized Load (CFDAMA-PB – Scenario 1).

The system throughput was excellent as a result of an almost ideal situation offered, i.e. there are no channel conditions that adversely affect the throughput. The offered load was almost completely transmitted as can be seen in the deference between the generated and successfully transmitted packet diagram,

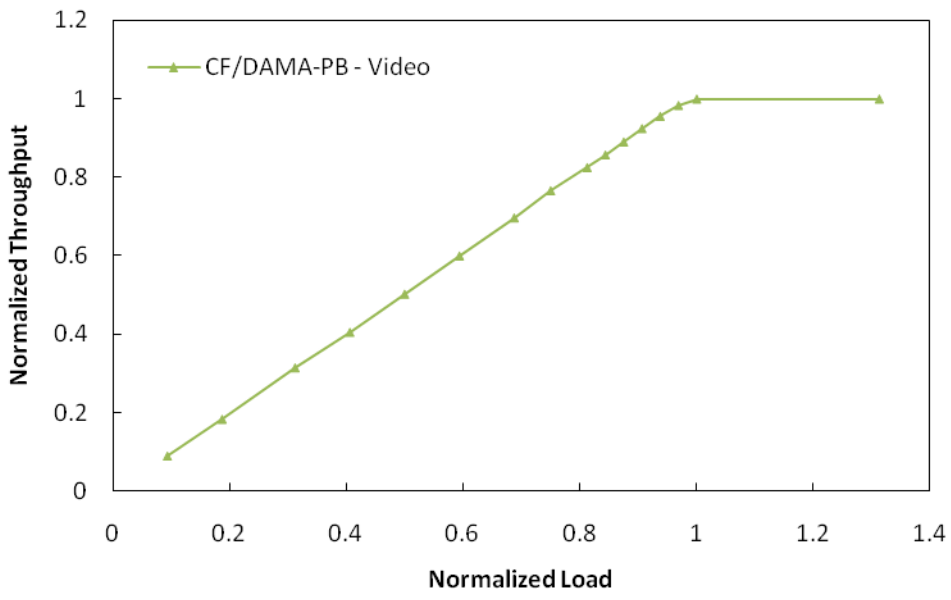

Figure 53: Normalized Throughput vs. Normalized Load (CFDAMA-PB – Scenario 1).

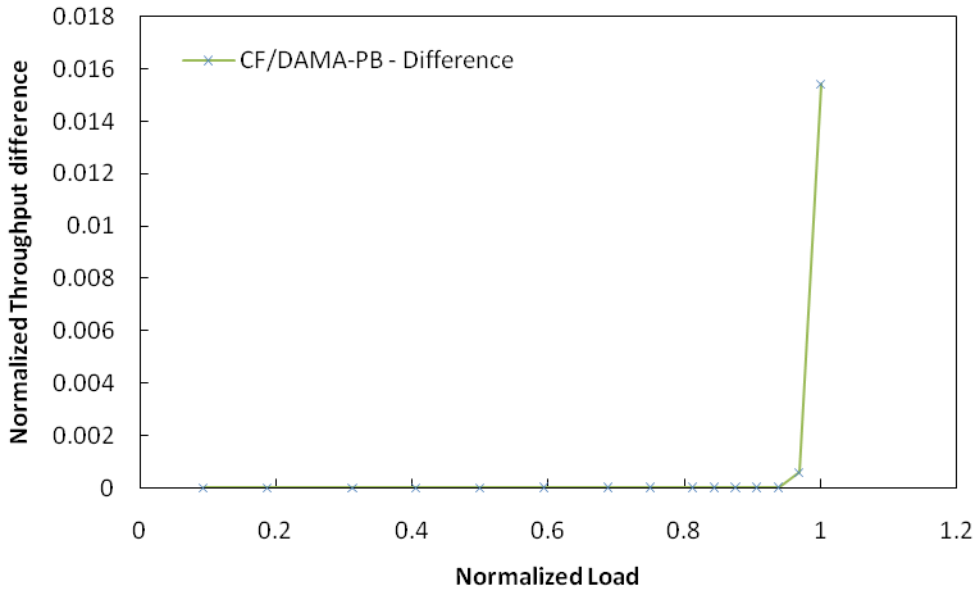

Figure 54: Video throughput difference between successful packet transmission and generated packets.

The throughput was not perfect, but extremely close to perfect as a result of free assignment as well as demand assignment. The scheme seems to decrease the throughput significantly as it approaches a load of 1.

Scenario 2

In scenario 2, a voice traffic set of sources was used without video and web traffic classes; this is to investigate the system performance when presented with a purely voice traffic load.

The average packet delay of the voice was presented in Figure 55. The graph depicts a low average delay close to the RTD of 270 ms for loads below 0.4. This was due to the free assignment of slots to RCSTs that did not request them, more free assigned slots were as a result of a smaller number of voice connections, and thus less demand assigned slots. The system average packet delay starts to increase at higher loads as a result of RCSTs competing for resource allocation, but also mainly due to the fact that less free assignment was made giving RCSTs less opportunity to request capacity if they have not been assigned it through demand assignment. The average delay rises more quickly than the video delay did, as the number of voice connections that make up the same capacity requirement as the video was much larger, reducing the frequency of receiving free assigned slots.

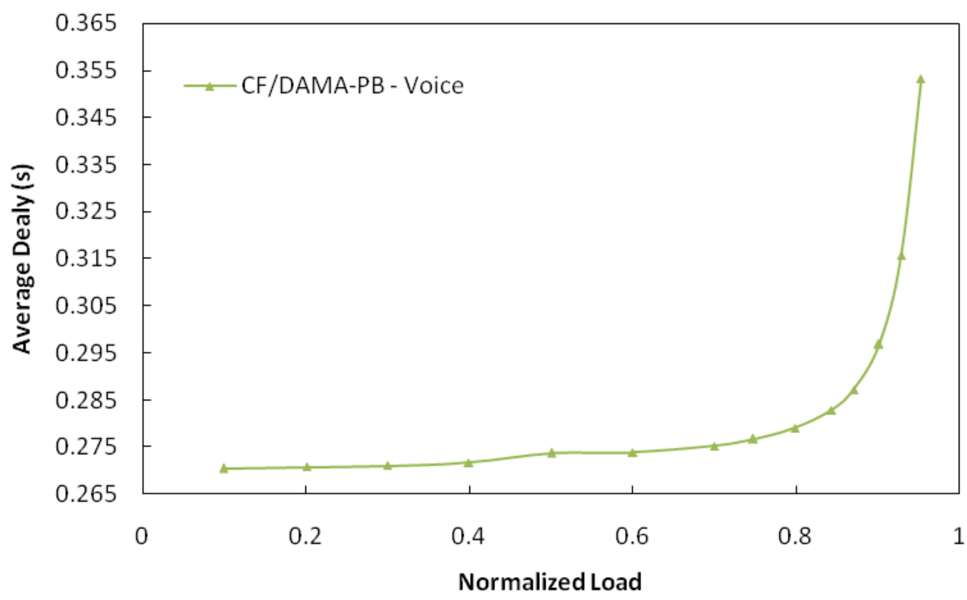

Figure 55: Average Voice Delay vs. Normalized Load (CFDAMA-PB – Scenario 2).

The throughput of the voice traffic was also excellent in the loss-free channel situation, as seen in Figure 56.

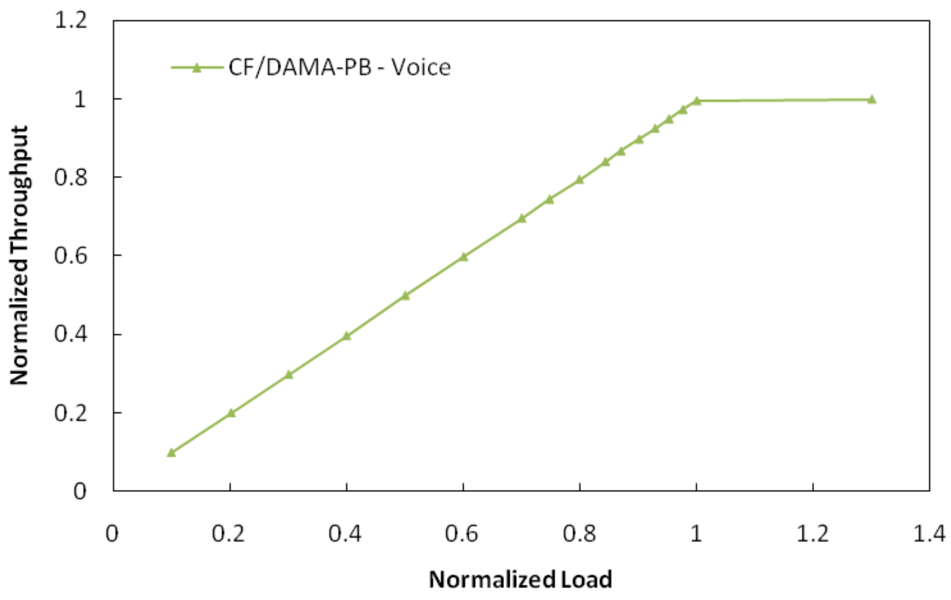

Figure 56: Normalized Load vs. Normalized Load (CFDAMA-PB – Scenario 2).

One can more clearly see the throughput performance by looking at the difference between the generated and the successful packet transmission, as in the video case, seen in Figure 57. The throughput was less than that of scenario 1 as in this scenario, voice has far greater number of connections for the same capacity thus at higher loads the number of free assigned slots will be far less than in scenario 1 thus reducing throughput overall and for smaller loads than scenario 1.

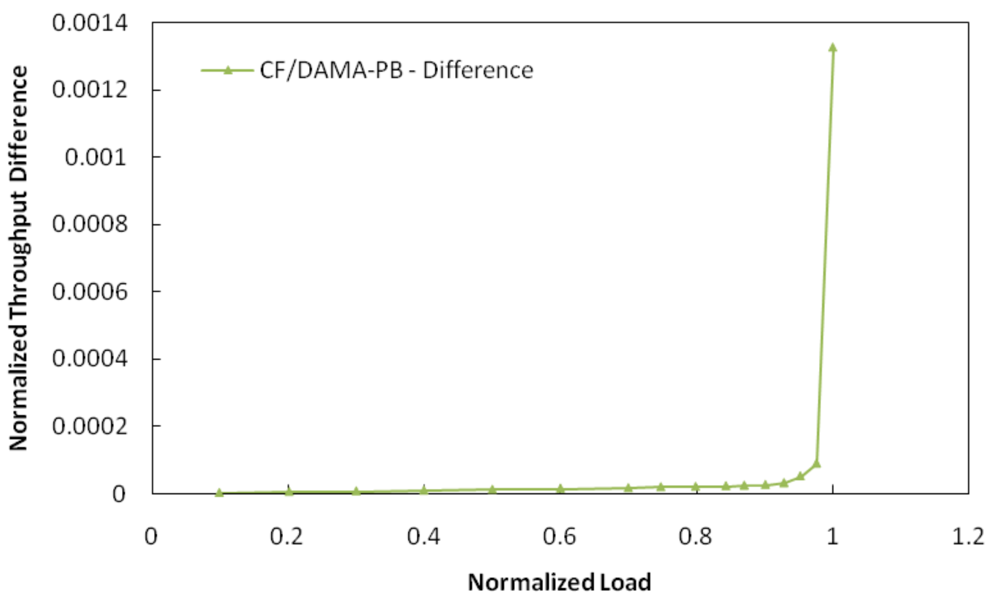

Figure 57: Voice throughput difference between successful packet transmission and generated packets.

Scenario 3

In scenario 3, a web traffic set of sources was used without voice and video traffic classes; this was to investigate the system performance when presented with a purely web traffic load.

The average packet delay of the web traffic was presented in Figure 58. The graph depicts a low average delay at low loads but as opposed to voice and video, immediately increases as the load increases. This was due to a substantially lower number of free assigned slots than web connections for the same capacity as that in voice and video. Packet size of each traffic type was dictated by the packet size of an MPEG packet. Varying packet size does not affect the system. However the number of packets does affect the system. As a result, the number of free assigned slots decrease far sooner, in terms of load, than seen in the voice and video scenarios above.

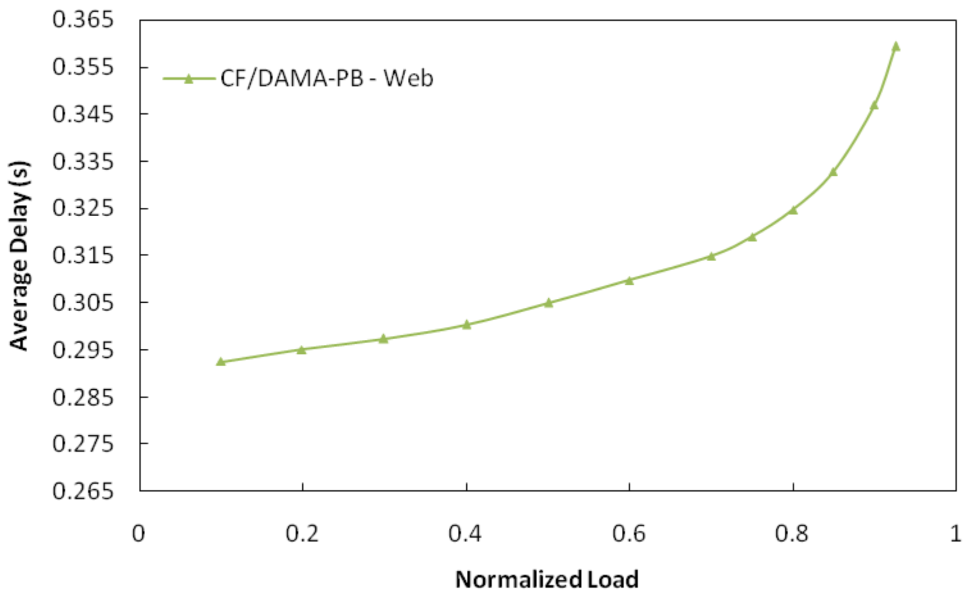

Figure 58: Average Web Delay vs. Normalized Load (CFDAMA-PB – Scenario 3).

The throughput of the web traffic was also excellent in the loss-free channel situation, as seen in Figure 59. The throughput was affected by the large number of web connections, which increases the number of requests needed at lower loads.

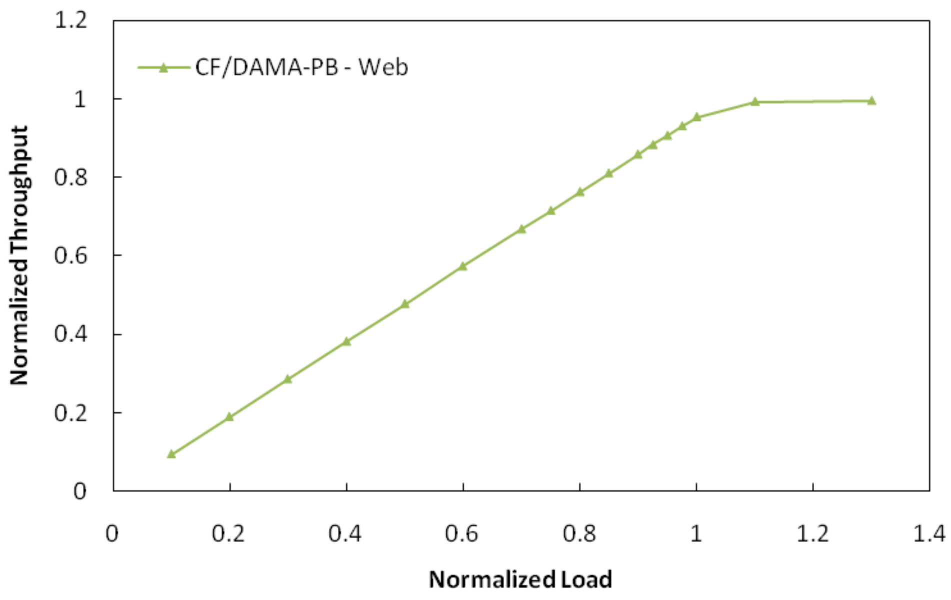

Figure 59: Normalized Load vs. Normalized Load (CFDAMA-PB – Scenario 3).

Web has a small mean bit-rate, thus requiring far more connections to make up the same capacity as the voice and video do with relatively small number of connections. This places pressure on the CFDAMA-PB system in that there were far fewer slots in a frame than there are number of web connections, resulting in less opportunity for free assigned slots and more frequent requests being made in the assigned slot allocation. The large Reading state periods in web traffic adversely affect throughput.

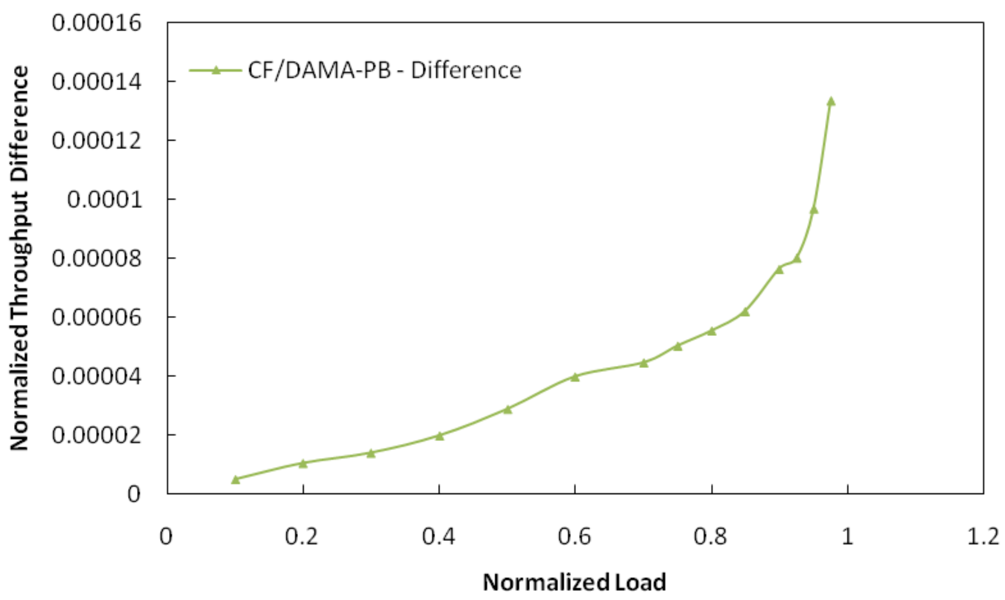

Figure 60: Web throughput difference between successful packet transmission and generated packets.

The following graph, Figure 61, clearly shows that the difference between generated and successful packets increase, i.e. throughput decreases, as the number of connections increase.

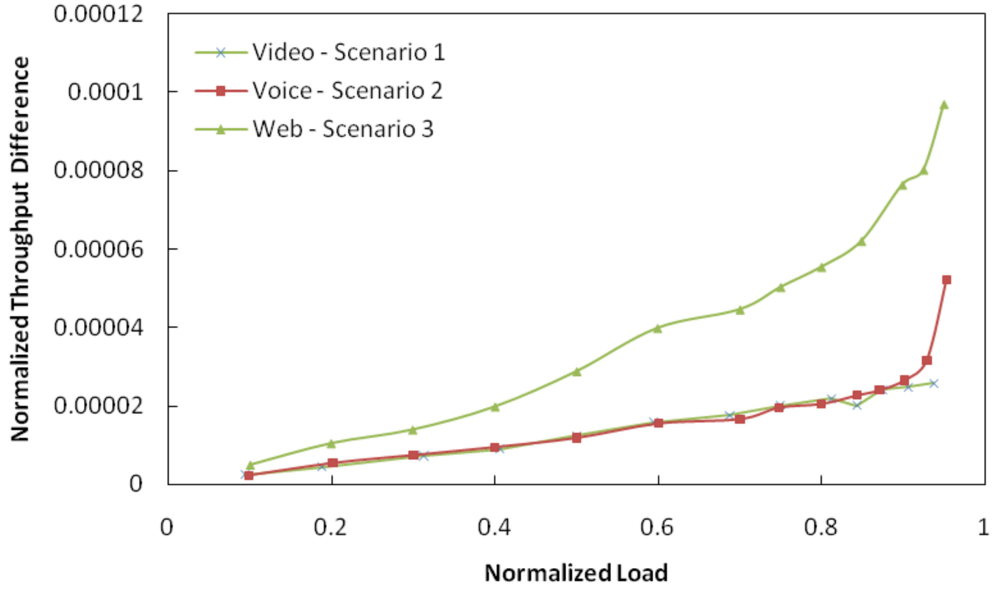

Figure 61: Throughput difference between successful packet transmission and generated packets comparison of scenarios 1-3. CF/DAMA-PB

Scenario 4

In scenario 4, the traffic was evenly divided between voice, video and web traffic classes, i.e. 1/3 voice, 1/3 web and 1/3 video traffic making up the total offered traffic load.

The average packet delay of the scenario 4 traffic is presented in Figure 62. The three traffic classes, each having equal contributions in terms of offered data rate, have different average packet delays. The voice delay at low load was better than that of the video traffic and substantially better than that of the web traffic. At higher load, loads above 0.8, the video traffic performs better than that of the voice traffic; this was due to there being a greater number of voice connections sharing the free assigned slots across all 128 RCSTs as opposed to in the video connections being far less than the number of RCSTs. One reason why the web traffic performs significantly worse than the voice and video across the offered load was that it had substantially greater number of connections causing far less regular successive assignments. The video and voice connections were fewer and hence had more regular allocation of capacity.

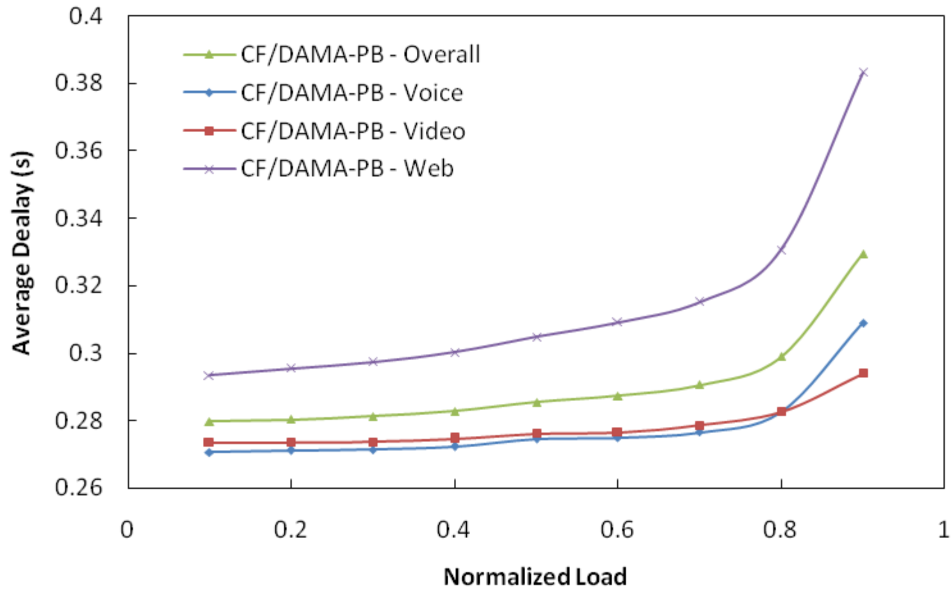

Figure 62: Average Delay vs. Normalized Load (Scenario 4). CF/DAMA-PB

The CF/DAMA-PB protocol seemed to perform in a similar fashion in multimedia traffic as it did in homogeneous traffic types. In throughput, CF/DAMA-PB performs well over the whole range of offered load. As expected, the traffic scenario dictated the fraction of contribution a traffic source had in the multimedia traffic source throughput. This was seen in Figure 63. Each traffic class was contributing a third of the total throughput, varying slightly at traffic loads greater than one.

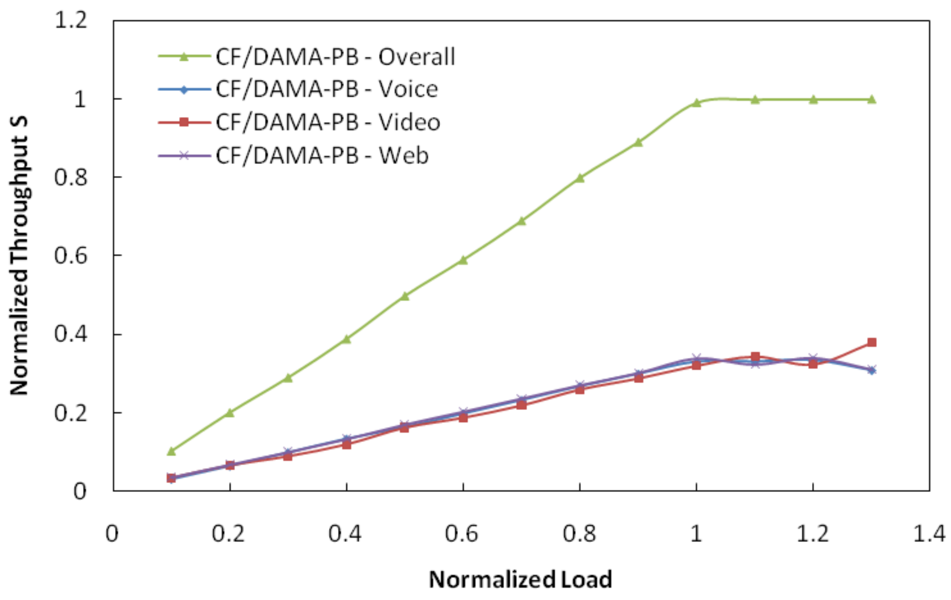

Figure 63: Normalized Throughput vs. Normalized Load (Scenario 4). CF/DAMA-PB

Scenario 5

In scenario 5, the traffic was divided between voice, video and web traffic classes making the video a dominant traffic class in terms of offered load, i.e. 1/6 voice, 1/6 web and 2/3 video traffic making up the total offered traffic.

Figure 64 shows that the individual traffic classes have very similar average packet delays to that in scenario one, but due to the video class offering a greater percentage of traffic to the other classes and having lower average packet delay, it thus reduced the overall average packet delay significantly. This indicates that each traffic class was not significantly affected by each other in the CF/DAMA-PB scheme and that the characteristics of the dominant traffic class became the dominant factor affecting overall system delay performance.

In Figure 56, the system throughput was also representative of the traffic contribution between 0.1 and 1.0 after which the video traffic started to dominate the throughput placing voice and web traffic below their expected throughput, this was due to the arrival rate of packets from each traffic class, the least frequent being that of the web traffic class, thus having the least throughput, then the voice and finally the video arrival rate was so significantly larger than the other two traffic classes causing them to suffer as a result.

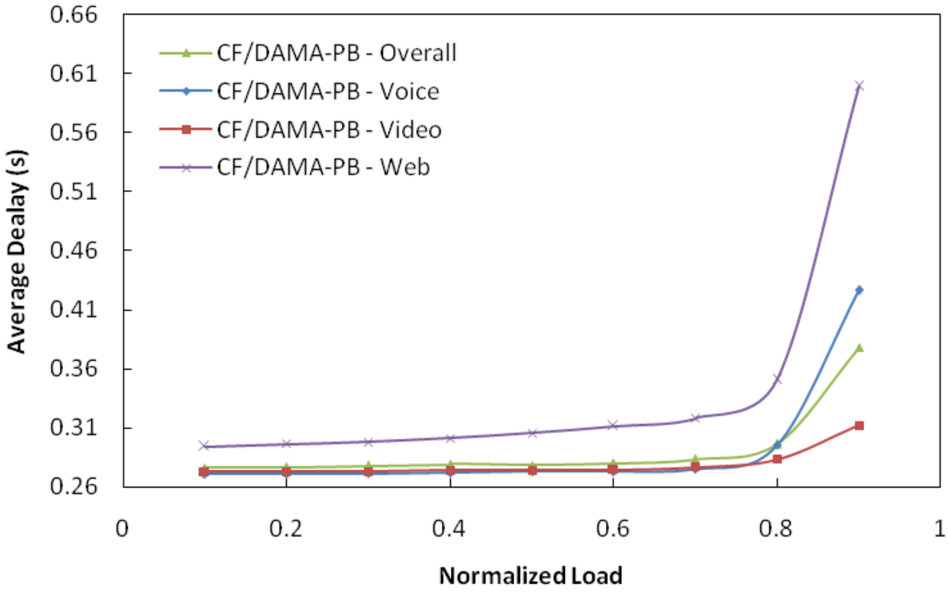

Figure 64: Average Delay vs. Normalized Load (Scenario 5) CF/DAMA-PB

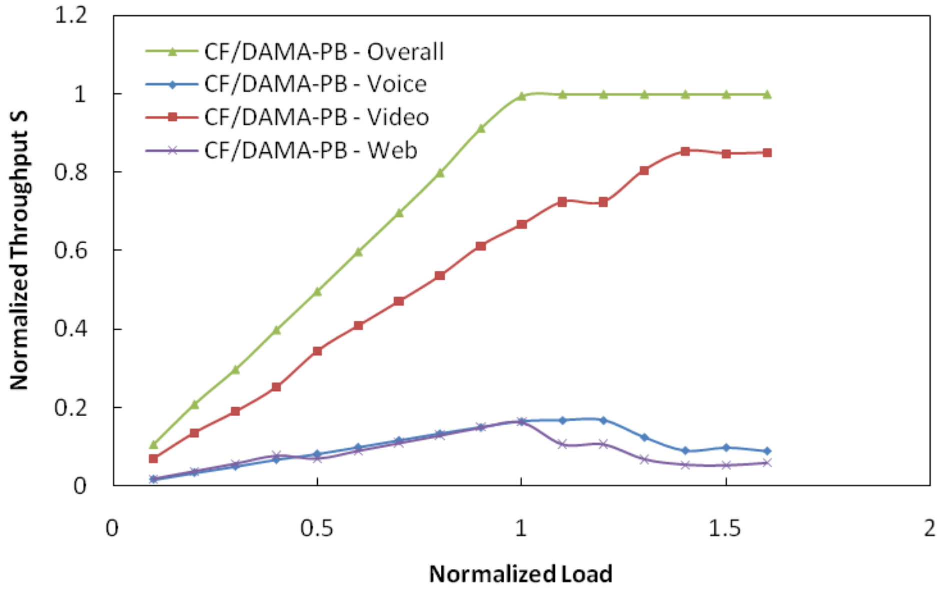

Figure 65: Normalized Throughput vs. Normalized Load (Scenario 5) CF/DAMA-PB

3.9.2 CF/DAMA-PB-PD Protocol

The CF/DAMA-PB-PD Protocol aims to reduce the average packet delay in the system for traffic classes such as the voice and video. All five traffic scenarios will be presented and compared to the results of the CF/DAMA-PB protocol to determine if any improvements have been made on that of the CF/DAMA-PB scheme.

Scenario 1

In scenario 1, a video traffic set of sources was used without voice and web traffic classes; this was to investigate the system performance when presented with a purely video traffic load.

Figure 66 depicts the difference in the CF/DAMA-PB-PD protocol and the CF/DAMA-PB protocol. The packet dropping due to stringent deadlines on video packets significantly improved the average packet delay starting at a 0.5 load. The deadline created an upper bound for loads greater than 1.0 compared to the increasing average packet delay experienced in the CF/DAMA-PB scheme.

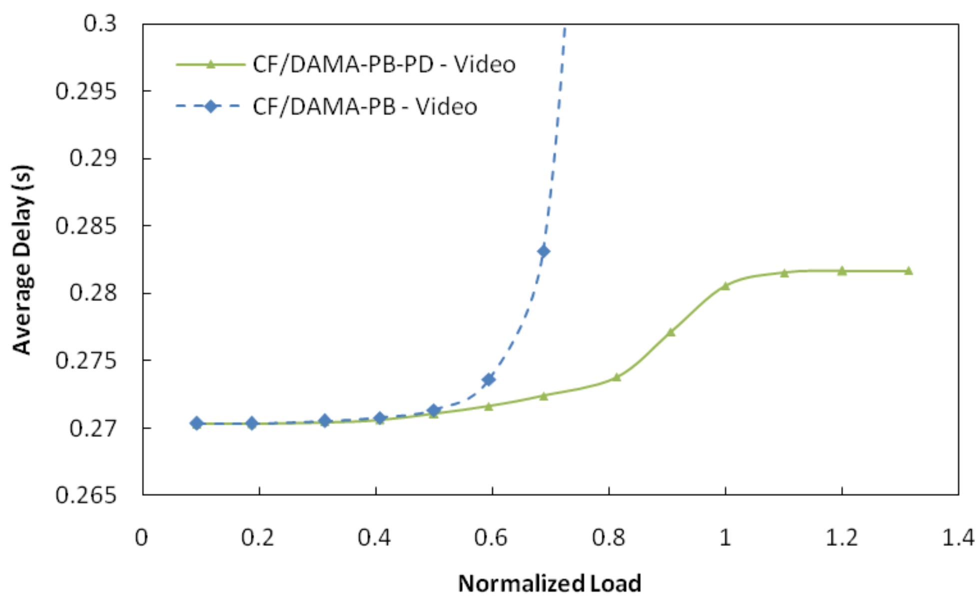

Figure 66: Average Video Delay vs. Normalized Load (CFDAMA-PB-PD – Scenario 1).

The throughput, depicted in Figure 67, showed a significant decrease in the CF/DAMA-PB protocol. This decrease can be attributed to the increase in packet dropping as a result of stringent deadlines on video. Above loads of 0.9 the throughput performance of the system decreased significantly.

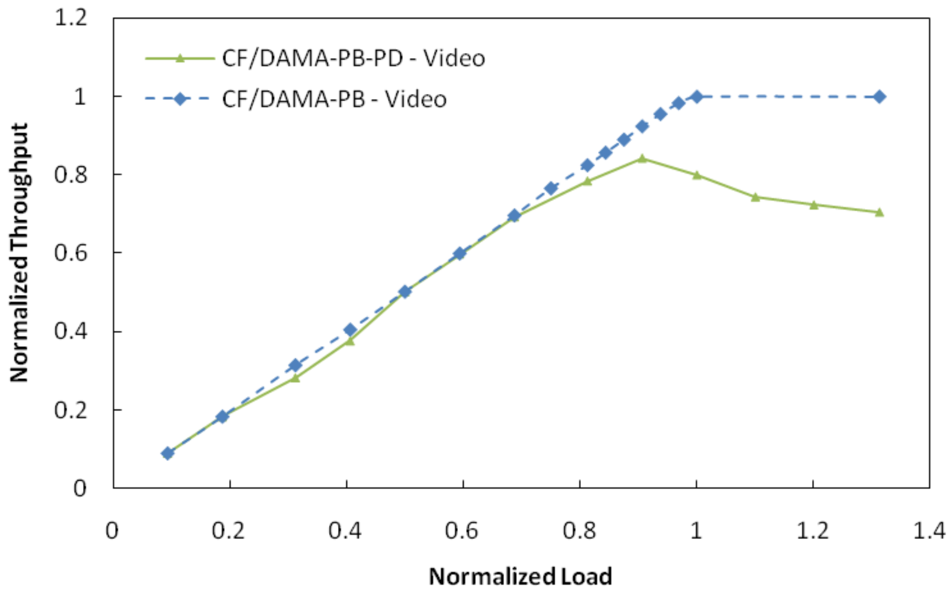

Figure 67: Normalized Throughput vs. Normalized Load (CFDAMA-PB-PD – Scenario 1).

Scenario 2

In scenario 2, a voice traffic set of sources is used without video and web traffic classes; this is to investigate the system performance when presented with a purely voice traffic load.

Figure 68 depicts the difference in the CF/DAMA-PB-PD protocol and the CF/DAMA-PB protocol. The packet dropping due to stringent deadlines on voice packets significantly improves the average packet delay starting at a load of 0.6. The deadline creates an upper bound for loads greater than 1.2 compared to the increasing average packet delay experienced in the CF/DAMA-PB scheme. Due to the relatively large number of connections compared to the video traffic scenario 1 and the less strict deadline, the performance will vary and hence have a higher upper limit to the average packet delay experienced.

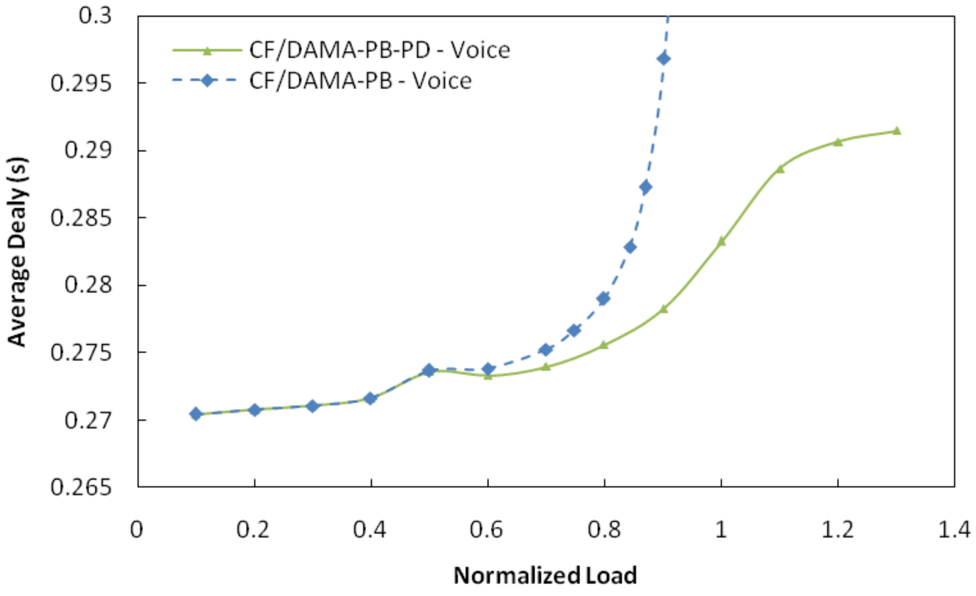

Figure 68: Average Voice Delay vs. Normalized Load (CFDAMA-PB-PD – Scenario 2).

The throughput, depicted in Figure 69, showed a small decrease compared to the CF/DAMA-PB protocol. This decrease can be attributed to the increase in packet dropping as a result of the deadlines on voice packets. Above loads of 0.8 the throughput performance of the system decreases. The better throughput experienced here compared to the video traffic scenario 1 can be attributed to the number of sources being greater in voice traffic for the same bit-rate and the extra frame period in the deadline which allowed for less packets being dropped.

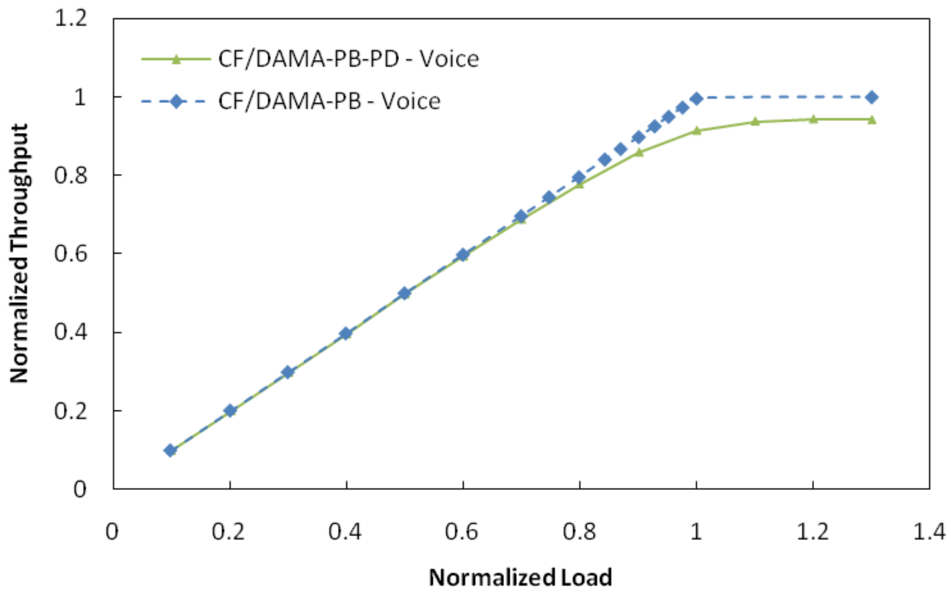

Figure 69: Normalized Throughput vs. Normalized Load (CFDAMA-PB-PD – Scenario 2).

Scenario 3

In scenario 3, a web traffic set of sources was used without voice and video traffic classes; this was to investigate the system performance when presented with a purely web traffic load.

As can be seen in Figure 70, there was little to no difference when comparing the CF/DAMA-PB-PD protocol with the CF/DAMA-PB protocol as the deadlines of the web traffic source are virtual and hence do not cause any packets to be dropped. Without any dropped packets, the delay will not be reduced and hence stays the same as expected.

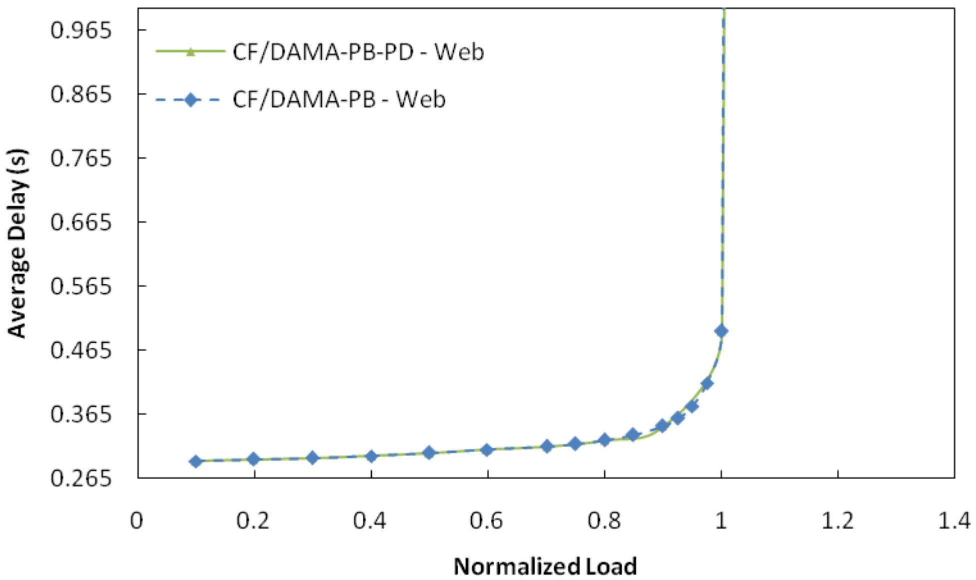

Figure 70: Average Web Delay vs. Normalized Load (CFDAMA-PB-PD – Scenario 3).

We can see in Figure 71 below, the performance of the two protocols were similar. This is attributed to the fact that packet dropping only affects the voice and video traffic classes which were not present in this scenario. The web traffic sources will benefit in a heterogeneous traffic source, Scenarios 4 and 5, as the web packets can benefit from the dropped packets of the voice and video traffic classes.

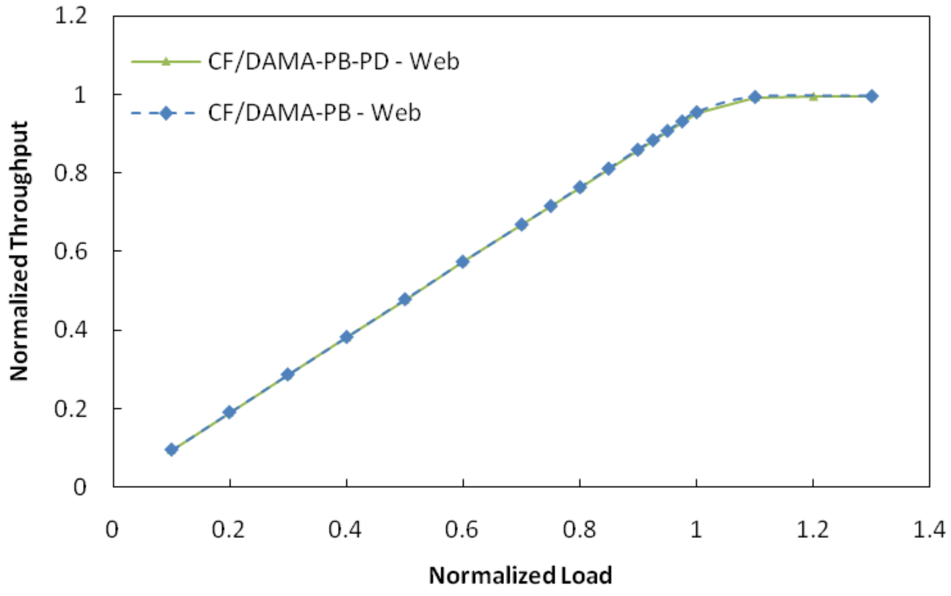

Figure 71: Normalized Throughput vs. Normalized Load (CFDAMA-PB-PD – Scenario 3).

Scenario 4

The average packet delay of the system in scenario 4 is depicted in Figure 72 and a comparison to the average packet delay of CF/DAMA-PB is given in Figure 73. From these two figures, it was clear that the average packet delay decreases below that of the CF/DAMA-PB scheme as the offered load increases.

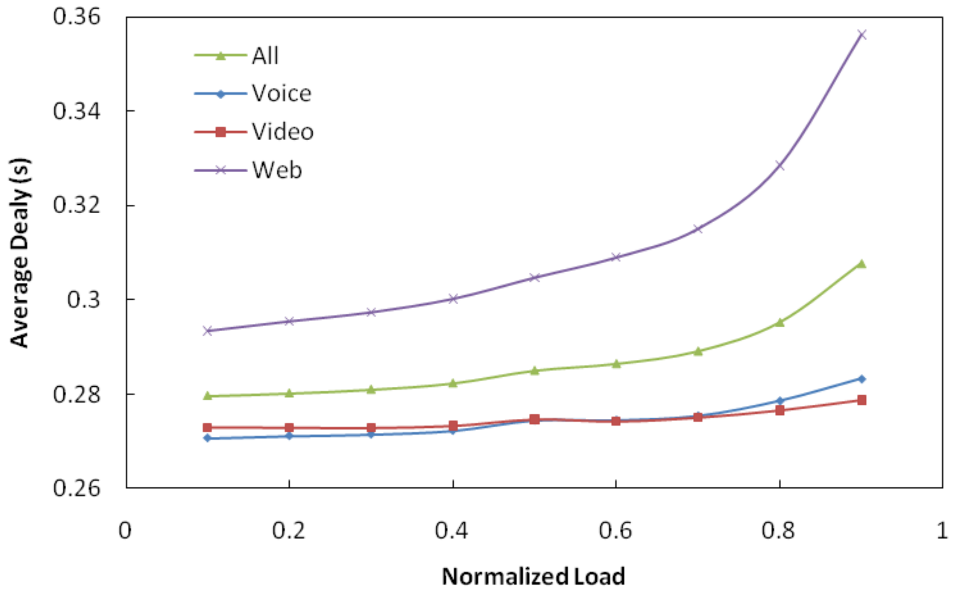

Figure 72: Average Delay vs. Normalized Load (Scenario 4) - CF/DAMA-PB-PD.

Once the load increased above an offered load of 1.0, the deadlines of the voice and video packets created a limiting factor or upper delay bound. The CFDAMA-PB scheme seemed to hit a wall at around 0.95 offered load as a result of too many packets being generated in the system than can be accommodated for, thus queues increase faster than they are serviced. In the packet dropping scheme, as packets that miss deadlines were dropped, allocated slots that were not required but were used by the traffic nevertheless benefit the overall system performance further by ensuring less packets were dropped than if one had to drop requests to compensate for 'over' requesting slots.

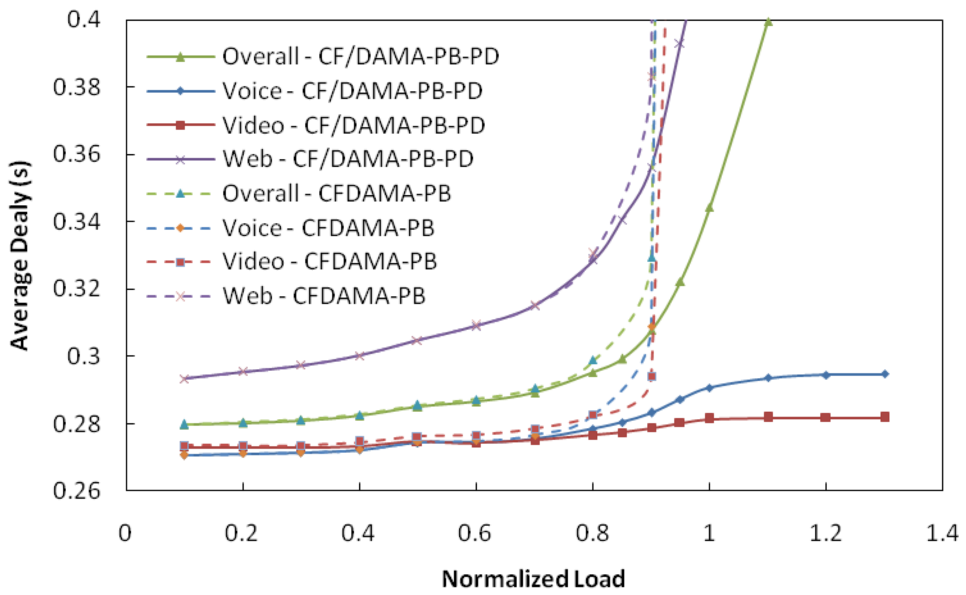

Figure 73: Average Delay vs. Normalized Load (Scenario 4) - CF/DAMA-PB-PD.

The average packet delay was significantly reduced above a load of 0.8, but at the cost of throughput, as can be seen in Figure 74 below. The increasing number of dropped packets as the load increases above 0.7 significantly reduced the overall system throughput. This is due to larger numbers of packets being added to the queue, i.e. arrival rate increases, but having a fixed service rate upper bound which was below the arrival rate. This in turn results in greater average packet delays and thus a greater number of packets exceeding the traffic class deadlines.

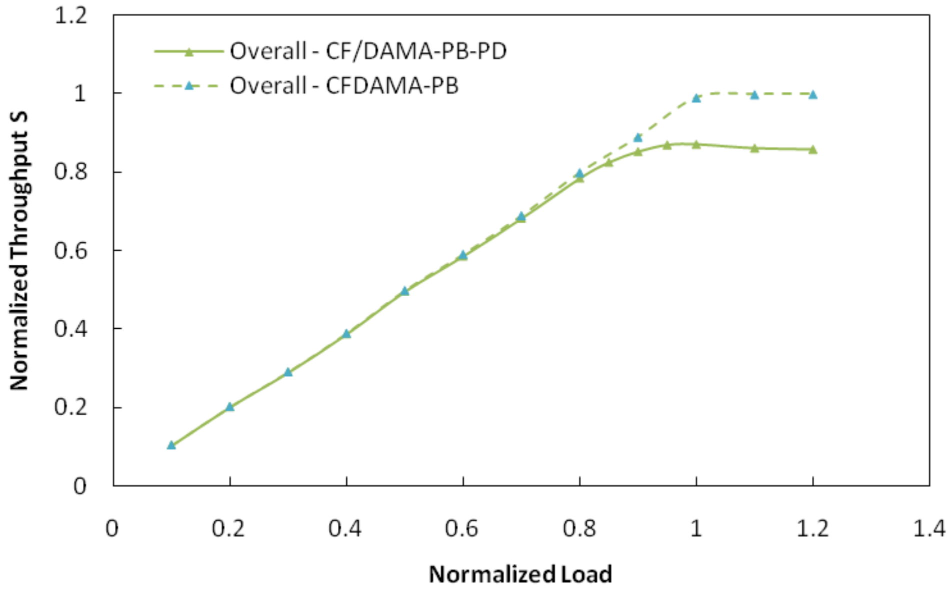

Figure 74: Normalized Throughput vs. Normalized Load (Scenario 4) - CF/DAMA-PB-PD.

In Figure 75 the traffic classes that contributed to the overall system throughput are depicted. The web traffic class's throughput was not affected by the dropping of the voice and video class packets and thus had similar, almost identical, throughput behaviour as in the CF/DAMA-PB scheme. The packet dropping of voice and video packets did however reduce the throughput of these classes and hence affect the overall throughput. Above a load of 0.8 the throughput of the video and voice classes started to decrease and above a load of 0.9 they decrease quite significantly, as mentioned before this was due to high arrival rates, low service rates and an increasing average packet delay exceeding the class deadlines as a result.

Another important measure was that of the percentage of dropped packets in the system. This determines the QoS that can be supported by the system. In Figure 76 the voice and video percentage packet dropping is depicted. The percentage of dropped packets is the number of packets dropped with respect to the packets generated in a class. For the video class, this is the dropped video packets as a percentage of the generated video packets, and for the voice class, this is the dropped voice packets as a percentage of the generated voice packets. The percentages for both voice and video were kept below 3 % for loads under 0.85, which was good and ensured an acceptable QoS for the traffic classes while reducing the average packet delay.

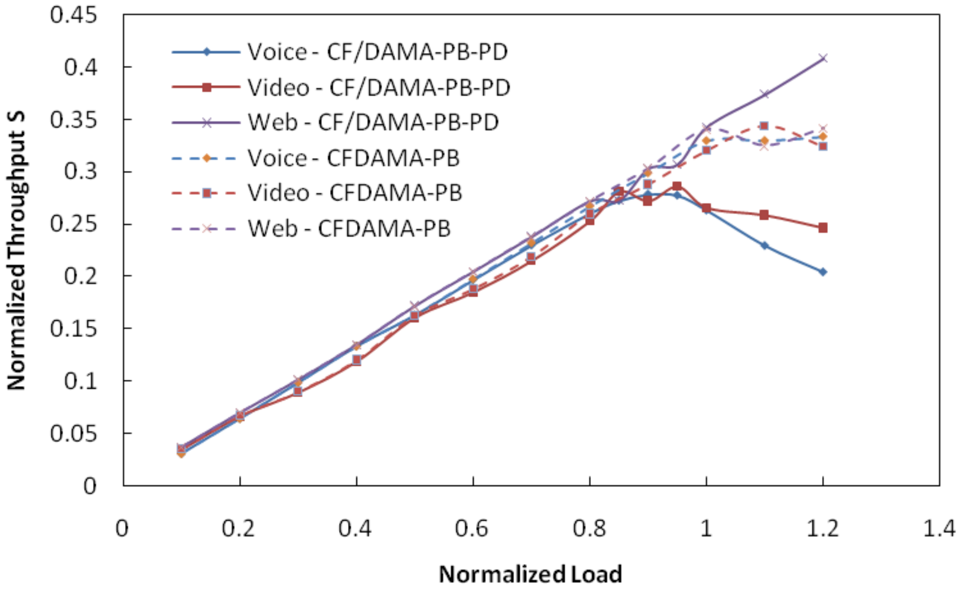

Figure 75: Normalized Throughput vs. Normalized Load (Scenario 4) - CF/DAMA-PB-PD.

The packet dropping was seen to affect the video class for lower loads more than the voice class, as a more stringent deadline of 1 frame is given to this class.

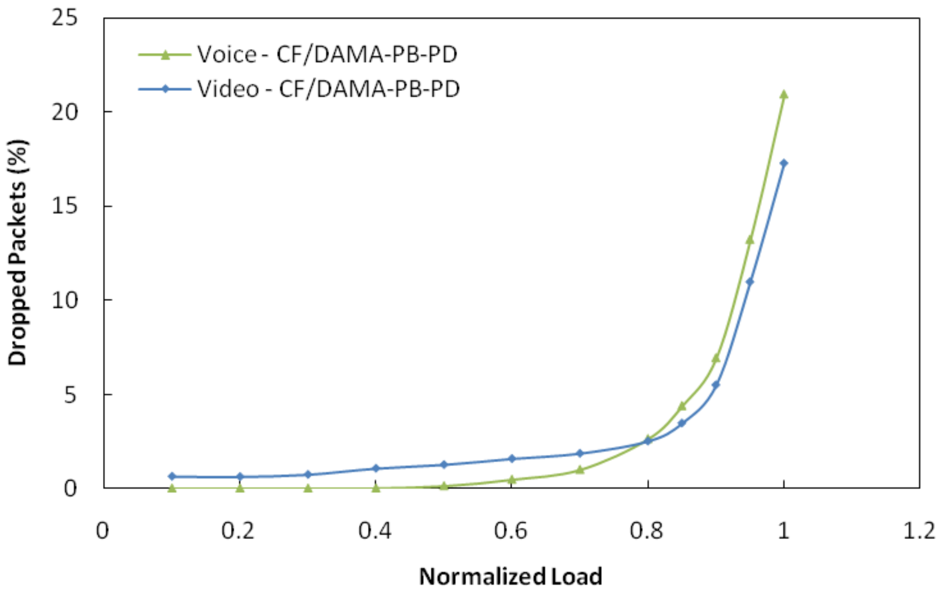

Figure 76: Percentage of dropped packets in CF/DAMA-PB-PD (Scenario 4).

Scenario 5

In scenario 5, with a larger percentage of video connections to that of voice and web, a similar average packet delay graph is achieved in Figure 77 to that in Figure 73, scenario 4.

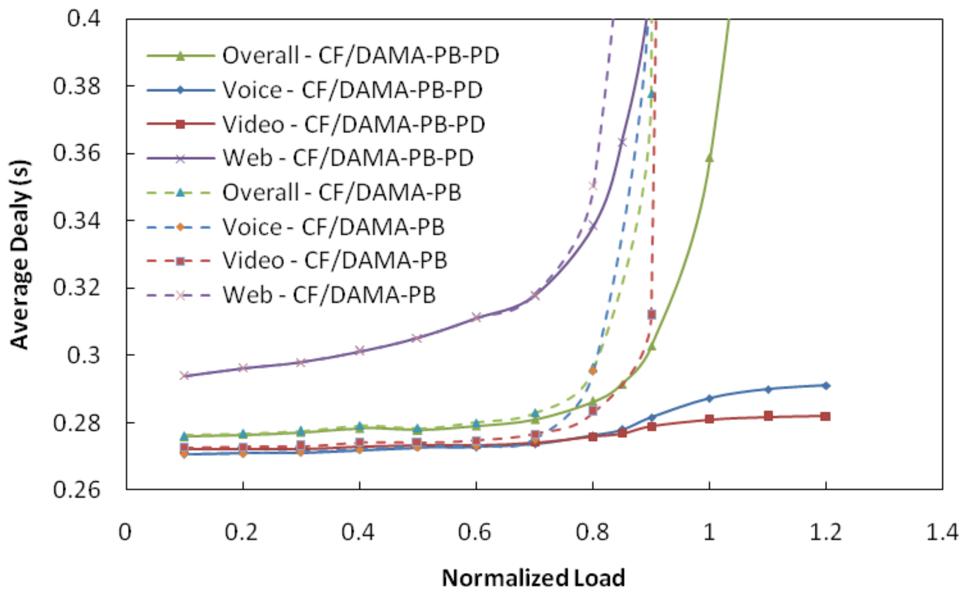

Figure 77: Average Delay vs. Normalized Load (Scenario 5) - CF/DAMA-PB-PD.

Once again the voice and video average packet delays were limited by the deadline for the respective classes while the web benefits from their dropped packets and thus extra slot allocation. This noticeably reduces the overall delay above loads of 0.6.

Due to there being a large ratio of video data compared to the voice and web data in scenario 5, the video contribution to packet dropping will be relatively large too. This will impact on the overall throughput by dropping more packets; therefore less generated packets will be transmitted, reducing the number of successfully transmitted packets. The result of this was seen in Figure 78 and Figure 79.

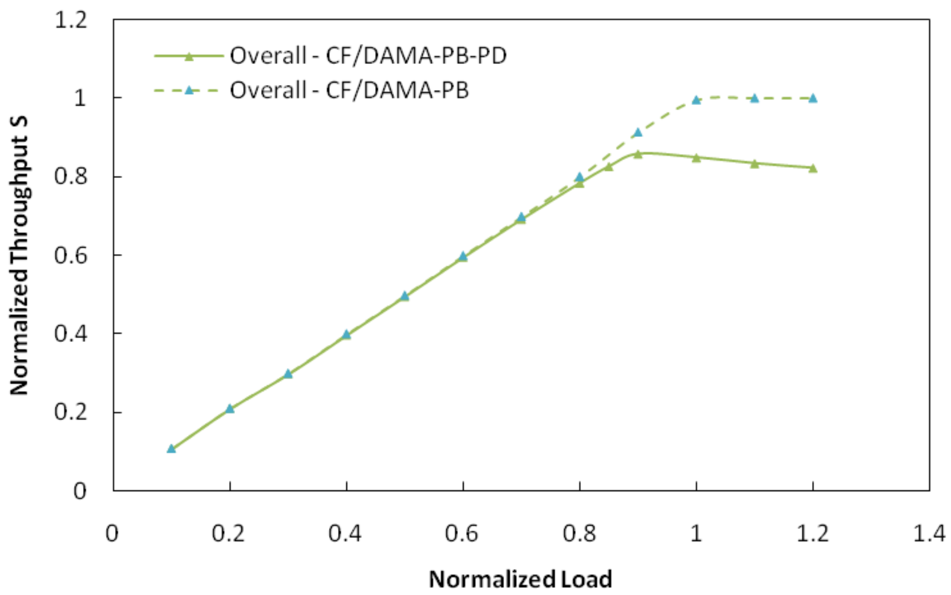

Figure 78: Normalized Throughput vs. Normalized Load (Scenario 5) - CF/DAMA-PB-PD.

Figure 79 depicts the individual traffic class' normalized throughput vs. Normalized load which indicate that scenario 5 was being used as the video class contributes 2/3 of the overall throughput below a load of 0.8 while the other two classes have 1/6 contributions to throughput each. Above this load packet dropping started to noticeably increase its impact on the traffic throughput. The throughput of web traffic was increased as a result of the slot allocations made available by dropped packets. Both the voice and video had deceased throughput compared with the CF/DAMA-PB scheme due to dropped packets.

Similar to scenario 4, the percentage of dropped packets was the number of packets dropped with respect to the packets generated in a class. For the video class, this was the dropped video packets as a percentage of the generated video packets, and for the voice class, this was the dropped voice packets as a percentage of the generated voice packets. The percentage of dropped packets is depicted in Figure 80. Again, the 3% packet dropping below 0.85 was maintained and due to the more stringent deadline on video, the packet dropping was higher at low loads.

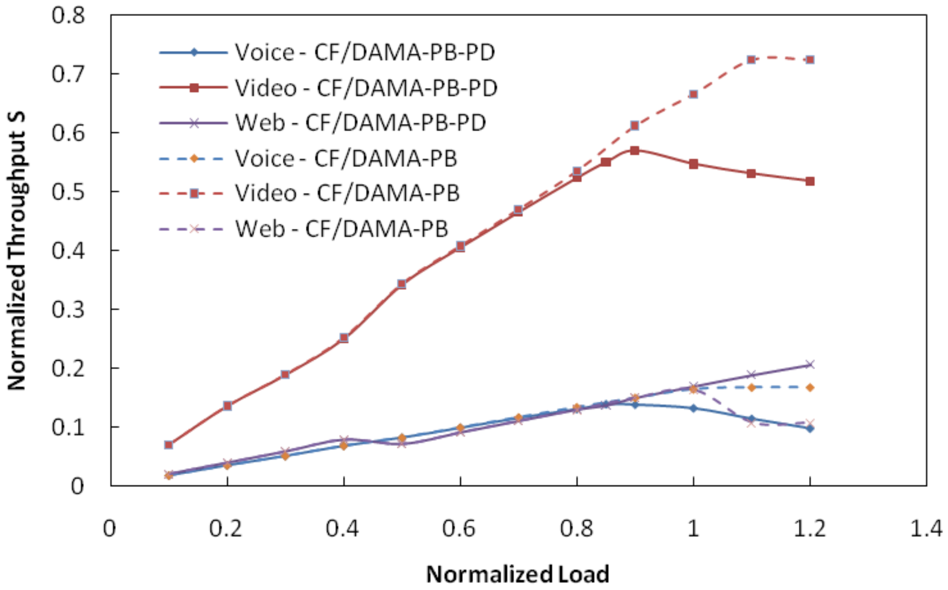

Figure 79: Normalized Throughput vs. Normalized Load (Scenario 5) - CF/DAMA-PB-PD.

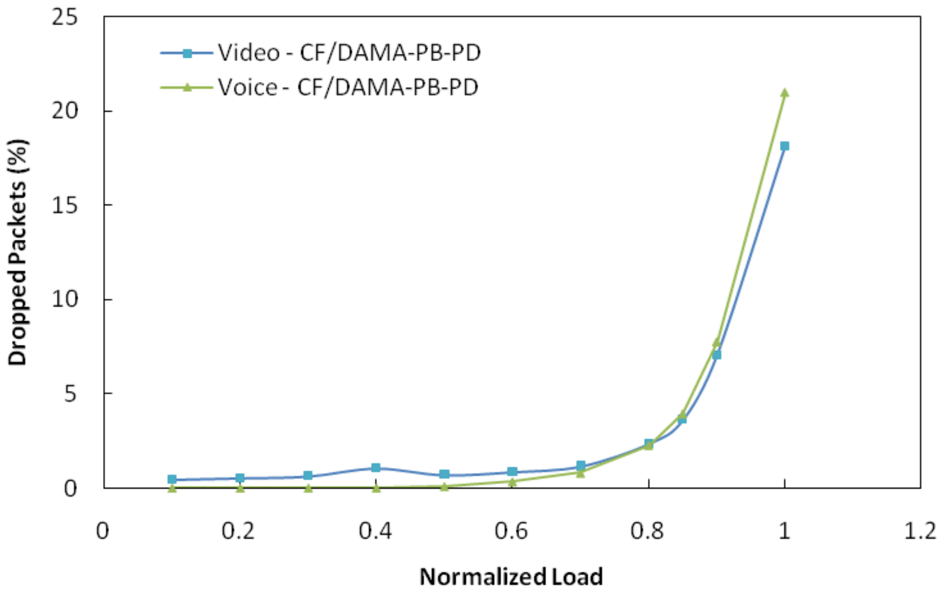

Figure 80: Percentage of dropped packets CF/DAMA-PB-PD (Scenario 5).

3.9.3 CF/DAMA-PB-PEDF Protocol

CFDAMA-PB-PEDF ensured that all deadlines are kept given that it was possible to keep the deadlines. The CFDAMA-PB-PEDF protocol thus will ensure that if any packet deadlines were missed that could have been avoided in the CFDAMA-PB-PD protocol, then this protocol will take care of it and in so doing reduce the number of packets being dropped.

Scenario 4

In scenario 4, the average packet delay for the CFDAMA-PB-PEDF protocol is shown in Figure 81.

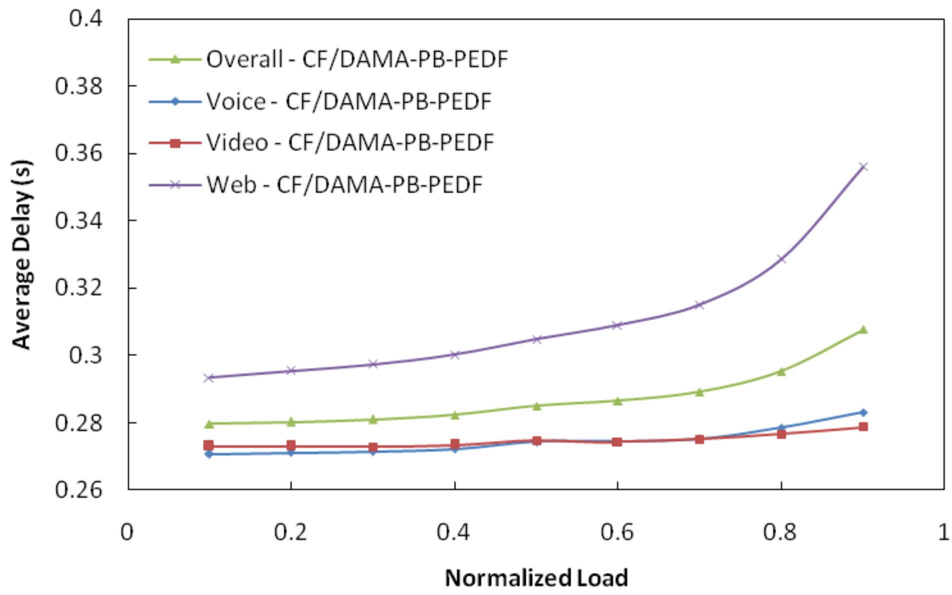

Figure 81: Average Delay vs. Normalized Load (Scenario 4) – CF/DAMA-PB-PEDF.

A comparison of the CFDAMA-PB-PD protocol and the CFDAMA-PB-PEDF protocol is far more useful in determining the performance of the CFDAMA-PB-PEDF over CFDAMA-PB-PD, as depicted in Figure 82. From the figure, it was quite clear that the CFDAMA-PB-PEDF protocol had made no delay improvements on the CFDAMA-PB-PD Protocol, as the graphs were identical. The stringent deadlines for the voice and video classes leave very little room for much, if any, improvement on packet dropping and average delay. Once a packet was allocated a slot and placed on the channel within its deadline, all other packets that had not been placed had to be dropped in the case of video and in the case of voice had one frame more to try and gain a slot or else they too had to be dropped. This left too little room for CFDAMA-PB-PEDF to help improve on performance.

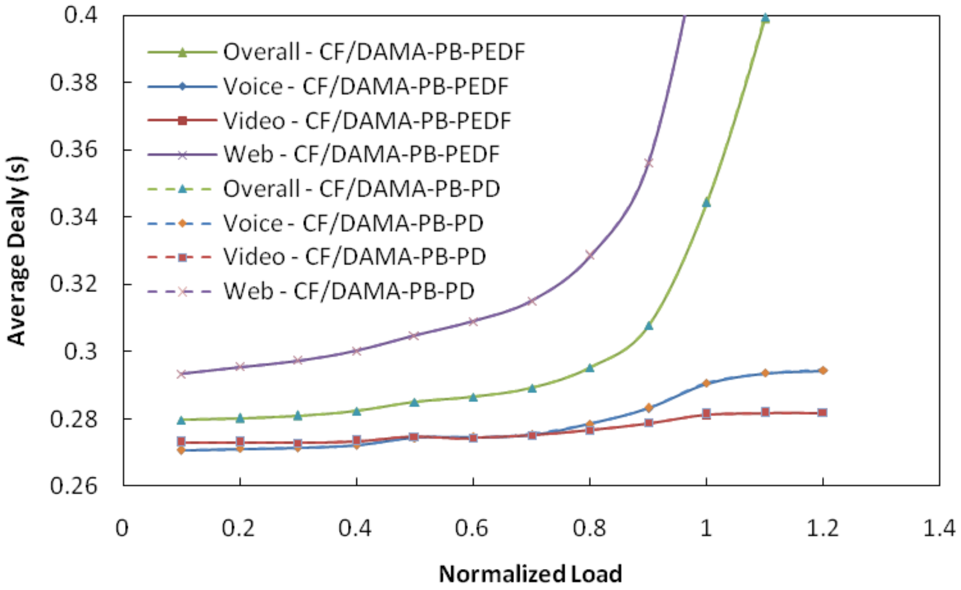

Figure 82: Average Delay vs. Normalized Load (Scenario 4) – CF/DAMA-PB-PEDF.

The throughput was also shown to back the reasoning given, i.e. that the deadlines had very little impact on the effectiveness of the CFDAMA-PB-PEDF protocol. Figure 83 and Figure 84 depict no increase in throughput and hence no performance benefits from the CFDAMA-PB-PEDF protocol.

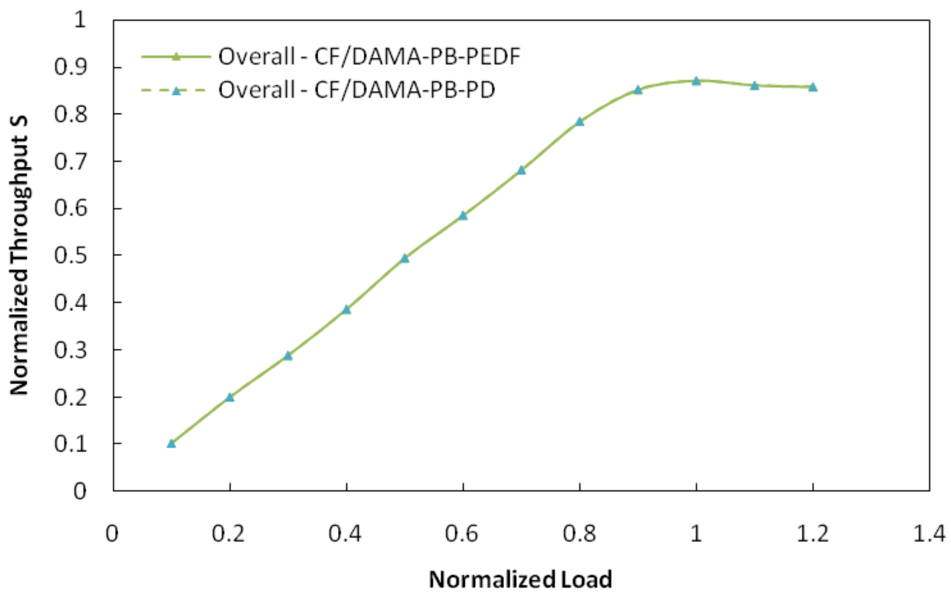

Figure 83: Normalized Throughput vs. Normalized Load (Scenario 4) – CF/DAMA-PB-PEDF.

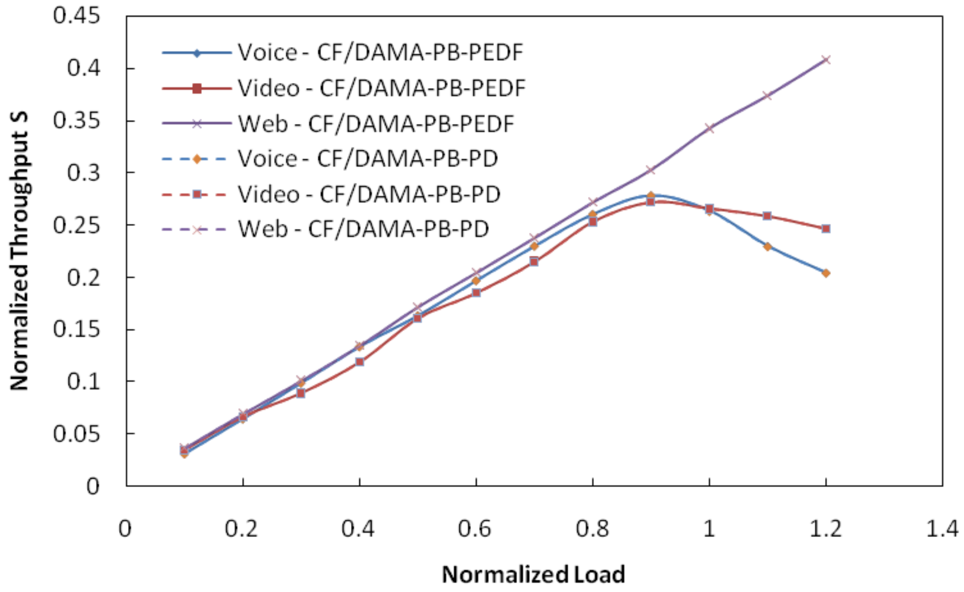

Figure 84: Normalized Throughput vs. Normalized Load (Scenario 4) – CF/DAMA-PB-PEDF.

From the results of throughput one can be assured that the percentage of dropped packets would also remain the same as in the CFDAMA-PB-PD protocol. The protocols were similar in performance.

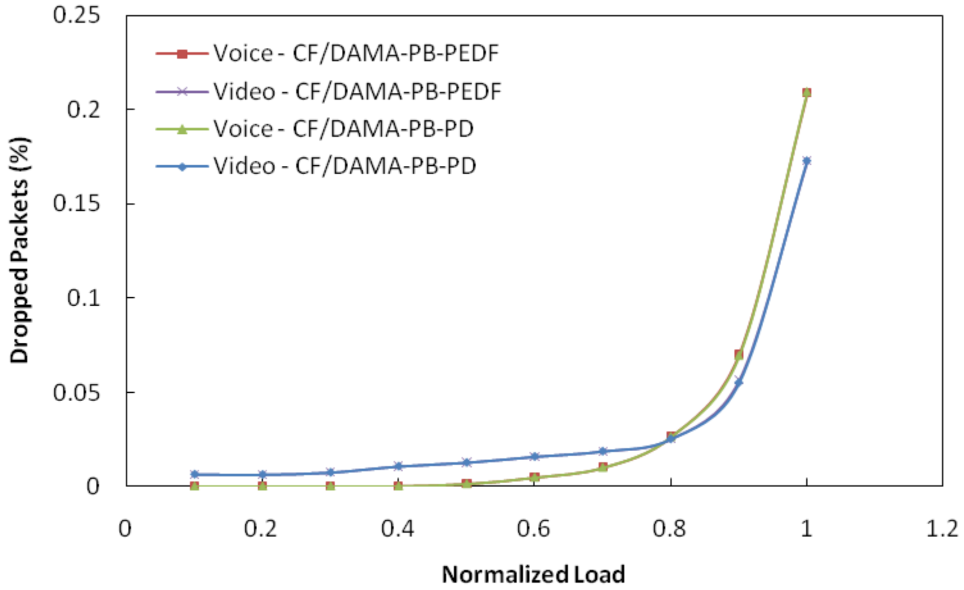

Figure 85: Percentage of dropped packets in CFDAMA-PB-PEDF (Scenario 4).

Scenario 5

In scenario 5 the same results were obtained as those in the CFDAMA-PB-PD protocol under scenario 5 once again showing that the deadlines create little to no flexibility for any possible benefits the CFDAMA-PB-PEDF may have introduced given some more time to play with.

The average packet delay had no improvements, see Figure 86 below. The protocols were similar in performance.

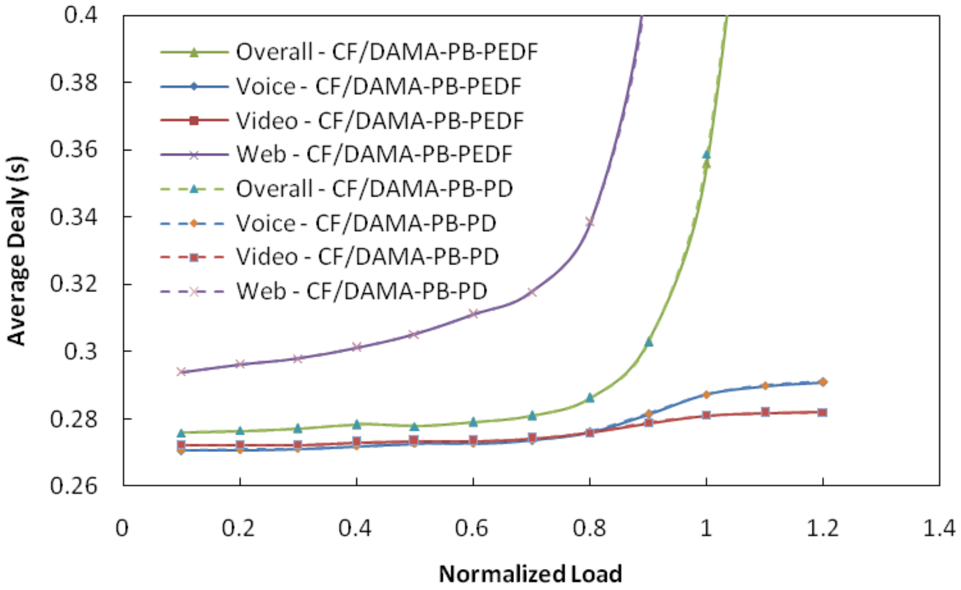

Figure 86: Average Delay vs. Normalized Load (Scenario 5) - CFDAMA-PB-PEDF.

The system throughput, as well as the individual throughput contributions of each class, had no improvements to offer on the CFDAMA-PB-PD protocol, see Figure 87 and Figure 88 below.

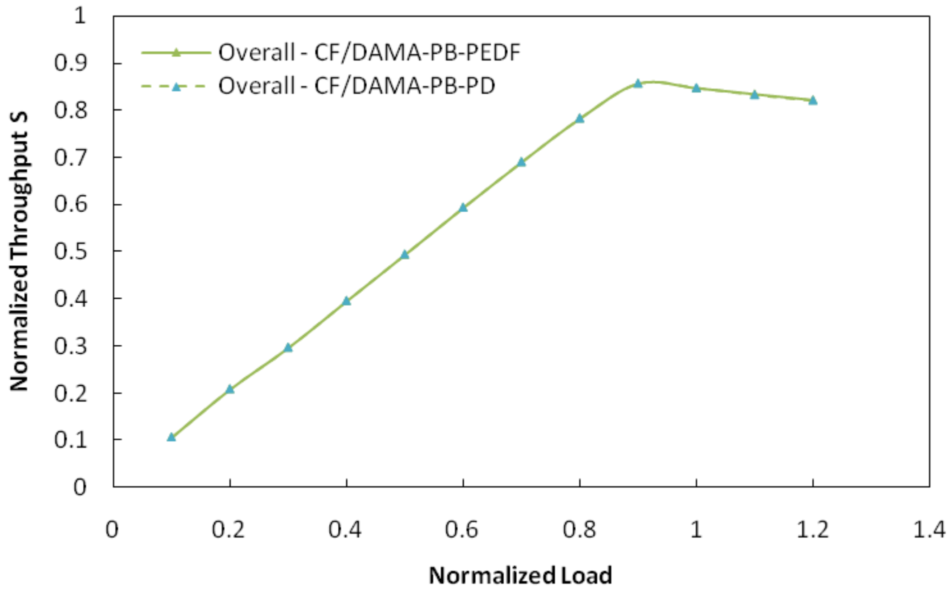

Figure 87: Normalized Throughput vs. Normalized Load (Scenario 5) - CFDAMA-PB-PEDF.

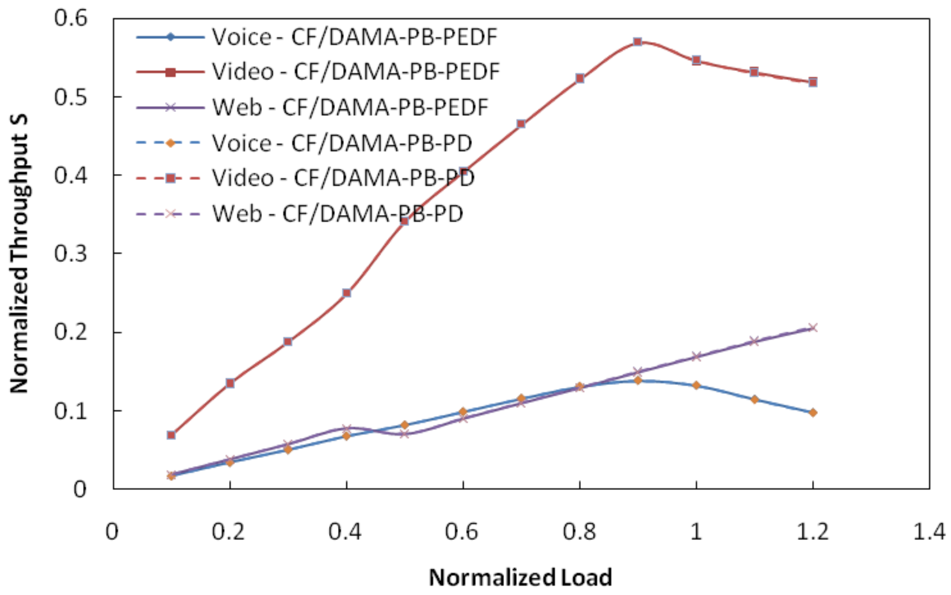

Figure 88: Normalized Throughput vs. Normalized Load (Scenario 5) - CFDAMA-PB-PEDF.

Finally, in Figure 89, once again the packet dropping for voice and video was the same as that of the CFDAMA-PB-PD protocol showing no noticeable improvements in packet dropping.

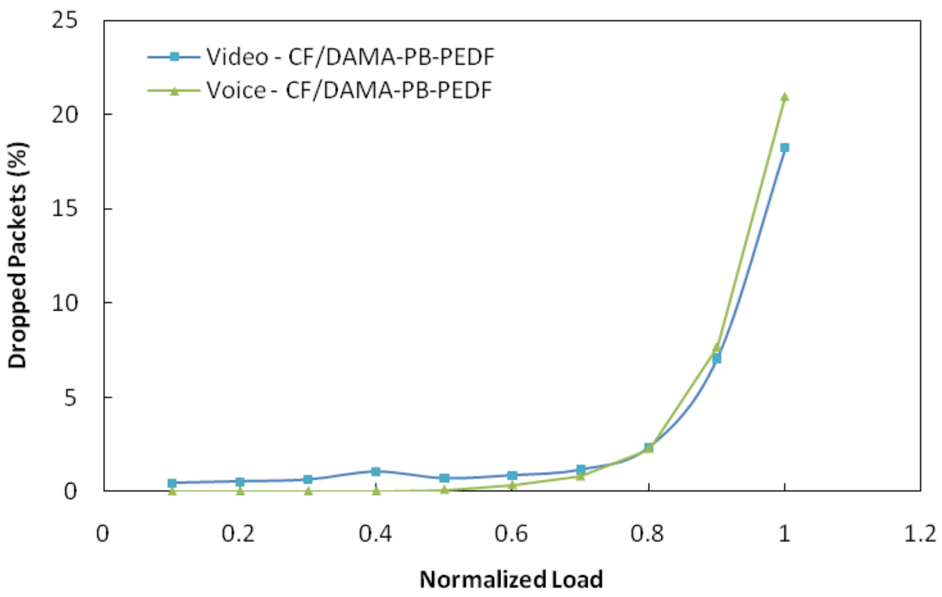

Figure 89: Percentage of dropped packets in CFDAMA-PB-PEDF (Scenario 5).

3.10 Conclusions

This chapter discussed MAC protocols used in DVB satellite communications. Two proposed protocols, the CFDAMA-PB-PD and CFDAMA-PB-PEDF, were described in detail with flow charts to simplify the explanation to some extent. The traffic models used; voice, video and web traffic models, were introduced and described ending in an attempt to ensure that the models implemented did result in the appropriate data rates and statistical distributions. The performance measures, delay and throughput, are plotted as a function of the input load and offered load. The performance measures were defined in order to ensure clarity in the explanations of the simulation results.

The performance of the CF/DAMA-PB, CFDAMA-PB-PD and CFDAMA-PB-PEDF protocols were simulated under different traffic scenarios, these being the even distribution of traffic, i.e. 1/3 voice, 1/3 video and 1/3 web traffic, the majority video traffic, i.e. 1/6 voice, 1/6 web and 2/3 video traffic, and then the homogeneous traffic scenarios of pure video, pure voice and pure web traffic. From these different scenarios the following was determined; video outperforms voice and web in average packet delay vs. Normalized load as the number of video sources required to make up the total system capacity was significantly smaller than voice and even more so than web traffic sources. This was also affected by the fact that there was always less video traffic sources than RCSTs, while the other traffic class's varied between less than and significantly greater than the number of RCSTs depending on the offered load. This ensured more free assignment being available to the video connections as opposed to the other traffic class scenarios.

The above conclusion suggests that the CFDAMA-PB-PD and CFDAMA-PB-PEDF protocols prefer fewer connections with high data rates than more connections with lower data rates for the same system capacity. In order to ensure that the system resources are used effectively while still offering the user good throughput, the system should be kept at a load of around 0.9 for optimal throughput. Hence, the optimal load is dependent on the system user's traffic ratios and each network administrator should use this as a guideline - increasing resources according to traffic ratios and offered load of system users.

The Packet CFDAMA-PB-PD was found to improve the average packet delay in the system significantly by introducing the stringent packet dropping deadlines. This ensured an upper limit for delay in the two traffic classes, i.e. video and voice, which are sensitive to delay. The percentages of packets dropped were kept relatively low, ensuring an acceptable QoS, for offered traffic loads below 0.85.

In the CFDAMA-PB-PEDF protocol, the stringent traffic class deadlines reduced the ability of the P-EDF scheme to improve on system performance, both in average packet delay and overall throughput.

4 Channel State Aided MAC Protocols

4.1 Introduction

This chapter introduces three cross-layer based protocols; Combined Free/Demand Assigned Multiple Access with Piggy Backing - Dynamic Deadlines (CF/DAMA-PB-DD), Combined Free/Demand Assigned Multiple Access with Piggy Backing - Best Signal to Noise Ratio First (CF/DAMA-PB-BSNRF) and the hybrid Combined Free/Demand Assigned Multiple Access with Piggy Backing – Best Signal to Noise Ratio First and Dynamic Deadlines (CF/DAMA-PB-BSNRF+DD). Two channel models are discussed, the Good-Bad Markovian channel model and the 3-state Markovian channel model, the later being the one used in the simulations. The performance of the CF/DAMA-PB and CF/DAMA-PB-PD protocols under the channel model is presented. The 3 proposed protocols aim to improve on the throughput while not increasing the end-to-end delay of the system significantly.

Section 4.2 presents the cross-layer design approach and the possible benefits that may be obtained. Section 4.3 discusses two channel models and how the 3 state channel model was implemented for the DVB-RCS system simulation. Section 4.4 introduces the first of the three cross-layer protocols, the CF/DAMA-PB-BSNRF protocol. Section 4.5 and 4.6 describe the details of the CF/DAMA-PB-DD and CF/DAMA-PB-BSNRF+DD protocols respectively. Section 4.7 details the system parameters used in simulating the presented protocols. In Section 4.8 five protocols are presented, compared and contrasted to determine the various performance benefits of the different protocols in terms of throughput, average packet delay and percentage of dropped packets. A conclusion is presented in Section 4.9.

4.2 Cross-layer design

ETSI Technical Committee – Satellites Standards Earth Stations and Systems/Broadband Satellite Multimedia (ETSI TC-SES/BSM) defines the IP-based satellite network architecture seen in Figure 5. The architecture defines three abstract layers, the satellite dependent, the satellite independent layers and the external layers. A SI-SAP is used as the interface between the two layers, i.e. satellite dependent and independent layers. The satellite dependent layer is further broken into sub-layers, these being:

- Satellite link layer
- Satellite Medium Access Control layer (SMAC)
- Satellite physical layer

In cross-layer design emphasis is placed on trying to utilize the information that is specific to certain layers in the other layers of the system in order to improve the system without completely destroying the stack architecture.

Cross-layer design has two basic types of design approaches:

- Implicit cross-layer design
- Explicit cross-layer design

The implicit method is one in which all the cross-layer information exchange is not done during the operation of the system but rather the interactions are considered and implemented during design time in a way that tries to optimize the system performance.

In the explicit method, a dynamic approach is used during the system operation, the information exchange between adjacent and non-adjacent layers is considered during the operation of the system and accounted for in real time as far as possible. In many cases, there is the implementation of a coordinator to ensure the correct exchange of information is made without destroying the protocol stack. The co-ordinator may be a global one, or be placed within the functionality of a layer, e.g. can be an Application-centric coordinator which gives the control of the interaction of layers to the protocols in the application layer or a MAC-centric coordinator, which is the approach used in this work, that gives control of the interaction of layers to the MAC protocol.

The implicit method of cross-layer design is relatively simple to implement as it does not destroy the implementation and benefits of having the layered stack approach while still adding to performance of the system using cross-layer information exchange.

Another possible solution is to implement both layer-centric and global coordinators, thus benefiting from both approaches. The NCC, in the DVB-RCS system implementation, ensures that the information required for the interaction exchange is available to the scheduler.

4.2.1 Cross-layer Interactions

There are quite a few cross-layer interactions, [51], that could be used to improve the performance of the system, some being,

- Physical and MAC layer interactions,
- Physical and Network layer interactions,
- MAC and Network layer interactions,
- MAC and Transport layer interactions.

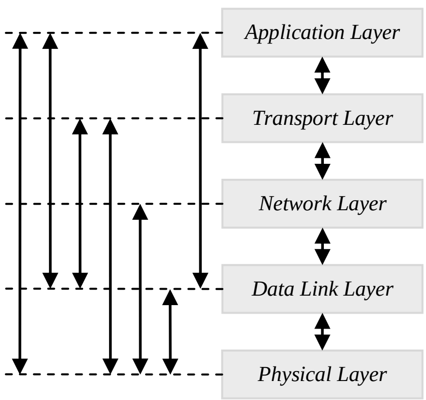

Figure 90: Cross-layer interactions.

Physical and MAC layer interactions

In systems supporting Adaptive Coding and Modulation (ACM), if the scheduling and medium access protocols have access to the physical layer properties, one is able to make better choices in the protocols to ensure a performance increase. DVB-S2 supports the ACM options and hence benefits from such interactions. By being able to vary the coding and modulation while operation of the system is underway, the negative effects of atmospheric losses can be reduced significantly, although ACM can have an undesirable effect of reducing the channel capacity if not correctly implemented. One is also able to use the channel information of the physical layer to reassign the channel capacity in such a manner as to increase the utilization. This is done by using the signal to noise ratio (SNR) of the users in the system. These kinds of interactions have positive effects on wireless communication systems.

Physical and network layer interactions

In these types of interactions one can utilise the physical layer to inform the network layer of its properties. The network layer may then make appropriate decisions regarding routing of calls from one satellite to the next. This affects mobile users in the GEO satellite scenario and the users in a LEO and MEO satellite network where routing occurs quite often according to the speed and distance of the satellite overhead. If these characteristics are not used correctly, a greater end-to-end delay of traffic is experienced during rerouting which can make the connection very undesirable and break QoS guarantees.

MAC and network layer interactions

In order to benefit from the interactions, the scheduler is required to be informed of any changes in the system as early as possible so as not to waste capacity by assigning resources to users already assigned by another scheduler due to handovers, etc. Management of traffic in the link-layer needs to be coordinated with that in the network layer; ways to do this is by using Integrated Services (IntServ) and Differentiated Services (DiffServ).

MAC and transport layer interactions

The standard TCP congestion control mechanisms are poorly designed in terms of systems with large propagation delays and hence RTD. Thus interactions can have a positive effect on these transport layer protocols. This is shown to work in a TCP Reno system that makes use of ACK messages to try on improving the overall system performance by varying its congestion window. The interactions can also be used to increase fairness among users.

Many more interactions can be used in improving system performance, one needs to weigh out the pros and cons of the various interactions and hence make design decisions that better suit that of the system's requirements.

4.2.2 Channel Losses

The Bit Error Rate (BER) in satellite networks is generally quite high as a result of the harsh channel conditions caused by atmospheric effects and the great propagation distance of signals from the Earth stations to the GEO satellites. The following effects contribute to the BER of the channel:

- Free space loss
- Atmospheric gases
- Rain Fade
- Fog and Clouds
- Scintillation

Atmospheric gases

The electromagnetic attenuation of the signals is affected by oxygen and water vapour in the air. Air atmospheric gasses mainly affect those signals below the 10 GHz frequency while water vapour is the main factor affecting frequencies above the 150 GHz range.

Rain Fade

Out of all the atmospheric effects rain fade affects the signal the most. It starts becoming a huge factor above 8 GHz and becomes more serious over 11 GHz. The rate of the rainfall, the size of each droplet and the length of signal exposure to the rain fade all contribute to this atmospheric effect.

Fog and Clouds

The affects of this are mainly seen at frequencies above 30 Hz. Rayleigh scattering is a result of clouds that scatter the signal.

Scintillation

This effect is seen above the 10 degree elevation angle and below the 10 GHz frequency. This is caused by irregularities in the troposphere which cause a varying refractive index of the atmosphere on the signal.

DVB-RCS tries to overcome many of these effects by adding protection to the packets and frame structures. The use of the FEC codes, RS and convolution, help to reduce the effects of atmospheric channel losses and a CRC is used to check if significant losses do occur.

4.3 Channel Model

A channel model is required in order to have a more realistic simulation and hence performance evaluation of the MAC protocols in a GEO satellite environment.

There are two channel models discussed in this section, the Good-Bad model and a 3-state model; this is to show that there are different models that can be used. The model chosen for this work is that of the 3-state model as it is a better modelling of the variations in a satellite system compared to simply having two states deciding whether the system is in a Line of Sight (LOS) or non-LOS link.

There are two main factors that contribute to channel propagation in radio communications,

- Frequency
- The environment

By environment, we consider the following to be a possible set of the environments,

- Urban area
- Suburban area
- Open Area
- Intermediate tree shadow area
- Heavy tree shadow area

The reason the environment has an impact on the various channel losses that can occur is that differing environments have differing obstacles to overcome, such as trees, buildings, dense areas or otherwise, etc.

An example of a path loss in the open areas is the free path loss defined by the following equation,

$$L = \left(\frac{4\pi Df}{c}\right)^2 = \left(\frac{4\pi D}{\lambda}\right)^2 \tag{4.1}$$

Where L is the loss in decibels (dB), D is the distance, f is the frequency and c is the speed of light in a vacuum.

In a mobile situation this equation needs to be rethought, such as in the urban and suburban test environment in [52],

$$\begin{aligned} L = 40(1 - 4 \times 10^{-3}\Delta h)log_{10}(D) - 18log_{10}(f) \\ + 21log_{10}(\Delta h) + 80\ dB \end{aligned} \tag{4.2}$$

Where L is the loss in dB, Δh is the base station antenna height in metres and f is the carrier frequency of 2000 MHz.

One typically experiences significantly higher path loss in non-LOS environments than the LOS environment. The 3 state model tries to accomplish a more realistic variation in channel path loss over time by adding an extra state.

Shadowing attenuation is modelled using the log-normally distributed shadow fading.

The two state model is a simplistic method to model the channel losses, and often is adequate for cellular network evaluation; hence, in order to achieve slightly closer to real situation, a three state model is preferred.

4.3.1 Good-Bad model

In a Good-Bad channel model [53] there are two states each having an exponentially distributed sojourn time. Once the current state ends its sojourn time, the state change occurs according to probabilities depicted in Figure 91,

- P_g – the probability of moving to the Good state, given a Good state
- P_b – the probability of moving to the Bad state, given a Bad state
- P_{gb} – the probability of moving to a Bad state, given a Good state
- P_{bg} – the probability of moving to a Good state, given a Bad state

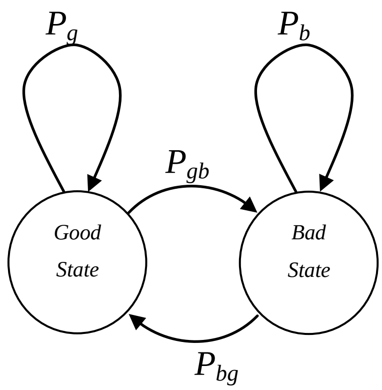

Figure 91: Good-Bad channel Model.

These probabilities may be derived as described by Andreadis and Giambene in [53] ,

$$\lambda_{gb} P_g = \lambda_{bg} P_b \tag{4.3}$$

And,

$$P_g + P_b = 1 \tag{4.4}$$

Thus if we solve for the two equations above, we obtain the following,

$$P_g = \frac{\lambda_{bg}}{\lambda_{gb} + \lambda_{bg}} \tag{4.5}$$

And,

$$P_b = \frac{\lambda_{gb}}{\lambda_{gb} + \lambda_{bg}} \tag{4.6}$$

From the probabilities one can find the sojourn times of the Good and Bad states as found in [53].

This model was not used in this work, as the 3 state model provides a better option to that of the two state model, as it achieves a more realistic channel.

4.3.2 3 State model

The 3 state model allows for a greater variation of signal states compared to that of the Good-Bad model, achieving a closer approximation to the actual channel fading characteristics. This model is used as it is closer to reality than the Good-Bad model. The protocols do not depend on the model used. The 3 state model consists of the following states,

- Line of Sight (LOS)
- Light Shadowing (LS)
- Heavy Shadowing (HS)

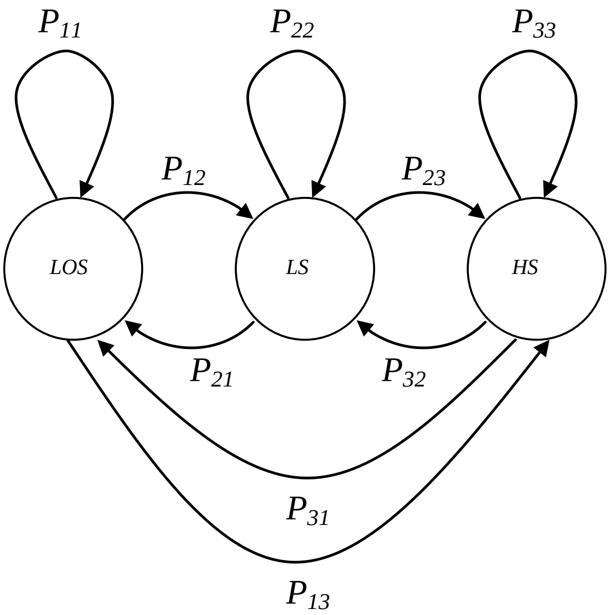

Figure 92: 3 State Markov chain channel model.

Similar to that of the Good-Bad model, each state has an exponentially distributed sojourn time and at the end of each sojourn time, the state that follows is determined by a set of probabilities, these probabilities being summarised in Table 10 and depicted in Figure 92.

Table 10: 3 State channel model probability matrix.

P_{11}	P_{12}	P_{13}
P_{21}	P_{22}	P_{23}
P_{31}	P_{32}	P_{33}
W_1	W_2	W_3

The probabilities work as follows; P_{ij} is the probability of moving to state j, given state i.. W_i is the overall probability of being in state i. The matrices, **[*P*]** and **[*W*]**, have the following properties [54],

- The sum of the elements in a row of the matrix **[*P*]** must add up to one
- The sum of the elements in a row of the matrix **[*W*]** must add up to one
- The asymptotic behaviour of the Markov chain is characterised by,

$$[W][P] = [W] \tag{4.7}$$

Each environment is defined by three states. The following environments and their associated experimental average probabilities were taken from [54]:

Table 11: 3 State channel model probabilities.

Environments	P_{11} / P_{21} / P_{31} / W_1	P_{12} / P_{22} / P_{32} / W_2	P_{13} / P_{23} / P_{33} / W_3
Open area	*95.86*	*3.22*	*0.92*
	5.71	*92.38*	*1.91*
	3.90	*1.29*	*94.81*
	25	*50*	*25*
Suburban area	*79.97*	*15.02*	*5.01*
	13.64	*77.84*	*8.52*
	12.20	*12.20*	*75.60*
	14.28	*42.86*	*42.86*
Intermediate Tree-shadowing	*78.26*	*19.95*	*6.79*
	24.81	*68.99*	*6.20*
	24.04	*6.01*	*69.95*
	42.85	*42.85*	*14.3*
Heavy Tree-shadowing	*81.75*	*7.30*	*10.95*
	2.01	*94.65*	*3.34*
	3.32	*9.98*	*86.70*
	41.67	*25*	*33.33*
Urban area	*88.28*	*6.62*	*5.10*
	14.47	*81.39*	*4.14*
	8.48	*4.71*	*86.81*
	50	*20*	*30*

4.3.3 SNR in a State

The instantaneous SNR in a state is assumed to be log-normally distributed with a mean value of $\overline{SNR}_i$. To obtain relatively accurate signal (dB) values that define each states average and instantaneous SNR in dB's, the use of experimental data obtained in [55] by Alberto Gotta and Paolo Barsocchi was used.

The Ec/No vs. PER was used to translate a SNR value to the Packet Error Rate (PER). This was achieved through a linear approximation of the experimental data depicted in Figure 93 from [55].

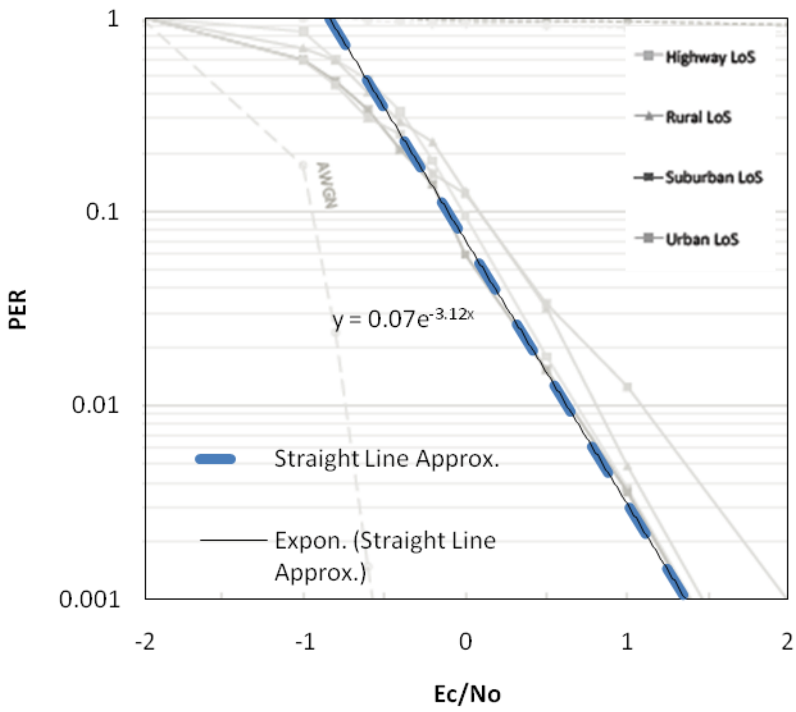

Figure 93: Experimental DVB-RCS PER vs. Ec/No.

Due to the exponential nature of the graph, an exponential equation results in the linear approximation depicted in Figure 93. The equation being of the form,

$$PER = be^{(c)(SNR)} \tag{4.8}$$

Where b = 0.07 and c = -3.12 define the equation that approximates the average of the environments giving,

$$PER = 0.07e^{(-3.12)(SNR)} \tag{4.9}$$

Each channel state is defined in terms of a mean SNR value and a standard deviation. The SNR has a log-normal distribution in each state. The following list of states and their parameters are chosen according to Figure 93 and [54] and the PER is calculated using equation (4.9),

Table 12: Channel states and associated SNR.

State	*Mean SNR [dB]*	σ_{state} *[dB]*	*Mean PER*
LOS	*1.5*	*0.16*	0.65×10^{-3}
LS	*1.0*	*0.15*	3.09×10^{-3}
HS	*0.5*	*0.27*	14.71×10^{-3}

In each state a log-normal random number with the mean SNR and standard deviation (σ_{state}) is used to define the instantaneous SNR on a per frame basis. An instantaneous PER is then determined using equation (4.9). This PER determines the packet loss on a per frame basis according to the state a terminal is currently in.

Each terminal, RCST, has a state that is statistically independent from one another as each terminal is statistically in differing degrees of shadowing according to its particular environment and channel loss characteristics.

The instantaneous channel state of each RCST is available to itself, while a channel state of a RCST is only available to the satellite if the satellite has received a packet from the RCST. Due to the propagation delay of GEO satellites, it is preferable to obtain a mean channel state at the satellite rather than the instantaneous channel state. In addition this makes it more desirable to implement channel state enhancements, as far as possible and where feasible, at the RCST rather than at the satellite scheduler.

Each state has a mean exponential sojourn time, with this time chosen to be significantly larger than the frame time, so as to ensure that the channel state of a RCST does not vary too often, as this would be unrealistic in a GEO satellite environment with the channel losses experienced in a fixed earth RCST. The frame time is 0.0235 s and therefore a mean exponential channel state sojourn time of 10 s was chosen.

4.4 CF/DAMA-PB-BSNRF Protocol

In a system with varying channel conditions affecting the signal quality, one is able to make use of the information at the physical layer in order to make decisions on improving system performance. The CF/DAMA-PB-Best Signal-to-Noise Ratio First (CF/DAMA-PB-BSNRF) Protocol takes advantage of the better, in terms of the SNR, connections between the RCSTs and the satellite and thus improves the overall throughput of the system.

The CF/DAMA-PB scheduling is used to decide on how requests are made at the RCST and the CF/DAMA-PB-PD protocol ensures that the time critical traffic classes keep within their deadlines while the BSNRF adds to this by ensuring the best system utilization.

The satellite scheduler takes into account the mean SNR of the RCSTs that have submitted a request and reorders the RRQ such that the RCST that had the best SNR is given a greater priority of being allocated.

The SNR information is accessed by the satellite when a RCST sends packets to the satellite. According to the RCSTs most recent state the scheduler allocates the system resources, requested by the RCST, in a BSNRF manner, thus ensuring that fewer packets are dropped due to bad channel conditions. This is depicted in Figure 94.

The channel propagation delay can affect the effectiveness of the BSNRF Protocol as the RCST allocation according to the SNR may be different to the RCST’s actual channel state once it receives the allocation. This is due to there being a 270 ms delay between the satellite having allocated the slots and the RCST receiving the slot allocation. However this should have minimal effect as the mean sojourn time of a state is significantly larger than the channel propagation.

The CF/DAMA-PB-BSNRF protocol has a centralised scheduling algorithm and a terminal requesting algorithm at each terminal. The scheduling algorithm can be situated at the satellite or an NCC on the earth. The implementation used is that of the scheduling algorithm at the satellite, thus reducing the number of hops to be made in order for a request to be allocated. In this case a minimum of one satellite hop is required to request channel capacity and have the request acknowledged.

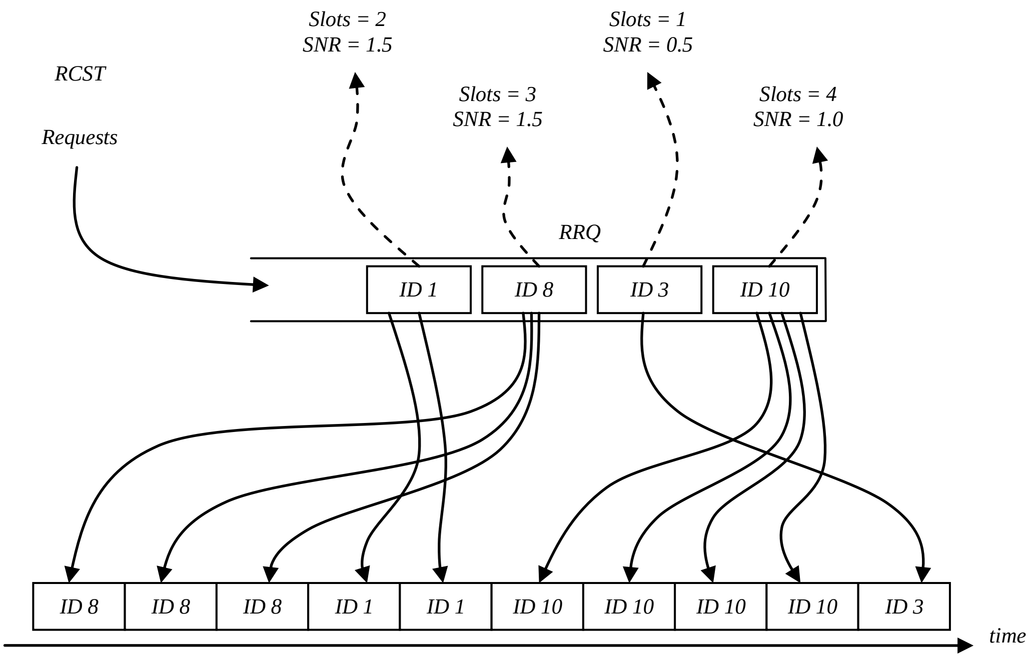

Figure 94: BSNRF resource allocation.

The scheduling algorithm is depicted in Figure 95. The scheduler has two main tables or queues, the Reservation Request Queue (RRQ) and the Free Assignment Queue (FAQ). The FAQ is populated with the addresses of each of the terminals connected to the satellite. The RRQ is used to hold the requests made by the earth terminals or RCSTs. The RRQ holds the source address of the RCST that made the request, as well as the number of slots requested by the RCST and the most recent mean channel state for the RCST. Each received request is placed into the RRQ in a FIFO fashion and served in a BSNRF fashion. The scheduler allocates the slots of the TDMA on a frame-by-frame basis and then sends the assignment to the RCST using a packet on the downlink frame. The number of slots in a TDMA frame is equal to the number of RCSTs, effectively giving each a turn to initially request resource capacity. In the MF-TDMA version of this work, the number of slots shared among all the frames in a frame time period is equal to the number of RCSTs.

The scheduler first queries the RRQ to determine whether there are any requests made. If the RRQ is not empty then the scheduler assigns a slot in a frame to the RCST request with the highest SNR value and subtracts one from the number of requested slots of the RCST in question. This is repeated, if the RCST request of the BSNRF RCST becomes zero, the RCST is removed from the queue and the corresponding FAQ entry is moved to the tail end of the FAQ. The slot allocation is repeated until the frame is full, at which point it waits to allocate the following frames slots, or if the RRQ is empty then the slots are assigned to the RCST at the head of the FAQ. The RCST then will be removed from the head of the FAQ and placed at the tail end of the queue. The FAQ is serviced in a round-robin fashion ensuring that the free assignment is fairly allocated to the RCSTs.

The terminal algorithm for the RCSTs is illustrated in Figure 96 and discussed.

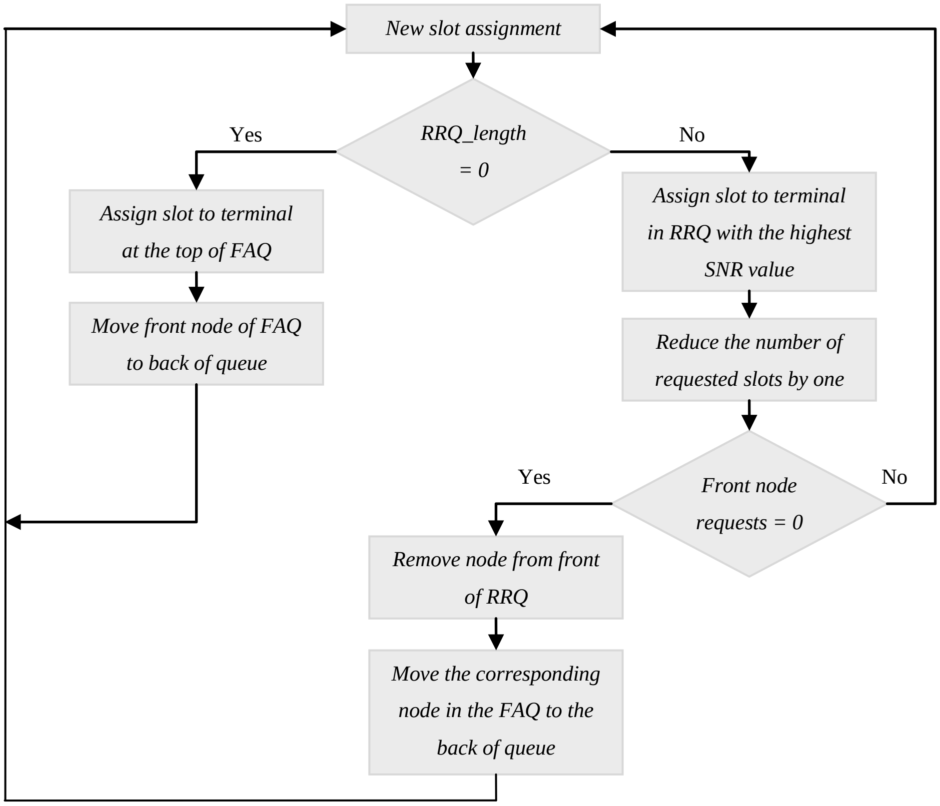

Figure 95: CFDAMA-PB-BSNRF scheduling algorithm.

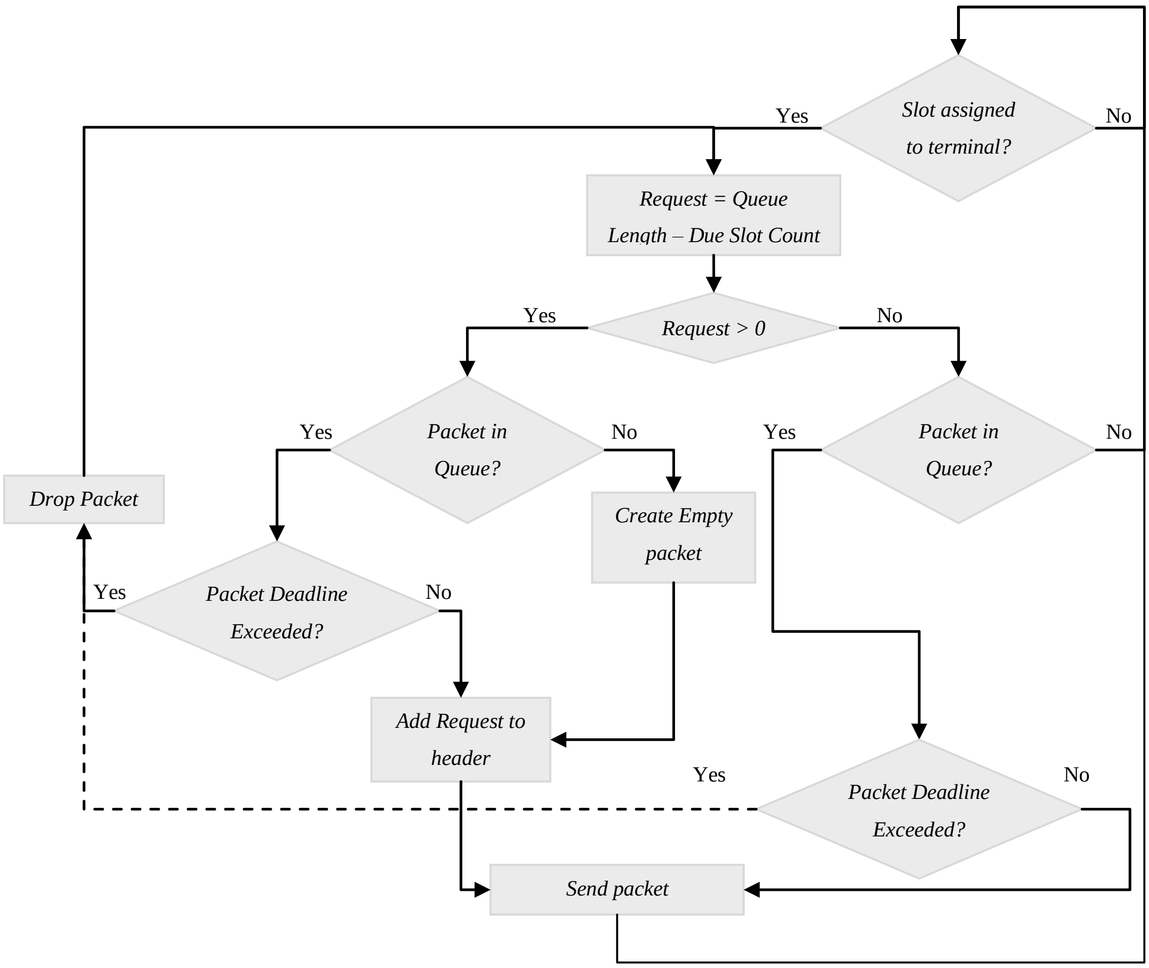

Figure 96: CF/DAMA-PB-BSNRF terminal algorithm.

The RCST waits for a slot assignment. Once a slot has been assigned to the RCST, either by free assignment or demand assignment, the RCST then calculates if it needs to request any additional slots for queued packets. If a request is necessary the RCST determines if the packet at the head of the queue has exceeded its deadline. If the deadline has been exceeded for the packet in question, this packet could be a voice, video or web packet, the packet is dropped and one returns to calculating if a request needs to be made. If the deadline was not exceeded, the RCST adds the request to the packet at the head of its traffic queue and sends the packet. If there are no packets in the queue and the RCST has a request and a slot assigned to it, then the RCST creates an empty packet, adds the request to that packet and sends the packet in the allocated slot. If the RCST does not have any additional requests to make, but has packets queued in the traffic queue, the packet at the head of the queue is checked if it has exceeded the deadline and if not it is sent in the allocated slot. On the instance of not receiving a slot for multiple frames, the request is accumulated until the RCST receives a slot allocation to send the request. This ensures that a request is made once per frame at maximum per RCST.

4.5 CF/DAMA-PB-DD Protocol

The CF/DAMA-PB-PD protocol takes advantage of the traffic classes that do not require zero packet loss in transmission and thus reduces the end-to-end delay of the system for the video and voice traffic.

The channel introduces packet loss by means of errors caused due to non-ideal channel characteristics experienced in a real system. This increases the overall packet loss, especially when a RCST is in the HS state.

In order to compensate for the channel contribution to loss of packets and still maintain a relatively low end-to-end delay of the system, it is possible to use the physical layer channel state information to decide on the deadline of packets to be dropped.

In the CF/DAMA-PB-DD Protocol the idea is to loosen up on the traffic class deadlines in RCSTs with a lower SNR value and tighten the traffic class deadlines in the RCSTs with a high SNR. The unit used to determine deadlines is that of the number of frames on top of the RTD and therefore it makes sense to vary these deadlines by means of number of frames. The traffic classes, video, voice and web, each have a deadline specific to their class and thus each deadline must be varied independently of each other but dependent on the SNR of the RCST.

In the simulation, a channel state with a SNR value is assigned to an RCST, this mean SNR is then used to obtain the instantaneous SNR value by means of a log-normal distribution. The instantaneous SNR dictates the SNR value for a frame period of 0.0235 s. The deadline changes are thus made on a per frame basis and according to the instantaneous SNR. The instantaneous SNR value is used to determine the packet loss due to the channel.

The deadline variation is tabulated in Table 13 below.

Table 13: Dynamic Deadline traffic class deadline variation according to channel state.

Deadline	*LOS*	*LS*	*HS*
T_{dvoice}	*0.587 s*	*0.6105 s*	*0.634 s*
T_{dvideo}	*0.5635 s*	*0.587 s*	*0.6105 s*
T_{dweb}	*1.5505 s*	*1.574 s*	*1.5975 s*

The scheduling algorithm is depicted in Figure 97. The scheduler has two main tables or queues, the Reservation Request Queue (RRQ) and the Free Assignment Queue (FAQ). The FAQ is populated with the addresses of each of the terminals connected to the satellite. The RRQ is used to hold the requests made by the earth terminals or RCSTs. The RRQ holds the source address of the RCST that made the request, as well as the number of slots requested by the RCST. Each received request is placed into the RRQ in a FIFO fashion and served in a FIFO fashion. The scheduler allocates the slots of the TDMA on a frame-by-frame basis and then sends the assignment to the RCST using a packet on the downlink frame. The number of slots in a TDMA frame is equal to the number of RCSTs, effectively giving each a turn to initially request resource capacity. In the MF-TDMA version of this work, the number of slots shared among all the frames in a frame time period is equal to the number of RCSTs.

The scheduler first queries the RRQ to determine whether there are any requests made. If the RRQ is not empty then the scheduler assigns a slot in a frame to the RCST request at the head of the queue and subtracts one from the number of requested slots of the RCST in question. This is repeated, if the RCST request at the head of the RRQ becomes zero, the RCST is removed from the head of the queue and the corresponding FAQ entry is moved to the tail end of the FAQ. The slot allocation is repeated until the frame is full, at which point it waits to allocate the following frames slots, or if the RRQ is empty then the slots are assigned to the RCST at the head of the FAQ. The RCST then will be removed from the head of the FAQ and placed at the tail end of the queue. The FAQ is serviced in a round-robin fashion ensuring that the free assignment is fairly allocated to the RCSTs.

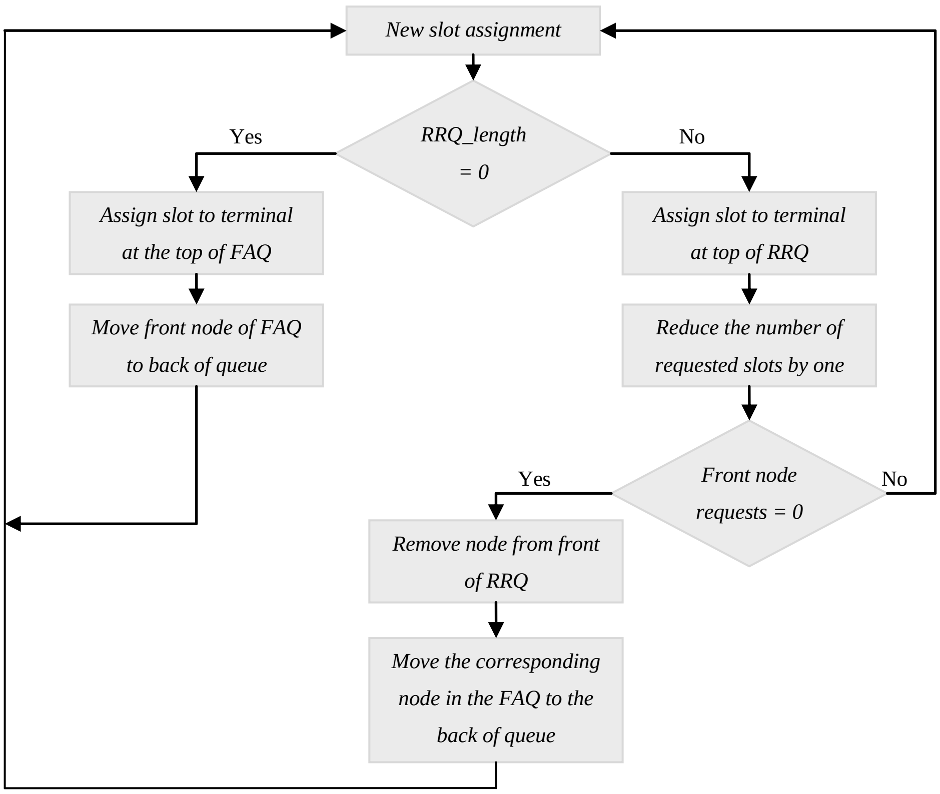

Figure 97: CFDAMA-PB-DD scheduling algorithm.

The terminal algorithm for the RCSTs is illustrated in Figure 98 and discussed below.

The RCST waits for a slot assignment. Once a slot has been assigned to the RCST, either by free assignment or demand assignment, the RCST then calculates if it needs to request any additional slots for queued packets. If a request is necessary the RCST determines if the packet at the head of the queue has exceeded its deadline. The deadline is set for each traffic class according to the mean SNR value the RCST has. If the deadline has been exceeded for the packet in question, this packet could be a voice, video or web (virtual deadline) packet, the packet is dropped and one returns to calculating if a request needs to be made.

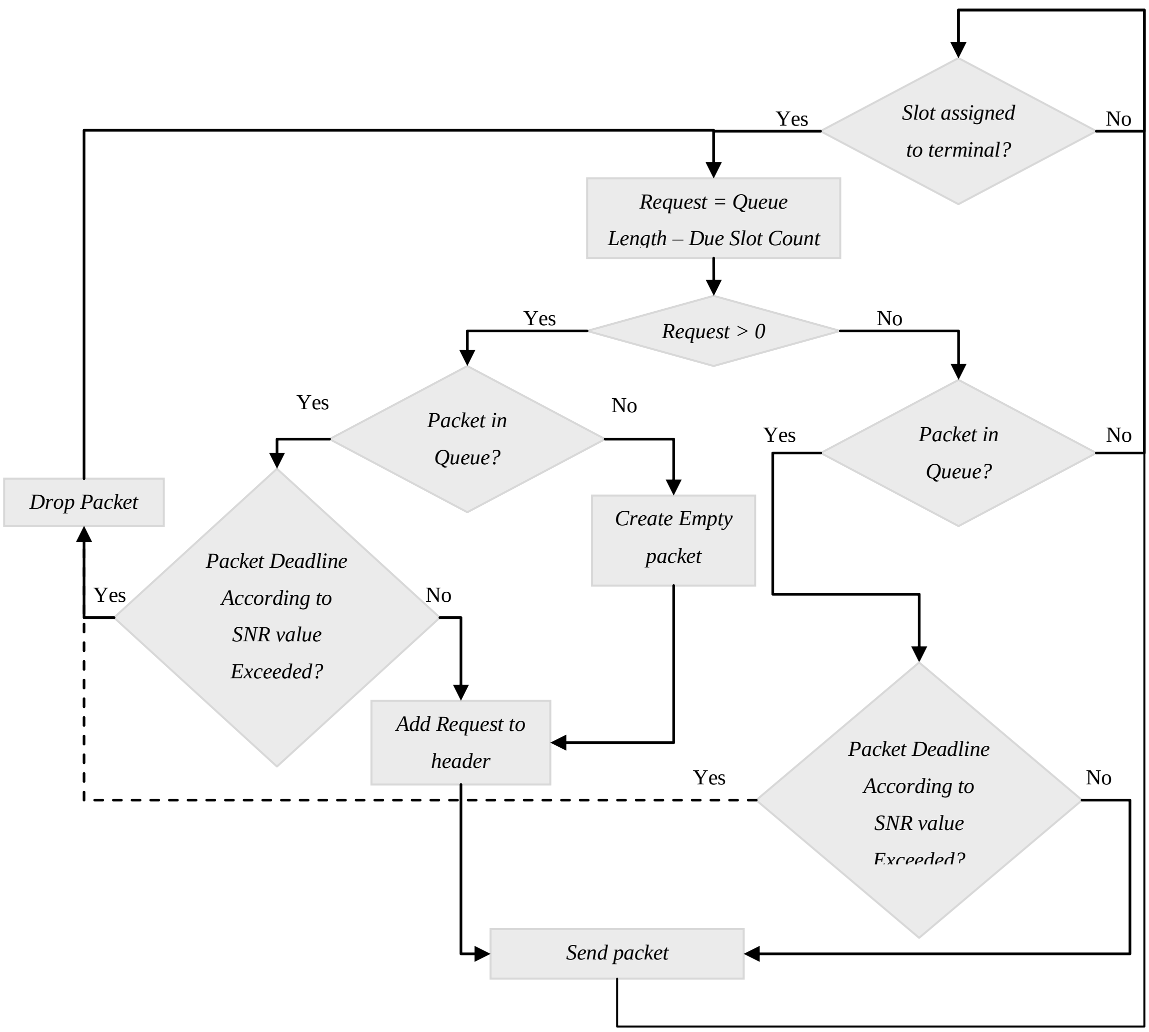

Figure 98: CF/DAMA-PB-DD terminal algorithm.

If the deadline was not exceeded, the RCST adds the request to the packet at the head of its traffic queue and sends the packet. If there are no packets in the queue and the RCST has a request and a slot assigned to it, then the RCST creates an empty packet, adds the request to that packet and sends the packet in the allocated slot. If the RCST does not have any additional requests to make, but has packets queued in the traffic queue, the packet at the head of the queue is checked if it has exceeded the deadline, which is set according to the RCST mean SNR value, and if not it is sent in the allocated slot. On the instance of not receiving a slot for multiple frames, the request is accumulated until the RCST receives a slot allocation to send the request. This ensures that a request is made once per frame at maximum per RCST.

4.6 CF/DAMA-PB-BSNRF+DD Protocol

In the CF/DAMA-PB-BSNRF+Dynamic Deadline Protocol, the aim is to combine the CF/DAMA-PB-BSNRF protocol with that of the CF/DAMA-PB-DD protocol. The CF/DAMA-PB-BSNRF aim was to try increase the system utilization while reducing the end-to-end delay by means of packet dropping and the CF/DAMA-PB-DD protocol's aim was to try loosening the deadline on bad channels so as to increase system throughput and compensate for the channel packet loss while not increasing the end-to-end delay significantly.

The hybrid protocol aim is to increase the system utilization above that of both the CF/DAMA-PB-BSNRF protocol and the CF/DAMA-PB-DD protocol without significantly affecting the end-to-end delay of the system.

This protocol implements the reordering of the RRQ resource allocation at the satellite in terms of the SNR values, these being the highest to lowest on a frame by frame basis. At the RCST the voice, video and web traffic class deadlines are varied according to the SNR value of each of the RCSTs thus decreasing the packet loss due to dropping by deadlines to compensate for the packet loss due to the channel.

Once again the channel propagation delay can affect the effectiveness of the CF/DAMA-PB-BSNRF and CF/DAMA-PB-DD protocol as the RCST channel state may be different to the RCST's channel state when the satellite made the allocation as it is a minimum of 270 ms after the satellite read the state of the RCST.

The scheduling algorithm is depicted in Figure 99. The scheduler has two main tables or queues, the Reservation Request Queue (RRQ) and the Free Assignment Queue (FAQ). The FAQ is populated with the addresses of each of the terminals connected to the satellite. The RRQ is used to hold the requests made by the earth terminals or RCSTs. The RRQ holds the source address of the RCST that made the request, as well as the number of slots requested by the RCST and the most recent mean channel state for the RCST. Each received request is placed into the RRQ in a FIFO fashion and served in a BSNRF fashion. The scheduler allocates the slots of the TDMA on a frame-by-frame basis and then sends the assignment to the RCST using a packet on the downlink frame. The number of slots in a TDMA frame is equal to the number of RCSTs, effectively giving each a turn to initially request resource capacity. In the MF-TDMA version of this work, the number of slots shared among all the frames in a frame time period is equal to the number of RCSTs.

The scheduler first queries the RRQ to determine whether there are any requests made. If the RRQ is not empty then the scheduler assigns a slot in a frame to the RCST request with the highest mean SNR value and subtracts one from the number of requested slots of the RCST in question. This is repeated, if the RCST request of the BSNRF RCST becomes zero, the RCST is removed from the queue and the corresponding FAQ entry is moved to the tail end of the FAQ. The slot allocation is repeated until the frame is full, at which point it waits to allocate the following frames slots, or if the RRQ is empty then the slots are assigned to the RCST at the head of the FAQ. The RCST then will be removed from the head of the FAQ and placed at the tail end of the queue. The FAQ is serviced in a round-robin fashion ensuring that the free assignment is fairly allocated to the RCSTs.

The terminal algorithm for the RCSTs is illustrated in Figure 100 and discussed.

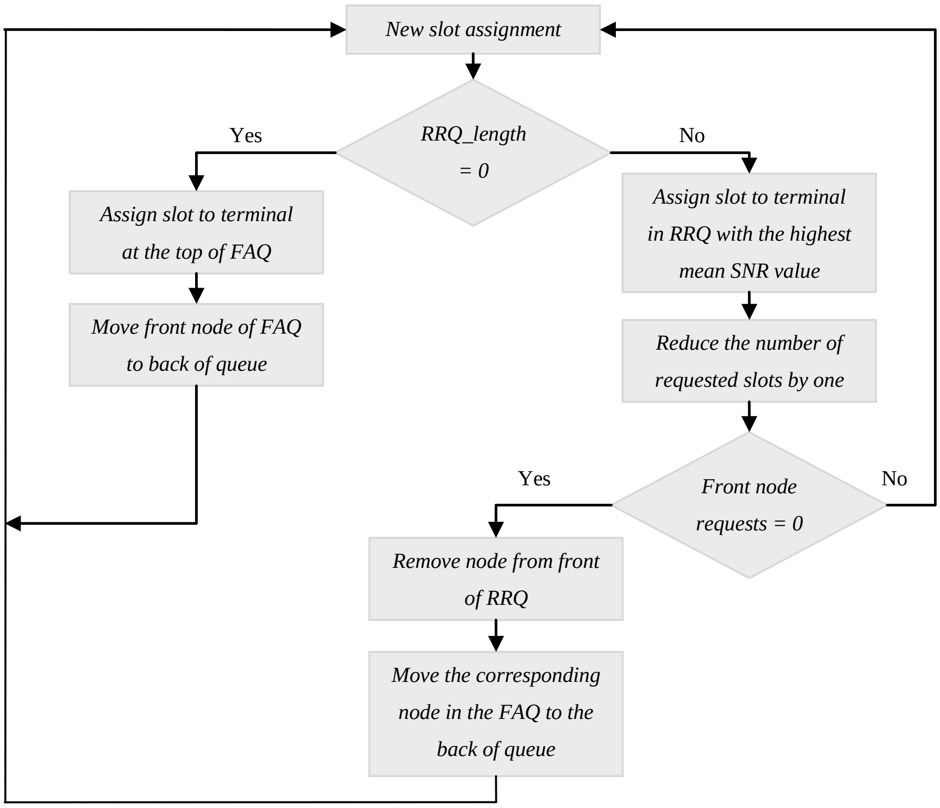

Figure 99: CFDAMA-PB-BSNRF+DD scheduling algorithm.

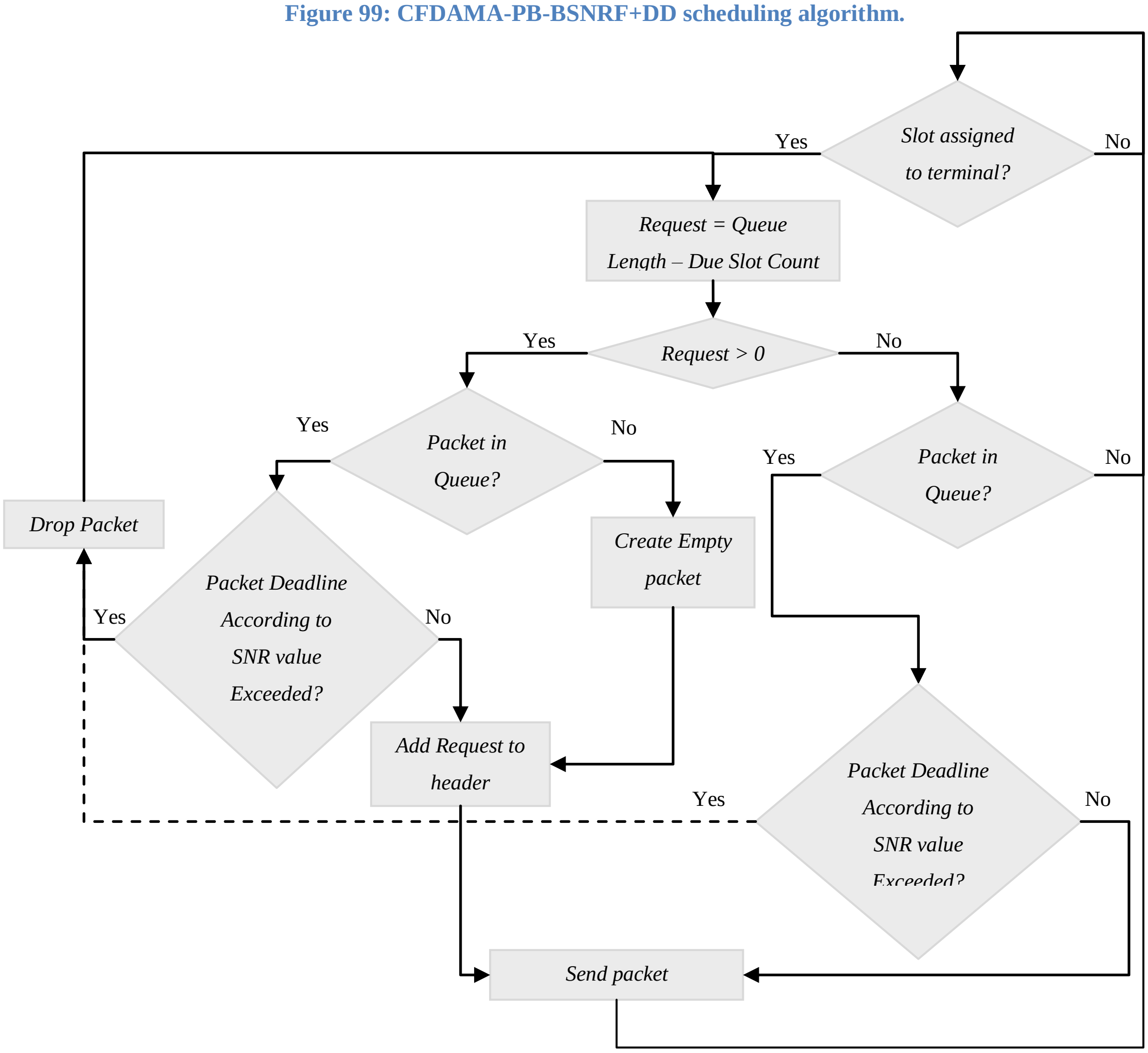

Figure 100: CF/DAMA-PB-BSNRF+DD terminal algorithm.

The RCST waits for a slot assignment. Once a slot has been assigned to the RCST, either by free assignment or demand assignment, the RCST then calculates if it needs to request any additional slots for queued packets. If a request is necessary the RCST determines if the packet at the head of the queue has exceeded its deadline. The deadline is set for each traffic class according to the mean SNR value the RCST has. If the deadline has been exceeded for the packet in question, this packet could be a voice, video or web (virtual deadline) packet, the packet is dropped and one returns to calculating if a request needs to be made. If the deadline was not exceeded, the RCST adds the request to the packet at the head of its traffic queue and sends the packet. If there are no packets in the queue and the RCST has a request and a slot assigned to it, then the RCST creates an empty packet, adds the request to that packet and sends the packet in the allocated slot. If the RCST does not have any additional requests to make, but has packets queued in the traffic queue, the packet at the head of the queue is checked if it has exceeded the deadline, which is set according to the RCST mean SNR value, and if not it is sent in the allocated slot. On the instance of not receiving a slot for multiple frames, the request is accumulated until the RCST receives a slot allocation to send the request. This ensures that a request is made once per frame at maximum per RCST.

4.7 Simulation Parameters

As in chapter 3, the simulations consisted of 1000 s runs for each load value, a resulting virtual time of between 10000 s and 17000 s simulation runs per set of throughput, average delay and packet dropping graphs. The GEO satellite DVB-RCS system consisted of a global rate of 8 Mbits/s and a carrier rate of 2 Mbits/s each. The total number of slots was equivalent to the number of RCST nodes, i.e. 128 slots and nodes. This equates to 32 slots per carrier, 4 carriers, imposing a limit of 32 slots being allocated to a node per frame period of 23.5 ms. A channel propagation delay of 270 ms due to the distance between an RCST and the GEO satellite. Each MPEG2 packet contains 188 bytes in total and 183 of these are for the payload. The main system parameters are summarised in Table 14.

Table 14: System parameters.

Parameter	*Value*
Simulation time	*1000 s*
Global rate	*8192000 bits/s*
Number of carriers	*4*
Number of RCSTs	*128*
Packet format	*MPEG2*
Packet size	*188 bytes*
Payload size	*183 bytes*
Rate per carrier	*2048000 bits/s*
Slots per carrier	*32*
Propagation delay	*270 ms*
Frame time	*23.5 ms*

The traffic used as input to the system was that of video, voice and web combinations, making up a multimedia traffic source. The scenarios used dictate the ratios of each traffic type that made up a multimedia traffic. The first scenario allowed for the evaluation of the system with even ratios of the three traffic classes. This does not necessarily occur in reality, thus a more close estimate of reality, being dominated by the video class, was also examined in scenario 2. These are summarised in Table 15.

Table 15: Simulation traffic scenarios.

Scenario	*Video*	*Voice*	*Web*
1	*1/3*	*1/3*	*1/3*
2	*2/3*	*1/6*	*1/6*

The main difference between the system in chapter 3 and the system in this chapter was a channel model was used to evaluate its effects on overall performance and hence have a more realistic system implementation.

The channel model used was the 3 State channel model described in this chapter and its parameters are summarised in Table 16.

Table 16: Channel state parameters.

Channel State	*Mean SNR*	*Standard Deviation*
LOS	*1.5 dB*	*0.16*
LS	*1.0 dB*	*0.15*
HS	*0.5 dB*	*0.27*

4.8 Results

The following simulations are presented:

- CF/DAMA-PB
- CF/DAMA-PB-Packet Dropping Protocol
- CF/DAMA-PB-BSNRF Protocol
- CF/DAMA-PB-Dynamic Deadline Protocol
- CF/DAMA-PB-BSNRF and Dynamic Deadline Protocol

4.8.1 CF/DAMA-PB Protocol

A 3 state channel model was added to the CF/DAMA-PB scheme to evaluate its performance in non-ideal situations closer to that of the GEO satellite environment.

Scenario 1

In scenario 1, the traffic was evenly divided between voice, video and web traffic classes, i.e. 1/3 voice, 1/3 web and 1/3 video traffic making up the total offered traffic load.

The average packet delay experienced in the CF/DAMA-PB with a channel model under scenario 1 traffic loads is depicted in Figure 101. The comparison with the CF/DAMA-PB performance of chapter 3 is shown. The channel affects the average packet delay negatively as it increases the delay slightly. This was as a result of the channel causing packets to be lost that could have contributed to the average delay being less, as in the ideal situation. It must be noted that once a packet misses its deadline and is dropped, due to too little time to retransmit, it is lost. All three traffic classes were affected by the increase in delay, the greatest of these being the web traffic class. The increase in delay was only seen when high loads were offered to the system as there was a greater chance of having increased number of channel losses when increasing the use of the channel capacity. One must note that the channel did not have a significantly noticeable effect on the average packet delay.

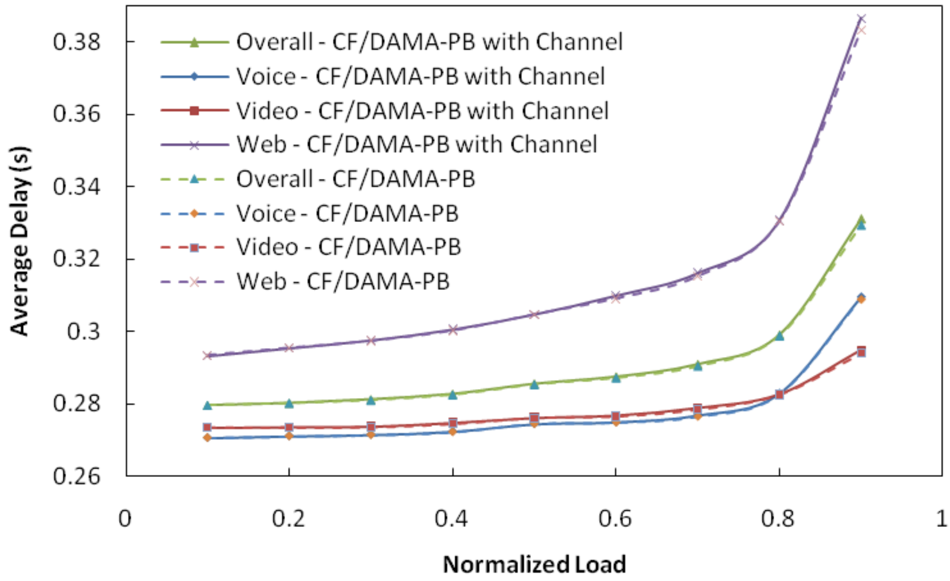

Figure 101: Average Delay vs. Normalized Load (Scenario 1) - CF/DAMA-PB with and without a channel model.

The throughput had no noticeable variation from the CF/DAMA-PB scheme without a channel model when looking at the overall throughput of the system as depicted in Figure 102. The channel did not have a large impact due to the relatively insignificant lost packets compared to the number of generated packets.

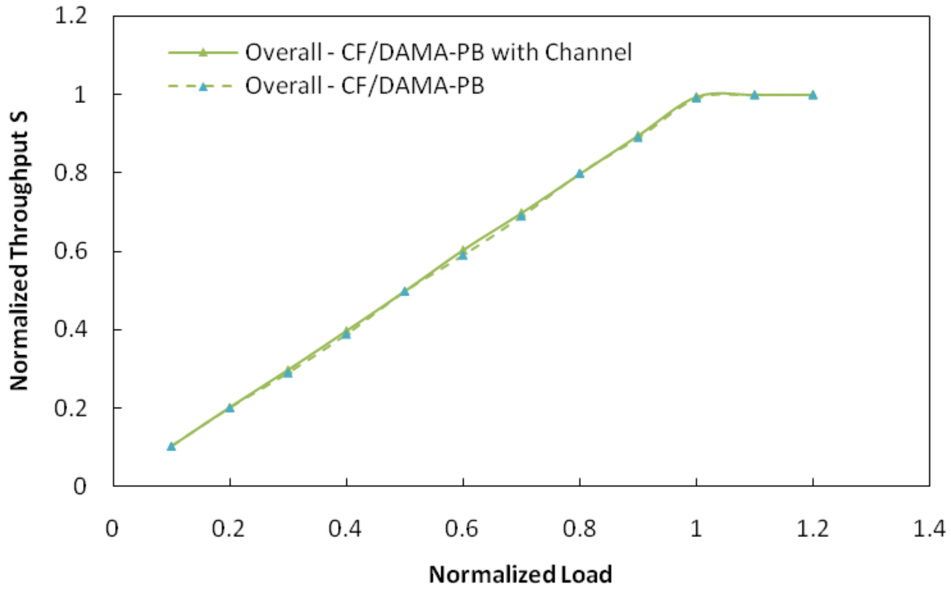

Figure 102: Normalized Throughput vs. Normalized Load (Scenario 1) CF/DAMA-PB with channel.

In Figure 103, there was a small difference seen in the throughput of the individual traffic classes, mainly above a throughput of 1.0. This may be accounted for by the relatively small packet error rate offered by the channel model. The slight increase in the video throughput was however compensated by the decrease in the web and voice traffic class throughputs.

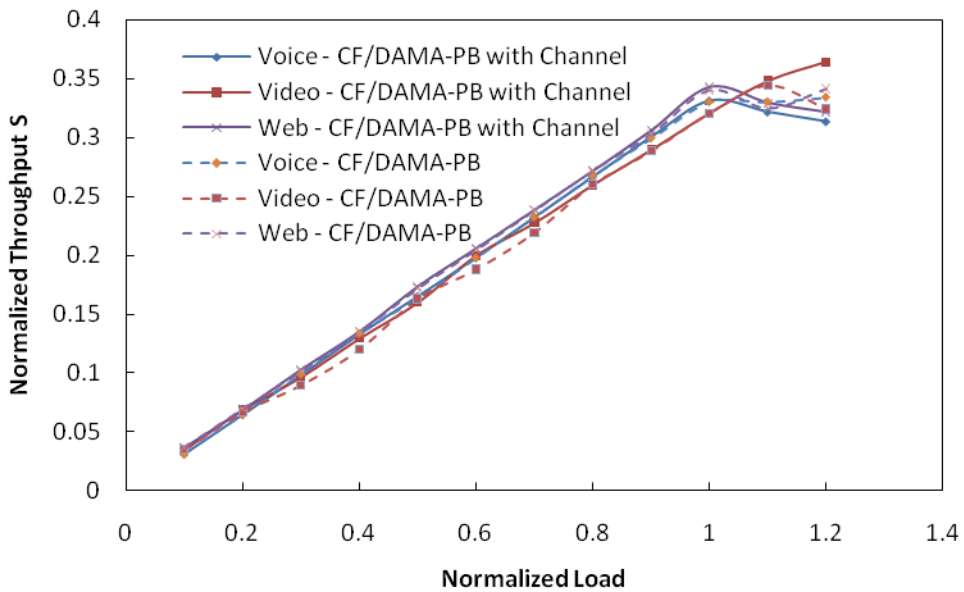

Figure 103: Normalized Throughput vs. Normalized Load (Scenario 1) CF/DAMA-PB with channel.

In order to see the channel losses one can record the number of lost packets due to the channel and thus have a better understanding of the extent of the impact that the channel may offer to the system performance, this is shown in Figure 104. With 6.5 million packets generated at a load of 1.2 and only 1244 packets lost due to the channel, this had a small, if any, impact on the overall system performance.

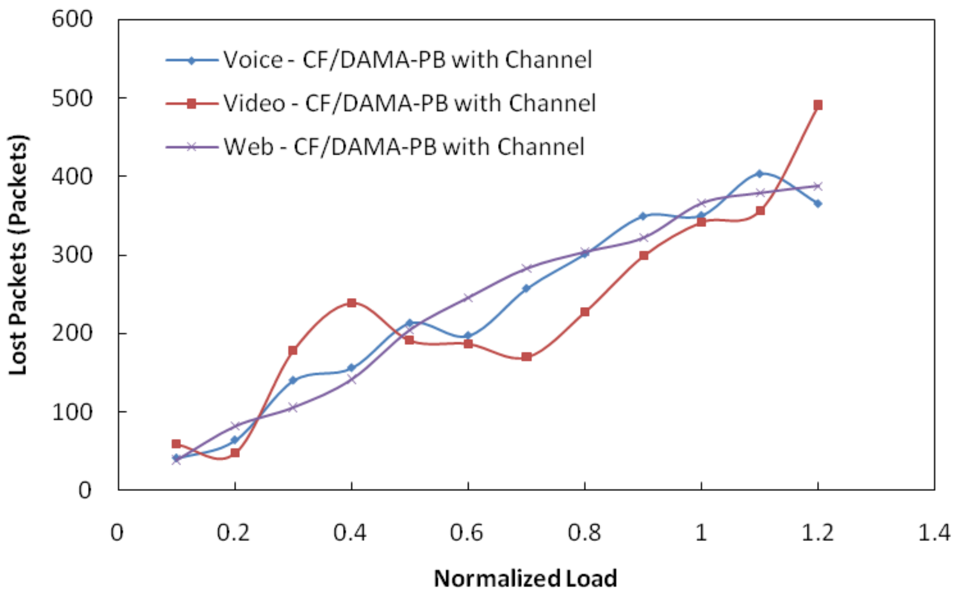

Figure 104: Lost Packets due to channel vs. Normalized Load (Scenario 1) CF/DAMA-PB with channel.

Scenario 2

In scenario 2, the traffic was divided between voice, video and web traffic classes with greater numbers of video sources than in scenario 1, i.e. 1/6 voice, 1/6 web and 2/3 video traffic making up the total offered traffic load.

The average packet delay experienced in the CF/DAMA-PB with a channel model is depicted in Figure 105. This was compared to the performance without a channel model.

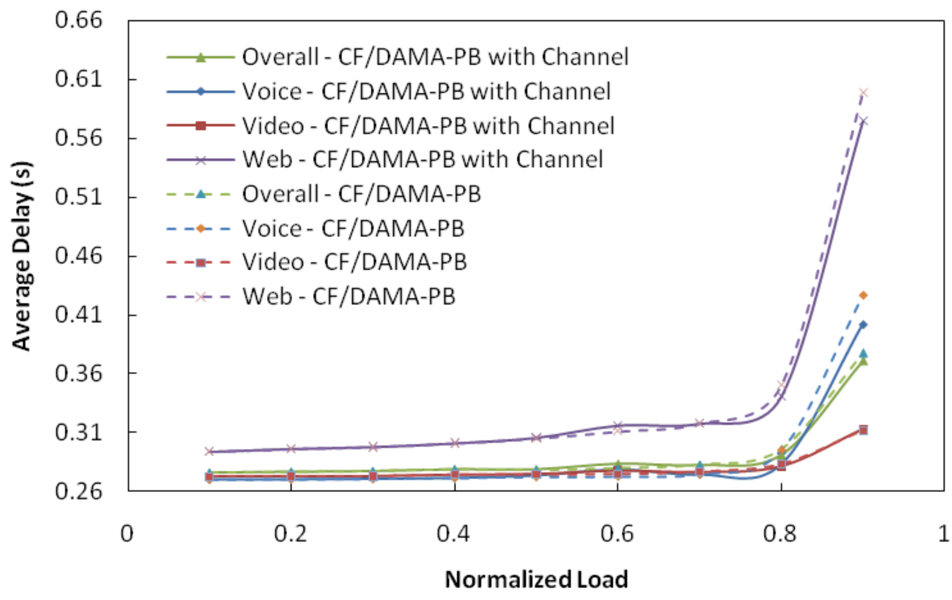

Figure 105: Average Delay vs. Normalized Load (Scenario 2) - CF/DAMA-PB with and without a channel model.

The CF/DAMA-PB scheme performed slightly worse than the CF/DAMA-PB scheme with a channel model in terms of the average packet delay. The packet loss due to the channel contributes to an increase in delay. The video class, being the dominant traffic class in this scenario has had little to no change in delay while the voice and web traffic classes had slightly differing delays with and without a channel model.

Once again, the overall throughput of the system had very little change with and without the channel model seen in Figure 106; this was attributed to the small number of lost packets relative to the number of generated packets.

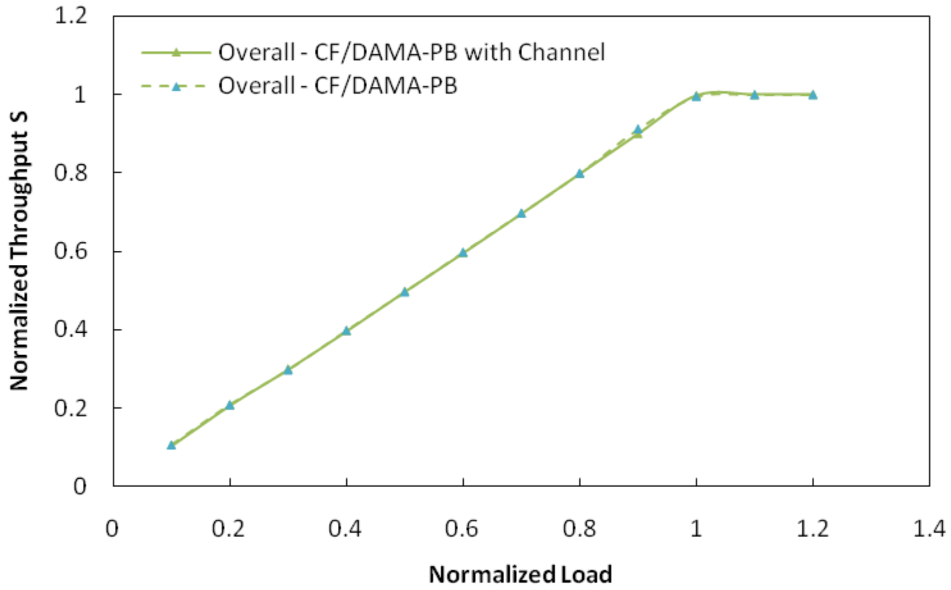

Figure 106: Normalized Throughput vs. Normalized Load (Scenario 2) CF/DAMA-PB with channel.

Similar to scenario 1, each traffic class had a contribution to the overall throughput of the system and thus the variations from the CF/DAMA-PB scheme can be attributed to statistical variations in the coding, while still given an overall same result due to the system compensating for a decrease in one traffic class by an increase in another. These being video decreased while web increased in Figure 107.

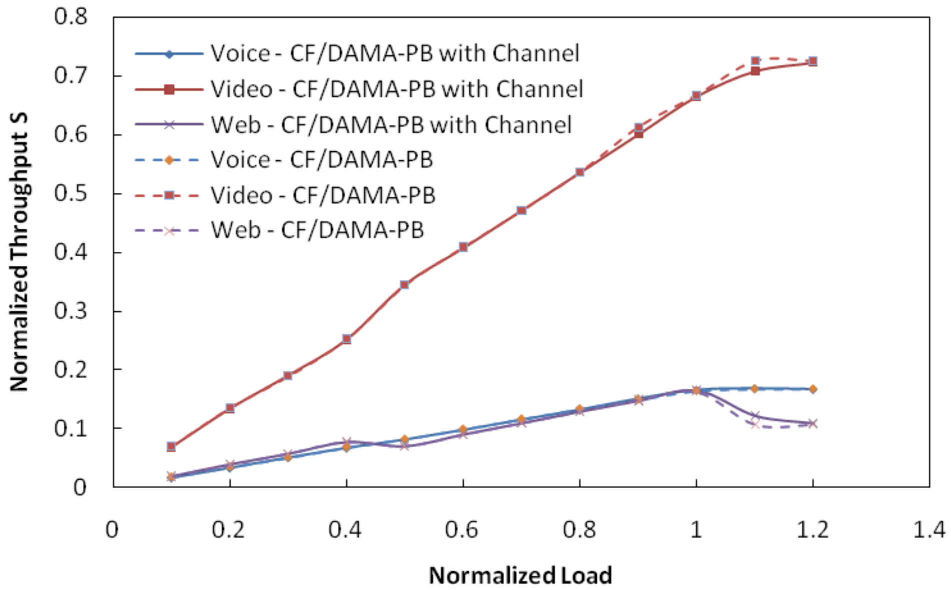

Figure 107: Normalized Throughput vs. Normalized Load (Scenario 2) CF/DAMA-PB with channel.

The effect the channel model had on the system is the number of lost packets due to the channel; this is shown in Figure 108. The lost packets did cause a decrease in the throughput of the system, but due to the relatively small number of lost packets compared to the generated packets, one cannot noticeably see this in the overall system performance depicted in Figure 106. The number of lost packets generally increased as the offered load increased, as expected. This is accounted for by the fact that if more packets are generated, the same percentage loss will result in more packets lost.

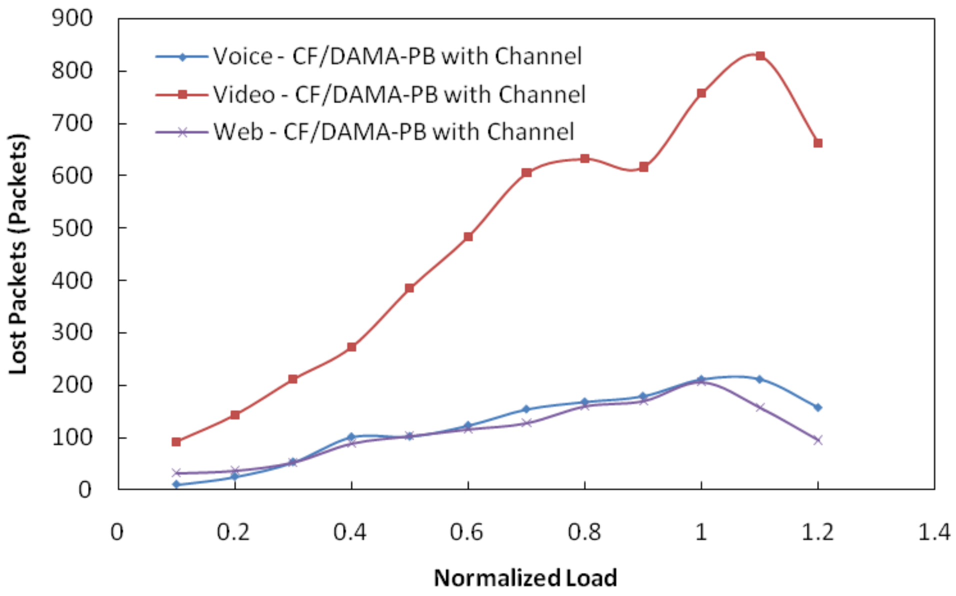

Figure 108: Lost Packets due to channel vs. Normalized Load (Scenario 2) CF/DAMA-PB with channel.

4.8.2 CF/DAMA-PB-PD Protocol

The CF/DAMA-PB-PD protocol was presented under channel conditions that are not ideal by adding the 3 state channel model to the system. This had the effect of increasing the number of packets dropped as a result of lost packets due to channel conditions.

Scenario 1

In scenario 1, once again, the offered load was equally divided into the three traffic classes, i.e. 1/3 voice, 1/3 video and 1/3 web traffic.

The CF/DAMA-PB-PD protocol outperformed the CF/DAMA-PB protocol in terms of the average packet delay for loads greater than 0.8. It had more of an effect on the packet delay of voice and video traffic, as the web traffic could not be dropped due to its virtual deadline. The dropping of the voice and video packets that exceed their deadlines, however, did have a positive effect on the web traffic packet delay, as seen in Figure 109, as the allocated slots of dropped packets may be used by the web packets as well as reducing the overall delay of web traffic. The deadlines of voice and video traffic created an upper limit to these traffic delays irrespective of the load applied to the system.

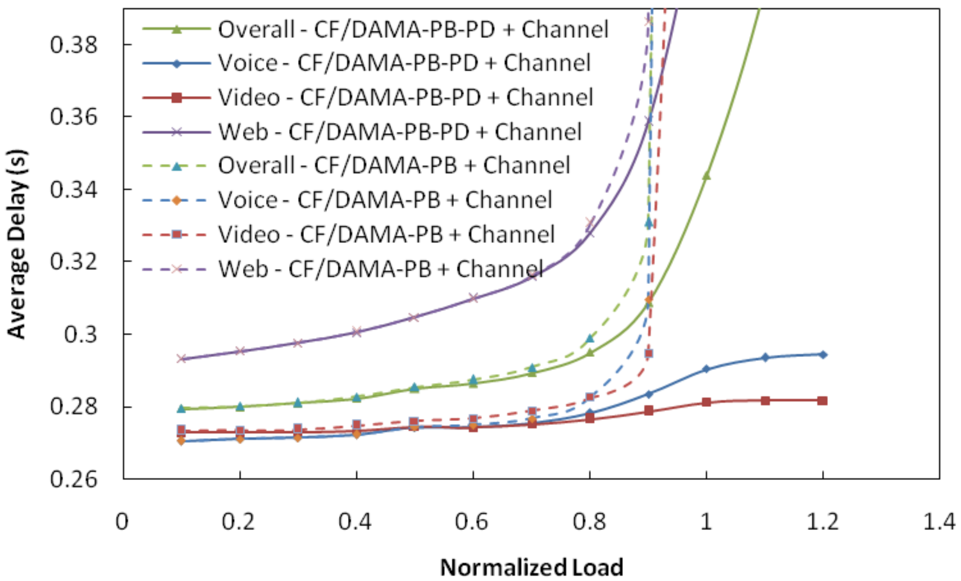

Figure 109: Average Delay vs. Normalized Load (Scenario 1) CF/DAMA-PB-PD with channel.

Due to the large number of dropped packets as a result of the deadlines and the lost packets caused by the channel losses, the overall system throughput was significantly reduced. This was effectively noticeable at loads in which the packet delay started to climb rapidly for the CF/DAMA-PB scheme, i.e. above loads of 0.8. This is depicted in Figure 110 below.

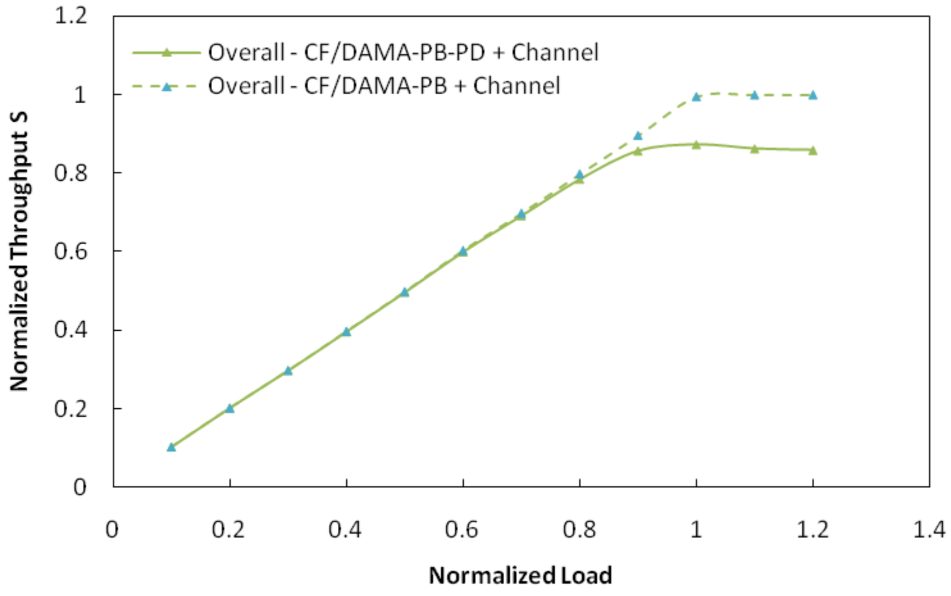

Figure 110: Normalized Throughput vs. Normalized Load (Scenario 1) CF/DAMA-PB-PD with channel.

The individual traffic class contributions to the overall throughput of the system are shown in Figure 111. This, similar to the ideal channel performance, indicated that the throughput drop was as a result of the packets dropped in the video and voice traffic classes and that once again the web achieved an improved throughput as it was able to utilise the allocated slots while benefiting from not having any packets dropped due to its virtual deadline. The system deteriorated for voice and video traffic compared to the system without packet dropping.

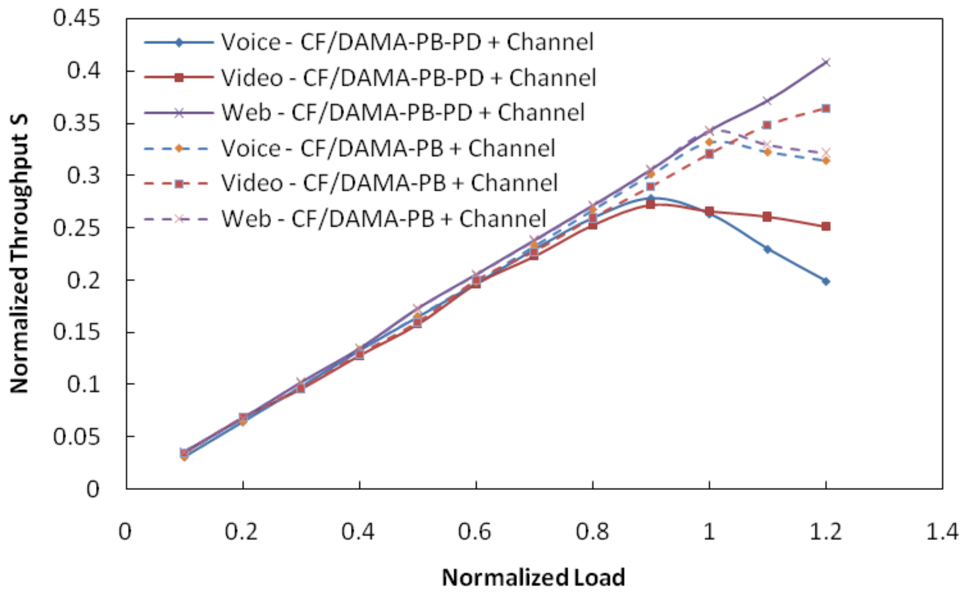

Figure 111: Normalized Throughput vs. Normalized Load (Scenario 1) CF/DAMA-PB-PD with channel.

The interesting set of results for this system was how much worse off the packet dropping was with a channel model compared to the ideal situation previously discussed. This is shown in Figure 112, where it was quite clear that packets lost to the channel do not have a huge impact on the overall performance, but do have a small part to play. There was not much change in dropped packets.

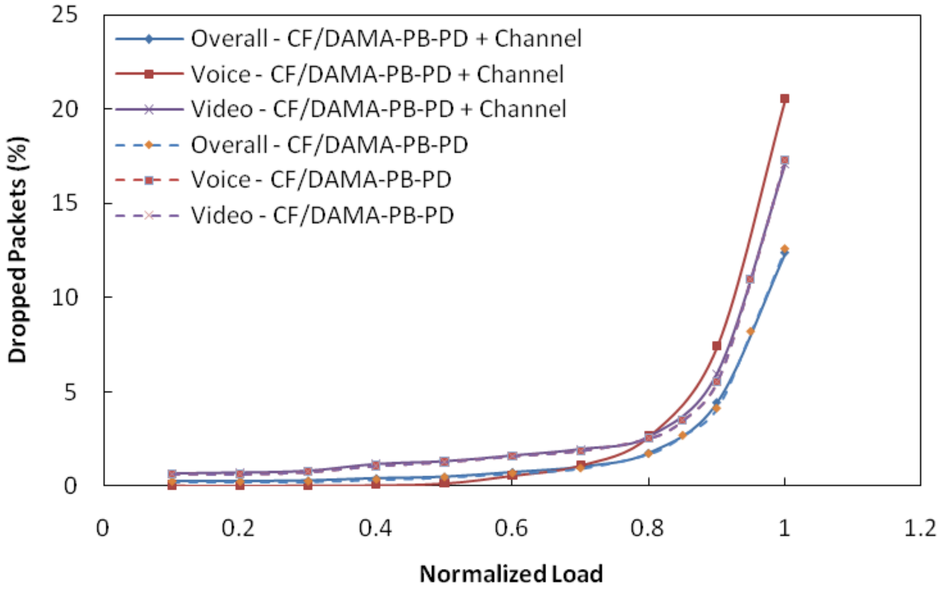

Figure 112: Percentage of dropped packets (Scenario 1) CF/DAMA-PB-PD with channel.

The overall percentage of dropped packets was a combination of the packets dropped due to deadlines and the packets lost to the channel all as a ratio to the total generated packets. This was seen to be smaller than the packet dropping for voice and video above loads of 0.7, as expected, as the web packets did not contribute to the dropped packets and hence lowered the percentage from the total generated packets.

Scenario 2

In this scenario, the offered load was majority video traffic, i.e. 1/6 voice 1/6 web and 2/3 video traffic make up the offered load.

Similar to the CF/DAMA-PB-PD protocol performance without a channel model, the average packet delay, depicted in Figure 113, improves on the overall and individual traffic packet delays. The deadlines created an upper limit for the voice and video traffic classes irrespective of the offered load. This helped reduce the web packet average delay by making slots allocated to the dropped packets available to the web packets, noting that these slots are also available to the queued video and voice packets as well. This gave a reduced average packet delay across the board and hence reduced the packet dropping slightly.

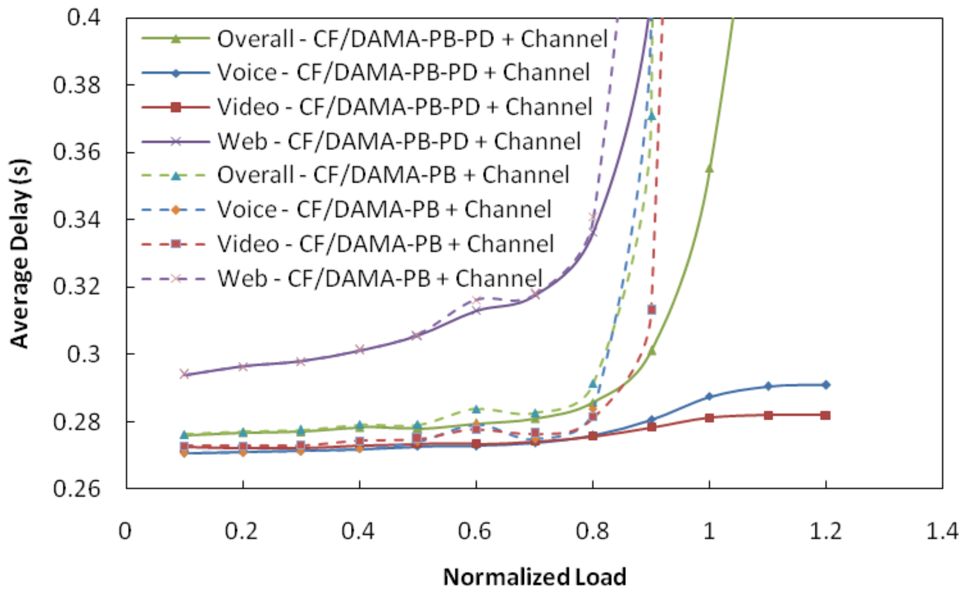

Figure 113: Average Delay vs. Normalized Load (Scenario 2) - CF/DAMA-PB-PD with channel.

The increased ratio of video packets to that of voice and web created more overall dropped packets as the offered load increased, which directly affected the system throughput, as seen in Figure 114. This is further backed by the graphs of Figure 115 illustrating that the video throughput had a large contribution to the overall throughput and as a result of increased number of dropped packets compared with scenario 1, a decrease in throughput occurred.

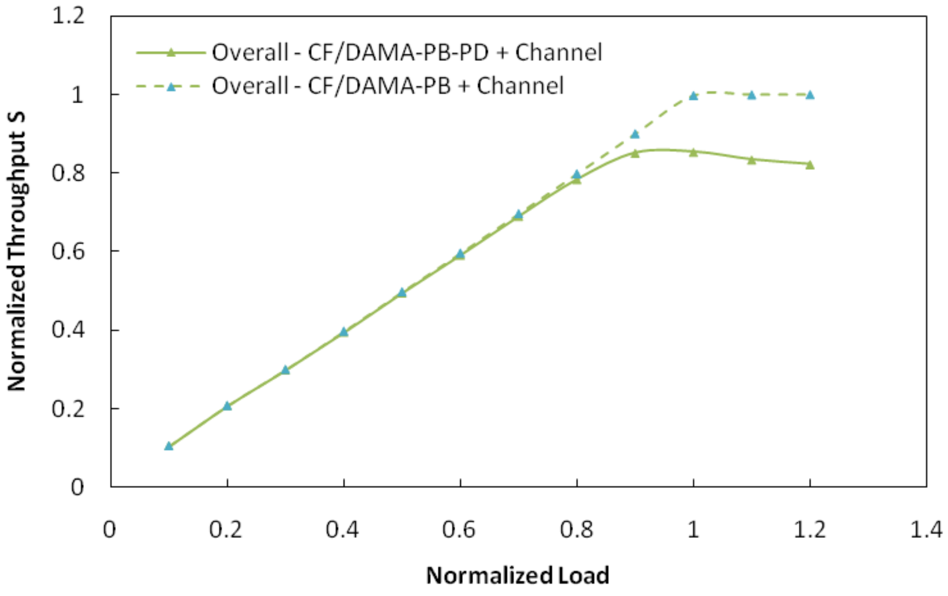

Figure 114: Normalized Throughput vs. Normalized Load (Scenario 2) CF/DAMA-PB-PD with channel.

Web traffic did slightly benefit in terms of the throughput, as the more dropped packets there were, the more slots were available to the traffic queued and hence a greater probability of transmitting web packets. Above an offered load of 0.8, the video class was significantly affected by the deadline assigned to it while the voice traffic was only significantly affected by its deadline above the 0.9 mark. The throughput reduced compared to that with no packet dropping.

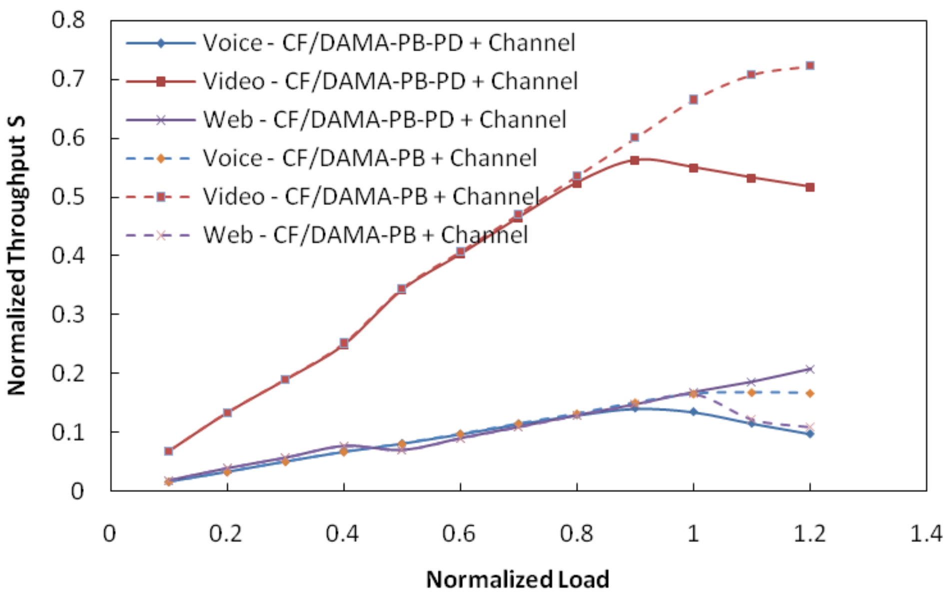

Figure 115: Normalized Throughput vs. Normalized Load (Scenario 2) CF/DAMA-PB-PD with channel.

4.8.3 CF/DAMA-PB-BSNRF Protocol

In this cross-layer design approach, the CF/DAMA-PB-BSNRF protocol, the channel state was taken into account and used to decide which RCSTs have priority over the others when allocating slots. The scheduler on the GEO satellites determined the channel conditions of the RCST making the request and compared it to the other RCSTs also making a request and decided whether it was allocated the slot in a highest SNR first basis.

Scenario 1

Scenario 1 divided the offered traffic load up into three equal amounts of voice, web and video class traffic, i.e. 1/3 voice, 1/3 video and 1/3 web traffic.

The graphs in Figure 116 below depict the average packet delay performance of the BSNRF protocol compared to that of the CF/DAMA-PB-PD protocol. Both with the same 3 state channel model.

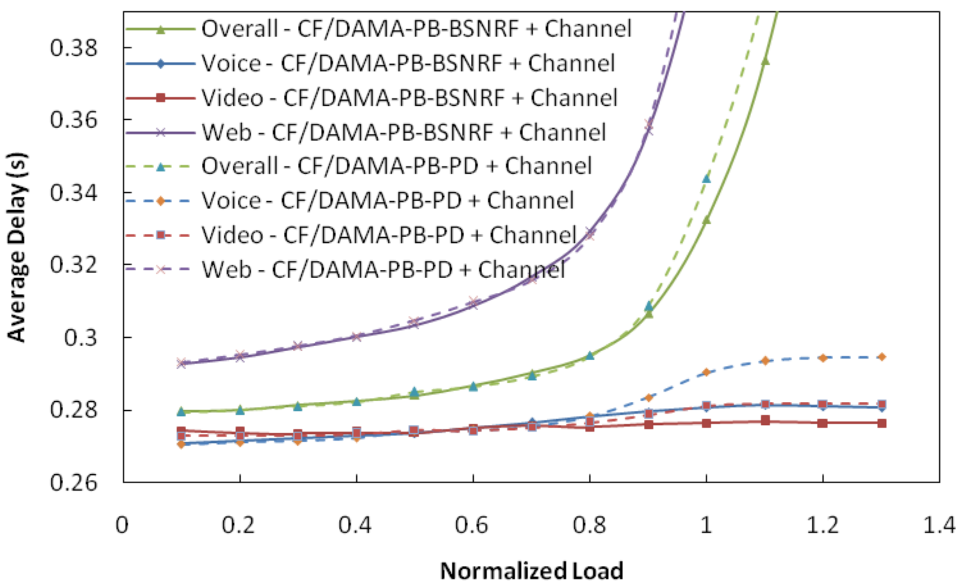

Figure 116: Average Delay vs. Normalized Load (Scenario 1) CF/DAMA-PB-BSNRF with channel.

The web traffic class was seen to have similar performance in both protocols. The video and voice traffic classes had a small decrease in average packet delay which increased system performance. The decrease in the average delay was attributed to prioritising the RCSTs with a better SNR, thus reducing the channel effects on lost packets and causing an increase in system performance. This however may have caused unfairness in the sense that if an RCST has a bad channel state for a prolonged period, those users will experience a decrease in system performance while the RCST with a constantly good state, LOS state, will have experienced better performance as it was given greater priority. The positive effect of this protocol was ensuring good system utilization; this is seen in Figure 117. The increased system performance compared to the CF/DAMA-PB-PD protocol was seen above loads of 1.0 when considering throughput and above loads of 0.9 when considering average packet delay.

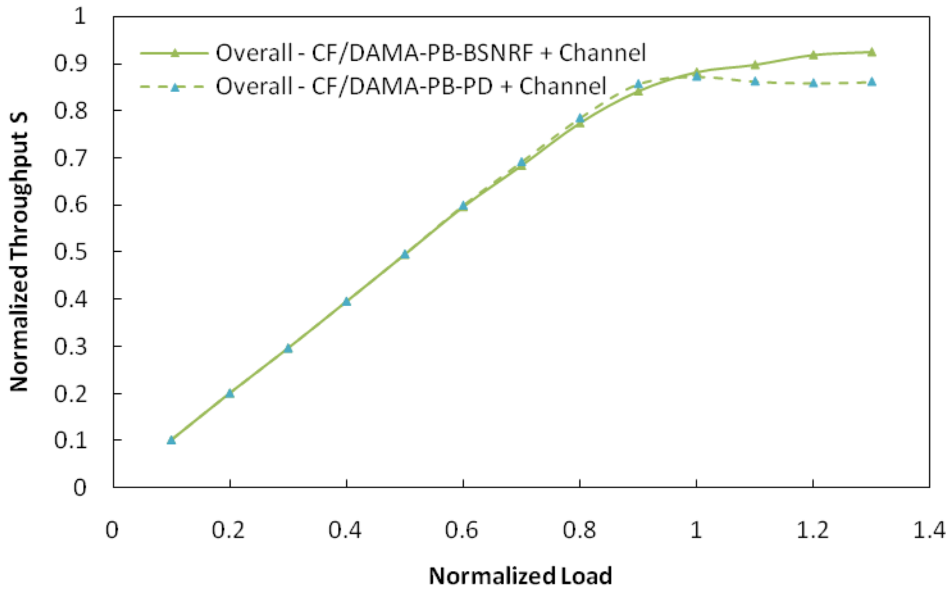

Figure 117: Normalized Throughput vs. Normalized Load (Scenario 1) CF/DAMA-PB-BSNRF with channel.

The main throughput enhancement was in the video traffic class, as shown in Figure 118. The interesting graphs were those of the percentage of dropped packets compared to that experienced in the CF/DAMA-PB-PD protocol, see Figure 119. The BSNRF performance in terms of packet dropping was decreased below loads of 1.0, after which the system benefited from the scheme by a reduction in dropped packets due to the channel and packet deadlines. The video traffic class achieved the best reduction in packet dropping; this supports the increase in throughput of video traffic, as seen in Figure 118.

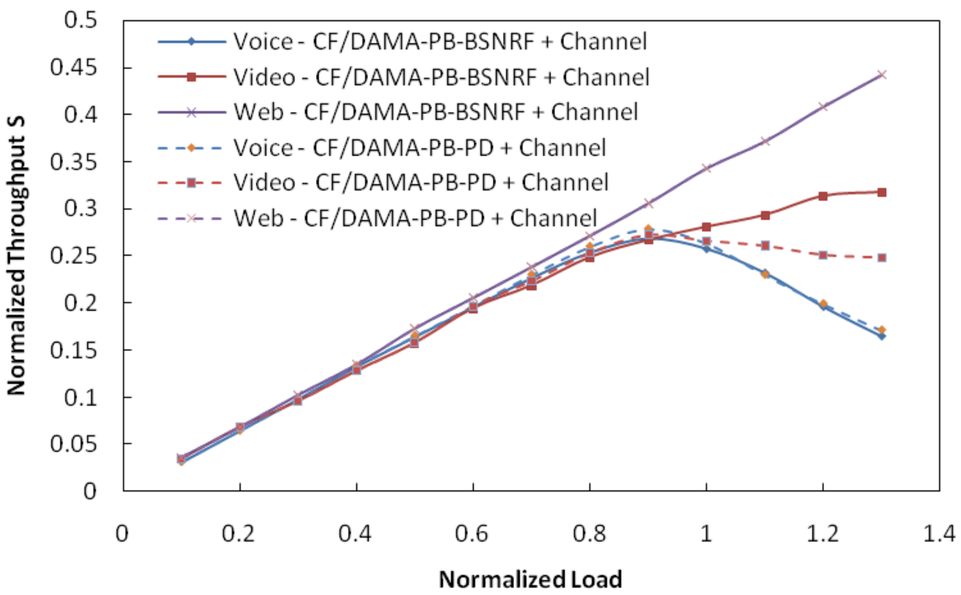

Figure 118: Normalized Throughput vs. Normalized Load (Scenario 1) CF/DAMA-PB-BSNRF with channel.

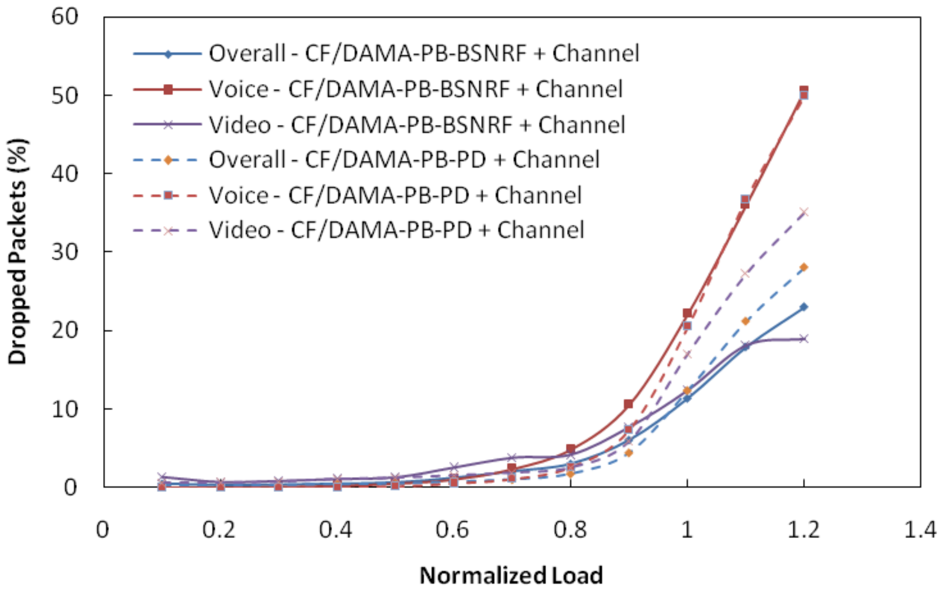

Figure 119: Percentage of dropped packets (Scenario 1) CF/DAMA-PB-BSNRF with channel.

The percentage of dropped packets became unacceptable above a load of 0.8 despite the improvement made at higher loads; the QoS that this protocol provides under scenario 1 is not good.

Scenario 2

The increased proportion of video class traffic to that of voice and web traffic did not offer any new system performance benefits that differ from that of scenario 1. The average packet delay, as depicted in Figure 120, of the voice and video packets was significantly improved, but still at the detriment of increasing the percentage of dropped packets to an unacceptable level for large loads. The web traffic did not achieve any noticeable improvement from this protocol compared to the CF/DAMA-PB-PD protocol.

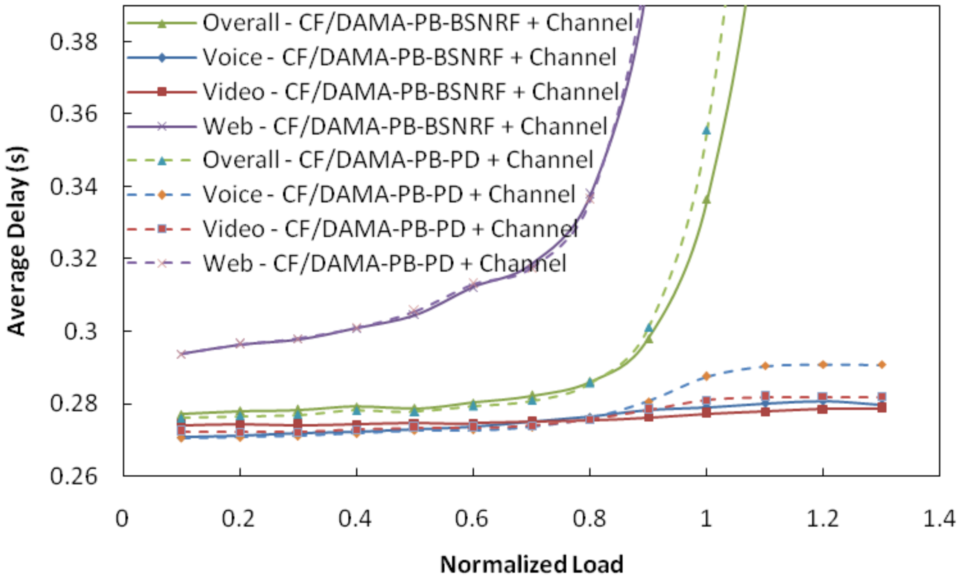

Figure 120: Average Delay vs. Normalized Load (Scenario 2) CF/DAMA-PB-BSNRF with channel.

The throughput was shown to increase as a result of CF/DAMA-PB-BSNRF, as seen in Figure 121 and Figure 122.

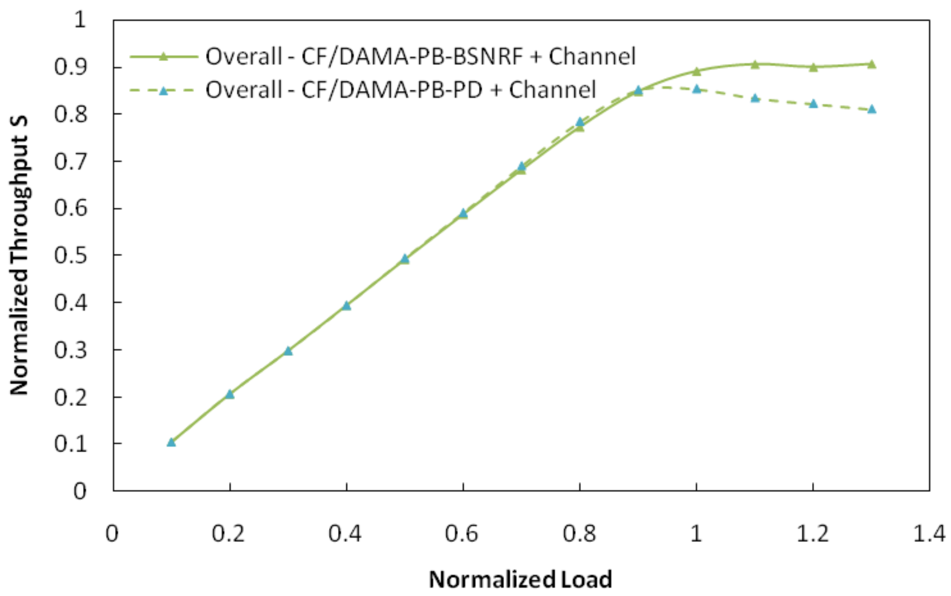

Figure 121: Normalized Throughput vs. Normalized Load (Scenario 2) CF/DAMA-PB-BSNRF with channel.

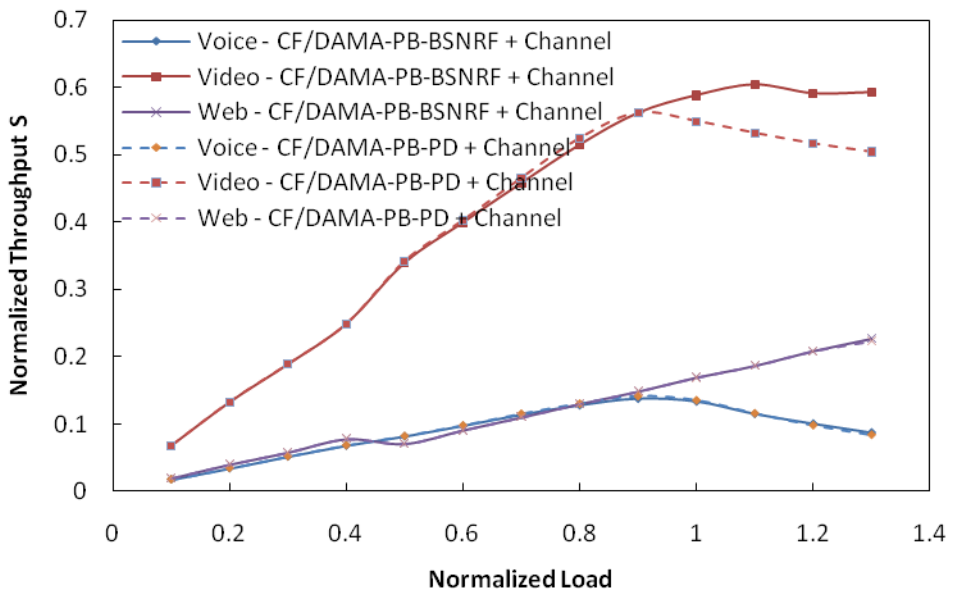

Figure 122: Normalized Throughput vs. Normalized Load (Scenario 2) CF/DAMA-PB-BSNRF with channel.

4.8.4 CF/DAMA-PB-DD Protocol

The CF/DAMA-PB-DD protocol accounts for the channel losses and in the process also reduces the packets lost to strict deadlines by making use of the cross-layer information made available to the data link layer by the physical layer. This information is the channel state information and the SNR associated with the states. For high SNR channel states, the deadline is tightened up, while for low SNR channel states the deadlines are slightly loosened. This also ensures that a minimal amount of unfairness is achieved when compared to the CF/DAMA-PB-BSNRF protocol.

Scenario 1

In scenario one the offered load was divided equally among the traffic classes, i.e. 1/3 voice, 1/3 video and 1/3 web traffic.

In Figure 123 the average packet delay vs. Offered load is depicted illustrating the difference between the CF/DAMA-PB-DD protocol and the CF/DAMA-PB-PD protocol under the 3 state channel model. The average packet delay was increased marginally by the CF/DAMA-PB-DD protocol when compared to the CF/DAMA-PB-PD protocol with greater increases seen above loads of 1.0. This is due to the increasing of deadlines from that of the deadlines applied in the CF/DAMA-PB-PD protocol according to SNR values.

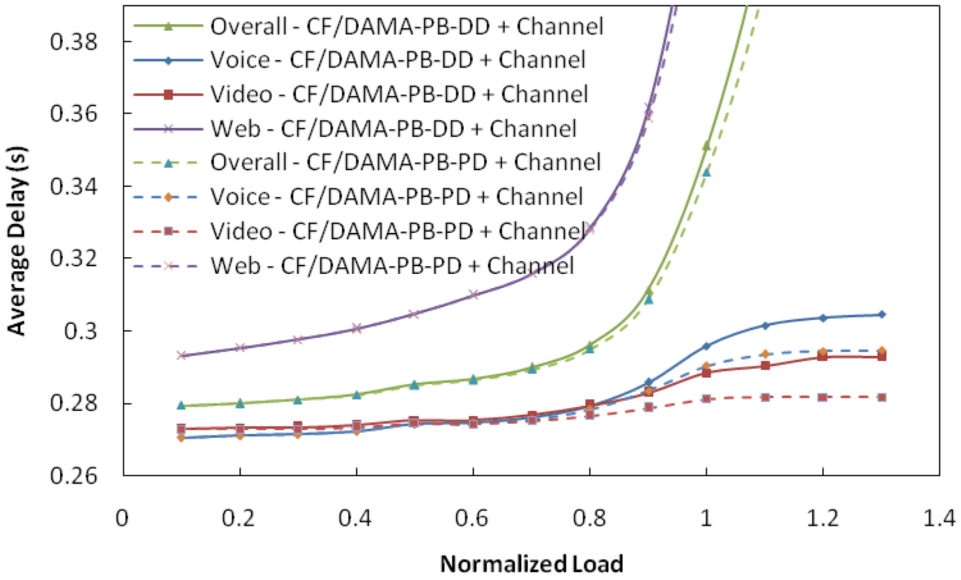

Figure 123: Average Delay vs. Normalized Load (Scenario 1) CF/DAMA-PB-DD with channel.

As a result of the increased average deadlines of voice and video traffic, the packets dropped would decrease, hence increasing the overall throughput of the video and voice traffic in the system as seen in Figure 124 and Figure 125. Figure 125 shows throughput of web traffic was unaffected as web traffic had virtual deadlines which do not drop packets (the largest impact on throughput).

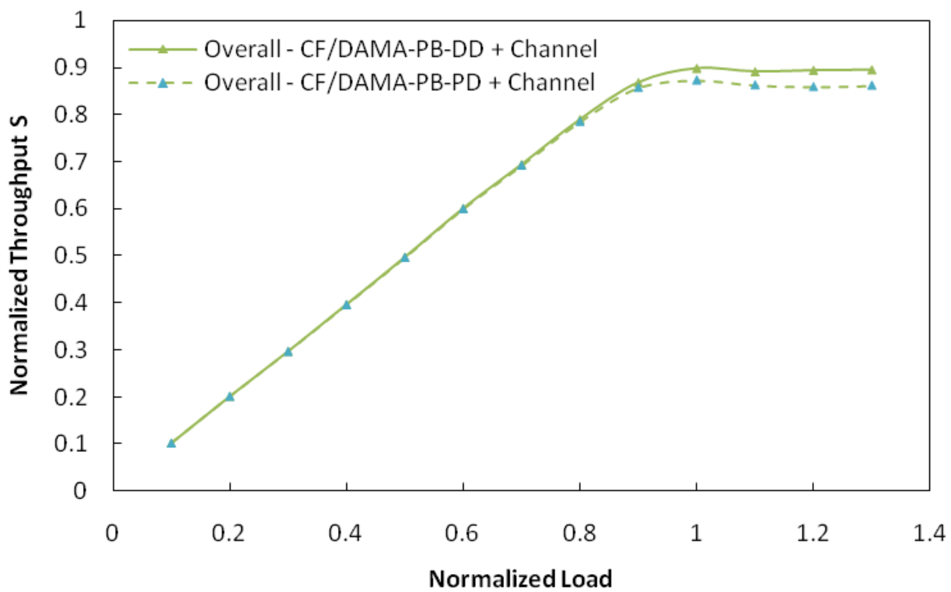

Figure 124: Normalized Throughput vs. Normalized Load (Scenario 1).

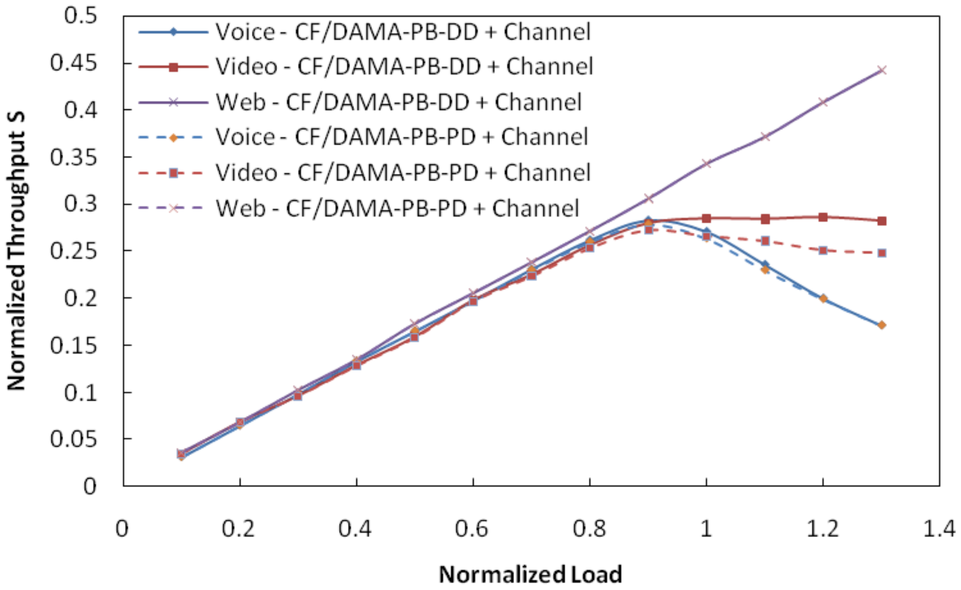

Figure 125: Normalized Throughput vs. Normalized Load (Scenario 1) CF/DAMA-PB-DD with channel.

The following figure, Figure 126, shows the extent of the reduction of packet dropping experienced in the CF/DAMA-PB-DD protocol compared to the CF/DAMA-PB-PD protocol. The overall reduction allowed for a better QoS guarantee for loads higher than those in the CF/DAMA-PB-PD protocol while also ensuring the individual traffic classes, voice and video, also have significant improvements too.

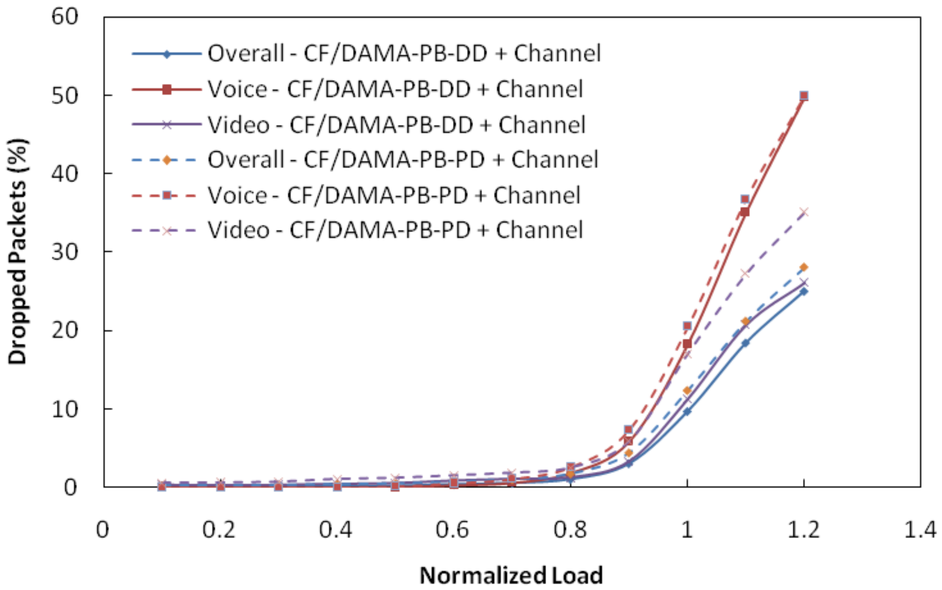

Figure 126: Percentage of dropped packets (Scenario 1) CF/DAMA-PB-DD with channel.

Scenario 2

In scenario 2 an uneven ratio of traffic classes made up the offered load, i.e. 1/6 voice, 1/6 web and 2/3 video traffic.

As in scenario 1, there was a small increase in the overall average packet delay, as well as in the individual traffic class's average packet delay, when compared to the CF/DAMA-PB-PD protocol, as seen in Figure 127. This was due to the average deadline of the CF/DAMA-PB-DD protocol being slightly higher than the fixed deadlines in the CF/DAMA-PB-PD protocol.

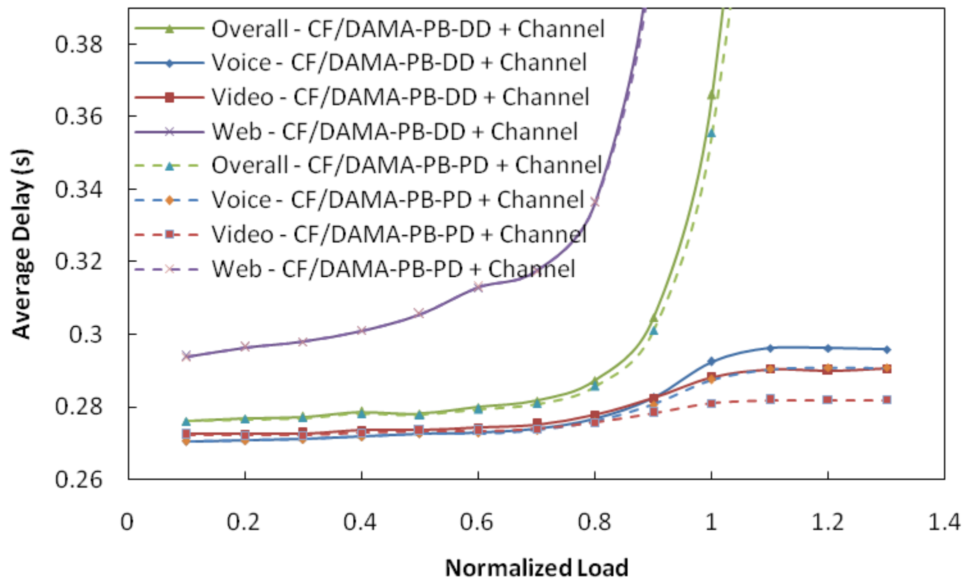

Figure 127: Average Delay vs. Normalized Load (Scenario 2) CF/DAMA-PB-DD with channel.

The slightly higher average deadlines of the CF/DAMA-PB-DD protocol are also the reason why the increased throughput was experienced in Figure 128. Figure 128 also shows that the traffic scenario used affects the extent to which an increase of throughput was experienced. This was due to the traffic classes that have packet dropping deadlines can contribute to the throughput change whereas the web traffic class could not as packets are not drop as a result of a virtual deadline. Thus, the greater ratio of video to web, as seen in Figure 129, increases the ability for the CF/DAMA-PB-DD protocol to improve on the overall system throughput. The increase in throughput was only made for loads that had higher average packet delays, as there was a higher probability of having a packet dropped due to deadlines. Due to the video traffic having had a significantly higher ratio traffic contribution than voice and web traffic, the effects of the system on the traffic would be better seen in the video traffic as depicted in Figure 129.

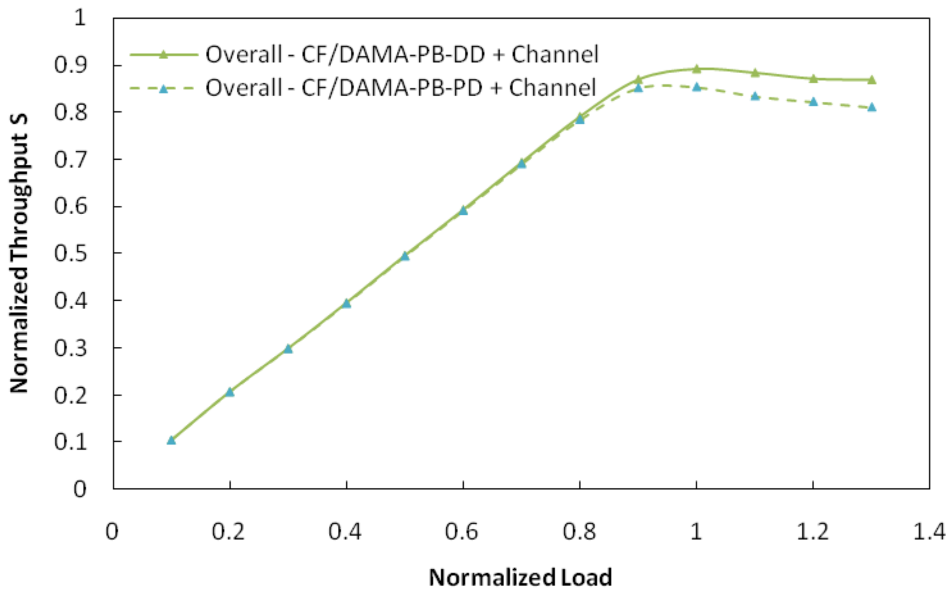

Figure 128: Normalized Throughput vs. Normalized Load (Scenario 2) CF/DAMA-PB-DD with channel.

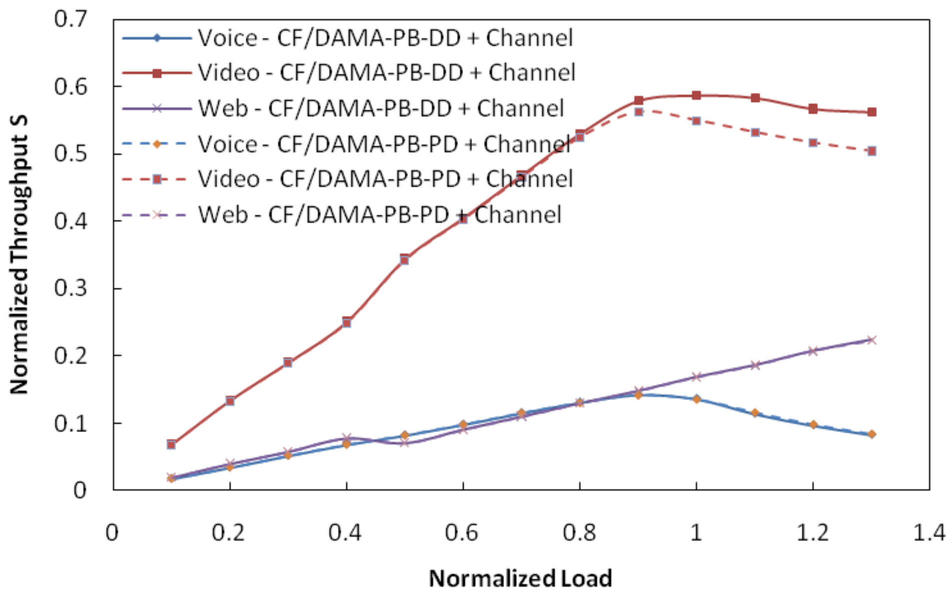

Figure 129: Normalized Throughput vs. Normalized Load (Scenario 2) CF/DAMA-PB-DD with channel.

The percentage of dropped packets experienced would have also decreased similar to that in scenario 1. This is illustrated with the results obtained for percentage of dropped packets in Figure 130, showing that video had the dominating effect on the overall packet dropping experienced across the loads.

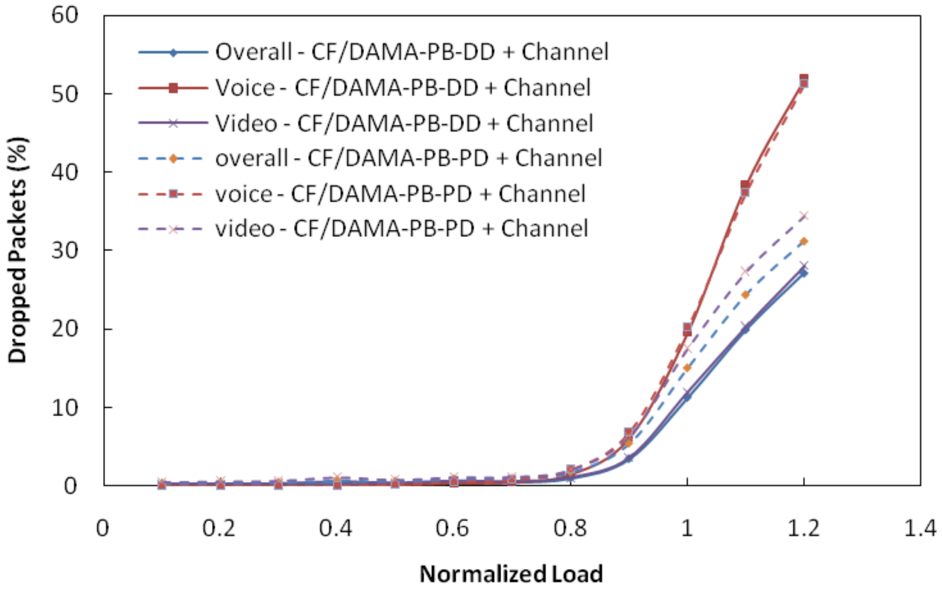

Figure 130: Percentage of dropped packets (Scenario 2) CF/DAMA-PB-DD with channel.

4.8.5 CF/DAMA-PB-BSNRF+DD Protocol

This protocol tries to combine the better average packet delay performance offered by the CF/DAMA-PB-BSNRF Protocol with that of the improved packet dropping performance of the CF/DAMA-PB- DD protocol. Two comparisons, each under scenario 1 and scenario 2 traffic conditions, will be shown, the comparison of the CF/DAMA-BSNRF and Dynamic Deadline Protocol (CF/DAMA-PB-BSNRF+DD) to that of the CF/DAMA-PB-DD protocol and the comparison of the CF/DAMA-PB-BSNRF+DD to that of the CF/DAMA-PB-BSNRF protocol performance. Again, the 3 state model was used for the channel.

Scenario 1

In scenario 1 the offered load was evenly divided into voice, web and video traffic. This created a multimedia traffic environment, i.e. 1/3 voice, 1/3 video and 1/3 web traffic.

The CF/DAMA-PB-BSNRF+DD protocol had very little difference to that of the average packet delay performance of CF/DAMA-PB-BSNRF protocol, depicted in Figure 131. A small increase in delay was seen for the voice and video traffic, this was due to the Dynamic Deadline approach of varying the deadline according to the SNR of each requesting RCST such that the average deadline was slightly higher than the fixed deadline.

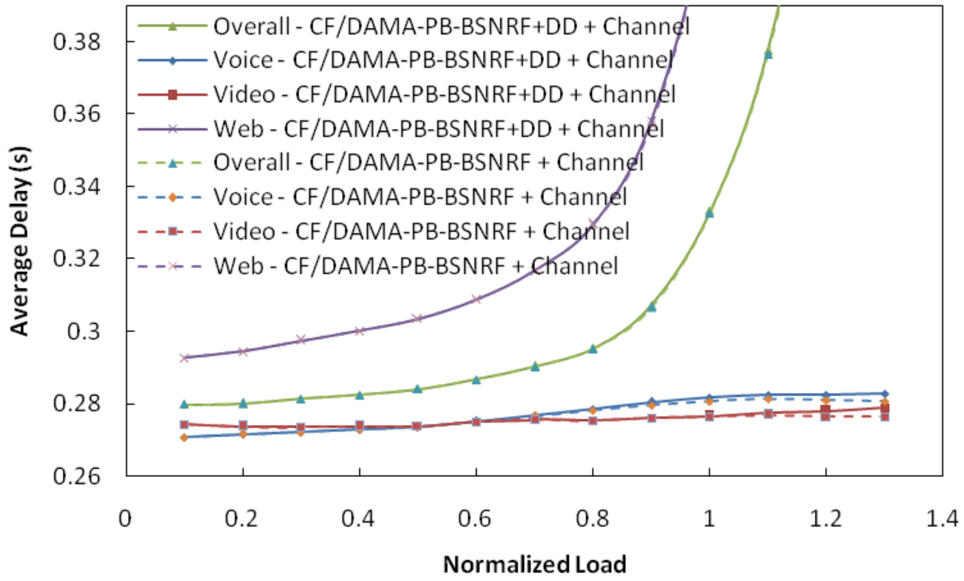

Figure 131: Average Delay vs. Normalized Load (Scenario 1) CF/DAMA-PB-BSNRF+DD with channel.

The throughput had small, but almost insignificant improvement compared to the CF/DAMA-PB-BSNRF protocol, as the CF/DAMA-PB-DD did not impact the throughput in a drastic fashion, but rather tried to increase the throughput across all RCSTs depending on the SNR values of the RCSTs, depicted in Figure 132 and Figure 133. Figure 133 depicts the contribution that each traffic type made to the overall throughput performance. The web traffic was unaffected by adding dynamic deadlines to the system as the web traffic had virtual deadlines which did not allow for the dropping of packets. Packet dropping had an impact on the throughput performance of the system.

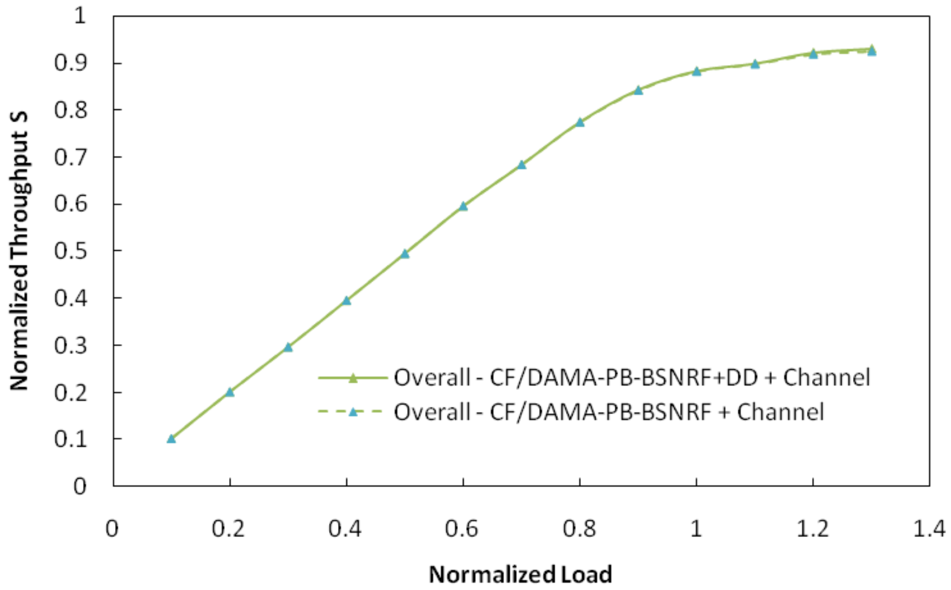

Figure 132: Normalized Throughput vs. Normalized Load (Scenario 1) CF/DAMA-PB-BSNRF+DD with channel.

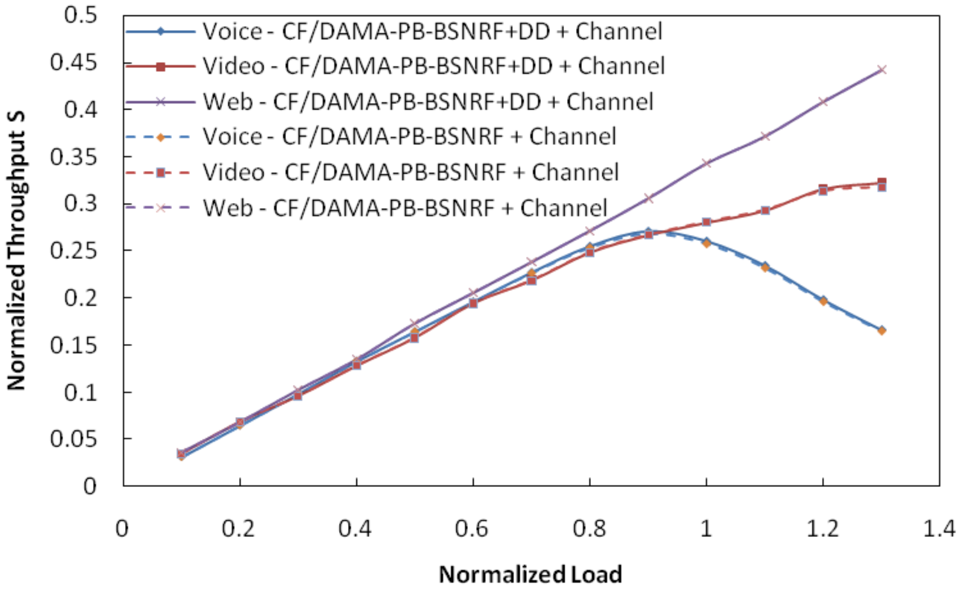

Figure 133: Normalized Throughput vs. Normalized Load (Scenario 1) CF/DAMA-PB-BSNRF+DD with channel.

CF/DAMA-PB-BSNRF+DD had a significant improvement in average packet delay experienced in the system compared to that of the CF/DAMA-PB-DD scheme. This is seen in Figure 134. This was attributed to the BSNRF prioritising the better channel state RCSTs over those with low SNR values. This decreased lost packets due to the channel but increased the dropped packets due to the deadlines.

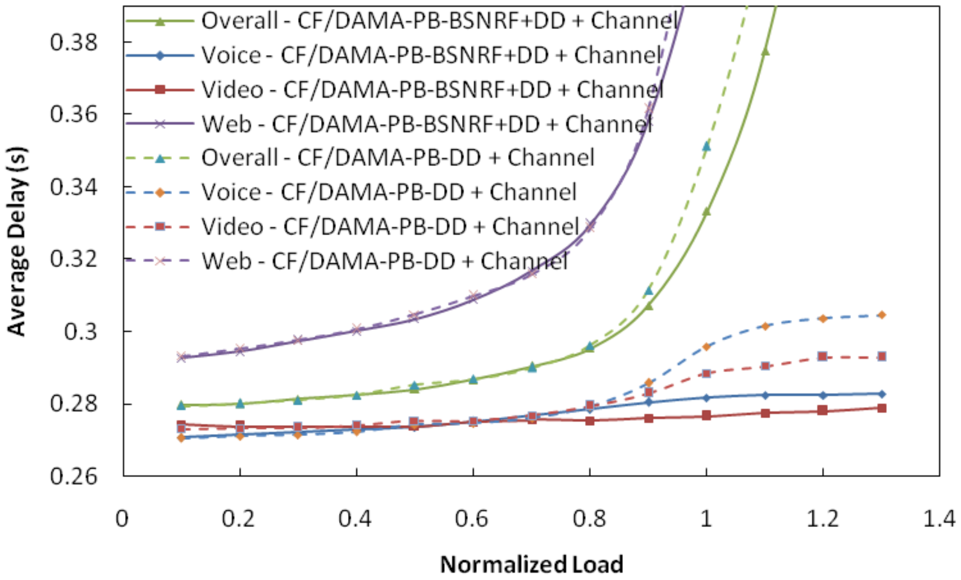

Figure 134: Average Delay vs. Normalized Load (Scenario 1) CF/DAMA-PB-BSNRF+DD with channel.

As can be seen in Figure 135 and Figure 136, the throughput of the system was reduced below a load of 1.1 while increased beyond this point. From Figure 136 it was clear that the voice and video class traffic were the contributing factor to the change in throughput. The BSNRF contributed to the increase of throughput at high loads but also was a negative aspect at the low loads as more packet dropping will be experienced if a RCST was in a bad channel state. This could be overcome by adding a call admission algorithm that took into account the probability of being allocated resources. In Figure 136 one can see an increase in video traffic throughput above a load of 1 and a small increase of voice traffic throughput above loads of 0.7. There was no difference, as expected, in web traffic throughput as web traffic had a virtual deadline which did not allow the dropping of packets. Once again, as in Figure 133, the web traffic throughput performance depicted in Figure 36 was unaffected by adding dynamic deadlines to the system as the web traffic had virtual deadlines which did not allow for the dropping of packets. Packet dropping had an impact on the throughput performance of the system.

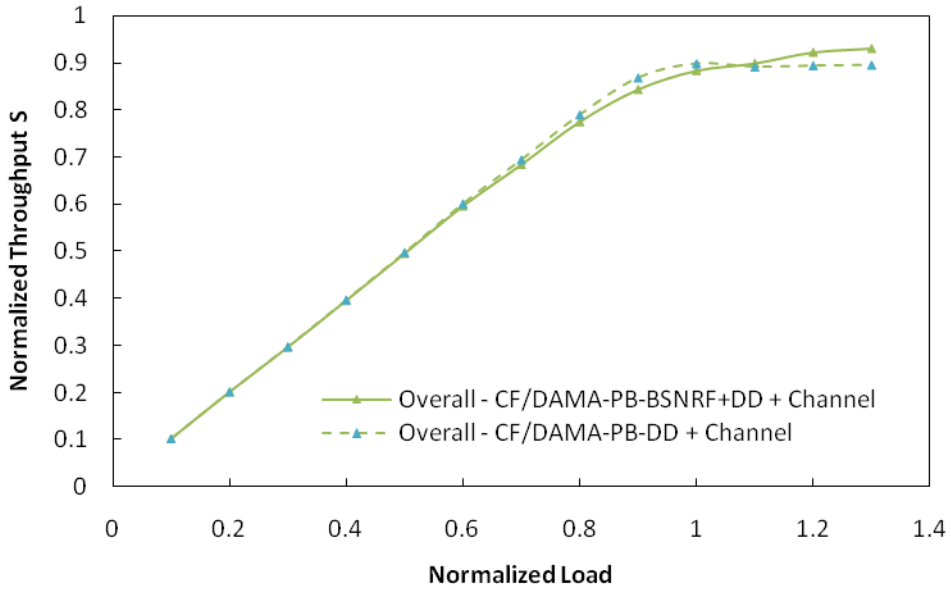

Figure 135: Normalized Throughput vs. Normalized Load (Scenario 1) CF/DAMA-PB-BSNRF+DD with channel.

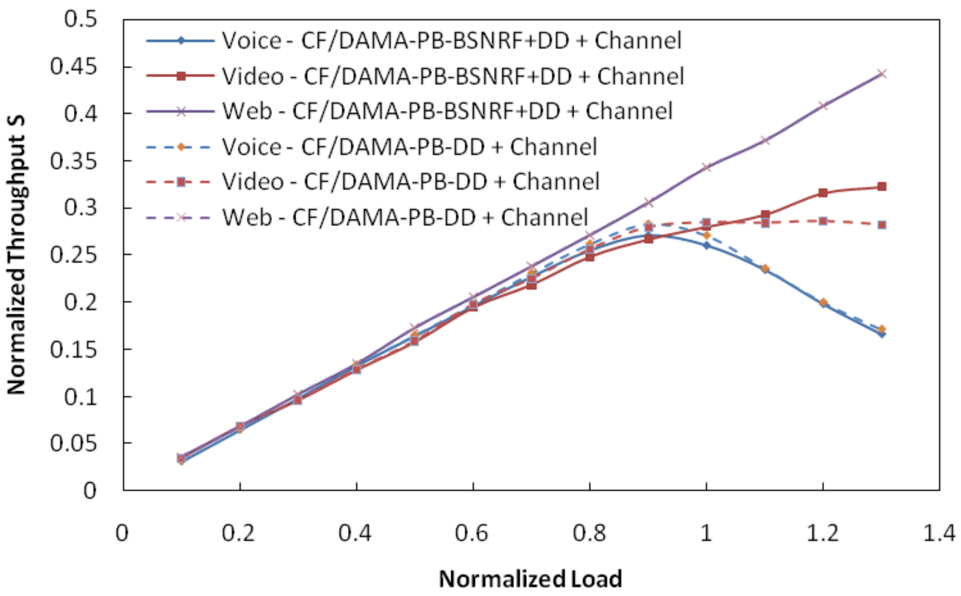

Figure 136: Normalized Throughput vs. Normalized Load (Scenario 1) CF/DAMA-PB-BSNRF+DD with channel.

Scenario 2

Scenario 2 traffic was divided into voice, video and web traffic such that video makes up the main capacity, i.e. 1/6 voice, 1/6 web and 2/3 video traffic.

Scenario 2 seemed to be similar to that of scenario 1, showing that the traffic did not have too much influence on the overall performance of the CF/DAMA-PB-BSNRF+DD protocol. The average delay was maintained when comparing the CF/DAMA-PB-BSNRF+DD to the CF/DAMA-PB-BSNRF protocol with slight increases in average packet delay above loads of 0.8 for the voice and video class traffic, see Figure 137. This was due to the increased average delay offered by the Dynamic Deadline Protocol.

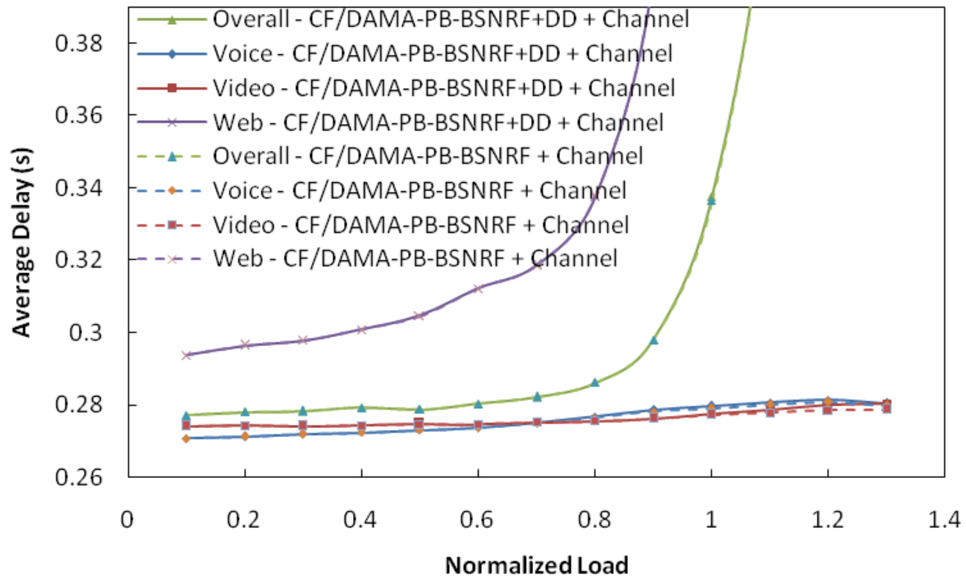

Figure 137: Average Delay vs. Normalized Load (Scenario 2) - CF/DAMA-PB-BSNRF+DD with channel.

No significant changes were seen in the overall and individual throughputs of the CF/DAMA-PB-BSNRF+DD when compared to the CF/DAMA-PB-BSNRF protocol performance. This is depicted in Figure 138 and Figure 139. This, once again, shows that the traffic scenario did not have a significant impact on the overall system performance. A traffic scenario that would have had a huge effect on the overall system performance would have been one in which web traffic was the only traffic or web traffic made up a large portion of the traffic in a network. This would reduce the effectiveness of packet dropping on delay as web traffic cannot be dropped. Figure 139 depicts an insignificant increase in video traffic throughput.

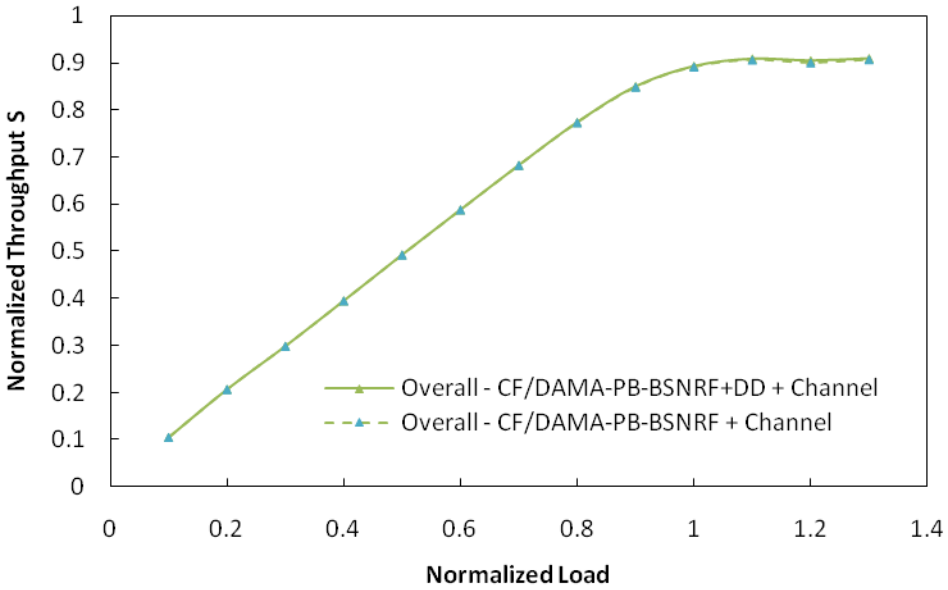

Figure 138: Normalized Throughput vs. Normalized Load (Scenario 2) CF/DAMA-PB-BSNRF+DD with channel.

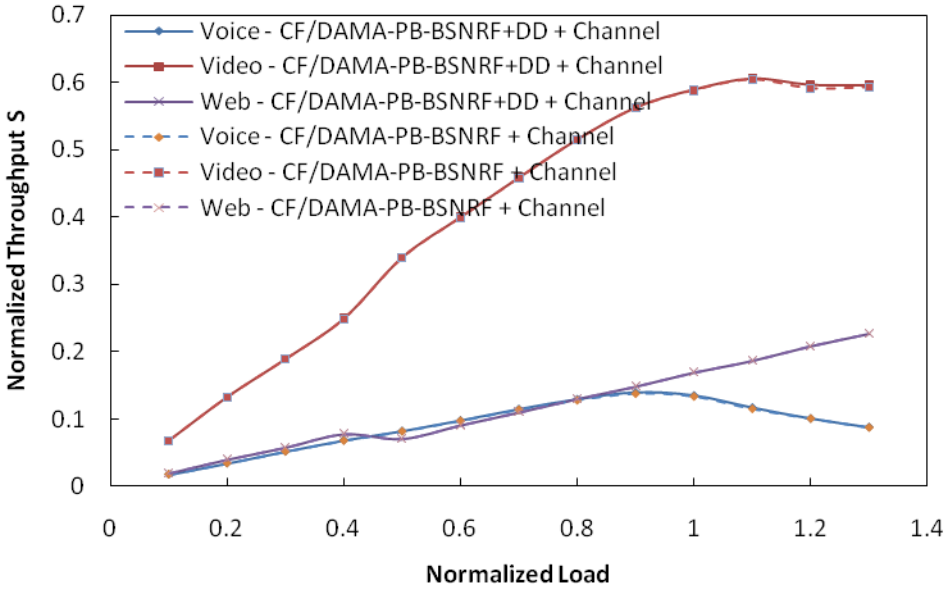

Figure 139: Normalized Throughput vs. Normalized Load (Scenario 2) CF/DAMA-PB-BSNRF+DD with channel.

Scenario 2 was similar to scenario 1 when comparing BSNRF DD and the CF/DAMA-PB-DD performances. The CF/DAMA-PB-BSNRF+DD had significant improvements in average packet delay over that of the CF/DAMA-PB-DD protocol; this is depicted in Figure 140.

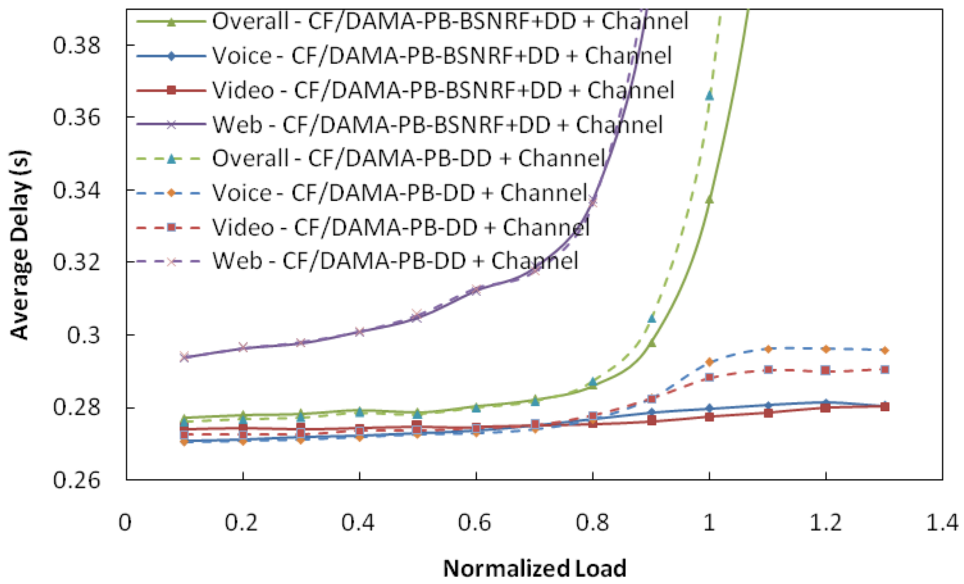

Figure 140: Average Delay vs. Normalized Load (Scenario 2) CF/DAMA-PB-BSNRF+DD with channel.

The system throughput was the same in scenario 2 as in scenario 1, decreased slightly below offered loads of 1.0 but increased above this. The increase was a result of the prioritising of RCSTs with higher SNR values over those with lower SNR values. See graphs in Figure 141 and Figure 142.

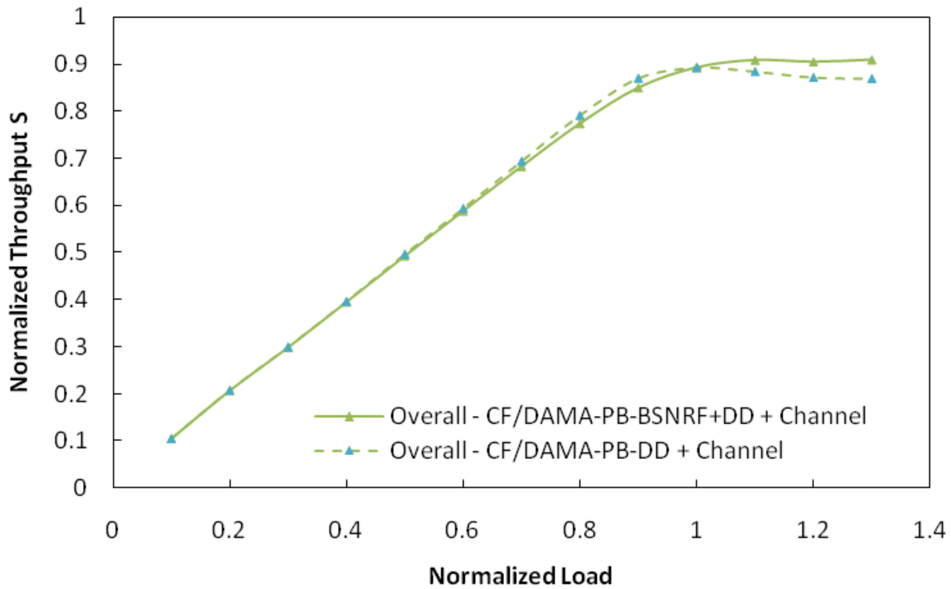

Figure 141: Normalized Throughput vs. Normalized Load (Scenario 2) CF/DAMA-PB-BSNRF+DD with channel.

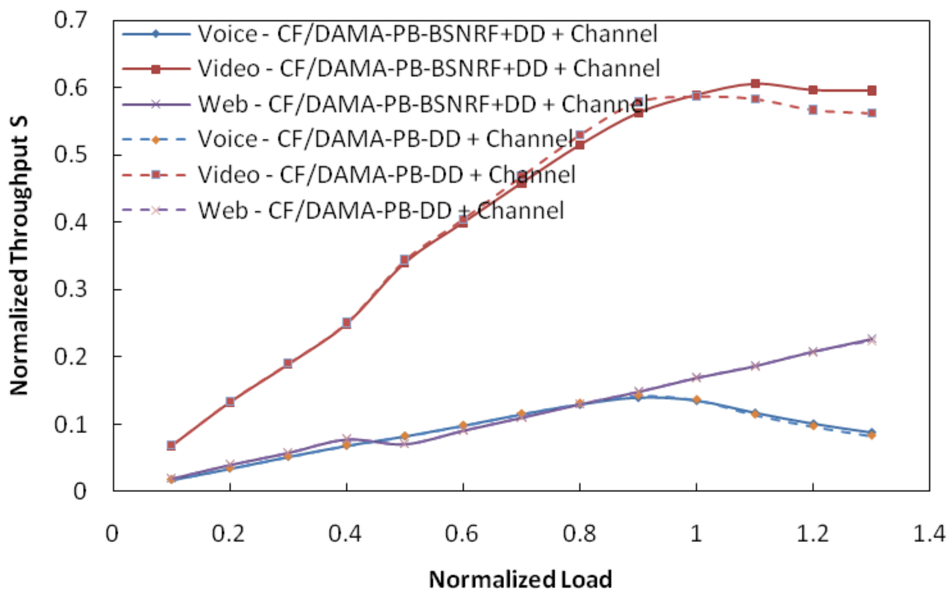

Figure 142: Normalized Throughput vs. Normalized Load (Scenario 2) CF/DAMA-PB-BSNRF+DD with channel.

The difference between the three protocols, CF/DAMA-PB-DD, CF/DAMA-PB-BSNRF and CF/DAMA-PB-BSNRF+DD Protocol, is illustrated in the following graphs of average packet delay vs. Normalized load, Normalized throughput vs. Normalized Load and Percentage dropped packets vs. Normalized load. Two traffic scenarios were used to compare the overall system performance of the three protocols, these being scenario 1 and scenario 2.

Scenario 1

Under scenario 1 traffic, the average packet delay experienced was the lowest in the CF/DAMA-PB-BSNRF and CF/DAMA-PB-BSNRF+DD protocols which are almost identical in performance as depicted in Figure 143. The CF/DAMA-PB-DD protocol had the worst, when compared to the other two protocols, performance in terms of average packet delay experienced in the same GEO satellite DVB-RCS system.

In Figure 144 the overall throughput of the CF/DAMA-PB-DD protocol was better than the other two for loads less than 1.0, while the other two, similar performing protocols, showed better results for loads greater than 1.0.

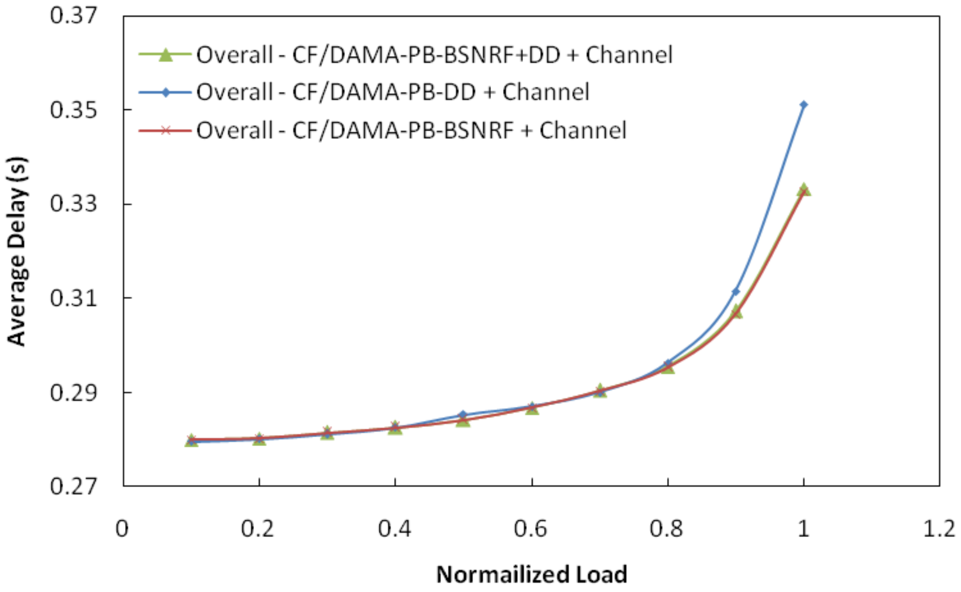

Figure 143: Average Delay vs. Normalized Load (Scenario 1) CF/DAMA-PB-BSNRF+DD with channel.

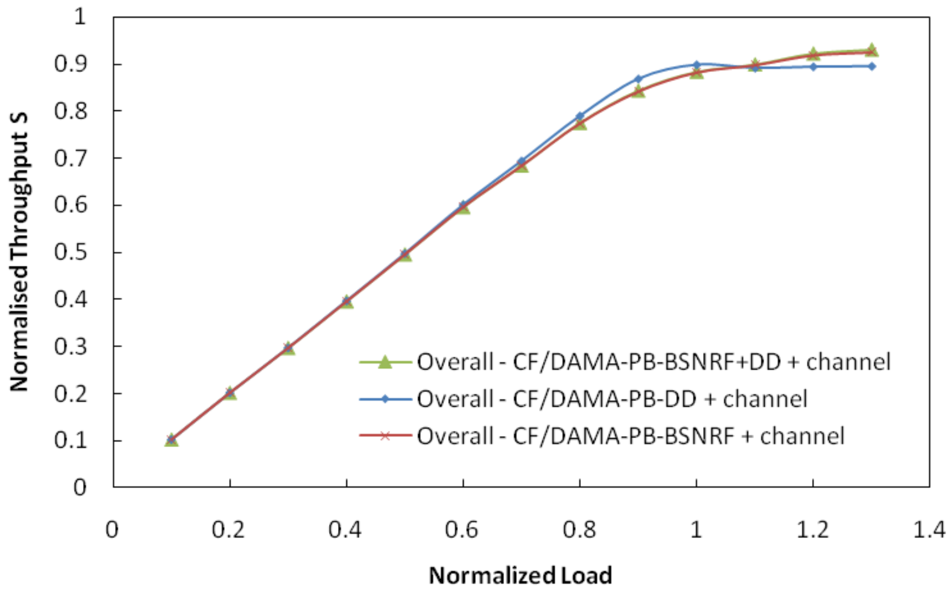

Figure 144: Normalized Throughput vs. Normalized Load (Scenario 1) CF/DAMA-PB-BSNRF+DD with channel.

The CF/DAMA-PB-BSNRF+DD performed slightly better than the CF/DAMA-PB-BSNRF protocol in terms of the percentage of dropped packets, shown in Figure 145. The reason was the less stringent deadlines from the CF/DAMA-PB-DD protocol, but yet this did not give a decrease in dropped packets like that of the CF/DAMA-PB-DD protocol which outperforms the other two protocols below loads of 1.0. The CF/DAMA-PB-BSNRF+DD and CF/DAMA-PB-BSNRF protocols are almost identical with respect to packet dropping.

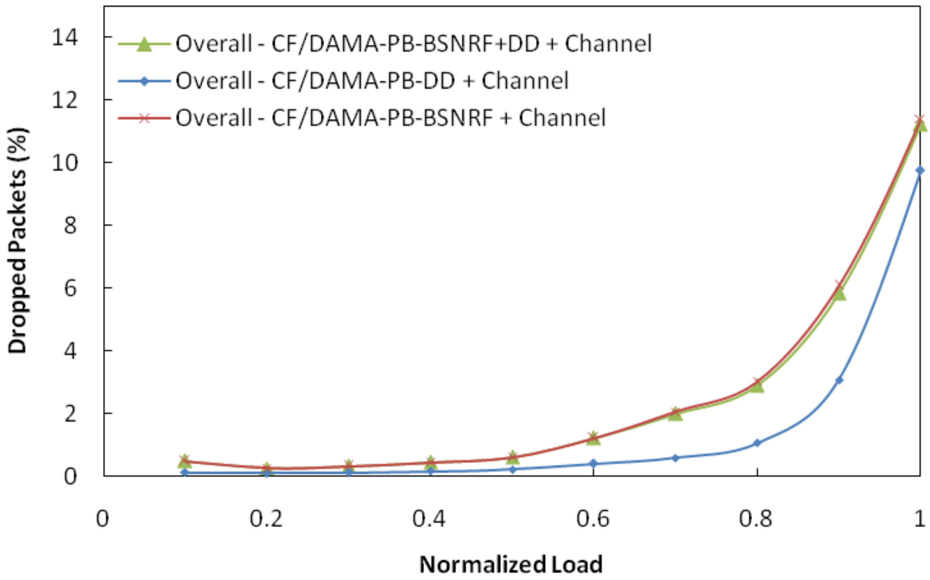

Figure 145: Percentage of dropped packets (Scenario 1) CF/DAMA-PB-BSNRF+DD with channel.

Scenario 2

In scenario 2 we saw a similar result to that in scenario 1. The CF/DAMA-PB-BSNRF and CF/DAMA-PB-BSNRF+DD protocols performed best when compared to CF/DAMA-PB-DD in terms of average packet delay.

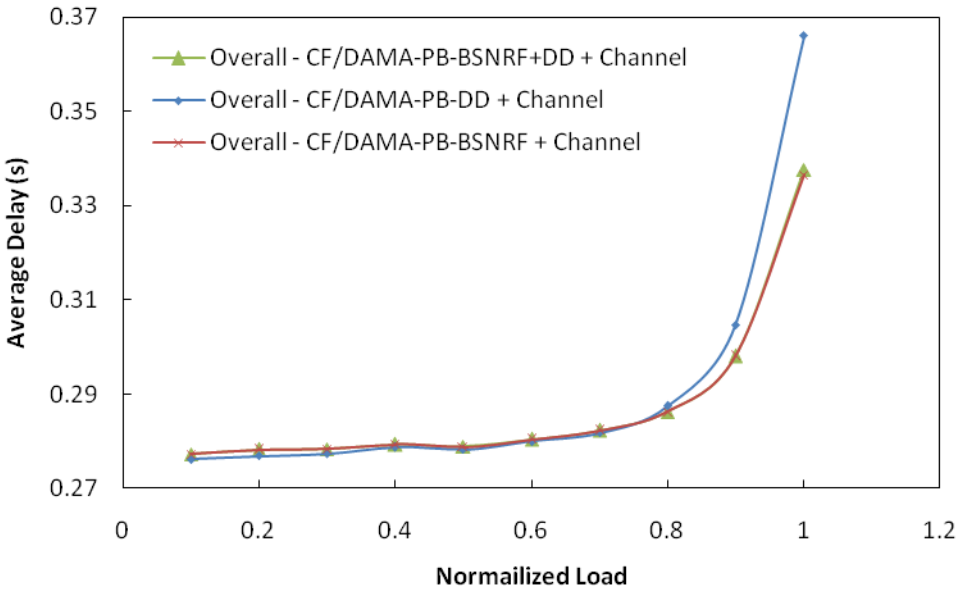

Figure 146: Average Delay vs. Normalized Load (Scenario 2) CF/DAMA-PB-BSNRF+DD with channel.

Again, the traffic scenarios, 1 and 2 had similar throughputs. The CF/DAMA-PB-DD throughput performed better than the other two below loads of 1.0 while the other two performed better at greater loads.

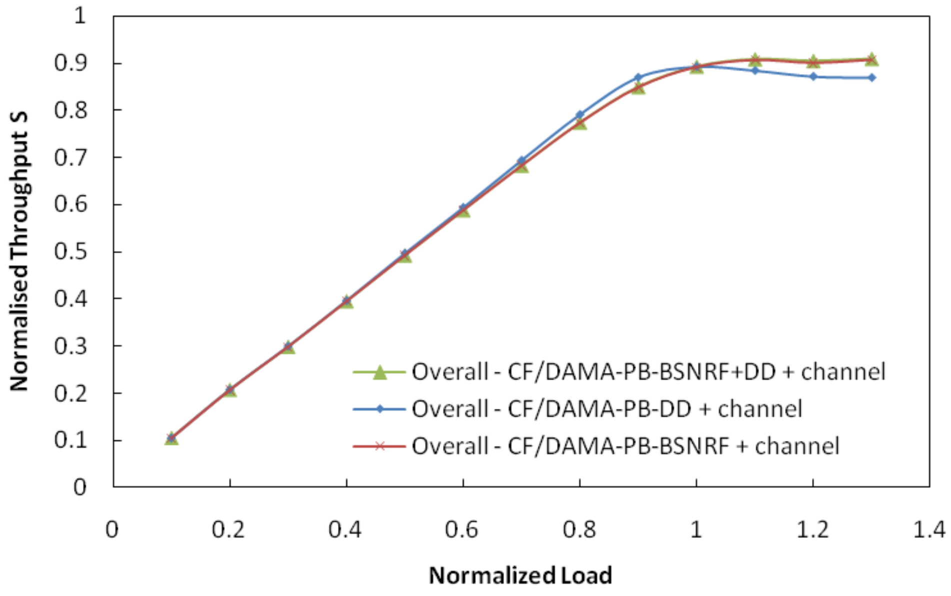

Figure 147: Normalized Throughput vs. Normalized Load (Scenario 2) CF/DAMA-PB-BSNRF+DD with channel.

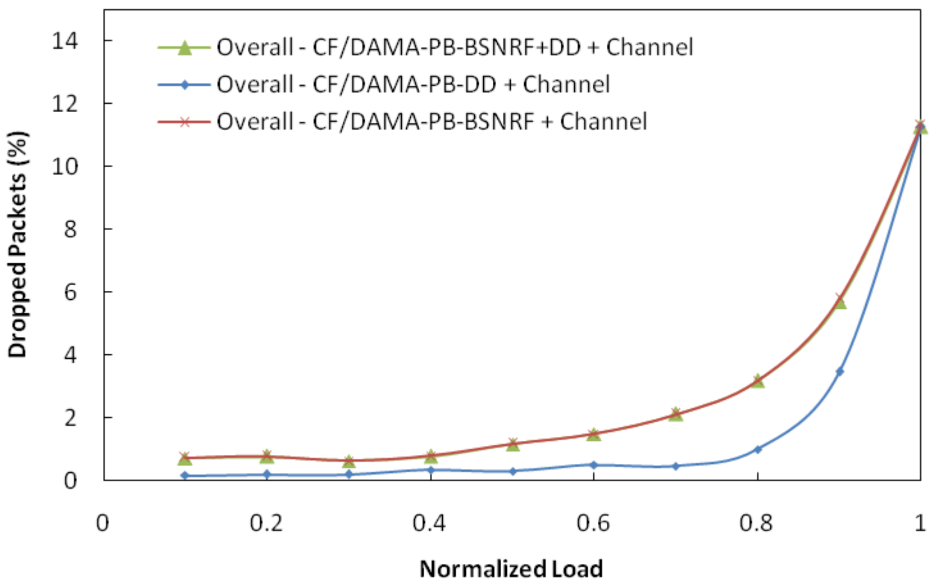

Figure 148: Percentage of dropped packets (Scenario 2) CF/DAMA-PB-BSNRF+DD with channel.

The CF/DAMA-PB-DD protocol outperformed the other two protocols in terms of percentage of dropped packets for loads less than 1.0. The performance in this respect ensured a reasonable QoS for CF/DAMA-PB-DD protocol but not for the other two protocols. This is illustrated in Figure 148.

4.9 Conclusions

This chapter introduced cross-layer design and the possible benefits that could be obtained when used appropriately, such as in the CF/DAMA-PB-DD protocol. The various issues that affect the channel performance and two methods that may be used to model the channel losses, i.e. the Good-Bad model and the 3 state model, were also discussed for simulation purposes.

The CF/DAMA-PB-PD protocol was seen to significantly improve the average packet delay of the system when compared to the CFDAMA-PB scheme. This also reduced the overall throughput of the system and increased the percentage of dropped packets. The percentage of packets dropped was kept within reasonable limits expected to maintain an acceptable QoS for loads less than 0.85. The packets lost through channel conditions were found to be relatively small compared to the packets dropped to deadlines and hence had almost no impact on the overall system performance.

The CF/DAMA-PB-BSNRF protocol took the CF/DAMA-PB-PD protocol a step further by adding priority by means of cross-layer interaction with the physical layer information. This proved to further decrease the overall packet dropping but at a greater cost of packet loss. The packet loss experienced reduced the acceptable QoS level to a maximum load of 0.85, but had a significant improvement in average packet delay for loads above this. The video and voice traffic class’s benefited the most from the delay improvements but were also affected the most by lost packets.

The CF/DAMA-PB-DD protocol improved slightly on the system throughput and percentage of dropped packets while giving up an insignificant amount in overall packet delay. This protocol is the most sensible to use if one wanted to have a acceptable QoS while still reducing the average packet delay experienced in the system.

Finally, the hybrid protocol that took both dynamic deadlines into account and the prioritised reordering of allocated slots was hoped to be promising but found to have failed to achieve the low percentage packet dropping that was offered by the CF/DAMA-PB-DD protocol. This protocol performed excellently in the average packet delay performance but to the detriment of QoS delivery.

5 Conclusion

Satellite communication plays an important role in digital communications by creating bridges for digital divides. Satellites ensure that global network connectivity is possible where previously impossible. In order to ensure that satellite communications is able to keep up with the demands of multimedia communications across the globe, the systems require continuous exploration and improvement. Medium access control protocols are one of the many aspects of satellite network design that can help in achieving better system performance.

This thesis has proposed five MAC protocols based on the CF/DAMA family of protocols, these being the CF/DAMA-PB-PD, the CF/DAMA-PB-PEDF, the CF/DAMA-PB-BSNRF, the CF/DAMA-PB-DD and the CF/DAMA-PB-BSNRF+DD protocols. CF/DAMA-PB-PD takes advantage of the different traffic classes and their ability to cope with packet loss by adding packet dropping to reduce the average packet delay significantly while still ensuring a reasonable QoS for each traffic class. The CF/DAMA-PB-PEDF protocol aimed at improving on the CF/DAMA-PB-PD protocol by trying to ensure deadlines are met where possible, and otherwise would not have been met, so as to reduce the percentage of dropped packets. This did not have any significant improvement as the stringent packet deadline requirements for voice and video traffic sources gave very little, if any, room for the P-EDF to have an effect.

The CF/DAMA-PB-BSNRF, CF/DAMA-PB-DD and CF/DAMA-PB-BSNRF+DD protocols took advantage of the cross-layer design approach. The interaction of the data-link layer and the physical layer proved to assist in the decision making of the protocols through the SNR information of each RCST. CF/DAMA-PB-BSNRF used the SNR information of the RCSTs requesting resource allocation by assigning the request with the highest SNR value slots first. This improved system throughput for high system loads and decreased the average packet delay of the system. Priority was given to the RCSTs that could ensure better system performance in terms of channel quality.

The CF/DAMA-PB-DD protocol compensates for bad channel noise by loosening up on the packet deadlines according to the RCST's instantaneous SNR values. This ensures that fewer packets are dropped as a result of packet deadlines in a bad channel environment to compensate for the packet loss due to the environment. By loosening the deadline, the RCSTs in bad environments also increased the overall packet delay experienced slightly while increasing system throughput too.

The hybrid protocol of the CF/DAMA-PB-BSNRF and CF/DAMA-PB-DD protocols, i.e. the CF/DAMA-PB-BSNRF+DD, incorporated both dynamic deadlines and BSNRF to try and gain from both protocols. As expected the BSNRF dominated the overall outcome of this protocol. By giving priority to the RCSTs with the better SNR values, the effect of the dynamic deadlines was reduced, as it favours the RCSTs with lower SNR values. Dynamic deadlines are effective in ensuring the QoS of a source is maintained while under varying conditions while the BSNRF tries to maximise the usage of the good channels over the bad channels and thus achieves better system throughput.

Both CF/DAMA-PB-BSNRF and CF/DAMA-PB-DD are effective at increasing throughput at high loads while CF/DAMA-PB-BSNRF is effective at reducing average packet delay and CF/DAMA-PB-DD forfeits some delay in order to increase system throughput. CF/DAMA-PB-PD is effective at reducing the average delay of the system while forfeiting the system throughput through packet dropping.

5.1 Future Work

The following are potential research projects that could build on the work presented in this thesis:

- The proposed MAC protocols could be analytically defined and evaluated.
- The fairness of the proposed algorithms could be determined.
- A Call Admission Control algorithm could be added and investigated in terms of the possible performance benefits to the system.
- The ratios of voice, video and web traffic could be simultaneously varied in real-time and hence the performance of the proposed MAC protocols determined under these conditions.

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