PACKET SCHEDULING IN SATELLITE HSDPA NETWORKS

By

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DECLARATION

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ABSTRACT

The continuous growth in wireless networks is not showing any sign of slowing down as new services, new technologies and new mobile users continue to emerge. Satellite networks are expected to complement the terrestrial network and be a valid option to provide broadband communications services to both fixed and mobile users in scenarios where terrestrial networks cannot be used due to technical and economical viability. In the current emerging satellite networks, where different users with varying traffic demands ranging from multimedia, voice to data and with limited capacity, Radio Resource Management (RRM) is considered as one of the most significant and challenging aspect needed to provide acceptable quality of service that will meet the requirements of the different mobile users. This dissertation considers Packet Scheduling in the Satellite High Speed Downlink Packet Access (S-HSDPA) network.

The main focus of this dissertation is to propose a new cross-layer designed packet scheduling scheme, which is one of the functions of RRM, called Queue Aware Channel Based (QACB) Scheduler. The proposed scheduler, which, attempts to sustain the quality of service requirements of different traffic requests, improves the system performance compared to the existing schedulers. The performance analysis comparison of the throughput, delay and fairness is determined through simulations. These metrics have been chosen they are three major performance indices used in wireless communications.

Due to long propagation delay in HSDPA via GEO satellite, there is misalignment between the instantaneous channel condition of the mobile user and the one reported to the base station (Node B) in S-HSDPA. This affects effectiveness of the channel based packet schedulers and leads to either under utilization of resource or loss of packets. Hence, this dissertation investigates the effect of the introduction of a Signal-to-Noise (SNR) Margin which is used to mitigate the effect of the long propagation delay on performance of S-HSDPA, and the appropriate SNR margin to be used to achieve the best performance is determined. This is determined using both a semi-analytical and a simulation approach. The results show that the SNR margin of 1.5 dB produces the best performance.

Finally, the dissertation investigates the effect of the different Radio Link Control (RLC) Transmission modes which are Acknowledged Mode (AM) and Unacknowledged Mode (UM) as it affects different traffic types and schedulers in S-HSDPA. Proportional fair (PF) scheduler and our proposed, QACB, scheduler have been considered as the schedulers for this investigation. The results show that traffic types are sensitive to the transmitting RLC modes and that the QACB scheduler provides better performance compared to PF scheduler in the two RLC modes considered.

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LIST OF ACRONYMS

2G	Second Generation Communications Systems
3G	Third Generation Communications Systems
3GPP	Third Generation Partnership Project
4G	Fourth Generation Communications Systems
AM	Acknowledged Mode
AMC	Adaptive Modulation and Coding
AMPS	Advanced Mobile Phone Systems
AP	Access Point
ARQ	Automatic Repeat Request
ATM	Asynchronous Transfer Mode
AVBDC	Absolute Volume Based Dynamic Capacity
BER	Bit Error Rate
BGAN	Broadband Global Area Network
BLER	Block Error Rate
BR	Booked Rate
BS	Base Station
CAC	Call Admission Control
CBQ	Class Based Queuing
CCM	Constant Coding and Modulation
CDMA	Code Division Multiple Access
CDMA 2000	Code Division Multiple Access 2000

CIF-Q	Channel Condition Independent Fair Queuing
CQI	Channel Quality Indicator
CSDPS	Channel State Dependent Packet Scheduling
CSI	Channel State Information
CT	Cordless Telephone
DAMA	Demand Assignment Multiple Access
DAMPS	Digital Advanced Mobile Phone System
DDBHP	Dynamic Doppler Based Handover Prioritization
DECT	Digital Enhanced Cordless Telecommunications
DGW	Destination Gateway
DRR	Deficit Round Robin
DS-CDMA	Direct Sequence Code Division Multiple Access
DTH	Direct-To-Home
DVB	Digital Video Broadcasting
DVB-S	Digital Video Broadcasting via Satellite
DVB-RCS	Digital Video Broadcasting Return Channel via Satellite
EDD	Earliest Due Date
EDF	Earliest Deadline First
EDGE	Enhanced Data GSM Environment
EURANE	Enhanced UMTS Radio Access Network Extensions
FCA	Fixed Capacity Assignment
FCFS	First Come First Serve

FDD	Frequency Division Duplex
FDMA	Frequency Division Multiple Access
FIFO	First In First Out
FM	Frequency Modulation
FSL	Free Space Loss
FTP	File Transfer Protocol
GEO	Geosynchronous Orbit
GGSN	Gateway GPRS Support Node
GPRS	General Packet Radio Service
GPS	Generalized Processor Sharing
GSM	Global Systems for Mobile Communications
HARQ	Hybrid Automatic Repeat Request
HDTV	High Definition Television
HEO	Highly Elliptical Orbit
HOL	Head of Line
HRR	Hierarchical Round Robin
HSDPA	High Speed Downlink Packet Access
HSPA	High Speed Packet Access
HSUPA	High Speed Uplink Packet Access
IEEE	Institute of Electrical and Electronics Engineers
IMR	Intermediate Module Repeater
IMT-2000	International Mobile Telecommunications 2000

IP	Internet Protocol
ISM	Industrial Scientific Medical
ITU	International Telecommunication Union
IWFQ	Idealized Wireless Fair Queuing
JFI	Jain Fairness Index
LAN	Local Area Network
LEO	Low Earth Orbit
LMS	Land Mobile Satellite
LMSS	Land Mobile Satellite Systems
LOS	Line of Sight
LPDC	Low Parity Density Check
LSM	Link State Monitor
LTE	Long Term Evolution
LTFS	Long Term Fairness Server
MAC	Medium Access Control
MAP	Mobile Application Part
MCS	Modulation and Coding Scheme
MEO	Medium Earth Orbit
MF-TDMA	Multiple Frequency Time Division Multiple Access
MIMO	Multiple Input Multiple Output
M-LWDF	Modified Largest Weighted Delay First
MMS	Multimedia Messaging Service

MPEG	Moving Picture Experts Group
MSC	Mobile Switching Centre
NCC	Network Control Centre
NMT	Nordic Mobile Telephone
NS-2	Network Simulator 2
OFDM	Orthogonal Frequency Division Multiplexing
OSI	Open System Interconnection
PDC	Personal Digital Cellular
PDF	Probability Density Function
P-EDF	Prioritized Earliest Deadline First
PDU	Protocol Data Unit
PEP	Performance Enhancement Proxies
PER	Packet Error Rate
PF	Proportional Fair
PHS	Personal Handyphone System
PSTN	Public Switching Telephone Networks
QACB	Queue Aware Channel Based
QoS	Quality of Service
RCQI	Relative Channel Quality Index
RCST	Return Channel Satellite Terminal
RLC	Radio Link Control
RNC	Radio Network Controller

RR	Round Robin
RRC	Radio Resource Control
RRM	Radio Resource Management
RRU	Radio Resource Unit
RTT	Round Trip Time
SBFA	Service Based Fairness Approach
SDMB	Satellite Digital Multimedia Broadcasting
SDU	Service Data Unit
SGSN	Serving GPRS Support Node
S-HSDPA	Satellite High Speed Downlink Packet Access
SIM	Subscriber Identity Module
SIR	Signal to Interference Ratio
S-MBMS	Satellite Multicast and Broadcast Multimedia Systems
SMS	Short Messaging Service
SNR	Signal to Noise Ratio
S-PCS	Satellite Personal Communications Services
SR	Static Rate
STFQ	Start Time Fair Queuing
S-UMTS	Satellite Universal Mobile Telecommunications Systems
TACS	Total Access Communications Systems
TBS	Transport Block Size
TCP	Transmission Control Protocol

TCRA	Time Based Channel Reservation Algorithm
TDD	Time Division Duplex
TDMA	Time Division Multiple Access
TES	Transform Expand Sample
TFRC	Transport Format and Resource Combination
TG	Traffic Gateway
TM	Transparent Mode
TTI	Transmission Time Interval
T-UMTS	Terrestrial Universal Mobile Telecommunications Systems
UE	User Equipment
UM	Unacknowledged Mode
UMTS	Universal Mobile Telecommunications Systems
U-NII	Unlicensed National Information Infrastructure
UTRAN	UMTS Terrestrial Radio Access Network
VBR	Variable Bit Rate
VCM	Variable Coding and Modulation
VOIP	Voice over Internet Protocol
VSF	Variable Spreading Factor
WAP	Wireless Application Protocol
WCDMA	Wideband Code Division Multiple Access
WFS	Wireless Fair Service
WFQ	Weighted Fair Queuing

WIMAX	Worldwide Interoperability of Microwave Access
WLAN	Wireless Local Area Network
WPAN	Wireless Personal Area Network
WRR	Weighted Round Robin

LIST OF SYMBOLS

a_k	QoS differentiation factor for User *k*
α	Mean of direct ray attenuation
δ_k	Probability of exceeding the delay deadline for User *k*
d_k	Waiting time of Head of Line packet in user queue *k*
D_k	Previous instantaneous data rate of user *k* before the present TTI
G_R	Gain of receiving antenna
G_T	Gain of transmitting antenna
h	Signal-to-Noise Ratio margin
L_{all}	Total loss
L_k	Instantaneous queue length of user queue *k*
L_s	Free Space Loss
L_{queue}	Maximum length of queue
M_i	Number of state frames corresponding to state *i*
$M_{i,j}$	Number of transitions from state *i* to *j*
MP	Average multipath power
n	Number of Transmission Time Intervals (TTIs)
N	Single Sided Noise Power Spectral Density
N_f	Total number of samples for an environment
N_k	Number of Samples corresponding to state *k*
$P_{i,j}$	Transition probabilities matrix from state *i* to *j*

P_r	Received power
$P(r)$	Loo's Probability density function
Ψ	Standard deviation of direct ray attenuation
R_k	Maximum allowable data rate by user *k*
S	Throughput
S_{av}	Average throughput
S_i	Instantaneous throughput
σ	Proportional factor for queue length
T_k	Average data rate of user *k* up to the present TTI
$T_{k,deadline}$	Delay deadline for the Head of Line packet in user queue *k*
W_i	State probabilities matrix for state *i*

Chapter 1

INTRODUCTION

1.1 Overview of Wireless Communications

The unrelenting growth in wireless communications has captured the imagination of the public. The world has witnessed rapid growth in wireless communications in the last decade as seen by the wide expansion of mobile systems across the globe. The recent advancements in internet technology ranging from e-commerce, social networking, video-on-demand and video conferencing, has significantly increased the internet traffic which has led to an exceptional growth in mobile internet. Also, at present, the Wireless Local Area Network (WLAN) is either complementing or replacing the existing wired network in various homes and businesses. The new emerging applications like wireless sensor networks, remote telemedicine and smart home appliances are also contributing towards the progress of wireless communications [1]. However, the continuous expansion of wireless systems, and emerging applications have also led to constant technical challenges that needed to be addressed to have a successful converged wireless network. In this chapter, the evolution of wireless communications is presented. Also, a more detailed look at satellite systems is presented and then the chapter is concluded with the motivations for this research, the research overview, original contributions in this research and publications.

1.2 Evolution of Wireless Communications

The success of wireless communications was not sudden but rather based on an evolutionary development. The evolution of mobile communications can be categorized into generations [2].

1.2.1 First Generation Wireless Communications

The first generation systems, as commonly called, were developed based on analog frequency modulation technology (FM) for its radio transmissions and designed for voice communications services. These systems which were deployed in the 1980s were incompatible with one another and this made roaming across countries unachievable [3]. Examples of these systems are Advanced Mobile Phone Systems (AMPS), Nordic Mobile Telephone (NMT), Total Access Communications Systems (TACS), Radiocom 2000, C-450 and C-Nets.

1.2.2 Second Generation Wireless Communications

The second generation (2G) cellular systems which were based on digital technology replaced the first generation system due to the fact that it could not meet up with the rapid growth in demand. The advancements in second generation digital systems that are responsible for higher bandwidth compared to first generation systems are low rate speech encoding and complex signal processing due to reduction in size of the mobile terminal equipment [4]. Also, one frequency can be divided among several users simultaneously either by code or by time. The Second generation system supports roaming which allow mobile phones to be used across national borders. Also, apart from digital voice communications, low data rates services like voice mail, short message service (SMS), mobile fax were added [2].

Although, many second generation systems such as DS-CDMA (IS-95), Digital Advanced Mobile Phone Systems (DAMPS) which are IS-54 and IS-136, Global Systems for Mobile Communications (GSM) and Personal Digital Cellular (PDC) exist, GSM leads others in terms of subscriber base. More services like Multimedia Messaging Service (MMS), group calls and Wireless Access Protocol (WAP) based internet services were added to upgrade the GSM networks, and the data rate were improved to 115kbps and 384kbps through the introduction of packet data technologies like General Packet Radio Service (GPRS) and Enhanced Data GSM Environment (EDGE) respectively [5].

1.2.3 Third Generation Wireless Communications

The third generation (3G) systems were designed to cater for more advanced services with higher data rate like multimedia services and video calls and also support seamless roaming in order for mobile users to have access to service anytime, anywhere. The higher data rate was achieved by improving the efficiency of the spectrum. The third generation systems known as International Mobile Telecommunications 2000 (IMT-2000) within the auspices of International Telecommunication Union (ITU) [6], is made up of different standards which includes Enhanced Data Rates for GSM Evolution (EDGE), Code Division Multiple Access 2000 (CDMA2000), Universal Mobile Telecommunication Services (UMTS) and Digital Enhanced Cordless Telecommunications (DECT). The IMT-2000 were allocated frequencies around 2 GHz and was capable of providing data rate from 384 kbps for pedestrian use to 2Mbps for indoor use [1]. The

UMTS is a general term for third generation radio technologies within the 3rd Generation Partnership Project (3GPP) that comprises of standardization bodies from Europe, China, Korea, USA and Japan. It comprises of the radio access network, the core network and user authentication via Subscriber Identity Module (SIM). The UMTS uses the Wideband Code Division Multiple Access (W-CDMA) which is the most widely used radio technology for 3G networks as its air interface and it supports both Frequency Divison Duplex(FDD) and Time Divison Duplex (TDD) variants [7][8]. The WCDMA which was the most widely adopted was renamed as UMTS Terrestrial Radio Access Network (UTRAN) and was attractive to existing GSM operators since the core network it uses, is based on the GSM Mobile Application Part (MAP) network [9].

High Speed Packet Access (HSPA) is a collection of two protocols namely High Speed Downlink Packet Access (HSDPA) and High Speed Uplink Packet Access (HSUPA). It is an upgrade of 3G systems that improves on the performance of WCDMA protocols. HSPA improves on the data rate through the introduction of a new shared channel called High Speed Downlink Shared Channel (HS-DSCH), Adaptive Modulation Coding (AMC), a fast packet scheduling through the reduction of Transmission Time Interval (TTI) reduction from 10ms to 2ms, an extensive multi-code operation and a fast and spectral efficient retransmission strategy called Fast Physical Layer Hybrid ARQ (F-L1 HARQ) [10] [11].The HSDPA and HSUPA are expected to have data rates of up to 10Mbps and 3-4Mbps respectively. The HSPA can be achieved by upgrading the existing software and can share the radio and core network elements in 3G networks including Base stations (Node B), Radio Network Controller (RNC), Serving GPRS Support Node (SGSN) and Gateway GPRS Support Node (GGSN) [12].

As indicated in Table 1.1, the HS-DSCH in HSDPA networks does not make use of a variable spreading factor as it is in DSCH in UMTS networks. It uses sixteen (16) orthogonal spreading codes with one code for signaling purposes. It also does not implement power control as indicated in Table 1.1 below. While the reduction of the TTI to 2 ms and positioning of the MAC protocol to Node B which is closer to User Equipment (UE) has contributed to faster packet scheduling process, hence, the higher data rates experienced in HSDPA networks as compared to UMTS networks. The AMC has also immensely contributed to high utilization of resources, since packets are transmitted in accordance to channel conditions.

As stated above, one of the objectives of 3G is for users to have access to service anytime, anywhere, therefore, satellite systems will form an essential part of 3G networks since terrestrial networks can only cover limited area due to economic and technical reasons. Satellite systems will provide coverage to mobile users in remote areas, in the air and on the sea. Continuous research work is being undertaken to integrate the terrestrial and satellite systems together [3].

Table 1.1 Comparison of fundamental properties of DSCH and HS-DSCH [10]

Feature	DSCH	HS-DSCH
Variable Spreading Factor (VSF)	Yes (4-256)	No (16)
Fast power control	Yes	No
Adaptive Modulation and Coding (AMC)	No	Yes
Fast L1 HARQ	No	Yes
Multi-code operation	Yes	Yes, extended
Transmission Time Interval (TTI)	10 or 20 ms	2 ms
Location of MAC	RNC	Node B

1.2.4 Fourth Generation Wireless Communications

The fourth generation (4G) system is the next level of advancement in the field of wireless communications. Due to the continuous emergence of high data rate multimedia services and applications like High Definition TV (HDTV), mobile TV, video chat, Multimedia Messaging service (MMS) and limited support for heterogeneous networks by 3G technology like Wireless Local Area Network (WLAN) and Wireless Personal Area Network (WPAN), 4G systems are being developed to address all these challenges. The 4G systems are expected to attain 100 Mbps in mobile situations and up to 1 Gbps in stationary positions. They are also expected to support global roaming and provide different Quality of service (QoS) requirements for different class of services across heterogeneous wireless networks including cellular networks, satellite networks, and WLAN networks. An all-Internet Protocol (IP) wireless is the most acceptable platform for 4G systems, therefore, both the access and the core networks are expected to be packet switched networks [13] [14]. The 3GPP Long Term Evolution (LTE) is the most widely adopted technology for 4G systems. It is based on Orthogonal Frequency Division Multiplexing (OFDM) because it delivers higher data rate due to the fact that it is more spectral efficient [15]. Worldwide Interoperability of Microwave Access (WIMAX) is also one of the candidates of 4G technologies.

1.3 Satellite Networks

Satellite Communications system is one of the major components of wireless communications. It makes use of the electronic device in the orbit called satellite to transmit information from one point (earth station or mobile station) to the other via the space [16]. It covers very wide area, therefore, used for international and global communications. The satellites systems provide support for various applications like video, voice and data for fixed, mobile and broadcast users [17]. Although, the most interesting use of satellite systems are for audio and video broadcasting over big geographical areas [1], they are also very relevant in the provision of mobile communication services.

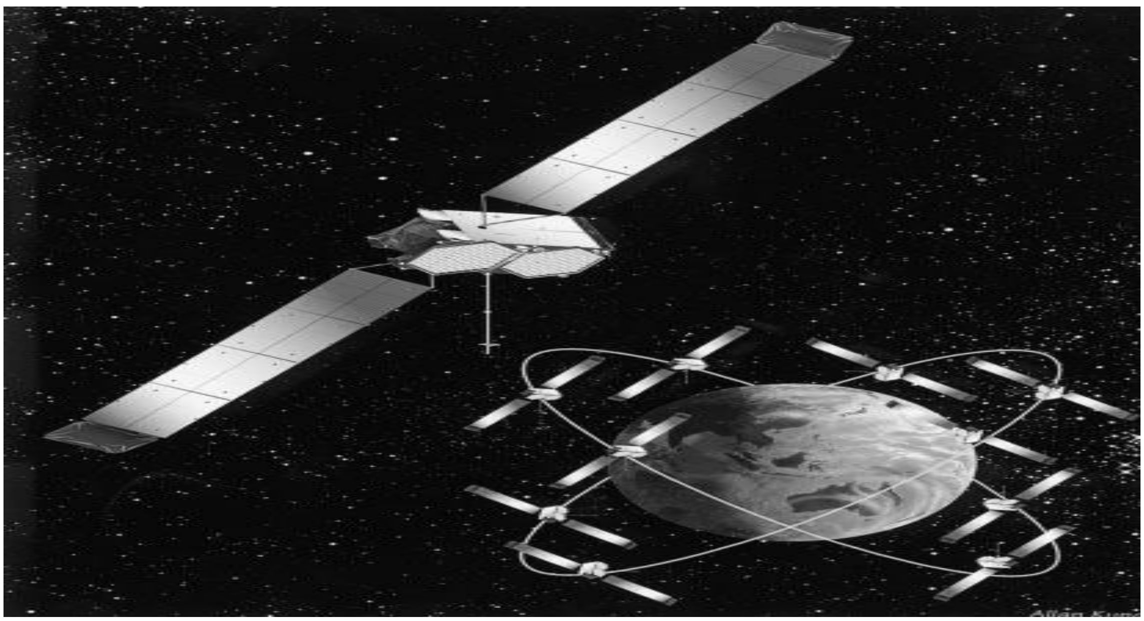

Figure 1.1 A space satellite and constellation of ten satellites [2]

There are four common types of satellite communications which are categorized based on their satellite orbits [17]. They are as follows;

1.3.1 Low Earth Orbit (LEO) Satellite Systems

Low Earth Orbit (LEO) satellite systems are very close to the surface of the Earth, up to 1,500 kilometers in altitude. Due to the relative short distance, they transmit signals with smaller delay of approximately 10 ms and can provide communications services. The LEO satellite is not stationary, therefore, the ground station must continuously track it. To provide a global coverage using LEO satellites, multiple constellation of inter-linked LEO satellites are needed. The major flaw of LEO satellite is its limited period of operation due to its constant movement in the sky [17].

1.3.2 Medium Earth Orbit (MEO) Satellite Systems

Medium Earth Orbit (MEO) satellite systems operate at about 10,000 to 20, 000 kilometers above the Earth, making it lower than GEO orbits but higher than LEO orbits. MEO satellites can be used to provide meteorological, remote sensing, navigation and global positioning services [17]. They also have larger capacity than LEOs.

1.3.3 Geostationary Orbit (GEO) Satellite Systems

Geostationary orbit (GEO) satellite systems are the most common satellites used for communication purposes and they operate at about 36,000 to 40,000 km above the earth. Each GEO satellite is stationary over a spot in the equatorial plane and as a result does not need any tracking antennas on the earth. GEO satellites are best used for multimedia and high-speed data transmissions. It has a wider coverage area and two to three are needed for global coverage. The major flaw of GEO satellites is its long propagation delay which is within the range of 260 to 280 ms [17]. The use of multi-spot beam coverage in GEO satellite orbit makes it possible for it to support mobile communications [2].

1.3.4 Highly elliptical orbit (HEO) satellite systems

Highly-elliptical orbit (HEO) satellite systems operate at an elliptical orbit and they are the only non-circular orbit satellites of the four types. They are 16000 to 35000 km above the earth and are used to provide better coverage to nations with higher southern or northern latitudes [16]. It has a wider view of the Earth and maximizes the amount of time each satellite spends in viewing populated areas. HEO satellite systems can be used for coverage in high latitude places where GEO satellites cannot cover, to provide communications services [17].

1.4 Mobile Satellite Communications Systems

Satellite communication systems have become a vital component of mobile communications. Mobile satellite systems have been in use for navigational purposes, systems tracking, safety and emergency communications purposes [18]. It is worthy to note that recent developments have now made it possible to use satellite systems to provide telecommunications services like voice calls and video streaming using portable mobile phones either as standalone satellite receivers or operating in dual modes for both cellular and satellite networks. This development was first achieved through the introduction of Satellite Personal Communication Services (S-PCS) using non-geostationary satellites like LEO and MEO satellite systems [18]. Recently, a mobile smart phone device is developed which has capabilities of connecting to 2G, 3G cellular networks and S-band Satellite networks [19].

The new developments in GEO satellite systems have made it more suitable for use to provide portable mobile communications facilities [2]. As depicted below in Fig. 1.2, the need to extend the telecommunication services like voice calls, video streaming or downloads, web browsing and file downloads provided by cellular networks to remote areas, low densely populated areas and also to users in the plane flying in the air or in the ship moving on the sea has made satellite communications a key element in the composition of next generation mobile networks.

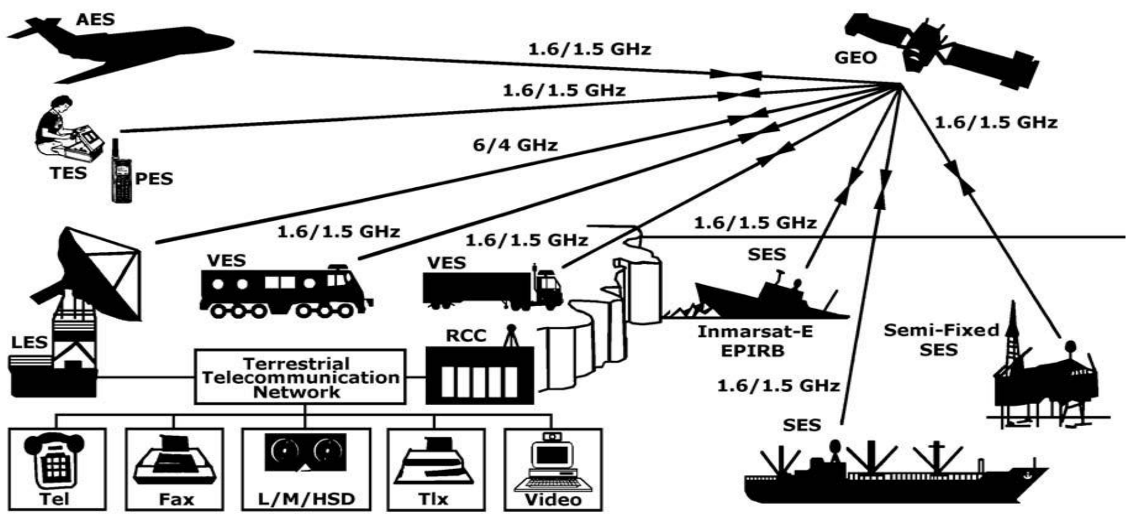

Figure 1.2 Mobile satellite service [18]

1.4.1 Satellite Universal Mobile Telecommunications System (S-UMTS)

Satellite Universal Mobile Telecommunications systems complement their terrestrial counterparts. Due to the global coverage that satellite systems possess, these systems will provide seamless global roaming and offer transparent handovers between the terrestrial and satellite networks. The UMTS is to use this to ensure mobile users have access to telecommunications services anytime, anywhere. The S-UMTS operates in two frequency bands which fall between 1.980 and 2.010 GHz and between 2.17 and 2.20 GHz. The three circular orbit satellite systems namely LEO, MEO and GEO satellite systems can be used to implement S-UMTS. The S-UMTS supports broadband applications and multimedia services inclusive [20]. The Satellite Universal

Mobile Telecommunications Systems (S-UMTS) Family G specification set aims at ensuring that the satellites interface that will be fully compatible with Terrestrial UMTS (T-UMTS)-based systems, even if some modifications will be made due to the differences between the terrestrial and the satellite channels [21]. As shown in Fig. 1.4, satellite cell type forms part of five basic cell types, as agreed by the IMT-2000, to provide a complete coverage in order to support all possible UMTS transmission environments. One of the existing satellite networks, the Broadband Global Access Network (BGAN) provided by INMARSAT can also provide UMTS services.

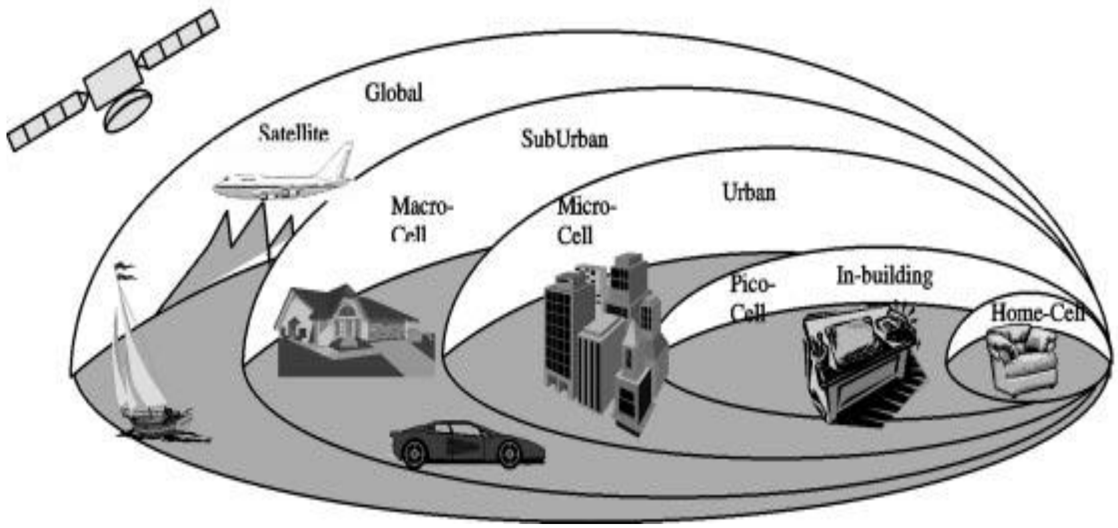

Figure 1.3 UMTS cell types [2]

The Satellite High Speed Downlink Packet Access (S-HSDPA) is one of the appealing S-UMTS services for point-to-point connections due to the inherent high data rate capacity of satellite. This can be used to provide high data rate services to mobile users. The improved delivery of higher data rate is achieved through the means of Adaptive Modulation Coding (AMC), a faster packet scheduling scheme at Transmission Time Interval (TTI) of 2ms, an extensive multi-code operation and a fast and spectral efficient retransmission strategy known as Fast Physical Layer Hybrid ARQ (F-L1 HARQ). Also, S-UMTS is used for point-to-multipoint connections for provision of multicast and broadcast multimedia services.

1.4.2 MBMS via S-UMTS

The 3G systems are expected to provide Multicast and Broadcast Multimedia services (MBMS) to mobile users and satellite systems will be able to provide this service conveniently due to its wider coverage and larger system capacity compare to terrestrial networks. The satellite system that provides MBMS is commonly known as Satellite MBMS (S-MBMS). The S-MBMS architecture as proposed by Satellite UMTS IP-based Packet Network (SATIN) is designed to also make use of terrestrial Intermediate Module Repeater (IMR) to provide access to indoor and urban areas where signal obstruction might be experienced. So, mobile users can either receive MBMS signals via satellite direct link or through IMR or both [22].

1.4.3 Satellite Digital Multimedia Broadcasting (SDMB) Systems

The multimedia service is seen as a key factor to drive the growth of 3G systems. The Satellite Digital Mobile Broadcasting (SDMB) system makes use of the natural feature of satellite systems to provide broadcasting services over global coverage. Therefore, SDMB system is able to provide inexpensive live streaming and video download services to mobile users. It is compatible with MBMS and can operate on both 2G and 3G networks. The SDMB systems operate at the S-UMTS frequency band and it also makes use of terrestrial IMRs. Mobile user located anywhere will have access to a download speed of 3.46 Mbps [23].

1.4.4 Digital Video Broadcasting (DVB)

The Digital Video Broadcasting (DVB) is a technology used in providing digital TV broadcasting services via cable (DVB-C), terrestrial (DVB-T) and satellite (DVB-S) transmitters. The DVB-S which is aimed at providing Direct-To-Home (DTH) services is appropriate for use on various satellite transponders bandwidth. The DVB-S was flexibly designed to allow the transmission capacity to provide different TV services configuration which includes data and voice services. The DVB-S is made to be compatible with Moving Pictures Expert Group 2 (MPEG-2) [24].

In order to provide a satellite return link for DVB-S, a Digital Video Broadcasting Return Channel Satellite (DVB-RCS) was developed. This is done by providing an interactive channel for GEO satellite networks with a fixed Return Channel Satellite Terminal (RCST) [21]. The return channel in DVB-RCS which is used for request by the user uses Multiple Frequency Time Division Multiple Access (MF-TDMA), while the DVB-S standard which uses Time Division Multiplexing is adopted as the forward channel. The DVB-RCS is composed of a certain number of RCSTs, GEO bent-pipe satellite, Network Control Centre (NCC), Traffic Gateway (TG) and the Feeder. The allocation methods used in DVB-RCS are Continuous Rate Assignment (CRA), Rate Based Dynamic Capacity (RBDC), Volume Based Dynamic Capacity (VBDC), Absolute Volume Based Dynamic Capacity (AVBDC) and Fixed Capacity Assignment (FCA). The extension of DVB-RCS to mobile users makes use of the same functional blocks except for the fact that the NCC, the mobile RCSTs and the gateway, which now have further features like spreading and blocking channel countermeasures, are specific to mobile environment. Also, it is worthy to note that some specific signaling have been introduced between the mobile RCSTs, NCC and gateway in order to manage handovers [25].

The second generation standard for an advanced satellite transmissions named DVB-S2 was recently developed. The DVB-S2 which has 30% greater spectral efficiency as compared to DVB-S provides more application services like broadcast services, interactive services and data content distribution. The advancement in DVB-S2 is a result of adoption of recent channel coding schemes like Low Density Parity Check (LPDC) and various modulation types like QPSK, 8PSK, 16APSK and 32APSK [26]. Depending on the application type, the DVB-S2 can operate in Constant Coding and Modulation (CCM), Variable Coding and Modulation (VCM) and Adaptive Coding and Modulation (ACM) [27]. The code rates when using VCM and ACM can be changed on a frame by frame basis. The DVB-S2 is used with the existing DVB-RCS standard to provide interactive services and it is compatible with the satellite decoders of DVB-S [21].

1.5 Issues with Satellite Systems

The Satellite systems are key solutions for the provision of global coverage and seamless global roaming with capabilities of providing broadband and multimedia services at an acceptable QoS. For this to be fulfilled there is need to address the technical challenges facing satellite systems.

1.5.1 Long Propagation Delay

The delay experienced along the link in satellite communications varies depending on the satellite orbit, the user‘s position on the earth and the satellite type. GEO satellite is situated at high altitude, so it has high propagation delay compared to LEO and MEO. This high delay causes many problems for delay sensitive applications like video and voice [21].

1.5.2 Atmospheric Effects

The atmospheric effects consist of the atmospheric gases, rain attenuation, scintillation, fog and clouds. The rain attenuation is the main effect among the atmospheric effects and can be neglected for frequency below 10 GHz. Scintillation affects communications at frequencies below 10 GHz and at an elevation angle of above 10°.The fog and clouds are significant at frequencies above 30 GHz [21].

1.5.3 Channel Losses

The atmospheric effects experienced in satellite communications make the Bit Error Rate (BER) very high. This makes the satellite link to experience rapid degradation which can lead to erroneous bits [21].

1.5.4 Satellite Lifetime

Due to the ageing process of the components used in building satellites, effect of radiations and other factors, the lifetime of satellites are very limited. The life span varies depending on the type of satellite. The GEO satellites have a lifetime of 10-15 years, MEO satellites functions well within 10-12 years while LEO satellites are useful for 5-8 years [21].

1.6 Motivation for Research

With the limited resources available coupled with the increasing demand of various applications, an efficient radio resource management mechanism is essential to ensure effective resource utilization. Radio Resource Management (RRM) is one of the most significant and challenging aspects in the provisioning of high resource utilization and an acceptable quality of service (QoS) in wireless communications networks [28].

The RRM functionality is a set of algorithms used to actualize an optimal utilization of the radio air interface and the hardware resources. These set of algorithms ensure the required quality of connection is attained, the network usage is optimized and low level of connection (call) blocking is actualized. The RRM functions are admission control, handover control, load control, packet scheduling and power control.

The design of an effective RRM function is very challenging in a dynamic wireless networks with varying traffic demands and channel conditions for both terrestrial and satellite systems. It becomes more challenging in satellite networks scenarios due to the different channel conditions and the long propagation delay experienced especially in Geostationary Orbit (GEO) satellite systems which have been used for this work. The multi spot-beam GEO satellite system has been considered for the Satellite High Speed Downlink Packet Access (S-HSDPA) network. The long propagation delay causes a misalignment between the reported Channel Quality Indicator (CQI) used by Node B in S-HSDPA systems for transmission and also for scheduling decisions, and the present CQI experienced by the mobile user.

This research focuses on packet scheduling schemes in S-HSDPA networks since packet scheduling is an important RRM function and also a key element in the design of S-HSDPA systems. Due to the long propagation delay experienced in GEO satellite system and its consequences to packet scheduling in S-HSDPA systems, the effect of the introduction of the Signal to Noise ratio (SNR) margin that is used to compensate for the long propagation delay is investigated and an optimal margin is recommended. Both a semi-analytical and a simulation approach are used to determine the best margin. The performances of various margins are compared.

The need for a packet scheduling scheme, which is proposed in this dissertation, that will provide a better trade-off between resource utilization, degree of fairness and an acceptable QoS, putting into consideration the available constraints is very vital in S-HSDPA and other wireless packet networks. This will ensure users‘ demands are met and better throughput is achieved. This has necessitated the proposal of a new packet scheduling scheme tagged Queue Aware Channel Based (QACB) in this dissertation. The scheduler is designed using a cross-layer based approach which uses both the channel state information (CSI) of the physical layer, application type (application layer) and the queuing factors (MAC sub-layer) while taking scheduling decisions. The comparison of several performance indexes of the proposed scheme with other various schedulers was achieved through the means of computer simulations.

The RLC mode used for transmission for each traffic type affects the performance of both terrestrial and satellite HSDPA networks. The performance evaluation has been conducted for terrestrial HSDPA networks in the literature. Since S-HSDPA will complement the terrestrial network, there is need for a performance evaluation of S-HSDPA networks as well. This has necessitated the investigation of the effects of the various Radio Link Control (RLC) transmission modes for various traffic types in S-HSDPA networks. Several performance aspects like throughput, delay and jitter are used for the comparison of the acknowledged mode (AM) and the unacknowledged mode (UM) RLC modes for the four Universal Mobile Telecommunications System (UMTS) traffic classes. Finally, the effect of using a non-delay sensitive and delay sensitive packet scheduler in the two RLC modes is investigated. An appropriate RLC mode is recommended for each traffic type in order to attain acceptable QoS and also utilize resources.

1.7 Dissertation Overview

This dissertation has been divided into six chapters. In chapter 1, an overview of the evolution of wireless communications networks is given and also an overview of various satellite systems is given. An overview of satellite mobile communications systems is presented, followed by the motivation for this research work. This is then followed by the dissertation overview, original contributions of this work and publications produced from this research work.

In chapter 2, an overview of radio resource management is presented. The discussion of the concept of cross-layer design as it affects RRM and some cross-layer interactions are then presented. This is followed by the discussion of RRM functions with examples including packet scheduling. Various schedulers in wired, terrestrial wireless and satellite communications are discussed.

Chapter 3 commences with a detailed look at channel models; both the common 2-state channel model and the 3-state channel model used in this work are given. The link budget analysis to determine SNR and computation to determine the CQI are given as well. The simulation set-up and results obtained from the investigation carried out on the effect of SNR margins on TCP performance in S-HSDPA are presented. The recommended optimal margin for suburban area is presented as well. Network Simulator 2 (NS-2) and Enhanced UMTS Radio Access Network Extension (EURANE) which is a UMTS module for NS-2 is used as the simulation software. A semi-analytical work and results are presented to confirm the simulation results. Discussion of results and drawn conclusions are stated.

In Chapter 4, the simulation set-up and results for the packet scheduling schemes are presented. Four different schedulers, including the proposed scheduler are considered for the simulation, and the throughput, delay and fairness performance are presented. The simulation results for a two state Markov channel model are also presented for validation. The simulation is also carried out using NS-2 and EURANE. Discussion of results and drawn conclusions are stated.

Finally, chapter 5 investigates the effect of RLC transmission modes for the four different UMTS traffic classes in S-HSDPA networks. The different performances like throughput, delay and jitter are measured for different RLC transmission modes, and the simulation set-up and results are presented. Discussion of results and drawn conclusions are stated as well.

In Chapter 6, the final conclusions of this dissertation are drawn and future directions are stated.

1.8 Original Contributions in this Research

The original contributions of this research work are stated as follows:

- Computation of an optimal SNR margin, used for delay compensation, in HSDPA via GEO satellite.
- Introduction of a new packet scheduling scheme to provide a good trade-off between resource utilization, degree of fairness and an acceptable QoS.
- Performance evaluation of RLC transmission modes for different UMTS traffic classes in HSDPA via GEO satellite.

1.9 Publications

Some of the works in this dissertation have been presented by the author as publications. The publications are as follows;

1. G. Aiyetoro and F. Takawira, ―Effect of SNR margin in TCP performance in HSDPA via GEO satellite," IEEE IWSSC 2009 Proceedings," September 2009.
2. G. Aiyetoro and F. Takawira, ―A Novel Packet Scheduling Scheme in HSPDA via GEO satellite," IEEE AFRICON 2009 Proceedings," September 2009.
3. G. Aiyetoro and F. Takawira, ―Effect of RLC transmission modes for differentiated traffic types in HSDPA via GEO satellite," SATNAC 2009 Proceedings, August – September 2009.
4. G. Aiyetoro and F. Takawira, ―A Novel Packet Scheduling Scheme in Satellite HSDPA networks," Draft journal paper

Chapter 2

RADIO RESOURCE MANAGEMENT

2.1 Introduction

The main aim of this chapter is to present a literature survey of the radio resource management functions with special focus on packet scheduling and resource allocation, which is the focus of this dissertation. An overview of each of the RRM functions and some of their algorithms are presented.

The chapter begins with the effect of cross-layer design in wireless communications and RRM functions in specific and a brief look at different cross-layer interactions. This is followed by definition of the RRM functions and some of their algorithms.

The rest of the chapter gives a detailed look at packet scheduling. The packet scheduling schemes are broadly classified into wired network and wireless network schedulers. The wireless network schedulers are further subdivided into terrestrial wireless network schedulers and satellite network schedulers with focus on Satellite High Speed Downlink Packet Access (S-HSDPA). The operations of various scheduling schemes under each category are explained as well. The scheduling schemes presented in this chapter by no means represent all the existing scheduling schemes. However, the ones provided give a good overview of the scheduling schemes.

2.2 Overview of Radio Resource Management

The mobile communication world is at present experiencing a continuous growth in size of mobile users and the emergence of different applications especially high data rate applications like multimedia. However, the Radio Resource Units (RRU); which are defined as the set of basic physical transmission parameters, needed to support a signal waveform that is used to convey end user information corresponding to a reference service [29], are limited. Examples of RRUs are radio spectrum, time slots and available codes. Despite the limited RRUs, the Quality of Service

(QoS) experienced by the mobile users and an effective utilization of resources cannot be compromised, therefore, an effective RRM is needed to be able to achieve this.

The RRM is regarded as one of the most important and demanding aspects in the provisioning of QoS that will be acceptable by the variety of mobile users and the effective utilization of resources that will satisfy the service providers for wireless network [30].

2.3 Cross-layer Design and RRM

The fast varying channel conditions experienced in a wireless network with different traffic types has made cross-layer design implementation a necessity. An example is the way TCP source reduces packet transmission rate when packet losses occur in a wireless channel, when the channel is in poor state. This is due to the fact that the TCP source only interprets this as congestion related issue. This therefore leads to low throughput [31]. Another example is that radio resources can be allocated to mobile users with bad channel state, since the resource allocations are done without physical layer information of the mobile user. These and many other reasons have made cross-layer design a vital concept. The need for cross-layer design implementation in satellite networks does not have to do with only different traffic demands and varying multimedia data rate but also the dynamic channel conditions and the long propagation delay experienced [32]. Cross-layer design can be referred to as a protocol design that is implemented to utilize the dependence through cross-layer interactions between the different layers in order to optimize the network [33]. Many cross-layer design approaches, which involve the architectural violation of the conventional independent layers, have been proposed in the literature. The two basic cross-layer design approaches are implicit cross-layer design which does not support information exchange during operations but only considers layer interactions at the design stage and explicit cross-layer design which allows interactions among non-adjacent layers [34]. The architectural violations as stated in [33] can be actualized by creation of new interfaces between adjacent and non-adjacent layers, merging of adjacent layers, design of coupling without interfaces and vertical calibration across layers. As stated in [35], there is a need for cautionary measures, so as not to have situations where an unanticipated consequence will be experienced on the overall performance due to interactions between conflicting cross-layer designs. RRM schemes are to provide acceptable users‘ satisfaction and effective resource utilization of the providers. In order to achieve this, the need for every Open System Interconnection (OSI) or TCP stack layer to co-operate is very essential [21]. These are some of the possible cross-layer interactions that can be utilized in satellite networks.

2.3.1 Physical and MAC Layer Interactions

There is need for resource allocation schemes to be aware of the characteristics of the physical layer (the channel conditions) before allocating resources. The usage of the Adaptive Modulation and Coding (AMC) scheme can be used to achieve good performance. This will allocate transmission rates of each user based on the channel conditions. Mobile users experiencing poor channel conditions will experience transmission at data rates of lower coding rate and lower order modulation while those with good channel conditions transmits at higher coding rate. Several authors in the literature have used this layer interaction to improve the system performance. A cross-layer concept of this type of interaction is proposed in [36] so as to improve the efficiency of the reliable multicast services provided via GEO satellite. A number of packet schedulers have been proposed in the literature using this concept as well, an example is proposed in [37] using a perfect prediction-based channel conditions.

2.3.2 Physical and Network Layer Interactions

In the implementation of an effective mobility management schemes at the network layer, there is need to put into consideration the mobility of the users in order to ensure a fast handover process especially for Non-GEO satellites systems where frequent handovers occurs. This will reduce the delay that is to be experienced by the Internet Protocol (IP) traffic so as to avoid large reduction in the TCP goodput [34].

2.3.3 MAC and Network Layer Interactions

Satellite IP based networks uses Integrated Services (IntServ) and Differentiated services (DiffServ) for the provisioning of QoS at the network layer as used in other IP based networks. The resource allocation and scheduling schemes at layer 2 need to allocate resources in a way that will be compatible to either of the two layer 3 QoS provisioning mechanisms. If the DiffServ mechanism is used at layer 3, there must be an appropriate QoS mapping at layer 2. This cross-layer interaction is utilized in the cross-layer RRM proposed in [38], where the scheduler assigns bandwidth in proportion to the request that is made. It uses the Dynamic Capacity Allocation (DCA) by computing each bandwidth request. The queues in the MAC layer are scheduled in such a way that the constraints in the IP-level queues are maintained [38].

2.3.4 MAC and Transport layer Interactions

In the IP world, the TCP is the prevailing transport layer protocol. It provides end-to-end reliable data transmission across the internet. Due to the long Round Trip Time (RTT) in satellite networks, the TCP congestion control scheme performs poorly. Therefore, cross-layer design approach is needed to design a TCP protocol that will improve the throughput performance. Examples are the Demand Assignment Multiple Access (DAMA) used in Digital Video Broadcast (DVB) systems and the adoption of Performance Enhancement Proxies (PEP) [39]. A cross-layer interaction between these two layers is also used to design a resource allocation scheme for DVB-RCS systems that was propose in [40]. The scheme ensures that the trend of request of resources tallies with the TCP transmission window.

2.3.5 MAC, Physical and Higher Layer Interactions

The emergence of real time traffic like multimedia amidst different traffic demanded by users not only in satellite networks but in wireless networks as a whole have necessitated the need for cross-layer interactions to extend to application layer. The different streams have different QoS requirements, therefore, the resource allocation scheme based at the MAC layer should use this information in allocating resources. Also, the AMC level that varies based on the channel status at the physical layer can be used by the application layer to dynamically adjust the traffic source generation data rate. This will prevent overflow of the queue and packet loss [34]. An example of this is proposed in [41], where the MAC layer protocol makes use of the Packet Error Rate (PER) of the physical layer and the type of service at the application layer to decide on which parameters to be used. It is worthy to note that the packet scheduler designed in this dissertation have employed the interaction between the physical, MAC and application layer.

2.4 Radio Resource Management Functions

Given that the aim of RRM is to ensure utilization of resources and provision of acceptable QoS to different mobile users, the RRM functions are used to achieve this set of objectives. The RRM functions which include admission control, handover control, power control, resource allocation and packet scheduling are in charge of allocating and managing the limited radio resources. They can be implemented in different algorithms and are not subjected to standardization [29]. Also, the RRM functions for every type of service or traffic might differ [42].

2.4.1 Connection Admission Control (CAC)

The connection admission control (CAC) schemes are used to ensure that the network is not at anytime overloaded and the accepted loads‘ QoS are not compromised. The CAC scheme grants or denies connections based on certain predefined criteria putting into considerations the network loads and QoS level as stated earlier [43]. The decision is commonly based on whether the new connection will not affect the QoS of the current streams and the network can provide the QoS level requested by the new connection [44]. However, in satellite networks and wireless networks as a whole, the CAC schemes are more complicated due to the peculiar nature of wireless communications especially in a multi-service network. The CAC schemes in satellite networks as stated in [43] are used:

- To guarantee a minimum transmission rate putting user mobility into consideration.
- To guarantee packet-level parameters by ensuring that the network is not overloaded.
- To control the handoff failure probability by ensuring some resources are reserved for handover connections.

The CAC is an important entity for load control and QoS provisioning. It is either centrally managed by the Network Control Centre (NCC) or in cases of non-GEO satellites, distributively managed on the board of the on-board processing satellites [21]. The CAC schemes can be categorized based on different premises like information, centralization, services, parameter or measurement [43].

2.4.1.1 Signal to Interference Ratio Based CAC

This CAC scheme makes decision using an admission criterion that will ensure the signal quality is not compromised. A certain predefined SIR value that is taken to be the threshold value is used to control the signal quality. The new connection is accepted if its SIR value is greater than the predefined threshold SIR value. There are many SIR based CAC that have been proposed in the literature. The SIR based CAC scheme proposed in [45], uses residual capacity for its admission criterion. The residual capacity is the remaining bandwidth. The admission criterion is defined as follows;

$$R_K = \left\lfloor \frac{1}{SIR_{th}} - \frac{1}{SIR_k} \right\rfloor \tag{2.1}$$

Where SIR_k is periodic SIR in cell *K*, SIR_{th} is the threshold SIR, $\lfloor \quad \rfloor$ is a floor function and R_K is the residual capacity. The residual capacity (R_K) which is determined periodically must be greater than zero before a new connection is admitted, else, the connection is blocked. Many other literature like [46] [47] have improved on this.

2.4.1.2 Bandwidth Based CAC

This admission control makes use of the effective bandwidth in making decisions whether to admit connections or reject it. These set of CAC schemes summarily check if there is bandwidth or resources available for the new connection before accepting it. This can be achieved using different approaches. In [48], a bandwidth based CAC is proposed for DVB-RCS systems which is an improvement to the one proposed for ATM satellites systems in [49]. The admission criterion is stated as follows;

$$B_{CRA}(k) + B_{RDBC}(k) \leq B_T \tag{2.2}$$

Where $B_{CRA}(k)$ is the total bandwidth allocated to continuous rate assignment (CRA) connections, $B_{RDBC}(k)$ is the total bandwidth allocated to all the Rate-Based Dynamic Capacity (RBDC) connections at *k-th* iteration of the scheme and B_T is the total bandwidth. The two bandwidths are updated periodically. The connection is only admitted if the total bandwidth of the CRA and RBDC connections is less than the total link bandwidth, else, it is rejected.

2.4.1.3 Cross-layer Design based CAC

A cross-layer design based CAC uses cross-layer interactions between different layers in deciding if a connection is to be admitted or rejected. This type of CAC considers not just availability of resources but distinguishes between various type of request based on their service type in order to provide QoS. A cross-layer design based CAC was proposed in [50] for GEO satellite networks which uses cross-layer interactions between physical, MAC and network layers to provide an end-to-end QoS. There are five different traffic classes with different Static Rate (SR) and Booked Rate (BR). Some traffic classes have BR values of zero while some are up to the SR (peak rate). The resources corresponding to the BR value are reserved for the traffic classes with BR values greater than zero. Once there is a service request from a user, the CAC checks the traffic class, it then search for which Destination Gateway (DGW) have the less cost from the routing tables. It then goes further to check if the chosen DGW has enough forward and return channel resources to cater for the service request based on the SR and BR of the traffic. The following admission criterion is used;

$$BR_{r_j} + SR_{r_j} + \sum_k BR_{r_k} + \sum_k SR_{r_k} \leq C_{r_{ab}} \tag{2.3}$$

$$BR_{f_j} + SR_{f_j} + \sum_k BR_{f_k} + \sum_k SR_{f_k} \leq C_{f_{ba}} \tag{2.4}$$

Where BR_{r_j} and BR_{f_j} are the allocated booked rate for the return and forward channel for connection j respectively, SR_{r_j} and SR_{f_j} are the statically allocated rate for return and forward channel for connection j respectively and $C_{r_{ab}}$ and $C_{f_{ba}}$ are the total provisioned return and forward channel capacity. The connection j will only be admitted if the summation of booked rate and static rate of connection j with the total book rate and static capacity already allocated to the ongoing connections is less than the provisioned channel capacity. This condition must be fulfilled for both forward and return channel else the connection is rejected.

2.4.2 Handover Control

Due to user mobility in wireless mobile networks, the handover control schemes are used to ensure that the user‘s call or service are not interrupted while the user moves from one cell or satellite to another. The handover process in terrestrial networks are driven by the movement of users from one location to another while in satellites networks it is often driven by the movement of satellite with respect to the earth surface. The types of satellites affected by this scenario are non-GEO satellites and their overlapping coverage areas are called the spot beams. So, due to this constant movement by the satellites, the mobile users must move from one spotbeam to another (intra-satellite handover) or from one satellite to another (inter-satellite handover) in order to retain connectivity. The CAC and the handover control are often inter-related, while CAC schemes reduces call blocking probabilities of new calls, the handover schemes minimizes call dropping probabilities for ongoing calls [21]. The handover control involves three stages; the initiation, the decision and the execution. Also, it has two modes of operation which are hard handover and soft handover. The hard handover ensures the resources used by the mobile user are released before new resources are allocated or used and are commonly referred to as break before make. While soft handover allows the mobile users to use both the current and new resources during the handover process and it‘s commonly referred to as make before break [51].

2.4.2.1 Intra-Satellite Handover Scheme

The continuous movement of non-GEO satellites and small spotbeam‘s coverage area have made intra-satellite handover, the most common handover experienced in non-GEO satellites networks. Many intra-satellite handover schemes have been proposed in the literature. The two major types are guaranteed handover schemes and prioritized handover schemes. The guaranteed handover schemes as the name implies, guarantees handovers for any call that is admitted, thus it reduces blocking probability to almost zero while the blocking probability is very high. This is due to the fact that a new call is only admitted if there are available resources not only in its current cell but also in the next transit cell. This leads to poor resource management and several schemes like elastic handover scheme, Time Based Channel Reservation Algorithm (TCRA) handover scheme and Dynamic Doppler Based Handover Prioritization (DDBHP) scheme have been proposed to improve on this.

The prioritized handover schemes were introduced to prioritize handovers and ensure handover failures are reduced. This can be achieved by reservation of carefully chosen optimized guard channels (reserved channels), queuing handover requests through different queuing schemes and implementation of channel rearrangement [52].

2.4.2.2 Inter-Satellite Handover Scheme

Due to the fact that in LEO satellite systems, the period of visibility can be as small as five minutes, the inter-satellite handover mechanism is of huge significance. There are few inter-satellites schemes as compared to intra-satellites schemes. One of the proposed schemes uses three criteria in taking handover decisions. According to the scheme, if the present position of the mobile terminal at the time of admission, shows that handover will take place in less than a predefined time t_{TH} then a channel is concurrently reserved at the satellite selected for the next handover, else the call is rejected. When the call is admitted, the current satellite that provides services to the mobile terminal does station monitoring [53]. The satellite for the next handover is selected using the following criteria;

- The satellite that will provide the maximum serving period. This will reduce the number of handovers.
- The satellite with the maximum number of free channels. This will ensure even distribution of load and will prevent overloading.
- The satellite closest to the mobile terminal is selected. This will prevent link failure.

It is very important to choose the right t_{TH} value because a high and low value will lead to high blocking and forced termination probabilities respectively.

2.4.2.3 Inter-Segment Handover Scheme

To achieve global coverage and seamless global roaming, the integration of terrestrial and satellite networks for mobile communication services is crucial. The inter-segment handover scheme is a fundamental solution, in order to successfully implement the terrestrial and satellite integration. The inter-segment handover process can either occur as a terrestrial-to-satellite handover or satellite-to-terrestrial handover. As stated in the literature that proposed inter-segment handover schemes, it is expected that the mobile terminal will operates in dual mode. It will be able to connect to both terrestrial and satellite networks. Once a change in QoS is noticed by the mobile terminal it sends a handover request through the satellite access network to the intelligent network. The home intelligent network of the mobile terminal finds a target terrestrial cell and sends the information to the satellite terrestrial networks which then makes a resource request to its terrestrial counterparts. Once the requested resource is available, the RNC reserves the resource and then the mobile terminal receives a handover command from the intelligent network of the satellite access network. The handover is then executed and the mobile terminal establishes a connection to the terrestrial networks [54]. A brief analysis of main inter-segment handovers schemes are provided in [55].

2.4.3 Power Control

Due to the fact that mobile users share the available limited resources to allow effective usage, there is bound to be interference. The power control is an important radio resource management function that minimizes the common interference experienced between mobile users, at same time compensating for the varying channel conditions. The main objectives of power control is to use constant power and variable coding and modulation to maximize transmission rate at good channel conditions and also to maintain constant Signal to Interference Ratio (SIR) thereby ensuring constant data rate [56]. The power control can be effected from both uplink and downlink direction. The uplink power control counteracts fading by dynamically increasing the transmitting power of carriers at the earth station [57]. The implementation of uplink power control can be achieved in several ways including these three different techniques [58];

- Open loop scheme is the least complex technique due to the fact that it can be installed at an earth station without any system wide considerations. It relies on the beacon signal fading measurement so as to control the uplink power.
- Closed loop scheme uses the transponded carrier of the earth station that is transmitting to estimate the fading experienced at the uplink. The closed loop scheme is sometimes not attainable since the availability of the transponded carrier depends on the configuration of the satellite network.
- Feedback loop scheme is more complex and can provide superior power control accuracy. A central control station is used for monitoring the power in such a way that the earth stations report their received signal strength to it. The central control station then commands the affected earth stations to adjust their power to compensate for fading experienced in their uplink.

The downlink power control increases the transmitting power of the carriers at the satellite to compensate for the fading. The downlink carrier power reduces and the sky noise temperature experienced by the earth stations increases when downlink fading occurs.

The power control algorithms uses two methods which are SIR based and distance based methods. Some of the SIR based methods which uses the received SIR at the transmitter to adjust the transmitting power are proposed in [59] [60], while some distance based methods which uses the distance between the transmitter and receiver to adjust the transmitting power are proposed in [61] [62]. Some proposed algorithms in literature have improved on these two approaches by using adaptive power control algorithms [63] [64].

2.4.4 Packet Scheduling

Packet scheduling is a key component towards implementing an effective radio resource management. Most networks are increasingly packet based networks which allows resource sharing. This leads to contention between the packets in order to claim the limited resources. The packet scheduler is therefore needed to decide which packet is to be allocated resources [21] [65]. The basic challenge of a packet scheduler is how to share the available resources to the group of users that are ready to transmit over the common link [12]. Multi-service networks with mixed traffic needs more advanced packet scheduler that will be able to prioritize users in order to provide acceptable QoS requirements without under-utilizing the available resources [65]. In order to achieve a guarantee QoS performance, a connection admission control is used together with a packet scheduling scheme. The combination of the two is known as service discipline [66].

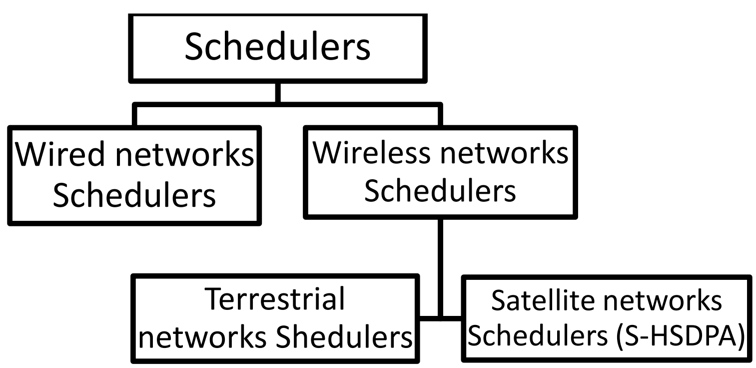

Figure 2.1 Classification of schedulers as discussed in this chapter

Many packet scheduling algorithms have been proposed for wired networks. However, these set of schedulers are not suitable for wireless networks due to the different nature of the two networks. An overview of the various packet scheduling schemes is provided in the rest of this chapter as classified in Fig. 2.1.

2.4.4.1 Wired Network Schedulers

Several wired network packet schedulers ranging from the best effort oriented ones to the ones that ensure QoS, have been proposed in the literature. The wired network schedulers are important set of schedulers that forms the basis of most wireless networks packet scheduler.Some of the wired network schedulers are presented as follows:

FIFO

The First-In-First-Out (FIFO) is the simplest packet scheduling scheme. This scheduler serves the packets in the queue in order of arrival and when the queue is full, it drops the packets that are just arriving [65]. This makes the scheduler effective and very easy to implement. The major setback is that it cannot differentiate between packets, therefore all packets experience the same delay, jitter and packet loss irrespective of which packet it is.

FIFO+

The FIFO+ was designed to improve on FIFO by reducing the jitter, since different queuing delays are experienced at different hops when using FIFO. The FIFO+ divides the traffic into classes based on priority. The average delay experienced by each packet in each class is measured at each hop, the difference between the measured delay of each packet and average delay of the class is computed. The difference is then used to compute the expected arrival time if the packet had experienced the average service at every hop. This computed arrival time is used to determine the order in which the packets will be scheduled in the queue [65] [67]. The main setback is its non-capability to provide fair sharing and also protect packets from being dropped.

Round Robin (RR)

The round robin scheduler is also a simple scheme which dedicates a queue to each user or class type and moves in a cyclic order to serve packets in queue that are not empty. It does treat all users equally but the major setback is that it does not have the capability to differentiate between various traffic types. This has lead to introduction of improved round robin schedulers that can provide QoS.

Weighted Round Robin (WRR)

This is the first improvement that was made to round robin scheduler. The Weighted Round Robin (WRR) scheduler allocates weight to each traffic class type. The WRR scheduler provides a user a share of the available resources based on its weight. It is worthy to note that if one of the traffic class types are not using part or all of the allocated resources, the unused resources is shared by other classes based on their weights.

Deficit Round Robin (DRR)

The Deficit Round Robin (DRR) [68] is another level of improvement on round robin scheduler. It does improve on WRR by catering for variable packet sizes in order to ensure fairness. It provides high level of fairness among users by ensuring that if a queue is not served in a round due to large packet size, it does add that portion to the other portion that is meant to be allocated to the queue in the next round, therefore compensating for the earlier round. This means that it keeps record of every deficit for each queue.

Hierarchical Round Robin (HRR)

The Hierarchical Round Robin (HRR) is also an extension to round robin scheduler. It does differentiate traffic class types into levels. The high level classes are allocated more bandwidth than the lower level classes so the frame time (the time it takes to service all slots at each level) for high levels is smaller than that of the low levels. This scheduler ensures each level has a constant share of bandwidth by providing round robin service to a fixed number of slots at each level, with each level allocated a certain number of slots. Its setback is that it still has a problem of combining delay and bandwidth allocation granularity [65] [69].

Earliest Due Date (EDD)

The Earliest Due Date (EDD) scheduling scheme allots deadline to each packet that arrives in a queue for service allocation. Services are therefore allocated to packets in order of increasing deadlines. Two modifications have been made to EDD. The Delay EDD assures each channel a hard delay bound provided the source does not go beyond the peak and the average sending rate as agreed with the server. In Delay EDD scheduling, the server sets a deadline for each packet, this deadline is determined using peak rate and the delay bound. The other modification is the Jitter EDD, which extends Delay EDD to provide delay jitter bounds by ensuring a certain minimum and maximum delay. This is achieved by stamping each packet at each server, with the difference between the deadline and the finishing time and each switch holds the packet for this period before it is scheduled [69].

Generalized processor Sharing (GPS)

The Generalized Processor Sharing (GPS) [65] [70] is a scheduling scheme which has become a prototype to achieve service differentiation in a multi-service communications network. It is designed on fluid flow principle and shares resources to different classes of traffic based on the assigned weight. The GPS provides a max-min fair share allocation by distributing the outgoing capacity to all backlogged sessions in proportion to their resource requirements. The basic procedure of GPS can be described using a mathematical concept. Let us assume we have *m* classes with each class having its own queue and the total service rate is *c*. Each class *i* is assigned a positive weight $\emptyset_i$ in such a way that the total weight of all classes is one (1) and each class is guaranteed a minimum of $\emptyset_i c$. If one or some of the classes does not have any backlog (does not require any service) then the excess is shared among the remaining classes with backlog in accordance with their respective weights. If a set of backlog classes is denoted by *B,* so a backlog class $i \in B$ is guaranteed of a service rate;

$$r_i = \frac{\emptyset_i}{\sum_{j \in B} \emptyset_j} c \geq \emptyset_i c \tag{2.5}$$

Many GPS based works have been done in the literature and some can be found in [71][72][(73]. The issue with GPS is that it is not practical to implement.

Weighted Fair Queuing (WFQ)

Since GPS can not be practically implemented, Weighted Fair Queuing (WFQ) was introduced for this purpose. WFQ also known as Packetized GPS is built on GPS principle by adapting it to packets scenario instead of fluids. The WFQ scheduling scheme selects among all the packets in the queues at every time *t*, the first packet that will complete its service in the corresponding GPS system assuming if no additional packets were to arrive the queue after time *t*. WFQ only uses the time it will take the packet to finish in GPS system, in making decision. Another GPS based scheduler proposed is called Worst-case Weighted Fair Queuing (WF^2Q), it improved on WFQ by using not only the finish time of the packet in GPS system but both the start and finish time of the packet in GPS system in order to achieve an exact emulation of GPS [74].

2.4.4.2 Wireless Network Schedulers

Several proposed wired network schedulers do not put into consideration several factors that are experienced in wireless networks scenarios. These set of factors include bursty errors, time varying wireless link, limited bandwidth, mobile users' mobility, varying channel capacity based on mobile users' location and power control [75]. All these factors have made the design or implementation of wireless networks schedulers very challenging. Therefore, an effective wireless networks scheduler must be aware of channel variations and utilize it and also provide QoS and fairness to all classes of users. All these must be achieved without too much complexity. Several wireless networks schedulers have been proposed, some of them are presented below.

Channel State Dependent Packet Scheduling (CSDPS)

The Channel State Dependent Packet Scheduler (CSDPS) is a wireless scheduling scheme that addresses bursty errors by ensuring that a mobile user is only allocated resources when the user is not tagged to be in a bad channel state. The principle it uses is simple, it keeps each mobile user in a separate queue and uses link status monitor (LSM) to keep track of the channel state of each mobile user. Once the mobile user detects that the link of a user is bad, it marks the user's queue and the user is not served for a fixed period of time after which it is unmarked. It does allow different service discipline to select the queue it will choose the next packet from. It improves channel utilization but the setbacks are that it does not guarantee bandwidth to mobile users and does not provide fair sharing of resources among users.

To address the problem of unfair sharing of bandwidth, a new scheduler was proposed which combines CSDPS and Class Based Queuing (CBQ). The CBQ is used to provide fair resource sharing by dividing the mobile users into classes and it ensures each class is allocated a certain portion of the available bandwidth, while the CSDPS handles wireless channel variation. Despite the fact that it addresses unfair resource sharing, it doesn't compensate for users who at some time were denied resources due to bursty errors and also doesn't guarantee QoS [75].

Idealized Wireless Fair Queuing (IWFQ)

The Idealized Wireless Fair Queuing (IWFQ) is a wireless version of Weighted Fair Queuing (WFQ). It is also built from the wireless version of Generalized Process Sharing (GPS) called Wireless Fluid Fair Queuing (WFFQ). Since WFFQ implementation is unrealistic because it is based on fluid model, IWFQ which is based on packets emulation was introduced. The scheduler uses the finish time of each packet to decide on which packet is to be served. This is similar to the

WFQ. The difference comes when the packet selected to be served is in a bad state, then the scheduler selects the next packet in another queue with the next smallest finish time. If the user with packet denied from transmitting due to bad channel state returns to good state, the user will have a smaller finish time compared to other queues, since the arrival time of packet does not change. This will enable this packet to have preference to be selected for transmission [75].

Channel-Condition-Independent Fair Queuing (CIF-Q)

Channel-Condition-Independent Fair Queuing (CIF-Q) is a scheduling scheme that was designed to majorly address the issue of both short-term and long-term fairness. It makes its scheduling decisions based on start time rather than using the finish time which IWFQ uses and therefore uses Start Time Fair Queuing (STFQ) as its error free service model. It is able to track by how much a session is leading or lagging its ideal service assuming error free channel by using lead and lag counter. It then uses a parameter α which falls within the range of 0 and 1 to control the rate at which the session that is leading will forsake its lead to the lagging session [76]. The major advantage CIF-Q has over IWFQ is its ability to guarantee both short-term and long-term fairness. However, since CIF-Q and IWFQ schedulers uses start and finish time respectively, both of them have the problem of determining the arrival time of uplink packets [75].

Server-Based Fairness Approach (SBFA)

Server-Based Fairness Approach (SBFA) uses Long-Term Fairness Server (LTFS) to compensate lagging sessions whose packets are not transmitted due to link errors [76]. This scheme uses two queues for each flow, one is the packet queue and the other is called slot queue. A virtual copy (slot) of every packet of a flow that arrives at the packet queue is also stored in the slot queue. The scheduler can use any of the wired network scheduler to select a slot in the slot queue and the corresponding Head of Line (HOL) packet of the flow for the selected packet‘s slot is transmitted provided the channel state is good. Otherwise, a slot for this flow is created and stored in LTFS queue. These set of flows that are denied transmission due to link errors are in due course compensated. This is achieved by constantly monitoring the loss of each flow in the LTFS queue and a dedication of part of the bandwidth to LTFS. It has a simple design and guarantees throughput, however, it does not restrict the service a flow can be allocated, thereby allowing excessive service to flows continuously in a good state. Another setback is that it assumes that all packets have a fixed size thus if the packet size varies, the scheduler will be ineffective [75]. While it addresses the problem of service deterioration of error-free flows due to compensation by using a reserved bandwidth, it does not give a fair share to flows that are to be compensated.

This is due to the fact that it uses First Come First Serve (FCFS). This was addressed in another proposed scheduler called Packetized Wireless General Processor Sharing (PWGPS). This proposed scheduler compensates each flows affected by bad channel state by not only using reserved bandwidth for compensation but also using increment in their service share to ensure fairness [77].

Wireless Fair Service (WFS)

The Wireless Fair Service (WFS) scheduling scheme provides fairness, delay and bandwidth guarantees. Each packet that arrives at the queue is timestamped and the service tag for each flow is set to be equal to the virtual finish time of the Head of Line (HOL) packet of the flow [76]. At every time t, the packet with the lowest service tag is selected among the packets whose virtual start time is less than $v(t) + \delta$. The $v(t)$ stands for the virtual start time while δ is the ‗lookahead' system parameter [78]. The δ is used to determine the schedulable interval and so if $\delta = 0$ at service rate Φ_i then it uses WF^2Q as its error free service model, while if $\delta = \infty$ the error free service model is WFQ. It achieves fairness among the flows and ensures gracious service degradation of leading flows by using lead and lag counter. Each flow has a lead bound of $l_{i,max}$ and lag bound of $b_{i,max}$. The leading flows give back the lead portion of its service, $l_i / l_{i,max}$ to the lagging flows while each lagging flows get a part of the total returned resources by the leading sessions in proportion to its lag due to link error. This proportion is determined using $b_i / \sum_{i \epsilon S} b$, where S is set of backlog flows [21].

2.4.4.3 Packet Schedulers for HSDPA via Satellite

Packet scheduling plays a very big role in the design of both terrestrial and satellite HSDPA and it's a key entity in the actualization of higher data rates delivery to mobile users. The design of a packet scheduler for S-HSDPA that will be able to achieve acceptable QoS to different mobile users without compromising effective resource utilization and fairness among users is a challenging one. The long propagation delay, slow link adaptation to channel conditions due to the distance between the mobile user and the scheduler makes it more challenging as compared to the terrestrial counterparts. Some packet schedulers used in HSDPA via satellite are presented below.

Proportional Fair (PF) Scheduler

The Proportional Fair (PF) scheduler as proposed in [79] and used in [80] [81] for S-HSDPA networks is said to be an opportunistic scheduling that uses the Relative Channel Quality Index

(RCQI) in making scheduling decisions. The User Equipment (UE) with the highest RCQI is allocated resources. The RCQI is said to be the ratio of the UE's maximum supported current data rate to the average data rate experienced in the past by the UE. The PF scheduler is generally regarded to be a scheduler that provides a good trade-off in allocation of resources between mobile users. It does this by balancing between maximizing the throughput and ensuring all users receives a certain level of service. This good trade-off is only feasible if the network is made up of a single traffic class. The setback is that it does not have the capability to differentiate between different traffic classes, therefore, it can provide acceptable QoS when delay-sensitive traffic is present [82].

Earliest Deadline First (EDF)

The Earliest Deadline First (EDF) is a dynamic priority packet scheduling scheme that allocates resources to mobile users or traffic based on urgency [21]. This scheduler has been used for S-HSDPA networks as presented in [80] [81]. Each packet has its own deadline and the deadline for each traffic class is set based on its delay sensitivity. The packet with the least residual time is allocated resources. It uses the ratio between the waiting time of the HOL packet in a queue and its corresponding deadline in making scheduling decisions. It is very suitable for real time traffic [82]. Its setback is that it does not put the channel conditions into consideration and therefore does not utilize the resources available and also non-real time traffic will be starved in a mixed traffic scenario.

Modified Largest Weighted Delay First (M-LWDF)

The Modified Largest Weighted Delay First (M-LWDF) is a scheduler that combines both channel conditions and the state of the queue with respect to delay in making scheduling decisions. This scheduler has been used as presented in [83] for both terrestrial and satellite HSDPA networks. It ensures that the probability of delay packets does not exceed the discarded bound below the maximum allowable Service Data Unit (SDU) error ratio.

$$\Pr(d_i > T_i) \leq \delta_i \tag{2.6}$$

Where d_i is the HOL packet delay of mobile user i, T_i is the delay bound and δ_i is the allowed percentage of discard packets. The scheduler allocates resources to the user with the maximum priority index which is made up of the product of the HOL packet delay of the user, the channel capacity with respect to flow and the QoS differentiating factor [84]. It provides better optimal performance compared to other schedulers but still favours delay-sensitive traffic at the detriment of non-real time traffic.

2.5 Summary

In this chapter, an overview of RRM and the discussion of the concept of cross-layer design as it is applicable to RRM were presented. The various cross-layer interactions that are used to achieve RRM were also presented. The details of each of the RRM functions with some examples were then discussed. Finally, packet scheduling schemes are presented in a systematic manner by dividing it into two main categories which are wired and wireless schedulers. The wireless schedulers are then further subdivided into terrestrial based and satellite based schedulers. Detailed discussions of these schedulers with their respective examples were presented.

Chapter 3

CHANNEL MODELING

3.1 Introduction

The main objective of this chapter is to present the three state Markov chain channel model that is used in subsequent chapters of this dissertation, to estimate the Signal-to-Noise Ratio and the corresponding Channel Quality Indicator (CQI) in a satellite communication system. The effects of the SNR margin that is introduced to compensate for the long propagation delay experienced in satellite channels on S-HSDPA performance are also presented.

The chapter begins with the analysis of the two state Markov channel model with its transition probabilities. The analysis of the three state Markov channel model which is used in the subsequent chapters is also presented. The determination of SNR using Link Budget and the analysis of how CQI is determined from the corresponding SNR are presented.

The rest of the chapter provides the simulation results and analysis of the effect of the introduction of SNR margin on HSDPA via GEO satellite channel.

The Section 3.7 of this chapter is an improvement on the work presented at the 5th IEEE International Workshop of Satellite and Space Communications 2009 (IWSSC 2009), Siena-Tuscany, Italy.

3.2 Overview of Channel Modeling in Satellite Mobile Networks

The channel modeling is very important in the prediction of the performance of the genuine service experienced in Land Mobile Satellite Systems (LMSS). The Land Mobile Satellite (LMS) channel can be divided into three different effects as provided in [85]. They are known as the ionospheric effects, the tropospheric effects and the effects caused by the interaction of the radio waves with elements within the region of the mobile terminal e.g. buildings and vegetation commonly called the local effects. The tropospheric effects are negligible at frequencies below 10

GHz. The received LMS signal consists of direct LOS wave, the diffuse wave and the specular ground reflection. The major propagation impairments that affect the direct component of the signals at the band which S-HSDPA uses, known as S-band are Free Space Loss (FSL) and shadowing. The diffuse component which comprises of the multipath signals has only a slight effect in mobile-satellite channel links as compared to the terrestrial channel links in practical environments [2].

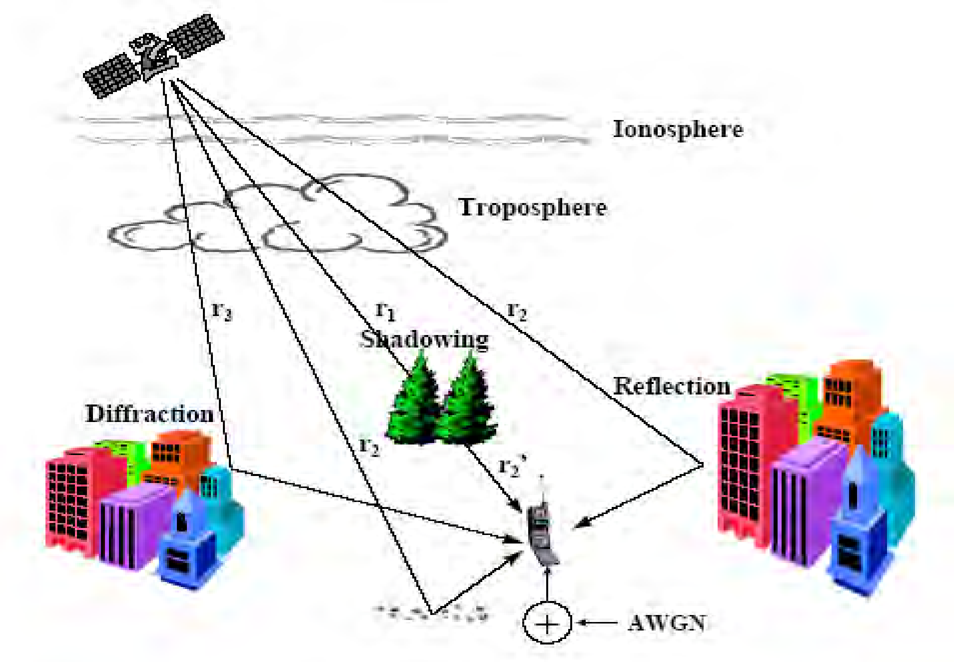

Figure 3.1 Mobile satellite channel [85]

There are several models that have been proposed in the literature that can be used to characterize the mobile-satellite channel for narrowband transmissions. They are categorized into empirical, statistical and geometric-analytical. The statistical models can easily predict the system performance under various conditions of coding and modulation compared to other model types [86]. The Markov multiple state models which falls under statistical models is the most regularly used channel models for LMS [87]. This is due to the fact that they can quickly switch from one signal level to another.

3.3 Two State Markov Channel Model

The two state Markov channel model is the simplest multistate Markov channel model. It has two states which are good and bad state. This distinguishes between the time the channel has high received signal power (good state) and low received signal power (bad state). As a result of this, it is expected that the two states will have different channel characteristics.

In [88], a two state satellite mobile channel model termed *Lutz* model was proposed. It has both good and bad states. According to the model, the good state refers to unshadowed areas which corresponds to state with direct Line of Sight (LOS) while bad state refers to shadowed areas which corresponds to areas where direct satellite signals are shadowed by obstacles like trees, high buildings and so on.

In good state, the total received signal is said to form a rician process and the probability density function of the momentary received power S as stated in [88] follows the rician probability density. This can be stated as:

$$P_{Rice}(s) = ce^{-c(S+1)} I_0(2c\sqrt{s}) \tag{3.1}$$

Where c is the direct-to-multipath signal power ratio (Rice factor), S is the received power and I_0 is the modified Bessel function of order of zero. For the bad state, when shadowing is present, it is assumed that the direct signal is totally obstructed and the multipath signal has a Rayleigh characteristic with mean received power of S_o. The probability density function of the received power conditioned on mean power S_o is stated as:

$$P_{Rayleig\ h}(S|S_0) = \frac{1}{S_0} e^{\left(-S/S_0\right)} \tag{3.2}$$

The time varying mean received power S_0 is assumed to be lognormally distributed due to the slow shadowing process,

$$P_{LN}(S_0) = \frac{10}{\sqrt{2\pi}\,\sigma \ln 10} \cdot \frac{1}{S_0} e^{\left[-\frac{(10 \log S_0 - \mu)^2}{2\sigma^2}\right]} \tag{3.3}$$

Where μ is the average power level due to fading and σ^2 is the variance of the power level due to shadowing. The resulting probability density function of the received signal which is made up of the above three stated densities can be stated as [2] [88]:

$$p(S) = (1 - A) \cdot p_{Rice}(S) + A \cdot \int_0^\infty p_{Rayleig\ h}(S|S_0) p_{LN}(S_0)\, d\, S_0 \tag{3.4}$$

The term A can be described as a shadowing factor that is used to determine the time sharing of the shadow fading.

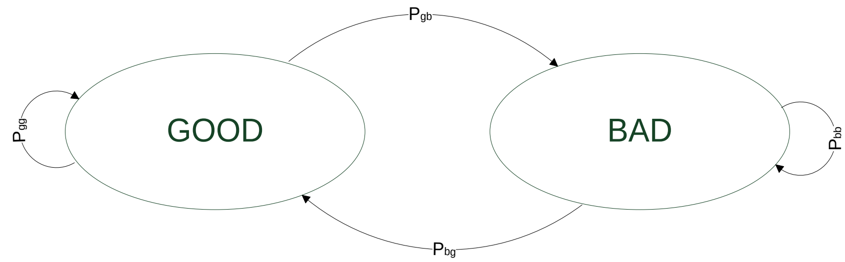

Figure 3.2 Two states Markov channel model

The switching process between a good state and a bad state can be characterized by a two state Markov model as shown in Fig. 3.2. The transition probability is used to switch the channel from one state to another. The threshold used to decide the good and bad state is determined in such a way that the time shared when received power signal is below this threshold must be equal to *A*. The threshold is determined by the probability density function of the received signal power. The duration at which the received signal power is greater or lower than the threshold are known as the good and bad states respectively. The probability that a good state remain in the same state is said to be the probability that a good channel state last longer than *n* bits duration (observation period). This also applies to the probability that a bad channel state remains in its state. The transition probabilities equations are presented below [88];

$$P_r(P_g\,(> n)) = P_{gg} \tag{3.5}$$

$$P_r(P_b\,(> n)) = P_{bb} \tag{3.6}$$

$$P_{gb} = 1 - P_{gg} \tag{3.7}$$

$$P_{bg} = 1 - P_{bb} \tag{3.8}$$

The P_{gg} is the probability that the channel remains in good state, P_{bb} is the probability that the channel remains in bad state, P_{gb} is the transition probability from good to bad state and P_{bg} is the transition probabilities from bad to good state. The transition probability can be related with the duration of each state and the duration can be determined with the knowledge of the user's velocity and bit rate. According to the Markov model, the mean duration for both good and bad channel state is stated as:

$$D_g = \frac{1}{P_{gb}} = \frac{R}{v} D_g\,(m) \tag{3.9}$$

$$D_b = \frac{1}{P_{bg}} = \frac{R}{v} D_b\ (m) \tag{3.10}$$

R is the transmission rate, v is the user's velocity, m in meters while D_g and D_b are the good and bad states respective durations measured in bits. The A used in (3.4) termed the shadowing factor can therefore be determined as follows:

$$A = \frac{D_b}{D_{g+D_b}} \tag{3.11}$$

3.4 Three State Markov Channel Model

The three state Markov channel model provides a more realistic channel characterization by increasing the numbers of possible states. It is very important to generate many received signal states. This will allow some level of dynamism in scheduling decisions made by packet schedulers and will no doubt increase the performance of the schedulers. The three possible states as proposed in [89] are LOS, moderate shadowing and deep shadowing states.

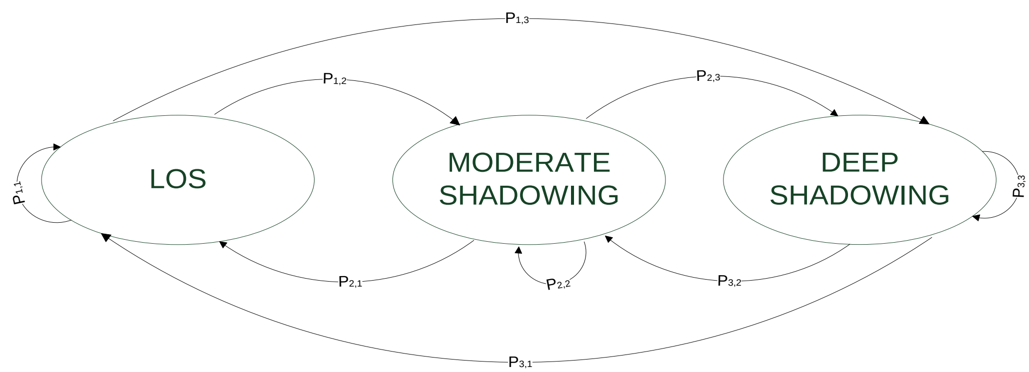

Figure 3.3 Three state Markov channel model

The Markov chain model depends only on the previous state to determine the next state and it is used to predict the next state out of the three states. The Markov chain model decides the given state using the state probability matrix $[W]$ and state transition probability matrix $[P]$. The matrix $[W]$ contains each state probability where each element W_i represents the probability of being in state i, while matrix $[P]$ contains the probability of changing from one state to another where each element $P_{i,j}$, represents the probability of changing from state i to state j. The conditions guiding the Markov chains are that the sum of all the elements in every row in matrix $[P]$ must be equal to one, the sum of all the elements in matrix $[W]$ must be equal to one and convergence property of the Markov chain is defined by $[W]\ [P] = [W]$. Each element of matrices $[W]$ and $[P]$ are determined as follows;

$$P_k\left(Env_j\right) = \frac{N_k}{N_f} \tag{3.12}$$

$$P_{i,j}\left(Env_k\right) = \frac{M_{i,j}}{M_i} \tag{3.13}$$

N_k is the number of samples corresponding to state k in an environment j and N_f is the total number of samples obtained in environment j. The number of samples are said to be the number of channel snapshots or signal series measured in a particular state at a particular elevation angle in a particular environment. The received signal series measured are sampled at a rate of 8ksamples/s from a comprehensive data set obtained at S-band. As presented in Table 3.1, the states are LOS, moderate shadow and deep shadow while the environments considered for these measurements are open, suburban, urban, intermediate tree shadow and heavy tree shadow. The $M_{i,j}$ is the number of transitions experienced from state i to state j from the available measurements, while M_i is the number of state frames corresponding to state *i* in an environment. The state frame is known as the state's minimum possible travel route length. The travelled route is made up of frames and state only changes at the beginning or the end of each frame. The measurements that are used to determine each element of both the state and transition probability matrices are obtained using a receiver and a data acquisition system for various elevation angles at different environments at S-band (frequency band). The state and transition probabilities for S-band as summarized in [90], is presented in Table 3.1.

Table 3.1 State and transition probabilities for the 3-state Markov channel model. [90]

State and Transition 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
Open	95.86	3.22	0.92
	5.71	92.38	1.91
	3.90	1.29	94.81
	54.42	26.27	19.29
Suburban	79.97	15.02	5.01
	13.64	77.84	8.52
	12.20	12.20	75.60

	39.58	38.76	21.66
Intermediate Tree-Shadow	78.26	14.95	6.79
	24.81	68.99	6.20
	24.04	6.01	69.95
	53.00	29.03	17.97
Heavy Tree-Shadow	81.75	7.30	10.95
	2.01	94.65	3.34
	3.32	9.98	86.70
	11.56	63.07	25.37
Urban	88.28	6.62	5.10
	14.47	81.39	4.14
	8.48	4.71	86.81
	49.18	24.24	26.58

The Loo‘s model is one of the two most widely used statistical models for LMSS [2]. The received signal of each state is determined using the Loo probability distribution which has shown reasonable level of agreement with measured data [90]. The Loo distribution consists of two components which are the direct ray attenuation and multipath. The direct ray attenuation is said to be lognormally distributed with a mean α (dB) and standard deviation ψ (dB) while the multipath power is said to be Rayleigh distributed with an average multipath power MP (dB).The Loo probability density function can be stated as:

$$P(r) = \frac{r}{b_{0\sqrt{2\pi d_0}}} \int_0^\infty \frac{1}{z} \exp\left[\frac{-(\ln z - \mu)^2}{2d_0} - \frac{(r^2+z^2)}{2b_0}\right] I_o\left(\frac{rz}{b_0}\right) \quad (3.14)$$

b_0 is the multipath power, d_0 is the variance in the direct ray attenuation‘s variance, μ is the mean value of the direct ray attenuation, *z* is lognormally distributed, *r* is the ricean vector and I_0 is the modified Bessel function of zeroth order. The more suitable values α, ψ and MP which are derived from μ, d_0 and b_0 respectively and are used as parameters for the purpose of generating the direct ray attenuation due to shadowing and variation due to multipath at every instant depending on the channel state. The parameters are derived as follows [90]:

$$\propto (dB) = 20 \log_{10}(e^{\mu}) \quad (3.15)$$

$$\varphi\ (dB) = 20 \log_{10}(e^{\sqrt{d_0}}) \quad (3.16)$$

$$MP\ (dB) = 20\log_{10}(2b_0) \tag{3.17}$$

The parameters for each environment at each elevation angle are presented in Table 3.2.

Table 3.2 Statistical parameters of Loo model for the three channel states [90]

Environment	Elev.	State1: LOS			State2:Moderate Shadow			State3: Deep Shadow		
		α(dB)	Ψ(dB)	MP(dB)	α(dB)	Ψ(dB)	MP(dB)	α(dB)	Ψ(dB)	MP(dB)
Open	40°	0.1	0.37	-22.0	-1.0	0.5	-22.0	-2.25	0.13	-21.2
	60°	0.0	0.12	-24.9	-0.7	0.12	-26.1	-1.4	0.25	-23.1
	70°	-0.1	0.25	-22.5	-0.5	0.28	-24.5	-0.75	0.37	-23.24
	80°	0.1	0.16	-22.4	-0.4	0.15	-23.5	-0.72	0.27	-22.0
Suburban	40°	-1.0	0.5	-13.0	-3.7	0.98	-12.2	-15.0	5.9	-13.0
	60°	-0.3	0.91	-15.7	-2.0	0.5	-13.0	-3.8	0.34	-13.2
	70°	-	-	-	-	-	-	-	-	-
	80°	-0.4	0.58	-13.7	-2.5	0.2	-16.0	-4.25	3.0	-25.0
Intermediate Tree shadow	40°	-0.4	1.5	-13.2	-8.2	3.9	-12.7	-17.0	3.14	-10.0
	60°	-0.2	0.75	-14.0	-3.1	1.9	-15.5	-	-	-
	70°	-0.8	0.75	-10.0	-3.3	1.1	-10.75	-7.7	2.9	-10.2
	80°	-0.6	1.87	-9.25	-2.5	1.55	-10.0	-4.6	2.0	-13.4
Heavy shadow	40°	-	-	-	-10.1	2.25	-10.0	-19.0	4.0	-10.0
	60°	-	-	-	-7.7	4.0	-10.1	-10.8	2.7	-10.0
	70°	-	-	-	-4.5	4.6	-12.1	-7.5	2.0	-7.0
	80°	-0.9	3.0	-9.1	-3.1	3.4	-9.0	-8.0	5.0	-7.0
Urban	40°	-0.3	0.73	-15.9	-8.0	4.5	-19.2	-24.4	4.5	-19.0
	60°	-0.35	0.26	-16.0	-6.3	1.4	-13.0	-15.2	5.0	-24.8
	70°	-0.5	1.0	-19.0	-5.6	1.2	-10.0	-12.3	4.1	-16.0
	80°	-0.25	0.87	-21.7	-6.6	2.3	-13.0	-11.0	8.75	-24.2

3.5 Link Budget

The CQI which is used to determine the transmission rate and also used for scheduling decisions by channel based schedulers in S-HSDPA networks is determined by the corresponding SNR. This SNR is obtained from the link budget analysis. The SNR is said to be the ratio of the received power (P_r) to noise power (N). The received power can be determined by the following equation:

$$P_r = P_T \cdot G_T \cdot G_R \cdot L_{all} \tag{3.18}$$

P_T is the transmit power, G_T is the gain of transmitting antenna, G_R is the gain of the receiving antenna and L_{all} is the total loss which includes free space loss L_s and other additional losses L_{others}. The additional loss considered in this dissertation is the direct ray attenuation due to shadowing, since the only propagation impairments that significantly affects the signal in S-band are free space loss and shadowing [2] . Other additional losses like rain attenuation, depointing losses, polarization mismatch losses and atmospheric losses are neglected since they are all negligible at 2 GHz frequency band (S-band). Therefore,

$$L_{all} = L_s + L_{others}\ (dB) \tag{3.19}$$

The free space loss is determined based on the distance between the transmitter and the receiver and also the carrier frequency that is being used. It is defined as the ratio of the received and transmitted power in a link between two isotropic antennas.

$$L_s = \frac{4\pi R}{\lambda} \tag{3.20}$$

λ is the wavelength which is the ratio of the speed of light (c= 3x10^8 m/s) and frequency (carrier frequency e.g. 2 GHz), and R is the distance between the two antennas. The SNR can then be stated as:

$$SNR = P_r - N \tag{3.21}$$

Where $N = KTB$, K is the Boltzmann's constant (K = -228.6 dBW/K/Hz), T is the noise temperature (in Kelvin) and B is the bandwidth of the noise. The SNR equation can then be rewritten as:

$$SNR_j = 10\log P_T G_T + 10\log G_R/T - 10\log KB - L_{all\,j}\ (dB) \tag{3.22}$$

$$and \quad L_{all\,j} = 20\log\frac{4\pi R}{\lambda} + L_{others_j}(dB) \tag{3.23}$$

The SNR_j varies with L_{others_j} , and the L_{others_j} value depends on the present channel state of the three state Markov channel model at every transmission time. It is generated using lognormal distribution as explained in section 3.3.

3.6 Channel Quality Indicator (CQI)

In both terrestrial and Satellite HSDPA networks, an up-to-date CQI value is required by the Node B (Base station in S-HSDPA) from the mobile users, to determine the appropriate Transport Block Size (TBS) at which it will transmit at every TTI. The corresponding TBS of the CQI value is determined from the lookup table presented in Table 3.3 below. The lookup table contains the modulation and coding used, and the numbers of code used. The CQI is also used by channel based schedulers in order to make optimal scheduling decisions.

Table 3.3 CQI mapping for TTI = 2 ms [21]

CQI value	Modulation and coding	Number of Codes used per TTI	Bits per TTI (TBS)
1	QPSK 1/3 (on each code 960 bits are Sent in a TTI)	1	137
2		1	173
3		1	233
4		1	317
5		1	377
6		1	461
7		2	650
8		2	792
9		2	931
10		3	1262
11		3	1483
12		3	1742
13		4	2279
14		4	2583
15		5	3319
16	16QAM 1/3 (on each code 1920 bits are Sent in a TTI)	5	3565
17		5	4189
18		5	4664
19		5	5287
20		5	5887
21		5	6554
22		5	7168
23		7	9719
24		8	11418
25		10	14411
26		12	17237
27		15	21754
28		15	23370
29		15	24222
30		15	25558

The CQI for every corresponding SNR at a certain Block Error Rate (BLER) can therefore be determined using the following equation as proposed in [91]:

$$SNR = \frac{\sqrt{3}-\log CQI}{2}\log(BLER^{0.7}-1) + 1.03CQI - 17.3 \qquad (3.24)$$

Equation (3.24) can further be simplified as follows [91];

$$CQI = \begin{pmatrix} 0 & if\ SNR \leq -16 \\ \frac{SNR}{1.02} + 16.02 & if - 16 \leq SNR \leq 14 \\ 30 & if\ 14 \leq SNR \end{pmatrix} \quad (3.25)$$

In this dissertation, the three state Markov channel model with the state and transition probabilities in Table 3.1, the link budget formula and the equation (3.25) that relates SNR and CQI are implemented using MATLAB script. The hidden Markov model function in MATLAB is used to generate the next state of the three possible states using the corresponding state and transition probabilities matrix depending on the environment considered. Depending on the state, the corresponding mean and standard deviation of the direct ray attenuation of the considered elevation angle in Table 3.2 is used to generate the shadow fading value. This is then added to the free space loss which adds up to give the total loss. The total loss is then substituted in the link budget formula which is used to compute the SNR. After computing the SNR, the CQI is then computed using (3.25). This is used to report the instantaneous CQI at every reporting interval in the simulations of our subsequent work.

3.7 Effect of SNR Margin in Satellite HSDPA Networks

There is a need for realistic channel modeling to determine the SNR. There is also need for an effective mechanism to ensure that the correct CQI corresponding to the instantaneous SNR experienced by the user is used for both scheduling decisions and resource allocation at the MAC sub-layer of the Node B in S-HSDPA networks. Due to the long propagation delay experienced in GEO Satellite, there is a misalignment between the SNR value reported at the Node B and the instantaneous SNR the User Equipment (UE) is experiencing at that time. This leads to either under utilization of resources or packet loss. The two common mitigation techniques that have been proposed are reduction of BLER to smaller values as compared to the one used in the terrestrial scenarios and the introduction of SNR margin, h [80]. The introduction of SNR margin will lead to a certain level of underutilization of resources, but since under utilizing resources is more acceptable as compared to loss of packets, the introduction of a margin, h, to be deducted from the SNR reported is considered as an option to compensate for the long propagation delay.

$$CQI = CQI(SNR - h, BLER) \quad (3.26)$$

The aim of this section is to investigate the effects of this margin, h on the performance of S-HSDPA and recommend the margin that will give the best performance.

3.7.1 Channel Model

The three state Markov channel model discussed in section 3.3 with LOS, moderate shadowing and deep shadowing states is adopted. The next state at each instance is decided using the state and transition probabilities matrix provided in Table 3.1 depending on the environment considered. This is obtained using MATLAB as explained in section 3.6. The SNR values and the corresponding CQI values for each instance are then used for the simulation. Since the simulation software considered for this work uses a preprocessed channel input trace file, a channel input trace file is generated for over 1000 seconds for the simulation by using MATLAB. Depending on the CQI, the corresponding TBS as presented in Table 2.1 is therefore used for transmission purposes. Since this work involves the investigation of the effect of different margins, the corresponding CQI are also determined for each SNR after deducting the corresponding margin, h. The SNR margins considered for this work are 0.0, 0.5, 1.0, 1.5, 2.0, 2.5, 3.0, 3.5 and 4.0 dB. The SNR and corresponding CQI values are generated using a MATLAB script for all these margin values and the channel trace inputs for over 1000 seconds are generated for the simulation. The CQI Probability Density Distributions (PDF) at margin of 0.0 and 4.0 dB of one of the users for the suburban environment obtained through simulation are presented Fig. 3.4 and 3.5:

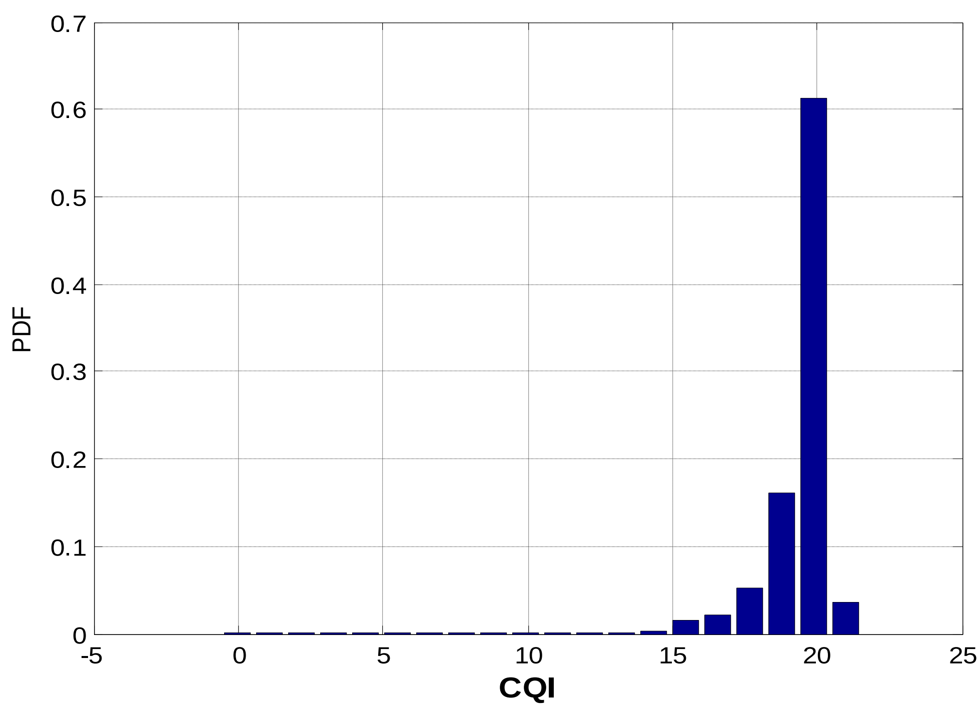

Figure 3.4 The CQI PDF distributions for a user at margin 0.0 dB in a suburban environment.

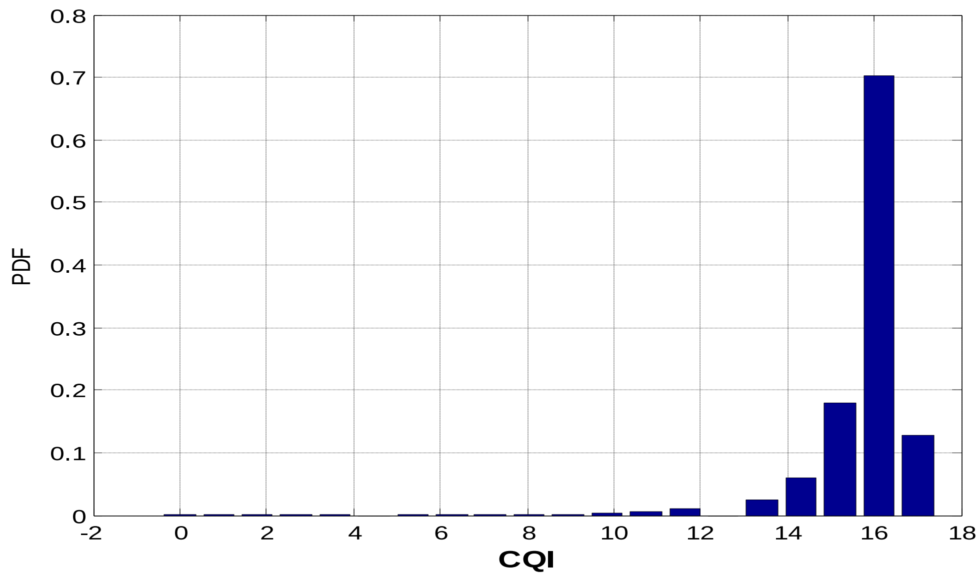

Figure 3.5 The CQI PDF distributions for a user at margin 4.0 dB in a suburban environment.

3.7.2 Traffic Model

The Web traffic has been considered as the TCP based traffic for this work and is modeled using M/Pareto ON/OFF model. Each web browsing download has web file size which is characterized by Pareto distribution with parameters α = 1.1, k=81.5 bytes and the maximum allowed packet size m = 66666 bytes [92] [93]. Where α is the Pareto index and k is the minimum packet size. The TCP version used for this simulation is TCP New Reno. See Appendix A2 for more details on web traffic model.

3.7.3 Simulation Set-up

The investigation of the effect of the SNR margin on the performance in S-HSDPA is implemented using discrete event simulation software called Network Simulator 2 (NS-2) [94]. Since NS-2 doesn‘t support UMTS and HSDPA by default, a UMTS/HSDPA extension to NS-2 called Enhanced UMTS Radio Access Network Extension (EURANE) module [95] is used on NS-2 platform. The maximum CQI Scheduler that selects user with the maximum CQI is used for the simulation. This is due to the fact that maximum CQI scheduler produces a good throughput performance for single traffic network. Since, only web traffic has been considered for this simulation, using another scheduler will not be relevant. It is assumed that the Scheduler only selects and allocates resources to a user at every TTI (2ms). The NS-2/EURANE has been adapted to satellite scenario by changing the physical layer characteristics and channel delay for the purpose of this work. This is achieved by replacing the preprocessed channel input trace file with that of satellite as explained in Section 3.7.1 and setting the channel delay in the tcl script to 280 ms. The Node B is connected to Wired Node 1 through RNC, Serving GPRS Support Node (SGSN), Gateway GPRS Support Node (GGSN) and Wired Node 2 as presented in Fig. 3.6. Every mobile terminal reports its CQI value at a certain reporting interval to Node B which is assumed to be every 40 ms (20 TTI) for this work as proposed in [80]. Depending on the CQI value which varies from 1 to 30 as shown in Table 3.3, the Node B transmits at the corresponding Modulation and Coding Scheme (MCS) level that will cope better with the channel conditions at every TTI. Each of the entities in the network below is represented as nodes in the tcl script written for the simulation.

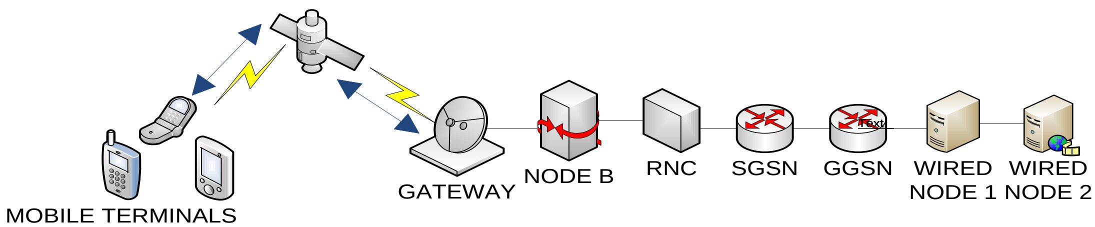

Figure 3.6 The S-HSDPA simulated network

The Simulations are conducted for three different traffic loads. The three different traffic loads considered are 384, 512 and 1024 kbps data rate of web traffic. The simulation is carried out by considering five web traffic users for each of the SNR margins, *h*, that ranges from 0.0 to 4.0 under a suburban area environment. The TCP version used for this simulation is TCP New Reno. This simulation is to provide the margin with best throughput and delay performance for each of the traffic load considered. The simulation details are provided in Table 3.4 below.

Table 3.4 Simulation details and parameters

Parameter	Values
Simulation Time	1000s
Numbers of Users	Up to 5
Transmission Time Interval (TTI)	2 ms
Channel Model	three state Markov channel model
CQI reporting interval	40 ms (20 TTI)
Channel Delay	280 ms (RTPD = 560ms)
Traffic Model	M/Pareto ON-OFF Model
Traffic Load	384,512,1024 Kbps
TCP Version	TCP New Reno
Packet Scheduler	Maximum CQI scheduler
RLC Mode	Acknowledged Mode (AM)
UE Category	5 & 6

3.7.4 Simulation Results

After discussing the simulation models and set-up, the simulation results are presented in this section. The goodput and delay performances of the various margins are compared. The simulations that are implemented produced huge event driven data results. To evaluate and analyze this huge result, the gawk programming language is used, and the Microsoft spreadsheet (Excel) is used to plot the graphs. The set of results for the three traffic loads are presented below.

3.7.4.1 Goodput and Delay Performance at Traffic Load 384 kbps

The goodput is the data rate of successful transmitted that is received by the user while the delay is the difference between the time at which the packet was received and the packet arrived the queue. The average goodput result as shown in Fig. 3.7 shows that the average goodput performance starts to increase from margin 0.0 dB till it gets to margin 1.0 dB where it reaches its peak and from there onwards, it starts to decrease till it gets to the last margin considered for this work which is margin 4.0 dB. The same set of observation can be said for average delay results as presented in Fig. 3.8, where the average delay performance decreases from margin 0.0 dB and reaches the minimum at margin 1.0 dB before it start to increase until it gets to the last margin considered for this work which is margin 4.0 dB. This shows that at this traffic load, the best margin is 1.0 dB.

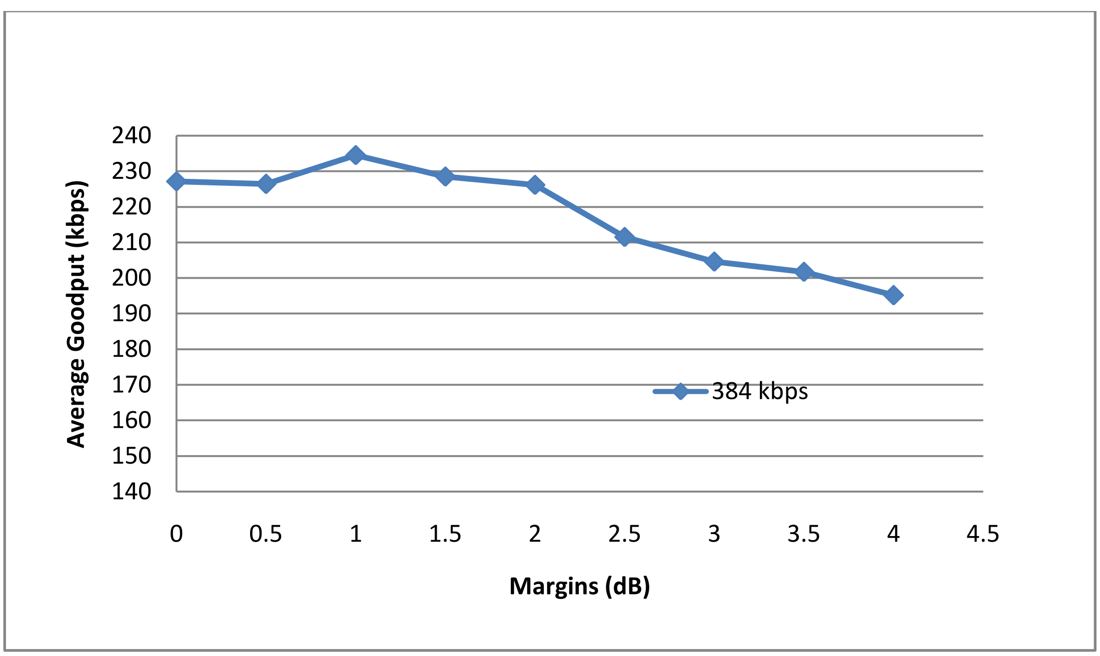

Figure 3.7 Average goodput for varying margins at 384 kbps

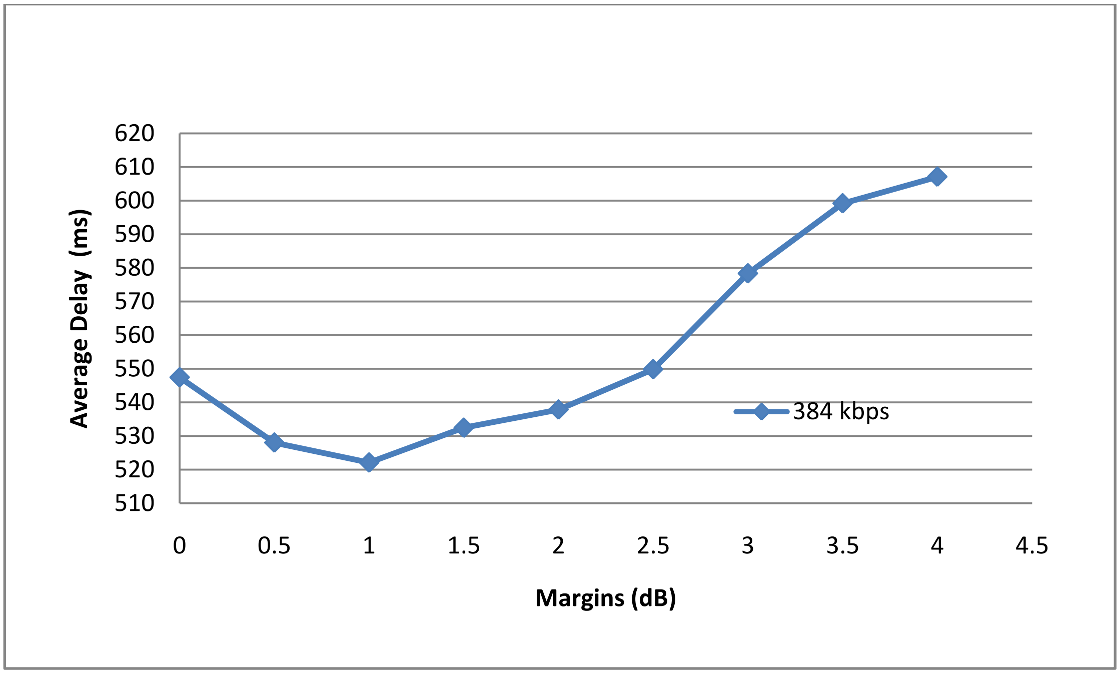

Figure 3.8 Average delay for varying margins at 384 kbps

3.7.4.2 Goodput and Delay Performance at Traffic Load of 512 kbps

For this traffic load, there is increment of the average goodput performance from margin 0.0 dB till it reaches its maximum at margin 1.5 dB as shown in Fig. 3.9. From margin 1.5 dB till margin 4.0 dB, the average goodput performance starts to decrease.

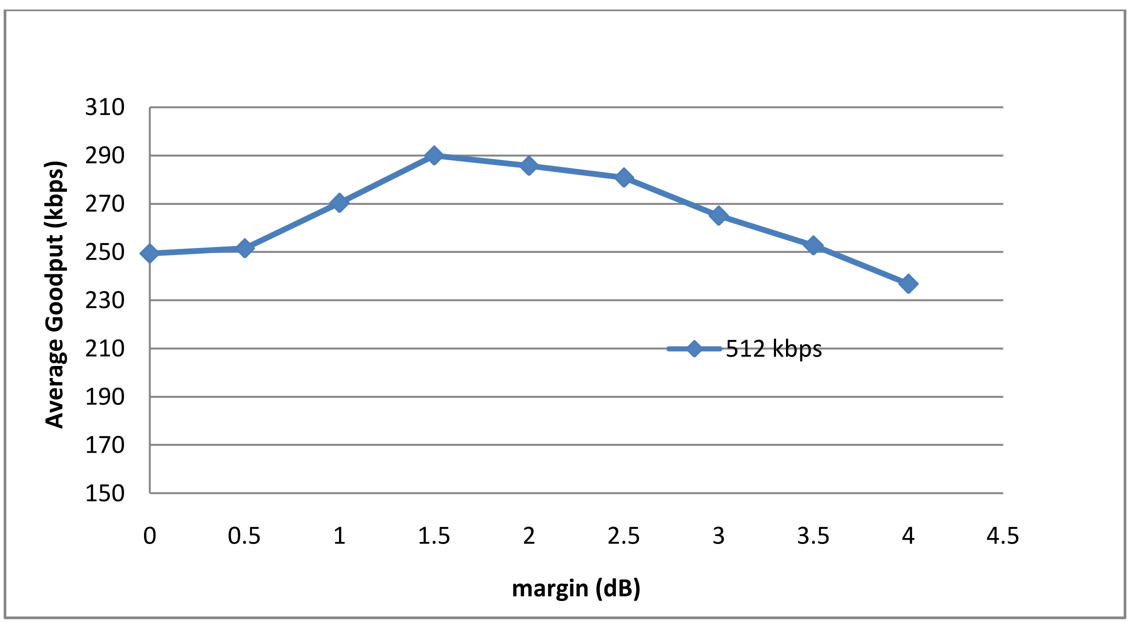

Figure 3.9 Average goodput for varying margins at 512 kbps.

As presented in Fig. 3.10, the average delay performance from margin 0.0 decreases till it gets to margin 1.5 dB and from this margin onwards, the average delay performance starts to increase. It is worth stating that from margin 3.0, there is a higher increment till it reaches margin 4.0. This set of results has similar trends to the set of results obtained for the first traffic load above. The major difference is that the best margin for this traffic load is 1.5 dB since the maximum average goodput and the minimum average delay performance is gotten at this margin.

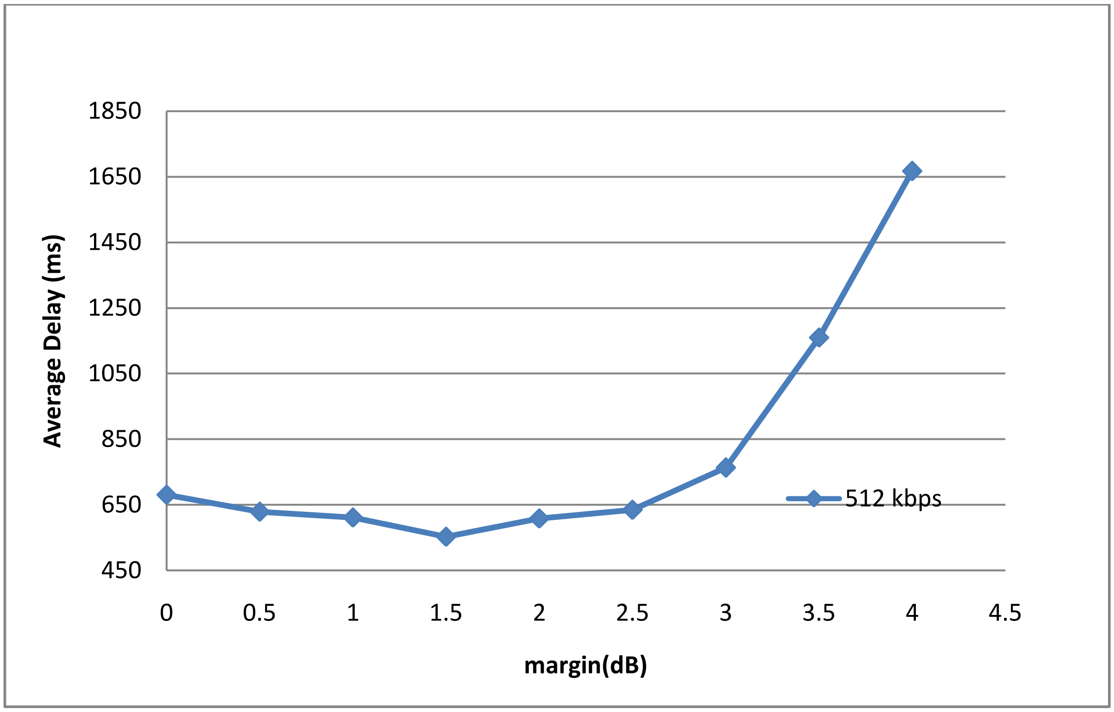

Figure 3.10 Average delay for varying margins at 512 kbps.

3.7.4.3 Goodput and Delay Performance at Traffic Load of 1024 kbps

The final set of simulation results as depicted in Fig. 3.11 and 3.12 shows that the average goodput and delay performances follows the same trend with the results of the other traffic loads. The average good performance just has an increment from margin 0.0 dB to 0.5 dB where it reaches its peak before it starts to decrease. The decrement continues till it gets to the last margin (4.0 dB) considered. In a similar trend, the average delay performance have an decrement from 0.0 dB to 0.5 dB where it reaches its minimum and thereafter, it start to increase till it gets to margin 4.0 dB. This shows that for this traffic load that the best performance is obtained at margin 0.5 dB. From the results obtained for the three different traffic loads, it shows that the best margin varies for different traffic loads.

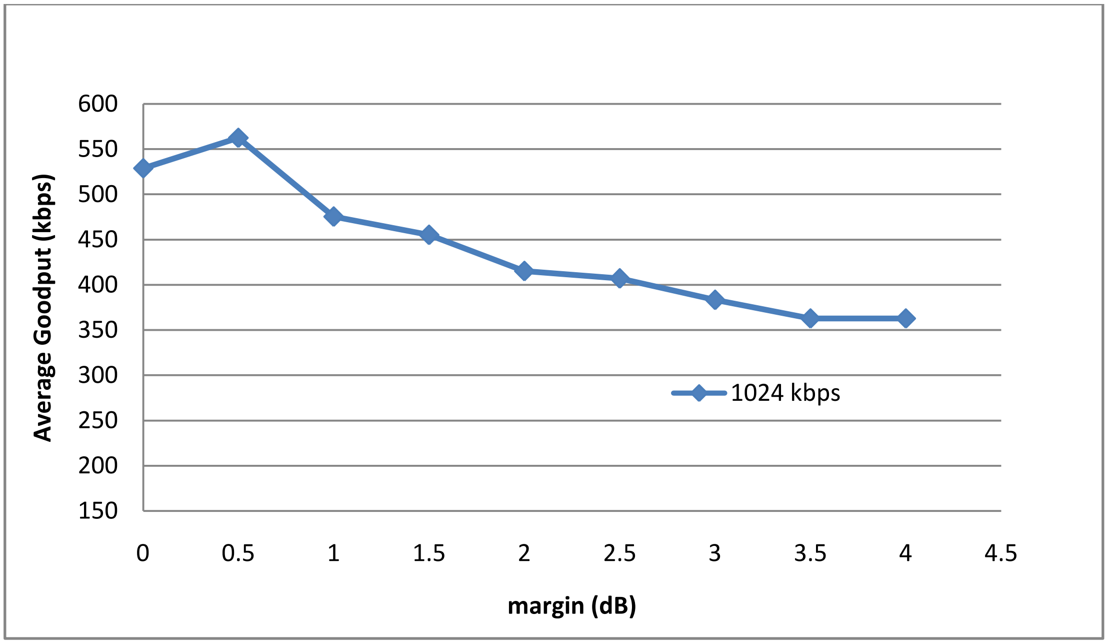

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Figure 3.11 Average goodput for varying margins at 1024 kbps.

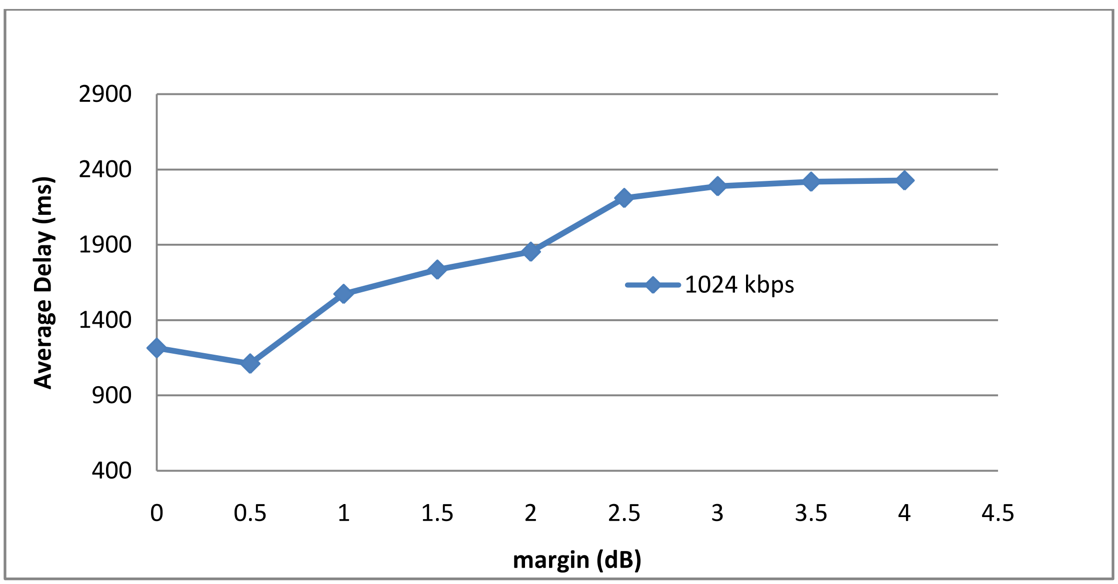

Figure 3.12 Average delayfor varying margins at 1024 kbps.

3.7.5 Semi-Analytical Approach

In order to confirm the simulation results presented above, a semi-analytical approach to determine the best SNR margin, *h*, is carried out. The throughput has been considered as the performance metric, since it is a key index that is used to measure performance in not just satellite communications systems but wireless communications systems as a whole. The best SNR margin, *h*, is obtained by analytically computing the throughput for different margins.

3.7.5.1 Throughput

The throughput computed is an approximate value as proposed in [96]. From [96], the throughput, S, can be theoretically computed for packet system that detect errors and allow retransmissions, as follows;

$$S(BLER(t), CQI(t)) = \left(1 - BLER(t)\right) \cdot Data\_Rate(t) \tag{3.27}$$

Since the data rate is a function of CQI as shown in Table 3.3,

$$Data\_Rate(t) = f(CQI(t)) \tag{3.28}$$

Neglecting the proportionality constant between $Data\_Rate$ and CQI, since the $Data\_Rate$ increases as the CQI increases, the proportionality constant can be said to be linear. The throughput, *S,* can therefore be computed as;

$$S(BLER(t), CQI(t)) = \left(1 - BLER(t)\right) \cdot CQI(t) \tag{3.29}$$

The Δ,which is the difference between instantaneous SNR and the delayed SNR, is computed as;

$$\Delta = SNR(t) - SNR(t - RTPD) \tag{3.30}$$

Hence, the $SNR(t)$ and $SNR(t - RTPD)$ can be expressed as;

$$SNR(t) = SNR(t - RTPD) + \Delta \tag{3.31}$$

And

$$SNR(t - RTPD) = SNR(t) - \Delta \tag{3.32}$$

The CQI computed at the Node B uses the reported SNR which is the delayed SNR that is $SNR(t - RTPD)$. With the introduction of SNR margin, *h,* the CQI can therefore be mathematically computed as a function of $SNR(t - RTPD)$ and *h*. This can be expressed as follows;

$$CQI(t) = f(h, SNR(t - RTPD)) = \begin{pmatrix} 0 & if\ SNR \leq -16 \\ \frac{SNR(t) - \Delta - h}{1.02} + 16.02 & if - 16 \leq SNR \leq 14 \\ 30 & if\ 14 \leq SNR \end{pmatrix} \tag{3.33}$$

Where *h* is the SNR margin. The SNR margin considered for this work varies from 0.0 to 4.0 dB.

The BLER that is used is derived from (3.24). The BLER is expected to be a function of the actual SNR that is $SNR(t)$ and the $CQI(t)$ which is computed using the delayed SNR, $SNR(t - RTPD)$. The BLER can therefore be expressed mathematically as;

$$BLER(t) = f(SNR(t), CQI(t)) = f(SNR(t), \Delta) = \left[10^{2^{\frac{SNR\,(t)-1.03CQI\,(t)+17.3}{\sqrt{3}-\log_{10} CQI\,(t)}}} + 1\right]^{-\frac{1}{0.7}} \qquad (3.34)$$

For the purpose of this analytical work, the throughput is computed over wide range of Δ values and $SNR(t)$ values which can be deduced from pdf obtained as shown in the pdf plots provided below in Section 3.7.5.3. This is to ensure that a more accurate throughput is computed for all margins.

The throughput S can be said to be a function of h, Δ and $SNR(t)$, that is;

$$S = f(h, \Delta, SNR(t)) \qquad (3.35)$$

To obtain an average *S(h),* there is need to average the equation (3.29) with respect to $SNR(t)$ and Δ. The throughput, *S*, can then be further expressed as;

$$S(h) = \iint_{SNR=0,\Delta=0}^{\infty,\infty} \big(1 - BLER(t)\big) \cdot CQI(t) \cdot f(\Delta) \cdot f(SNR) d\Delta\, d(SNR) \qquad (3.36)$$

The details on $f(SNR)$ and $f(\Delta)$ used in computing the throughput, S, are provided in the subsequent Sections.

3.7.5.2 Signal to Noise Ratio (SNR)

Since there is a Round Trip Delay of 560 ms, it is assumed that the SNR used at the Node B is the SNR of the mobile terminal at 560 ms earlier that is $SNR(t - 560)$ which might be different from the actual SNR experienced at time t by the mobile terminal, $SNR(t)$. A trace example of this scenario which portrays the difference between the actual SNR and the delayed SNR can be graphically expressed as follows;

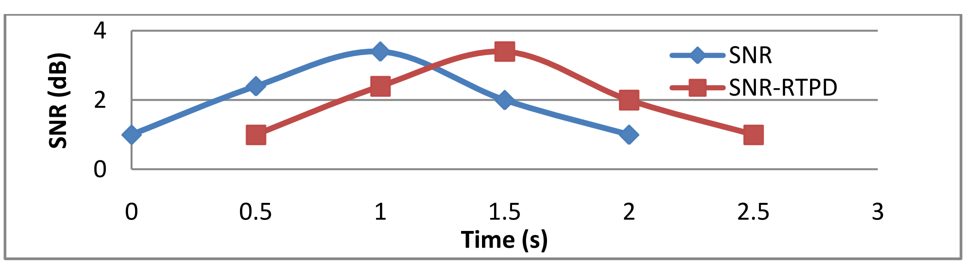

Figure 3.13 Representation of the difference between the SNR of UE and Node B.

$SNR(t)$ is randomly generated. The parameters provided for the suburban area at an angle of 60° in Table 3.2 as used for the simulation is also used for the analytical purpose. The $SNR(t)$ values are assumed to be the SNR experienced by the mobile terminal at every time t. The set of SNR used by the Node B are therefore assumed to be the SNR experienced by the mobile terminal, 560 ms before time *t* at every point in time. The convolution of the two SNR functions is obtained and can be expressed mathematically as;

$$f(\Delta\,) = f(SNR(t)) * f(SNR(t - RTPD)) \qquad (3.38)$$

The pdf plots of the two functions $f(SNR)$ and $f(\Delta)$ are presented in the next subsection which is the results section.

3.7.5.3 Semi-Analytical Results

PDF Plot

The probability density function (pdf) of $SNR(t)$ and Δ is computed using MATLAB. The plot for the pdf of $SNR(t)$ of the parameters provided in Table 3.2 at an elevation angle of 60° in the suburban area is presented below.

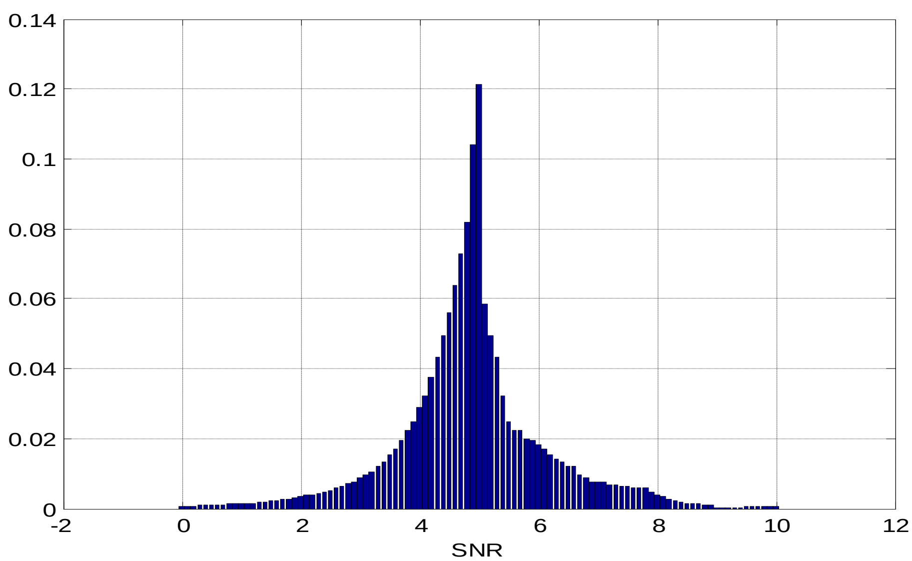

Figure 3.14 The pdf of SNR.

The plot of the pdf of Δ which is the convolution of $SNR(t)$ and $SNR(t - RTPD)$ is also provided as follows;

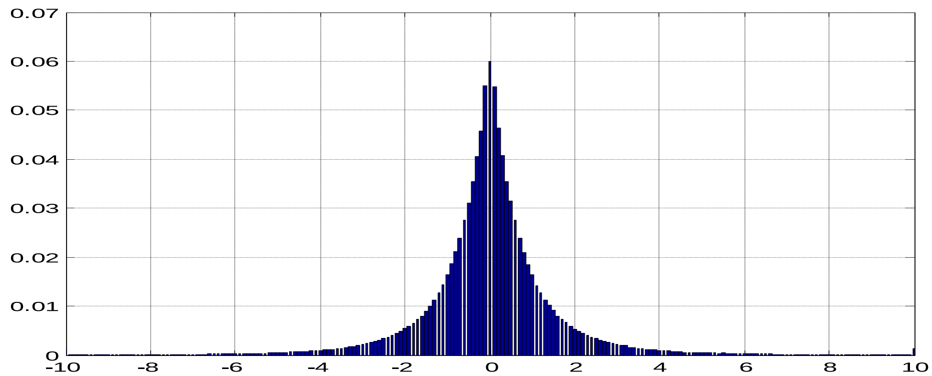

Figure 3.15 The pdf of convolution of $SNR(t)$ and $SNR(t - RTPD)$

Throughput

The computation of the throughput is obtained using C++ programming language. The developed iterative code will compute and sum the throughput for all values of Δ and then for all values of $SNR(t)$ for each margin. The margins considered ranges from 0.0 to 4.0 dB in step size of 0.5. The result obtained from this analytical work is presented below in Fig. 3.16.

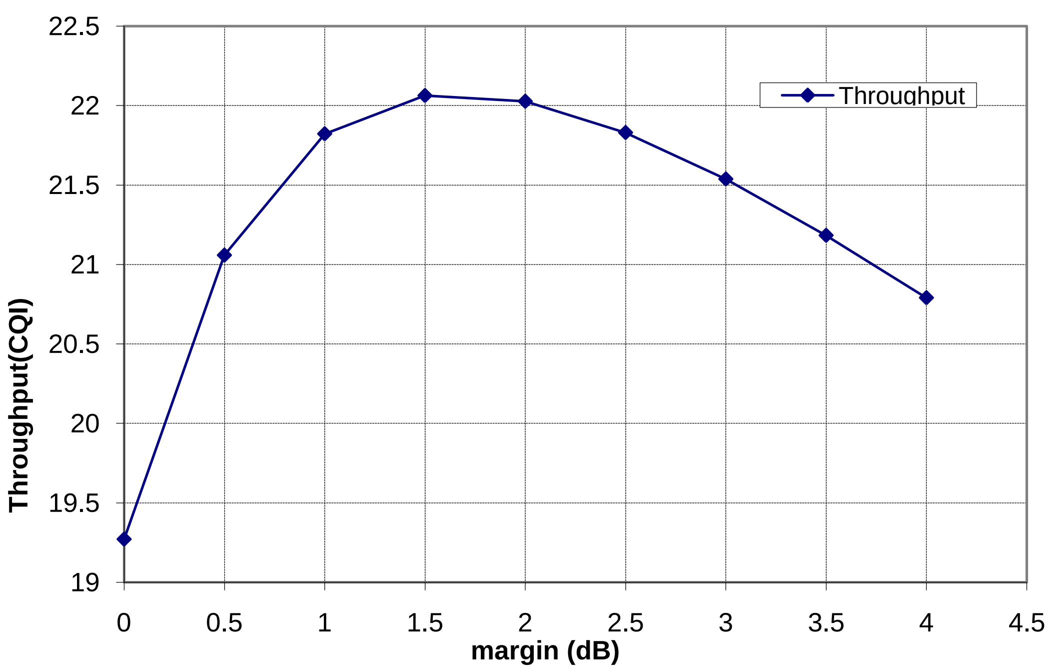

Figure 3.16 Analytical throughput for varying margins

The result as shown in Fig. 3.16, shows that the throughput increases from margin 0.0 dB to margin 1.5 dB when it reaches its peak, it then begins to decrease as the margin increases. From this result, the margin 1.5 dB provides the best throughput and can therefore be said to be the optimal margin from the semi-analytical approach. This result confirms the simulation results obtained for the traffic load where the data rate is 512 kbps as presented above in previous subsections.

3.8 Summary

This chapter has presented an overview of channel modeling in land mobile satellite communications and given detailed explanation of the widely used two and three state Markov channel models for research work. The state and transition probabilities matrices used to determine the channel state at every instance are presented as well. The link budget analysis used to compute the SNR and the formula used for CQI estimation in the S-HSDPA systems are presented.

In the rest of this chapter, the simulation set-up for the investigation of the effect of SNR margin on the performance of S-HSDPA which includes the architecture, the traffic model and channel model is introduced. The goodput and delay performance results from simulation that is carried out are presented and discussed. Finally, the semi-analytical work and results on the effect of SNR margin on the performance of S-HSDPA is presented.

The simulation models, set-up and the results have been discussed in this chapter. Also, the semi-analytical work and results have been presented in this chapter. The results shows that the introduction of margin to compensate for the long propagation delay has an impact on the performance of S-HSDPA systems and that there is a need to select the appropriate margin to compensate for the propagation delay, otherwise, it will affects the performance.

The best margins obtained from the simulation results for traffic loads 384, 512 and 1024 kbps are 1.0, 1.5 and 0.5 dB respectively. The simulation results therefore show that the best margin varies as the traffic load varies since the best margins obtained for each traffic load are not the same. It can be said that the change in arrival rates of the traffic will lead to a change in the best margin. However, from the semi-analytical work carried out which is not traffic dependent, it shows that the best margin is 1.5 dB. Thus, it can be concluded that the optimal margin is 1.5 dB since this can be deduced from the semi-analytical results and the results obtained from the simulation for the traffic load of 512 kbps. It is also worthy of note that 1.5 dB will not necessarily give the best performance for all traffic loads since the best margin varies from one traffic load to another as obtained from the simulation results.

Chapter 4

SIMULATION OF PACKET SCHEDULING SCHEMES IN S-HSDPA

4.1 Introduction

The aim of this chapter is to present a new scheduler tagged Queue Aware Channel Based (QACB) scheduler for Satellite HSDPA networks that will provide a good trade-off between resource utilization, degree of fairness and an acceptable QoS taking into consideration the available constraints in order to improve the overall performance. In order to evaluate the performance of the newly proposed scheduler, a simulation setup is used to compare various schedulers' performance in Satellite HSDPA networks with the newly proposed scheduler and the simulation results are presented. The throughput performance, delay performance and the fairness for the four schedulers considered are measured. The schedulers considered are Proportional Fair (PF), Earliest Deadline First (EDF), Modified Largest Weighted Delay First (M-LWDF) and QACB.

This chapter starts with an overview of scheduling in S-HSDPA networks, this is followed by the discussion of the four schedulers used in this simulation. The simulation setup and the accompanying channel and traffic models are presented. Finally, the simulation results including throughput, delay and fairness are presented and discussed.

The work in this chapter has been presented at the 9th Institute of Electrical and Electronics Engineers AFRICON Conference (IEEE AFRICON 2009), Nairobi, Kenya.

4.2 Packet Scheduling in S-HSDPA

The packet scheduling function is located in HSDPA specific MAC layer called MAC-hs at Node B in the S-HSDPA system architecture provided in Fig. 4.1. This makes packet scheduling process faster since it is closer to the mobile user as compared to being situated at the RNC. At the MAC-hs level, each user has its own queue and each queue is served based on the scheduling decisions made by the scheduler using either channel state information or QoS criterion or both. It is assumed that only one user can be scheduled at every TTI. The scheduler is responsible for the allocation of radio resources to mobile users and does this by selecting the mobile user that will transmit at every TTI based on its scheduling algorithm. It then allocates the selected mobile user resources at certain transmission rate which corresponds to the Adaptive Modulation and Coding (AMC) level. It is worthy to note that only one AMC level can be used at each TTI and that there are only sixteen (16) orthogonal spreading codes that are available, fifteen (15) for downlink transmissions and the remaining one (1) for signaling purposes [97]. The adequate AMC level is selected by Node B based on the mobile user's channel condition at that TTI. The channel condition status is determined using the CQI value that is sent by the mobile user at certain reporting intervals as requested by the RNC. The CQI contains Transport Format and Resource Combination (TFRC) mode, that is, the coding and modulation scheme to be used, the numbers of parallel codes that can be used for transmission and the Transport Block Size (TBS). The corresponding values for each CQI value are provided in Table 3.3. This CQI values are also used by channel based schedulers in making scheduling decisions. This concept is explained in the following section where details of each scheduler are discussed.

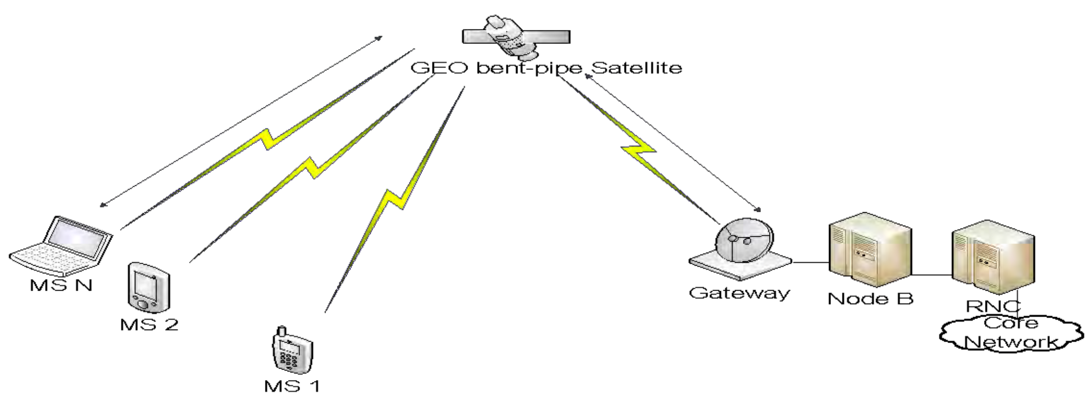

Figure 4.1 System architecture of S-HSDPA

4.3 Schedulers Considered

In S-HSDPA, an effective packet scheduler is needed to achieve high resource utilization and acceptable Quality of Service (QoS) for different UMTS traffic types such as conservational voice traffic, video streaming, interactive web traffic and background FTP download requested by different mobile users over varying channel conditions. The dynamic nature of the network and the long propagation delay experienced by mobile users over a GEO Satellite, which causes misalignment between the instantaneous Channel State Information (CSI) experienced by the mobile user and the last reported CSI, makes the design of an effective packet scheduler a major challenge.

4.3.1 Proportional Fair (PF)

The Proportional Fair (PF) scheduler is a channel based scheduler that is designed to address the problem of fairness among mobile users experiencing both good and bad channel conditions without compromising the overall throughput. It serves the user with the maximum Relative Channel Quality Indicator (RCQI). This is defined as presented in [80] [81] for S-HSDPA, as the ratio between the present maximum data rate supported by the User Equipment (UE) based on the CQI value and the average data rate of the UE's previous transmission till the present TTI. It therefore uses the following algorithm;

$$j = \max_k \left\{\frac{R_k[n]}{T_k[n]}\right\} \tag{4.1}$$

Where $$T_k[n] = \frac{\sum_{m=1}^{n} D_k(m)}{n \times TTI} \tag{4.2}$$

Where k is the UE index, n is the number of TTI units, $R_k[n]$ is the maximum allowable data rate by UE k, based on its CQI value (instantaneous data rate), $T_k[n]$ is the average data rate of UE k, up to the present TTI , $D_k(m)$ is the data rate for each previous TTI and j is the priority index. The user with the maximum priority index is scheduled. Based on the reported CQI value by the user, the lookup table presented in Table 3.3 is checked to determine the corresponding modulation and coding scheme to be used and also to determine the corresponding maximum allowable data rate. The setback of this scheduler is that it doesn't take into account QoS factors due to the fact that it cannot differentiated between different traffic types, therefore it is not suitable for Real Time (RT) services which are sensitive to delay especially in mixed traffic scenarios. Further details of this scheduler can be found in [98] [99].

4.3.2 Earliest Deadline First (EDF)

The EDF scheduler is a delay-sensitive scheduler and does not put channel conditions into consideration in making scheduling decisions. Each user's queue has a set deadline depending on the type of traffic. The scheduler then serves the oldest waiting packet in the queue with respect to its packet deadline, therefore, the packet with lowest residual lifetime in the queue in relation to its deadline is given the highest priority. So, users are served by increasing order of their deadline. The algorithm is defined as stated in [80] [97] [81] as the ratio of the waiting time of the oldest packet in the UE's queue to the packet deadline.

$$j = \max_k \left\{ \frac{d_k[n]}{T_{k,dea\ dline}} \right\} \tag{4.3}$$

Where $d_k[n]$ is the waiting time of the Head of Line (HOL) packet in user queue k at current TTI and $T_{k,deadline}$ is the delay deadline for the packet, this varies depending on the traffic type. In this dissertation, it is assumed that the $T_{k,deadline}$ for video packet is 160 ms while the $T_{k,deadline}$ for a web packet is 500 ms. The user with the maximum priority index, j, is scheduled. It is worthy to note that the user selected will transmit at the data rate corresponding to its present CQI value using the lookup table in Table 3.3. The Setback of EDF scheduler is that it doesn't take the channel conditions into consideration, therefore not taking into account the available capacity, which could lead to under utilization of resources. It might also be starved non-real time packets like web packets even when they have good channel conditions.

4.3.3 Modified Largest Weighted Delay First (M-LWDF)

The M-LWDF scheduler considers both channel conditions and the delay experienced by user's packet while making scheduling decisions. It provides good QoS performance with each user having its own probabilistic QoS requirement of the form as stated in [84] [100];

$$\Pr\ (d_k\ > T_{k,deadlin\ e}) \leq \delta_k \tag{4.4}$$

Where d_k is the waiting time of Head of Line (HOL) packet in user queue k, $T_{k,deadline}$ is the delay deadline for user k and δ_k is the probability of exceeding the delay deadline. The scheduler decides the next user to be allocated resources by using the following algorithm as presented in [83] [101] for HSDPA systems:

$$j = \max_k \left\{ \frac{\gamma_k d_k[n]}{R_k[n]} \right\} \tag{4.5}$$

$$\text{Where} \quad \gamma_k = a_k T_k[n] \tag{4.6}$$

$$\text{And} \quad a_k = \frac{-\log \delta_k}{T_{k,deadline}} \tag{4.7}$$

The algorithm can therefore be rewritten as follows;

$$j = \max_k \left\{ \frac{d_k[n]}{T_{k,deadline}} \ (-\log \delta_k) \frac{T_k[n]}{R_k[n]} \right\} \tag{4.8}$$

$R_k[n]$, $T_k[n]$ and $T_{k,dea\,dline}$ have the same meaning as stated in PF and EDF schedulers. The term a_k is used for QoS differentiation where δ_k varies based on the priority of the traffic been demanded by the user. The values of 0.01 and 0.1 are used to represent δ_k for video traffic and δ_k for web traffic respectively for the purpose of this dissertation. The user with the maximum priority index, j, is scheduled.

Since the waiting time, $d_k[n]$ has a significant impact on the scheduling decision of M-LWDF scheduler, packets with less $d_k[n]$ or high delay deadline (non real time packets e.g. web packets) will have to wait for long and may be starved in the process. Even if the number of non-real time packets in the queue is getting close to maximum queue length depending on the arrival rate, since they have high delay deadline as compared to real time traffic, real time packets are still given priority.

From the explanations above, it can be said that while PF scheduler provides to a large extent, a high capacity utilization and fairness, its inability to differentiate traffic types causes a major setback in providing QoS to varying users. The EDF scheduler is said to use the order of deadline, to differentiate traffic types by giving priority to delay sensitive traffic without any considerations to traffic users with good channel conditions, thereby underutilizing the available resources. The M-LWDF scheduler addresses the major shortcomings of these two schedulers by considering both the channel conditions and the QoS differentiating mechanism. The issue with M-LWDF is that the QoS differentiating mechanism only considers waiting time with respect to deadline. This will continuously give preference to delay sensitive traffic that is given smaller deadline. The need to extend the QoS differentiating factor to also consider the varying queuing length with respect to the maximum length of the queue has formed the basis of the introduction of the new scheduling scheme. So that both delay and non delay sensitive traffic with higher arrival rates that could result to the queue being filled up, will be taking into consideration. Therefore, queue congestion can be avoided and capacity also being utilized.

4.3.4 Queue Aware Channel Based (QACB) – Newly Proposed Scheduler

The proposed scheduler tagged QACB has been designed to improve on M-LWDF scheduler. This ensures that both the delay experienced with respect to the set deadline and the varying queue length of each user‘s queue, are both considered simultaneously in taking scheduling decisions. An algorithm that considers the ratio of the instantaneous queue length to the maximum length of the queue has been added to the existing M-LWDF algorithm. There is a proportional factor included in this algorithm to differentiate between traffic queues based on sensitivity to packet drop.

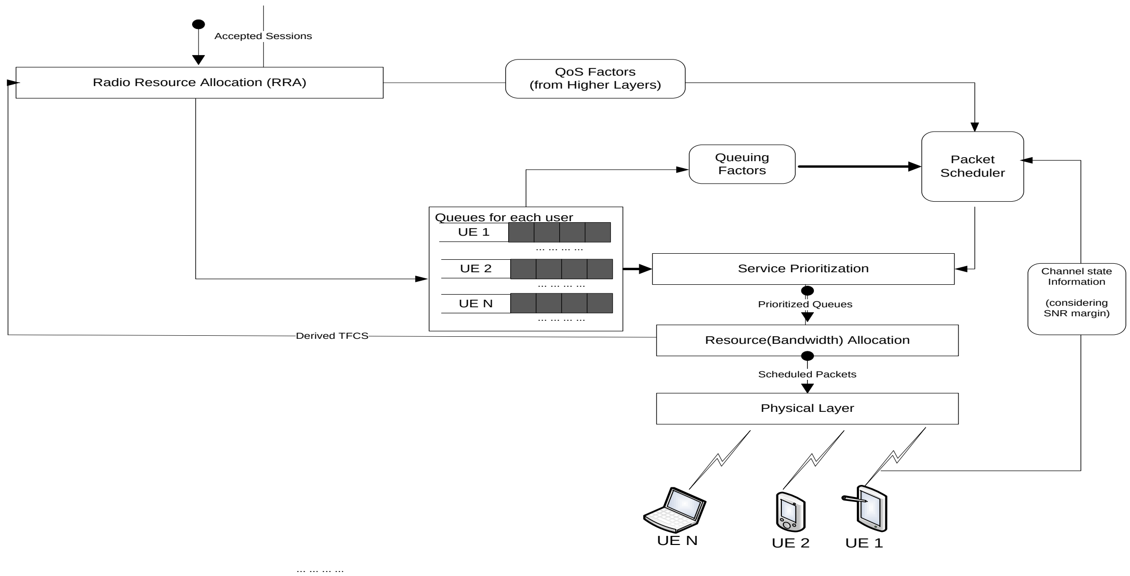

Figure 4.2 Proposed Scheduler Model

As shown in Fig. 4.2, the scheduler considers the application layer (using the time deadline and proportionality factor to differentiate traffic), the MAC layer (using queuing information like waiting time and queue length) and physical layer (CQI). This will increase the throughput experienced by Non Real Time (NRT) traffic users and also improve the level of fairness among users with different traffic types. The algorithm used to schedule the user is stated as follows;

$$j = \max_k \left\{\frac{T_k[n]}{R_k[n]} a_k d_k[n]\right\}\left\{\frac{L_K[n]}{\sigma L_{queue}}\right\} \tag{4.9}$$

$$a_k = \frac{-\log \delta_k}{T_{k,deadline}} \tag{4.10}$$

The algorithm can then be fully rewritten as follows;

$$j = \max_k \left\{\frac{T_k[n]}{R_k[n]} (-\log \delta_k) \frac{d_k[n]}{T_{k,deadline}} \frac{L_k[n]}{\sigma L_{qu\,eue}}\right\} \tag{4.11}$$

Where $L_k[n]$ is the instantaneous queue length, L_{queue} is the maximum length of queue and σ is the proportional factor for the queue length based on packet drop sensitivity. For this work, we have assumed 0.8 and 1 values for the proportional factors of video and web packets respectively. The $L_k[n]$ is deduced from the numbers of packets in the queue. All other terms have the same meaning as stated in other schedulers. The user with the maximum priority index, j, is scheduled.

4.4 Channel Model

The three state Markov channel model that was discussed in section 3.3 is used. The corresponding state and transition probabilities matrices for suburban area in Table 3.1 are used to predict the channel state of every user for every CQI reporting interval (20 TTI). The SNR at that channel state is then determined using (3.17) presented in chapter 3. The corresponding Loo parameters for this state are used to determine the SNR. The angle of elevation of 60° is considered. The corresponding CQI of the SNR is then determined using (3.20) in chapter 3. The NS-2/EURANE uses a preprocessed channel trace input file (physical layer parameters) during simulation, so the SNR and the corresponding CQI which constitutes the channel trace input files are generated using MATLAB script. The preprocessed input file is then imported to NS-2 for the simulation process.

4.5 Traffic Model

It is assumed that there are only two different traffic demands under the GEO satellite spot beam. They are the MPEG-4 video streamers and web traffic browsers.

4.5.1 MPEG-4 Video Traffic Model

The Transform-Expand-Sample (TES) model for a Variable Bit Rate (VBR) MPEG-4-coded video streaming traffic was used. The TES modeling procedure is made up of two stages. The first stage entails each frame type (I, P and B) being modeled by a TES process and the second stage involves interleaving the three TES models in the right order [102].The I frames (intra-coded picture), P frames (Predicted picture) and B frames (Bi-predictive picture) are used for video compression. The three different frames, I frame, P frame and B frame are generated in three different TES models [103]. The MPEG-4 video model is set to generate twenty five (25) frames every second. Since the NS-2/EURANE does not have this model by default, the model is integrated to the NS-2/EURANE simulation software and recompiled for the purpose of this work. The mpeg4_traffic.cc file is stored in the folder containing .cc files in NS-2 and recompiled. The I,P and B files for the video sample used are stored in a folder called Video model and stored in folder where the simulation will be ran. See Appendix A1 for more details on video traffic model.

4.5.2 Web Traffic Model

The Web traffic is modeled using M/Pareto ON/OFF model. Each web browsing download has web file size which is characterized by Pareto distribution with parameters $\alpha = 1.1$, k=81.5 bytes and the maximum allowed packet size m = 66666 bytes [92] [93]. Where α is the Pareto index and k is the minimum allowed packet size. This model is in built to Ns-2 by default. See Appendix A2 for details on web traffic model.

4.6 Simulation Setup

The performance evaluation of the newly proposed scheduler in comparison with the other schedulers is carried out using discrete event simulation software called Network Simulator 2 (NS-2) [94]. Since NS-2 alone doesn‘t support UMTS and HSDPA, Enhanced UMTS Radio Access Network Extension (EURANE) which is a UMTS/HSDPA extension to NS-2 [95] is used on NS-2 platform. As provided in the documentation of EURANE [104], there are only three schedulers (Round robin, C/I scheduling and Fair Channel Dependent Scheduling) that are provided in the EURANE module, so additional schedulers including the newly proposed scheduler are added to its source code and recompiled. This is achieved using the guidelines provided in [105]. The newly introduced schedulers to NS-2 which includes Max CQI, PF, EDF, P-EDF, M-LWDF and QACB schedulers are added into the hsdpalink.cc file as functions in c++ programming language and the hsdpalink.h header file is also edited in order to adjust to the modifications made in the hsdpalink.cc file. The schedulers are also included in the nsdefault.cc file so as to be able to call it during simulation. The whole NS-2 is then recompiled after making all the changes. The same simulated network used for the previous simulation in chapter 3 was also used for this simulation setup as seen in Fig. 4.3, where the mobile terminals are connected to Node B via the GEO bent-pipe Satellite and Node B is connected to Wired Node 1 through RNC, Serving GPRS Support Node (SGSN), Gateway GPRS Support Node (GGSN) and Wired Node 2.

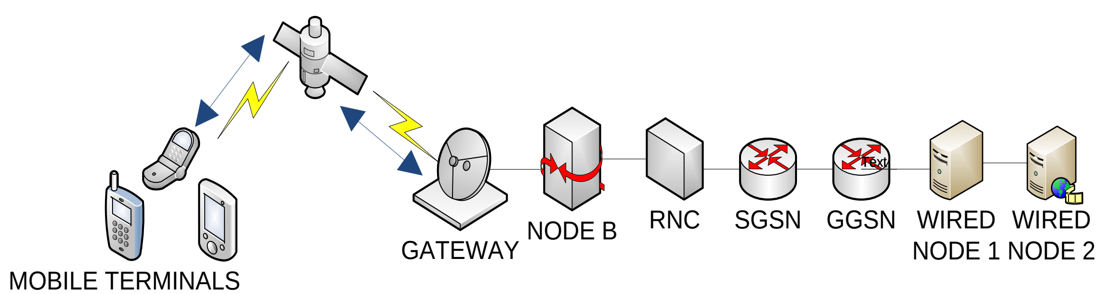

Figure 4.3 S-HSDPA Simulated Network

The Transmission Time Interval (TTI) is 2ms and the CQI reporting interval is assumed to be 40 ms (20 TTI). Only one user can be allocated resources at every TTI for all schedulers considered. For the purpose of this simulation study, ten mobile users made up of five video streamers and five web traffic users have been considered. Simulation details are provided in Table 4.1.

Table 4.1 Details of simulation parameters

Parameter	Values
Simulation Time	500s
Numbers of Video Users	5
Numbers of Web Users	5
Transmission Time Interval (TTI)	2 ms
Channel Model	three state Markov channel model
CQI reporting interval	40 ms (20 TTI)
Channel Delay	280 ms
Video Traffic Model	MPEG-4 TES Model
Web Traffic Model	M/Pareto ON-OFF Model
RLC Mode	Acknowledged Mode (AM)
UE Category	5 & 6

4.7 Simulation Results for two state Markov channel model

The three state Markov channel model have been considered for this work since it provides a more realistic channel scenario as compared to two state Markov channel model. But in order to validate the results obtained for the three state Markov channel model, a set of results that uses the two state Markov channel model are presented. This is due to the fact that the previous work done on comparison of scheduling algorithms in S-HSDPA was done using two state Markov channel model [21]. The two state Markov channel considered in this dissertation is the same as the one used in previous literature but with different traffic model [21]. It uses the GOOD/BAD Markovian channel model and assumes that the good state has a CQI of 25 while the bad state has a CQI of 15. The mean sojourn time of the good and bad state is 6s and 2s respectively. The schedulers considered are the ones considered in [21] which are the PF, EDF and Prioritized EDF (P-EDF). The prioritized EDF uses EDF for real time traffic (video) and PF for non real time traffic (web). The newly proposed scheduler, QACB is also considered. The throughput and delay performance results are presented. The 3 traffic loads are considered which are stated below.

Table 4.2 Traffic load representation for two state Markov channel model

TRAFFIC LOAD	1	2	3
VIDEO TRAFFIC (rate factor)	0.5	1.0	1.5
WEB TRAFFIC (Kbps)	384	512	768

4.7.1 Throughput

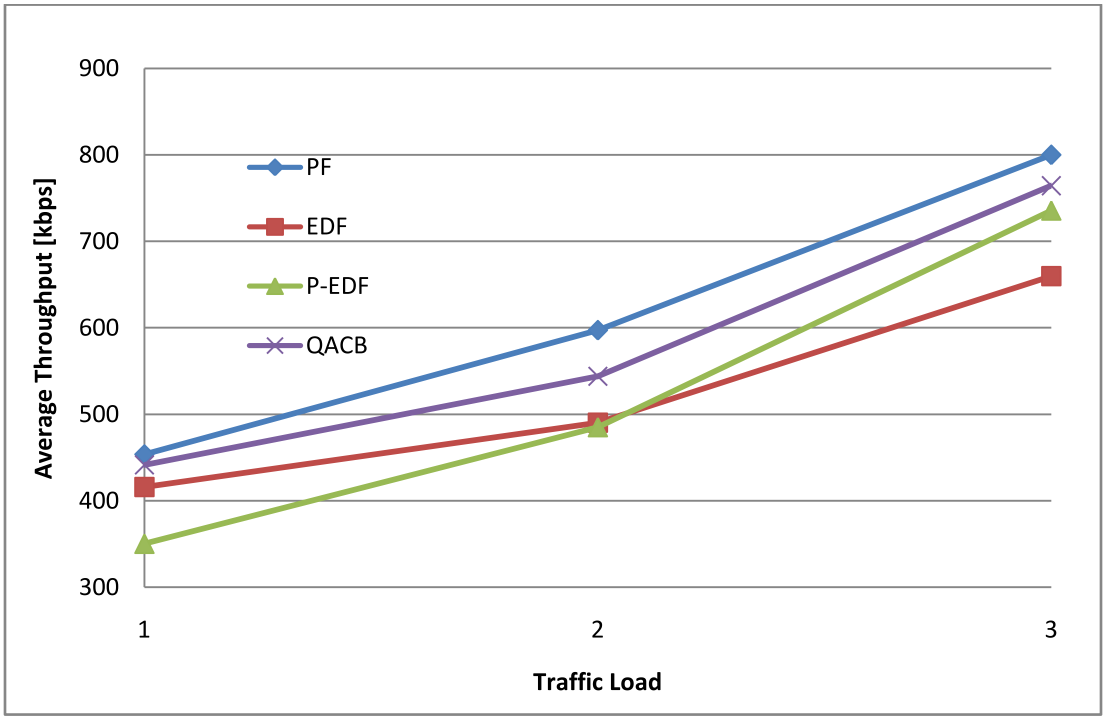

Figure 4.4 Average throughput of web packets for the two state channel

The throughput performance for the web packets presented in Fig. 4.4 shows that the PF produces the best performance, this is due to the fact the PF scheduler allocates to the user with the best channel state. The proposed QACB scheduler produces a better throughput performance as compared to EDF and P-EDF scheduler. Since, it considers the varying packet length in taking scheduling decisions. The P-EDF produces a better performance compared to EDF since it considers the channel status for web traffic as against the EDF which only uses the time spent in the queue with respect to the time deadline for both traffic types.

4.7.2 Delay

The PF scheduler produces the worst delay performance for video traffic and as shown in Fig. 4.5. Expectedly, the EDF produce the best delay performance for video traffic followed by P-EDF. While QACB produces a better delay performance than PF for video traffic and not too compromising performance as compared to EDF and P-EDF schedulers.

The web delay performance presented in Fig. 4.6 shows that P-EDF has a worse delay than EDF scheduler. This conforms to the delay performance for web traffic presented in [21] where a two state Markov channel model was used. While the PF scheduler produces the best delay performance for web traffic at the expense of video traffic due to its inability to differentiate traffic types. The QACB scheduler produces a better web delay performance (lower delay) compared to EDF and P-EDF scheduler.

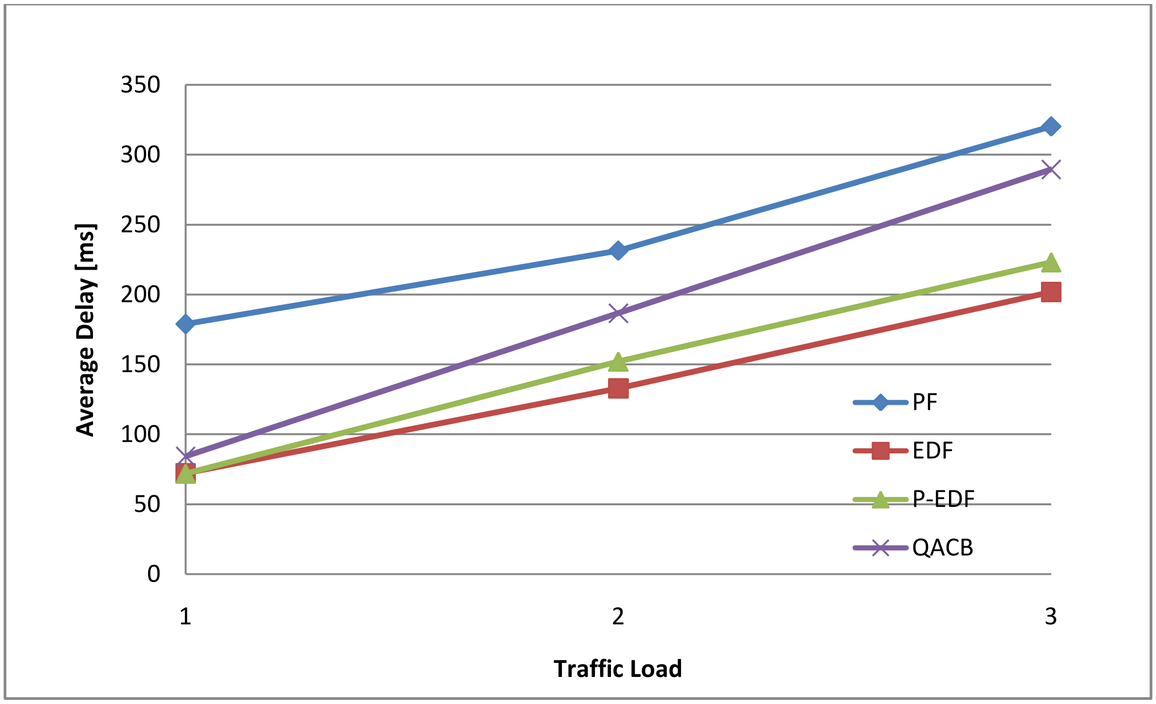

Figure 4.5 Average delay of video packets for the two state channel

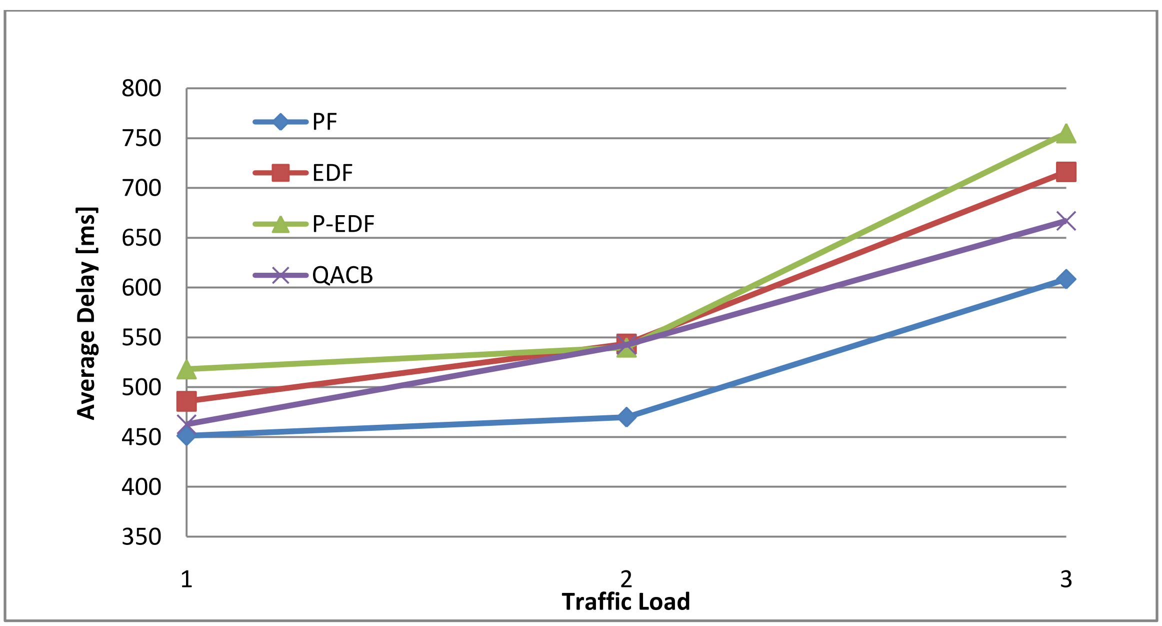

Figure 4.6 Average delay of web packets for the two state channel

4.8 Simulation Results for three state Markov channel model

The simulation results for the three state Markov channel model are presented in three parts. They are the throughput performance, delay performance and the fairness. The throughput performances comparison of the various schedulers considered is presented, their delay performances comparison is then presented and finally, the fairness using Jain fairness index is presented. The simulations that are implemented produce huge event driven data results. To evaluate and analyze this result, the gawk programming language is used and the Microsoft spreadsheet (Excel) is used to plot the graphs.

4.8.1 Throughput

The average throughput which can be referred to as the average number of packets transmitted with respect to time in Kbps. It is computed using gawk script and can be stated as;

$$S_{av.} = \frac{\sum S_i}{T} \tag{4.12}$$

Where $$S_i = N \times Packet\_Size \tag{4.13}$$

$S_{av.}$ is the average throughput, S_i is the instantaneous throughput at time i, T is the time interval considered and N is the numbers of packet transmitted. The traffic load is represented with data rates, so each traffic load depicted at the x axis of each graph presented is represented with a certain data rate. The details of the traffic load representation as recommended in [103] for video traffic and [92] for web traffic are presented below;

Table 4.3 Traffic load representation

TRAFFIC LOAD	1	2	3	4	5	6	7
VIDEO TRAFFIC (rate factor)	0.25	0.5	0.75	1.0	1.5	1.75	2.0
WEB TRAFFIC (Kbps)	8	32	64	128	384	512	768

As shown in Fig. 4.7 and Fig. 4.8 below, the PF scheduler expectedly has a good throughput performance both for web users and the total throughput, since it allocates resources to the user with the best Relative Channel Quality Index (RCQI). The QACB scheduler produces a better throughput performance for web users compared to M-LWDF and EDF schedulers especially at high traffic loads and has a close performance to PF scheduler at low traffic loads, but also produces better performance compared to PF scheduler at high traffic load.

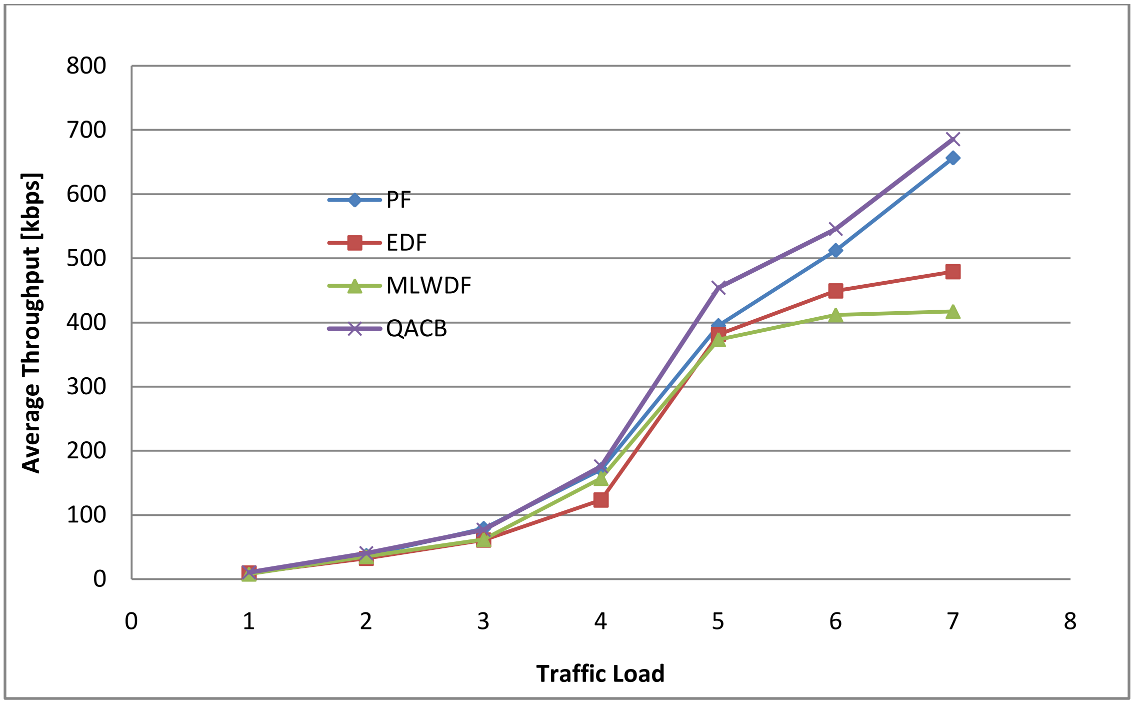

Figure 4.7 Average throughputs of web traffic users

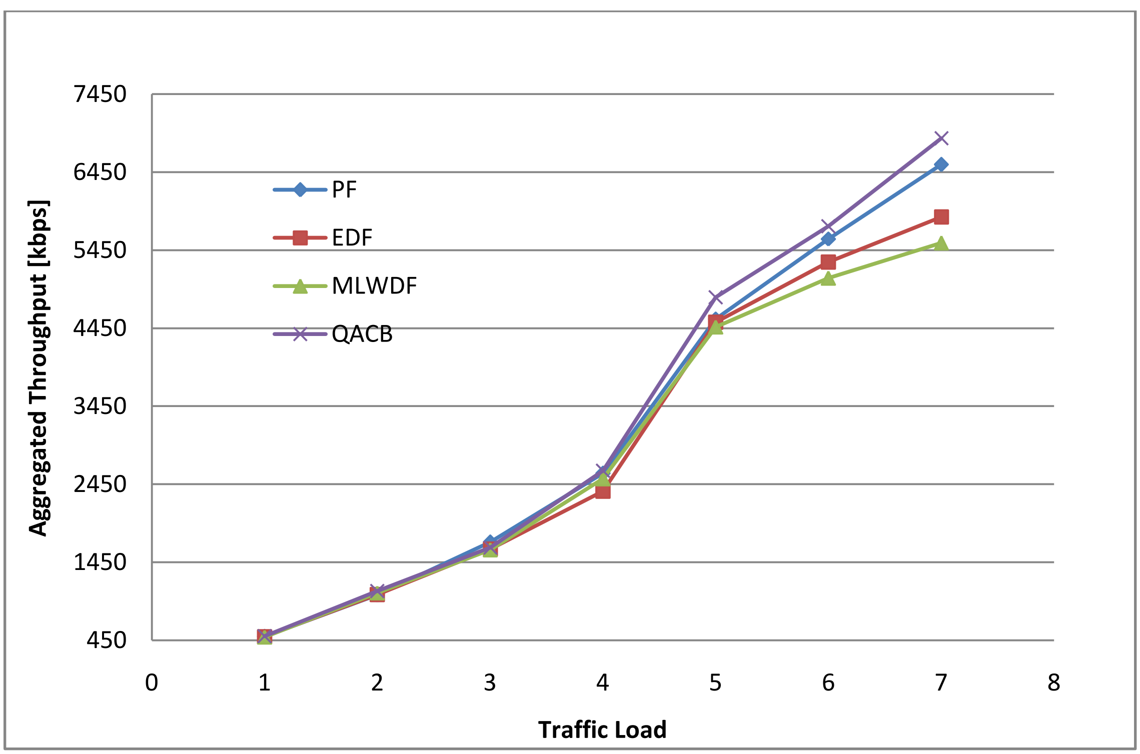

Figure 4.8 Aggregated throughputs of all users

For the total throughput, at lower traffic loads the four schedulers gave a close throughput performance. The proposed QACB scheduler, which considers the varying queue length in taking scheduling decisions, edges other schedulers at high traffic load followed by the PF scheduler.

4.8.2 Delay

The delay is an important performance metric in rating the performance of schedulers especially in a network that contains real time traffic. The average delay in the simulation from the event driven output results is determined as follows;

$$D_{av.} = \frac{\sum D_i}{N} \tag{4.14}$$

$$\text{Where} \quad D_i = T_{r_i} - T_{a_i} \tag{4.15}$$

$D_{av.}$ is the average delay, D_i is the instantaneous delay experienced by packet i, N is the numbers of packets, T_{a_i} is the time of arrival for packet *i* and T_{r_i} is the time packet *i* was received.

The comparisons of average delay experienced by the video and web users of the various schedulers are presented below in Fig. 4.9 and Fig. 4.10 respectively. Expectedly, it is observed that video users experienced the lowest delay (best delay performance) using EDF scheduler, this is due to the fact that EDF scheduler favours real time traffic in mixed traffic scenarios.

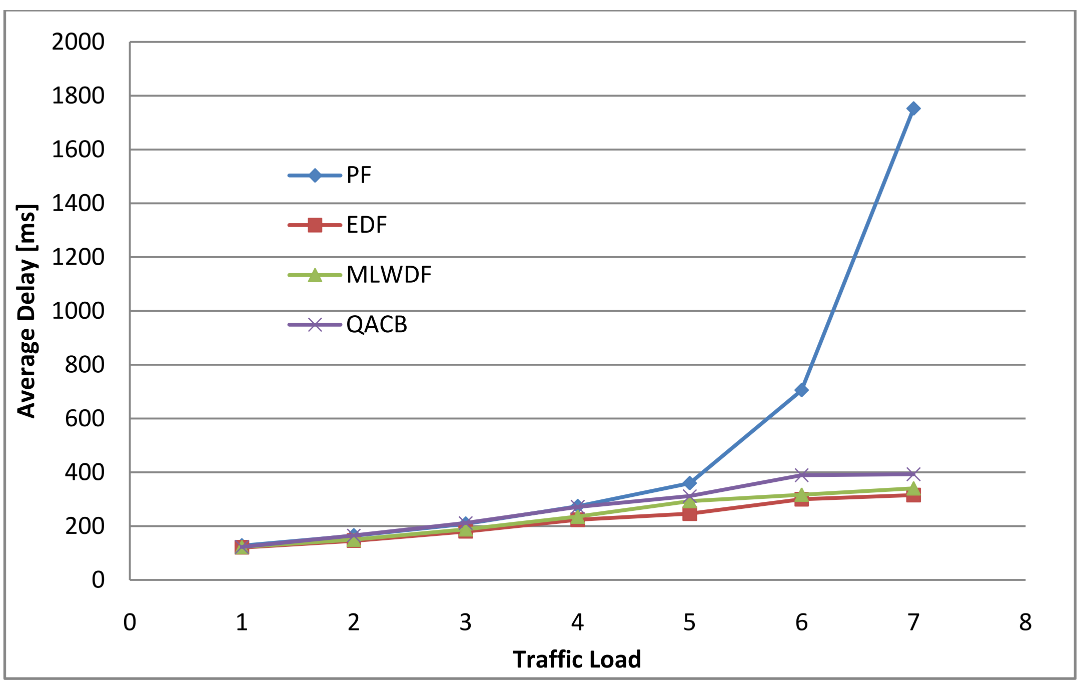

Figure 4.9 Average delay of video traffic users

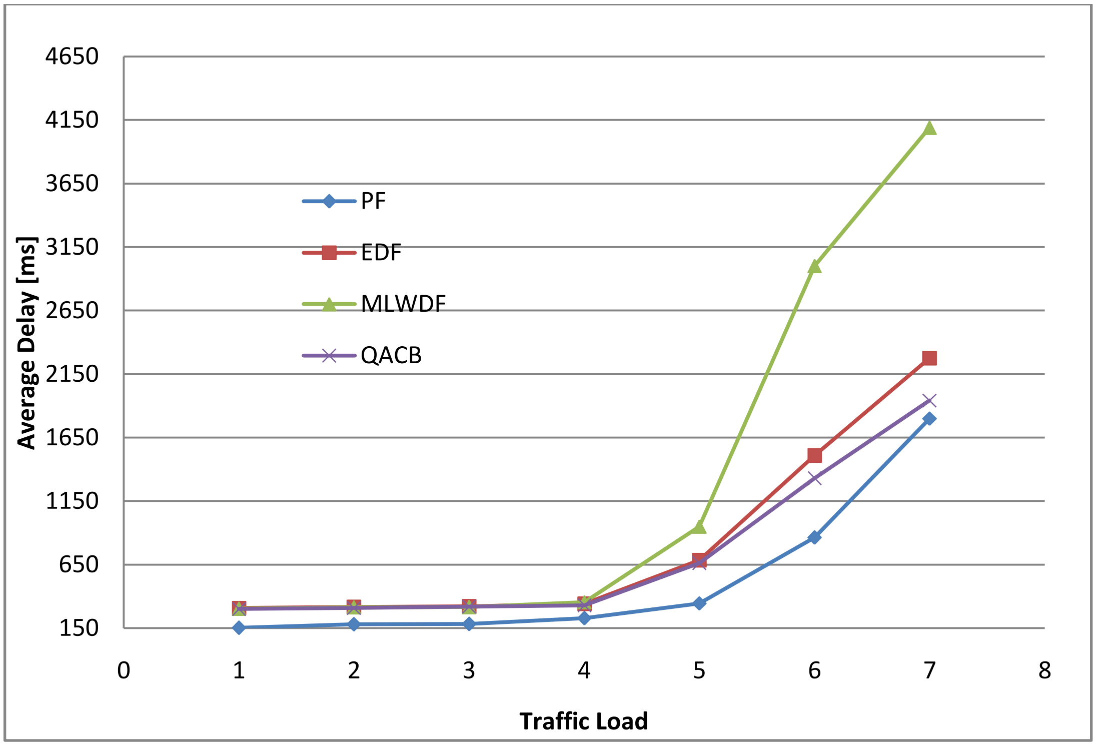

Figure 4.10 Average delay of web traffic users

Since, the PF scheduler can‘t differentiate between traffic types, it is observed that web users experience better delay performance and the video users experienced the worst performance as compared to other schedulers as shown above. This indicates that the PF scheduler produces a good delay performance for NRT traffic (web traffic) at the expense of real time traffic (video traffic). This shows that PF scheduler won‘t be suitable for real time traffic services especially in a mixed traffic scenario.

The M-LWDF produces closer delay performance for video users to EDF and both schedulers produces better performance as compared to the proposed QACB scheduler but the difference in the delay performance between the QACB scheduler and the other two schedulers as shown in Fig. 4.9 is not much. The delay performance for web traffic for the EDF, MLWDF and QACB schedulers at low traffic loads are close but at high traffic load, the delay performance of QACB scheduler produces a better performance than EDF and MLWDF. A worse delay performance of web traffic as the trend shows for MLWDF scheduler in the graph presented above will lead to web packets getting starved in queue. The PF scheduler as mentioned earlier produces the best delay performance for web traffic.

4.8.3 Fairness

The provision of fairness among varying users of different traffic types and under different channel conditions by schedulers is another important measure of performance. Each user, irrespective of its channel conditions and traffic type is expected to receive a fair share of the resources available. The degree of fairness for this simulation is measured using Jain Fairness Index (JFI). The formula used to compute the degree of fairness using JFI is stated as follows;

$$Fairness\ Index = \frac{(\sum S_i)^2}{n \sum S_i^2} \tag{4.16}$$

S_i is the normalized throughput at time i and n is the number of throughput considered.

The fairness index graph is presented above in Fig. 4.11. The fairness index performance for the different schedulers varies with the traffic loads. Considering all users, either PF or QACB schedulers produces the highest degree of fairness as shown in Fig. 4.11 for all the traffic loads. The EDF scheduler produces the lowest degree of fairness in most of the traffic load cases as compared to other schedulers. The M-LWDF scheduler degree of fairness performance is lower compared to PF and QACB scheduler. The QACB scheduler produces a better degree of fairness performance because it considers both queue length and waiting time, therefore, not favouring the delay sensitive traffic only.

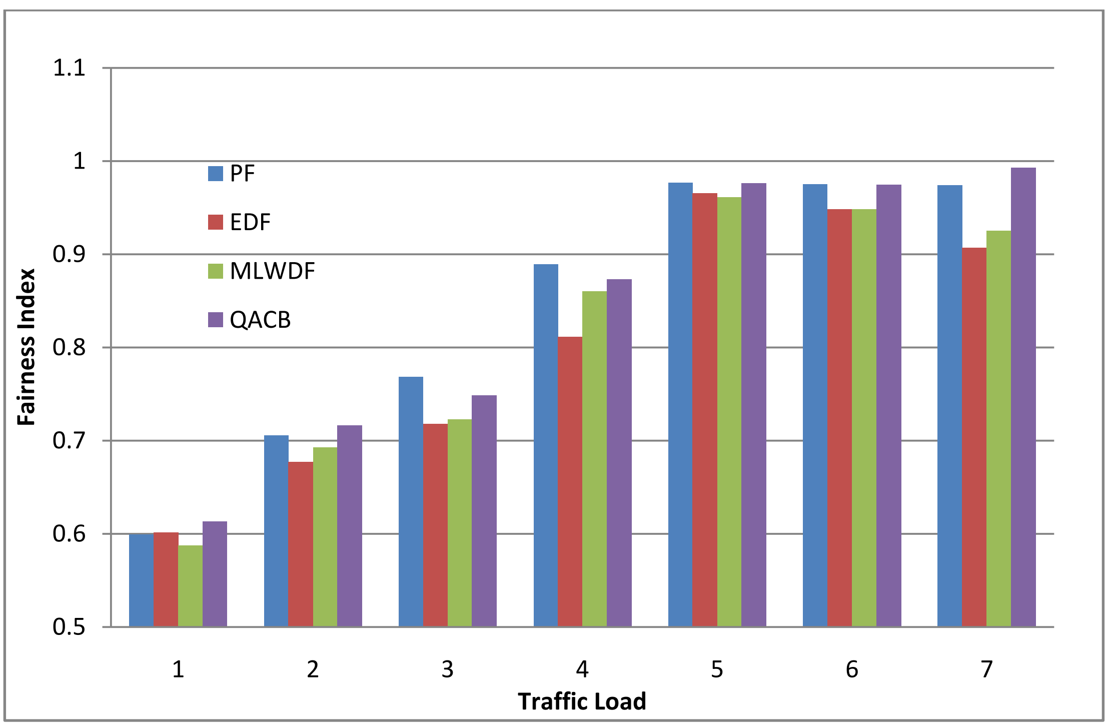

Figure 4.11 Fairness index for all users

4.9 Summary of Results

This chapter has presented an overview of packet scheduling in S-HSDPA environment and also the overview of the packet schedulers considered for this simulation which are PF, EDF, M-LWDF and QACB schedulers. The simulation setup, which includes the description of the simulated S-HSDPA network considered, the channel model and traffic model, is then presented. Since the aim of the simulation is to measure and compare the performance of the schedulers, the results in terms of throughput, delay and fairness are presented and discussed. The results for the two state Markov channel model are also presented and discussed for validation.

The EDF scheduler produces a better delay performance of real time traffic as expected, compared to other schedulers hence it does not favour non real time traffic since it doesn‘t consider channel status or any other factors. So, non real time traffic like web traffic may be starved in the queue with lower throughput. The PF scheduler produces a good throughput and fairness performance. Ironically, the PF scheduler produces a good delay performance to web traffic at the detriment of video traffic which happens to be a delay sensitive traffic, this is due to the fact that it cannot differentiate between different traffic types. Hence, the PF scheduler is not preferred in a mixed traffic scenario.

The M-LWDF produces a better delay performance for video traffic and very poor delay performance for web traffic, especially at high traffic loads as compared to QACB scheduler. The QACB scheduler produces better throughput performance compared to P-EDF for two state Markov channel model and M-LWDF for three state Markov channel model, since QACB considers the varying queue length and not just the channel conditions and waiting time in queue. The QACB scheduler also produces acceptable delay performance for both video and web traffic for both the two state and three state Markov channel model scenarios.

Overall, it can be said that QACB improves on the throughput and fairness of both the web and video traffic, as compared to P-EDF and M-LWDF for the two state and three state scenarios respectively without serious compromise to the video traffic‘s delay performance.

Chapter 5

SIMULATION ON RLC TRANSMISSION MODES IN S-HSDPA

5.1 Introduction

The aim of this chapter is to investigate the effects of the Radio Link Control (RLC) transmission modes on differentiated UMTS traffic types in Satellite HSDPA networks. The RLC sub-layer handles transmission in existing three different modes based on what is configured by the Radio Resource Control (RRC) and therefore plays a significant role in S-HSDPA system performance. In existing literature [108], this investigation was conducted for the terrestrial HSDPA network by presenting a performance evaluation for two RLC transmission modes. But the literature did not cover the satellite HSDPA network and did not use a delay sensitive scheduler in the work.

Due to these reasons, the investigation of the impact of RLC transmission modes based on performance for the four UMTS traffic types using both non delay sensitive scheduler (PF scheduler) and delay sensitive scheduler (QACB, our proposed scheduler in chapter 4) is carried out for satellite HSDPA network. A further investigation on the behaviour of the two schedulers used, in terms of performance for each RLC mode is then presented. The best mode for each traffic type in order to provide an acceptable QoS and utilize resources is recommended.

The existing RLC transmission modes are Acknowledged Mode (AM), Unacknowledged Mode (UM) and the Transparent Mode (TM). The considered modes are AM and UM for this simulation because TM is not applicable when using HSDPA transport channels. The different traffic types considered for this simulation are the four classes of UMTS traffic types. They are video streaming (MPEG-4), Voice over IP (VOIP), web browsing and FTP download.

The chapter commences with an overview of RLC transmission modes in S-HSDPA. This is followed by a detailed discussion of the existing RLC transmission modes which are AM, UM and TM. The simulation setup which includes channel model and traffic model are presented as well. Finally, the simulation results which include throughput, delay, jitter and packets sent are presented and discussed.

The work in this chapter has been presented at the Southern African Telecommunications Networks and Applications Conference 2009 (SATNAC 2009), Manzini, Swaziland.

5.2 Overview of RLC Transmission Modes in S-HSDPA

The different types of traffic such as conservational voice traffic, video streaming, interactive web traffic and background File Transfer Protocol (FTP) download can be transmitted in three different Radio Link Control (RLC) operation modes namely Acknowledged Mode (AM), Transparent Mode (TM) and Unacknowledged Mode (UM).

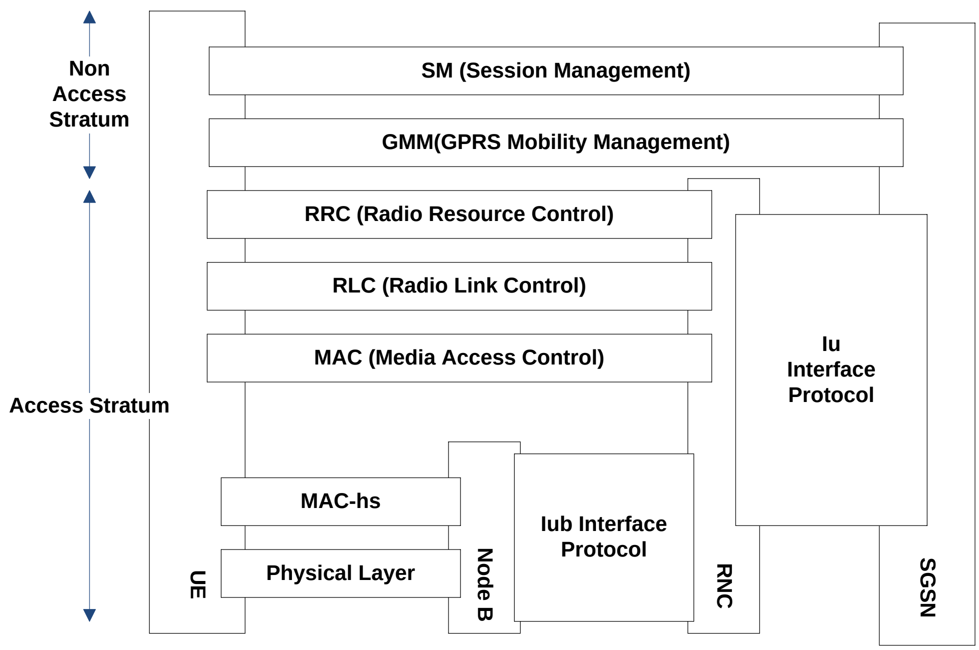

Figure 5.1 The layers and elements in S-HSDPA networks [106]

The RLC protocol is one of the major radio interface protocols which consist of flow control and error recovery. The link layer is made up of two sublayers which are the MAC and the RLC sublayers. The MAC sub-layer handles the scheduling of radio bearers, mapping of the logical channels to transport channels and provides data transfer services over logical channels in an unacknowledged mode. While the RLC sub-layer, which is above the MAC sublayer as shown in Fig. 5.1, provides the data transfer service of higher layer Protocol Data Units (PDU) and also handles error correction, segmentation and reassembly, padding and flow control. It also ensures that there is reliability of data transfers depending on the transmission modes. Each RLC instance is configured by Radio Resource Control (RRC) to transmits in any of the three modes (AM, UM and TM). These modes of transmission affect the performance of S-HSDPA and thus vary depending on the traffic type. The RLC therefore plays an important role in performance of S-HSDPA systems, since MAC layer only transfers data in an unacknowledged mode.

5.2.1 RLC Transmission Modes

As stated in the early sections, the RLC consists of three transmission modes. They are Acknowledge Mode (AM), Unacknowledged Mode (UM) and Transparent Mode (TM). The AM offers a reliable data delivery while UM and TM does not guarantee data delivery. Details of the three transmission modes are provided below [12].

5.2.1.1 Acknowledged Mode (AM)

The Acknowledged Mode (AM) ensures reliable transmissions to higher layers and across to RNC as well. The RLC entity in AM mode is bidirectional. It uses Automatic Repeat Request (ARQ) mechanism for error corrections. The RRC is used to control the performance of RLC through the configuration of the number of retransmissions. If the maximum number of transmissions configured is reached or transmission time is exceeded and the RLC is unable to deliver the data correctly, the upper layer is notified and the RLC Service Data Unit (SDU) is discarded. The RLC entity in this mode is bidirectional and it‘s capable of piggybacking which is an indication of the status of the link in the opposite direction into the user data.

5.2.1.2 Unacknowledged Mode (UM)

The Unacknowledged Mode (UM) does not guarantee data delivery since no retransmission protocol is used. The RLC entity in UM mode is unidirectional because association between the uplink and the downlink is not needed. Received erroneous data are either marked or discarded depending on the configuration. The sequence number of the PDU structure is used to ensure that the integrity of higher layer PDUs are observed.

5.2.1.3 Transparent Mode (TM)

The Transparent Mode (TM) does not guarantee data delivery as well and no protocol overhead is added to higher layer data when in this mode. RLC entity in TM mode is unidirectional because association between the uplink and downlink is not needed. Transmissions with limited or no segmentation can be accomplished in this mode. Erroneous PDUs can be discarded or are marked as erroneous. It is worthy to note that TM is not applicable when HSDPA transport channels are used. So, it is not considered in the simulation conducted for this dissertation.

5.3 Channel Model

The three state Markov channel model is also used for this simulation and the suburban area at an elevation angle has been considered as well. The process of determining SNR and CQI is the same as explained in section 4.3. The preprocessed channel input file (Physical layer parameters) used for the simulation is generated for twelve (12) users for over 500 seconds using MATLAB.

5.4 Traffic Model

For this simulation, the four different traffic types of UMTS which are the streaming, conversational, interactive and background traffic classes have been considered. The traffic considered are video streaming, VOIP, Web traffic and FTP downloads.

The traffic model used for this simulation for video (MPEG-4) and web traffic is the same as the one used in chapter 4 and explained in section 4.5.1 and 4.5.2 respectively while the FTP traffic is the default NS-2 FTP source.

The VOIP traffic is modeled using exponential ON/OFF model. Hence, the on and off state are determined using exponential distribution. It is worthy to note that all this traffic models are already built into NS-2 by default except for the video streaming (MPEG-4). The process is explained in chapter 4.

5.5 Simulation Setup

The investigation of the effects of RLC transmission modes on the performance of S-HSDPA systems for different traffic types is carried out using discrete event simulation software called Network Simulator 2 (NS-2) [94]. Since NS-2 alone doesn‘t support UMTS and HSDPA, Enhanced UMTS Radio Access Network Extension (EURANE) which is a UMTS/HSDPA extension to NS-2 [95] is used on NS-2 platform. The Proportional Fair (PF) and Queue Aware Channel Based (QACB) scheduler, that is proposed in chapter 4, are used for this simulation. The former is non delay sensitive, while the latter is delay sensitive scheduler. The network used for this simulation setup, as shown in Fig. 5.2, provides a scenario where the mobile terminals are connected to Node B via the GEO bent-pipe Satellite and Node B is connected to Wired Node 1 through RNC, Serving GPRS Support Node (SGSN), Gateway GPRS Support Node (GGSN) and Wired Node 2. The RLC transmission modes considered are AM and UM. The Transmission Time Interval (TTI) is 2ms and the CQI reporting interval is assumed to be 40 ms (20 TTI). Only one user can be allocated resources at every TTI for all schedulers considered.

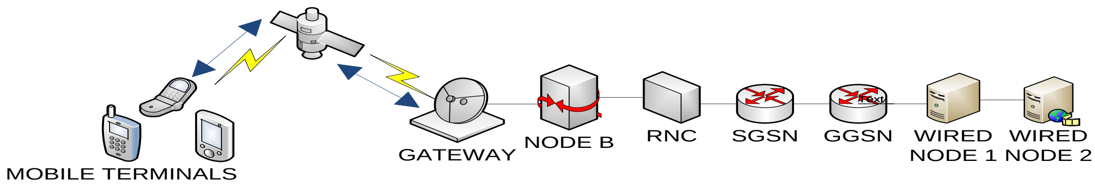

Figure 5.2 S-HSDPA simulated network

For the purpose of this simulation, twelve (12) mobile users made up of two (2) video streamers, four (4) VOIP users, four (4) web traffic users and two (2) FTP downloaders have been considered. Relevant details of the simulation scenario are provided in Table 5.1.

Table 5.1 Details of simulation parameters

Parameter	Values
Simulation Time	500s
Numbers of Video Users	2
Numbers of VOIP Users	4
Numbers of Web Users	4
Numbers of FTP Users	2
Transmission Time Interval (TTI)	2 ms
Channel Model	three state Markov channel model
CQI reporting interval	40 ms (20 TTI)
Channel Delay	280 ms
Video Traffic Model	MPEG-4 TES Model
VOIP Traffic Model	Exponential ON/OFF Model
Web Traffic Model	M/Pareto ON-OFF Model
IP Packet Size	1500 bytes
FTP Traffic Model	NS-2 Default FTP Source
FTP File Size	1MB
RLC Mode	AM & UM
UE Category	5 & 6

5.6 Simulation Results

The results for this investigation are presented in this section. The comparison of the throughputs, delay and the numbers of packets sent for all the four traffic types are presented for the two modes using two different schedulers. The jitter experienced by the real time traffic which are video and VOIP is also presented. The simulation produces huge event driven data results. To evaluate and analyze these set of results, the gawk programming language is used and the Microsoft spreadsheet (Excel) is used to plot the graphs. The traffic loads 1,2,3,4 and 5 represents different data rates, so each traffic load depicted at the x axis of each graph presented is represented with a certain data rate. The details of the traffic load representation are presented below as recommended in [103] for video traffic, [92] for web traffic and [107] for voice traffic;

Table 5.2 Traffic load representation

TRAFFIC LOAD	1	2	3	4	5
VIDEO TRAFFIC (rate factor)	0.25	0.5	0.75	1.0	1.5
VOICE TRAFFIC (kbps)	4.75	5.9	7.95	10.2	12.2
WEB TRAFFIC (kbps)	8	32	64	128	384
FTP TRAFFIC (MB)	128	256	512	768	1024

5.6.1 Delay

The average delay of voice and video traffic as shown in Fig. 5.3 and 5.4 depicts the fact that users in UM RLC mode experienced better delay performance compared to AM RLC mode, for both schedulers used. This is because the traffic sent in UM RLC operation mode does not wait for acknowledgements, therefore, a lower delay is experienced.

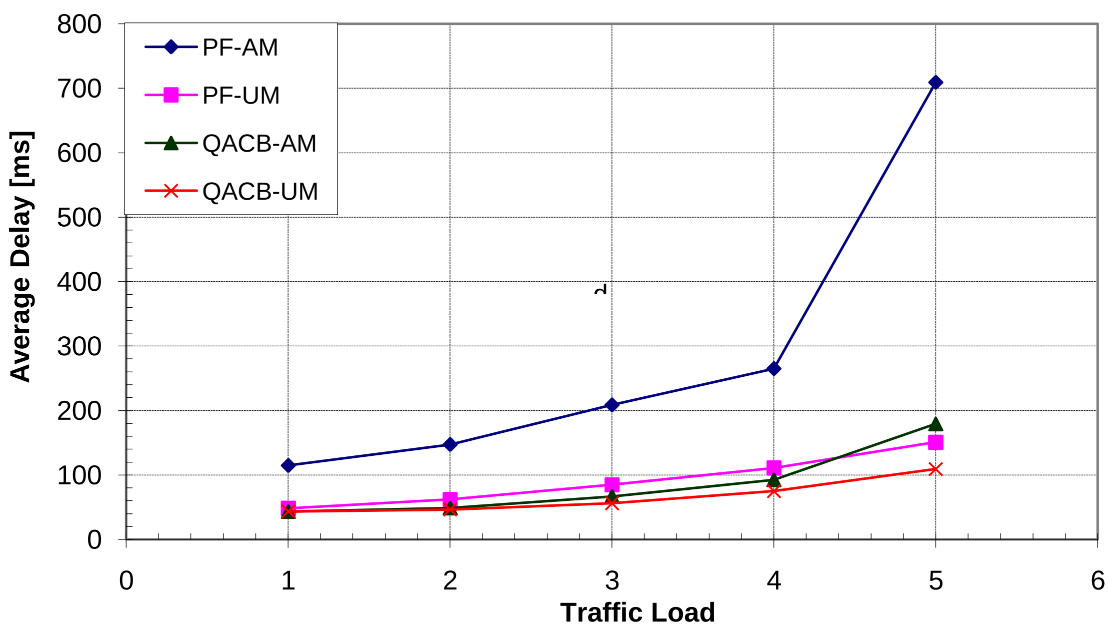

Figure 5.3 Average delay of video users for PF and QACB schedulers.

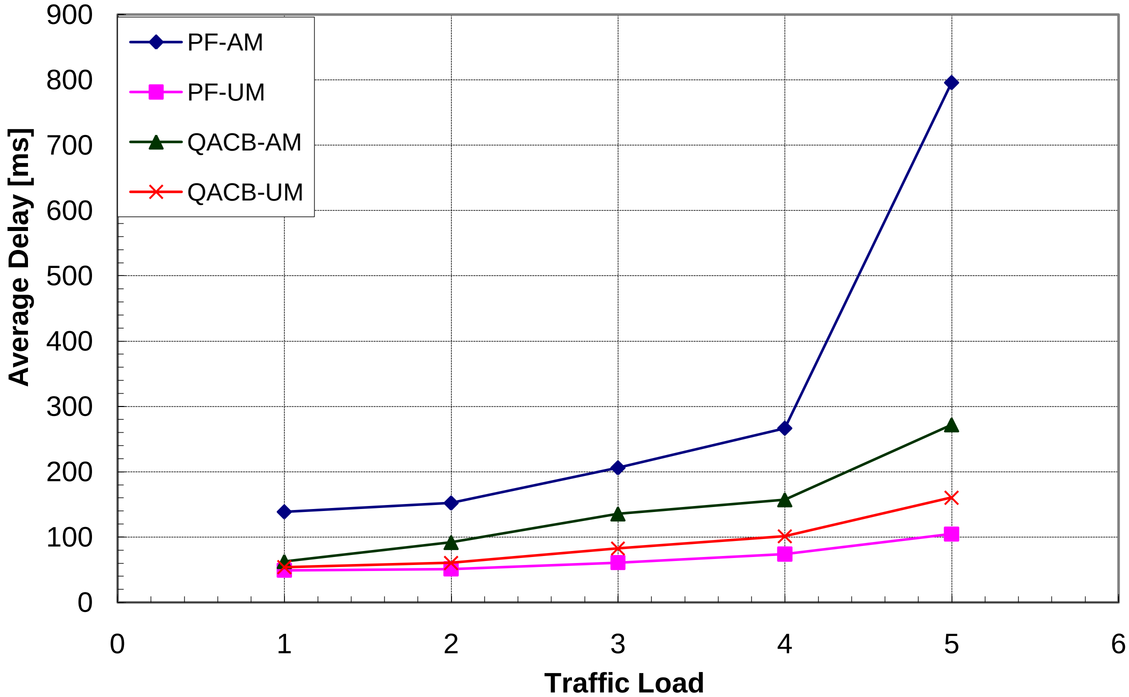

Figure 5.4 Average delay of voice users for PF and QACB schedulers.

The difference in the delay performance for AM and UM RLC modes when using QACB scheduler is minimal compared to the difference experienced when using PF scheduler. This can be linked to the fact that PF scheduler is strictly channel based, so it cannot differentiate between different traffic types while QACB scheduler has the capability to differentiate the traffic types using its QoS differentiation mechanism. Overall, the QACB scheduler has better delay performance compared to PF scheduler except for voice traffic in UM mode. This shows that the PF scheduler can also produce good performance for real time traffic. This is not on the basis that the scheduler is delay sensitive but that the voice users coincidentally have a better RCQI.

The average delay of web and FTP traffic experienced in UM mode for the two schedulers is close to zero as presented in Fig. 5.5 and 5.6. This is due to the fact that negligible numbers of packets of web and FTP traffic are sent through in this RLC mode. Since only negligible packets are sent through, the other packets experience an infinite delay. The zero average delay is as a result of the fact that the sum of delay experienced by the negligible packets that are sent through divided by the total number of packets that is to be sent will be close to zero as shown in equation (4.14). Results on the number of packet sent through are presented in Fig 5.17 and 5.18 for more clarification. The average delay experienced for web traffic in AM RLC mode for the two schedulers is close to each other with QACB scheduler producing a better delay performance at the highest traffic load while the PF scheduler produces a better delay performance compared to QACB for FTP traffic in RLC AM mode.

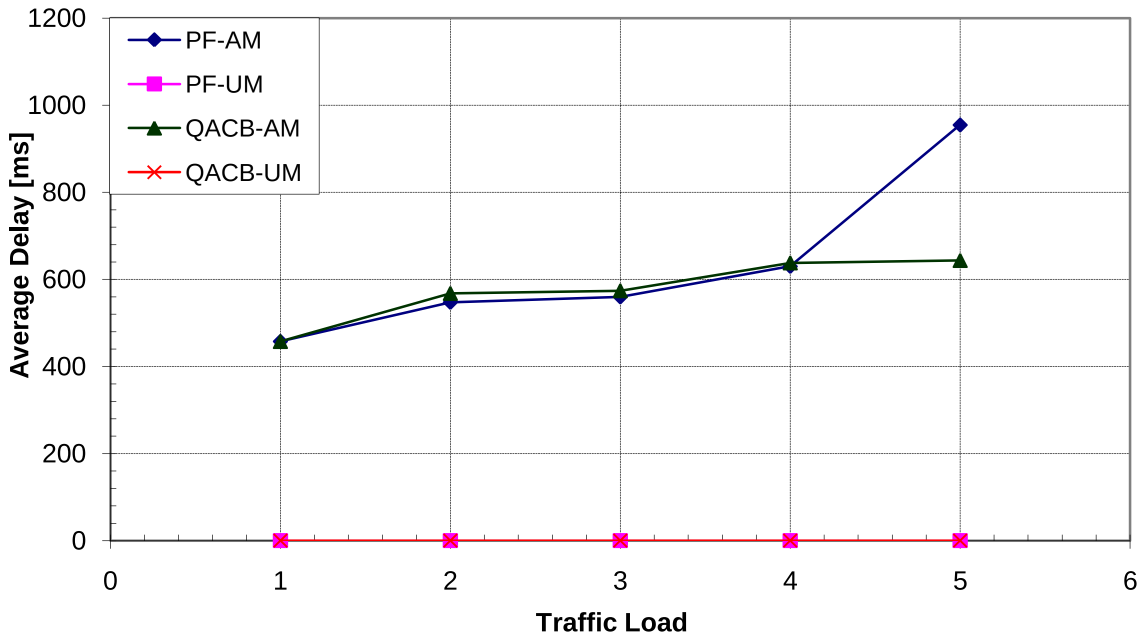

Figure 5.5 Average delay of web users for PF and QACB schedulers.

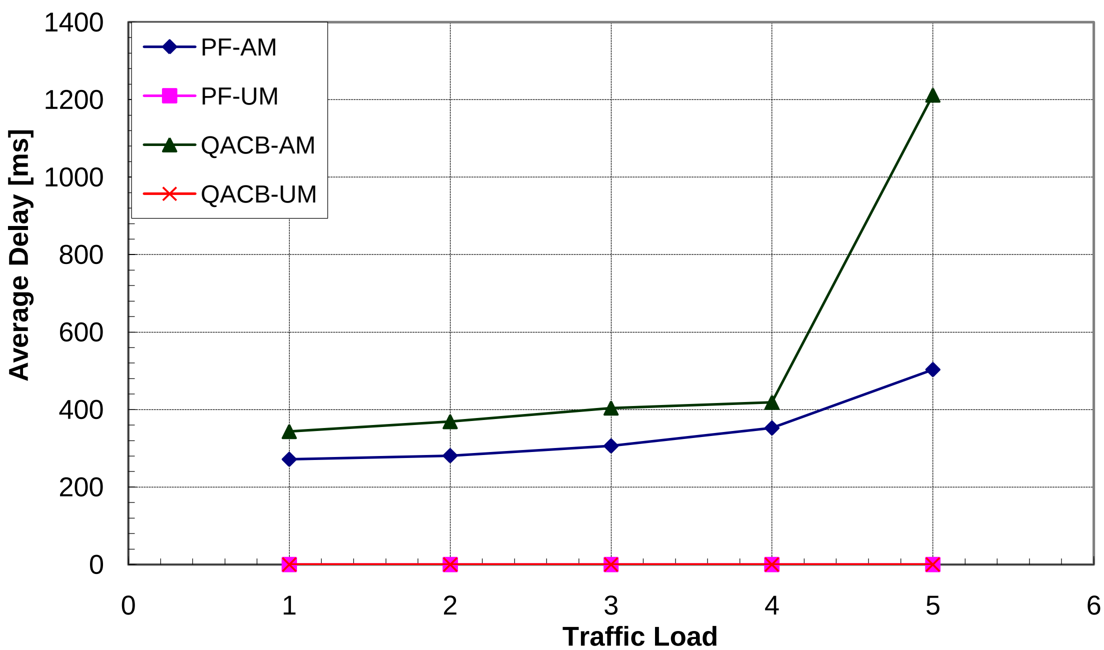

Figure 5.6 Average delay of FTP users for PF and QACB schedulers.

5.6.2 Jitter

The jitter is defined to be delay variation and can be presented mathematically as;

$$J_m = \frac{D_j - D_i}{n_j - n_i} \tag{5.1}$$

Where J_m is the instantaneous jitter at time m, D_j is the delay experienced by a packet with sequence number n_j and D_i is the delay experienced by a packet with a sequence number n_i. The statistical tool called easy fit is used to plot the PDF of the jitter results for specific voice and video users in both AM and UM RLC modes. This is done on order to obtain the range of jitter experienced for each scenario. The jitter performance is measured based on length of variation, so the smaller the range of variation, the better the jitter performance and wider the range of variation, the worse the jitter performance.

The jitter results for video traffic in both AM and UM mode are shown in Fig. 5.7 and 5.8. The jitter experienced for video traffic for PF scheduler is in a range of 200 ms and 45 ms for both AM and UM RLC mode respectively while that of QACB scheduler is in the range of 45 ms and 40 ms for both AM and UM RLC mode respectively. As shown in Fig. 5.7 and 5.8, the video traffic in UM mode for both schedulers experienced better jitter performance compared to AM mode because they have smaller range of variations compared to AM mode. Since there is no acknowledgements in UM RLC mode, this to an extent, will allow consistent delays.

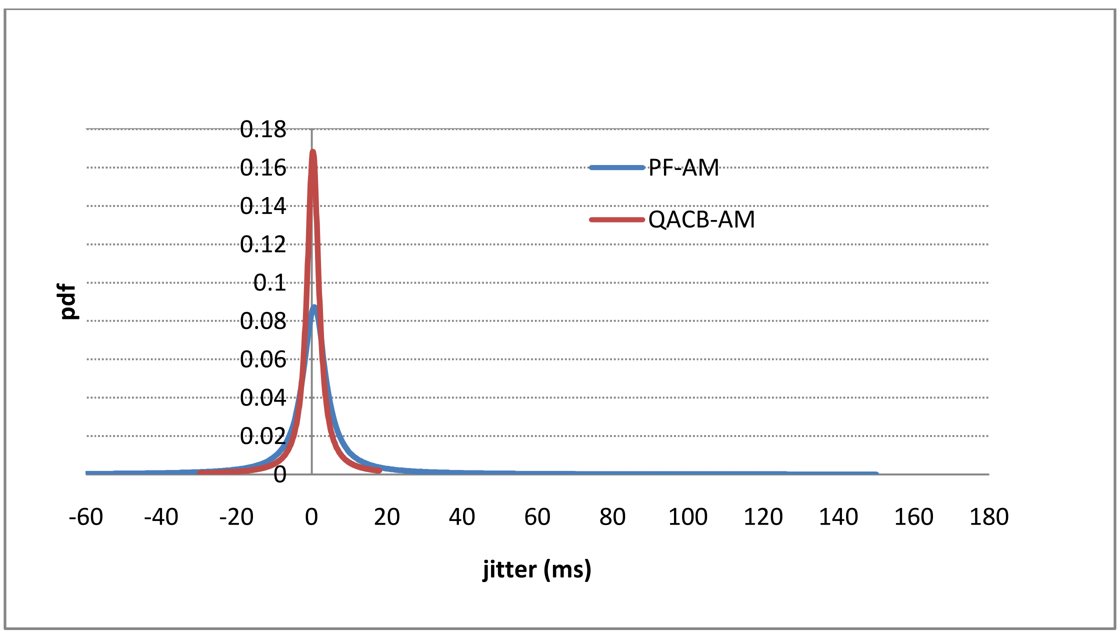

Figure 5.7 Average jitter of video traffic for PF and QACB schedulers in AM mode.

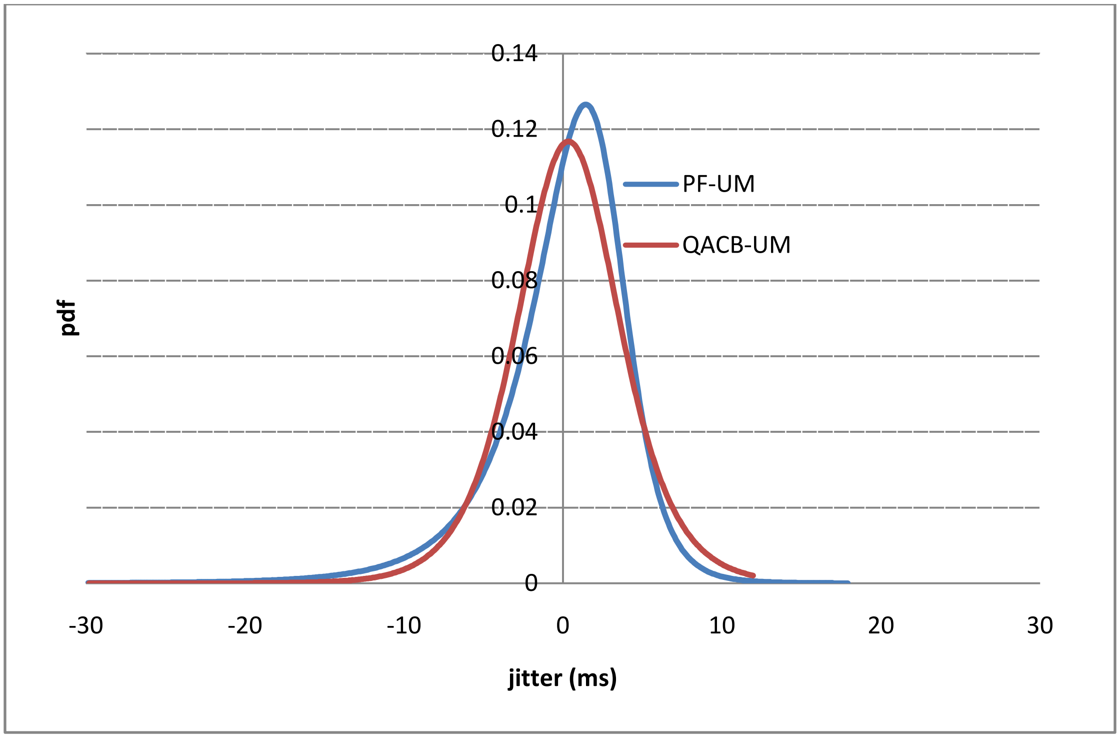

Figure 5.8 Average jitter of video traffic for PF and QACB schedulers in UM mode.

It is also observed that the jitter experienced by the video traffic for PF scheduler in AM mode is large compare to its UM mode and both RLC modes of QACB scheduler for the video traffic. This is because the range of variation of PF scheduler is AM mode is 200 ms as compared to 40 and 45 ms for others. The jitter results for both UM and AM RLC mode for QACB scheduler is close while there is a big difference for that of PF scheduler. Overall, the QACB scheduler produces a better jitter performance compared to PF scheduler for video traffic, this is as a results of the fact that the PF scheduler is not able to distinguish delay sensitive traffic from other traffic types. This shortcoming is noticeable in the AM mode.

The jitter experienced by voice traffic for PF scheduler in AM and UM RLC modes are in the range of 220 ms and 32 ms respectively while that of QACB scheduler are in the range of 64 ms and 56 ms respectively as presented in Fig. 5.9 and 5.10. The jitter performance for the voice traffic follows the same trend with that of video traffic except that the PF scheduler in the UM mode produces a better performance than both AM and UM RLC modes for QACB scheduler. This conforms to the delay performance presented in Subsection 5.6.1 above. The UM RLC mode performances for both schedulers are better than the AM RLC mode since they both have smaller variation as compared to AM mode for the two schedulers.

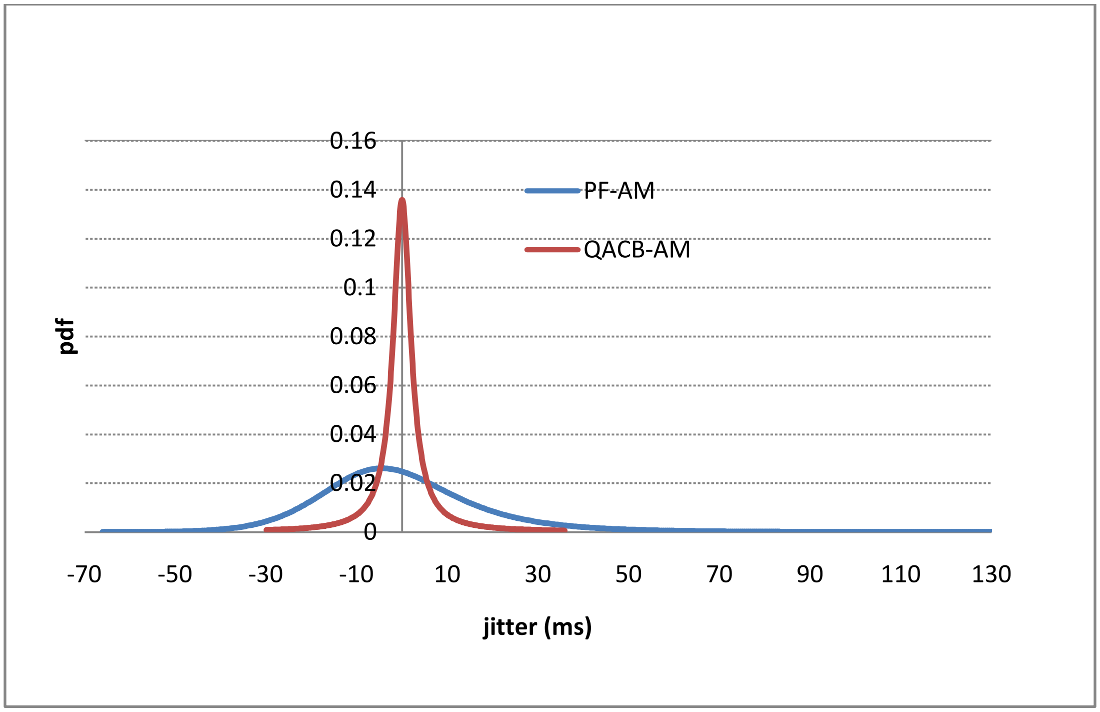

Figure 5.9 Average jitter of voice users for PF and QACB schedulers in AM mode.

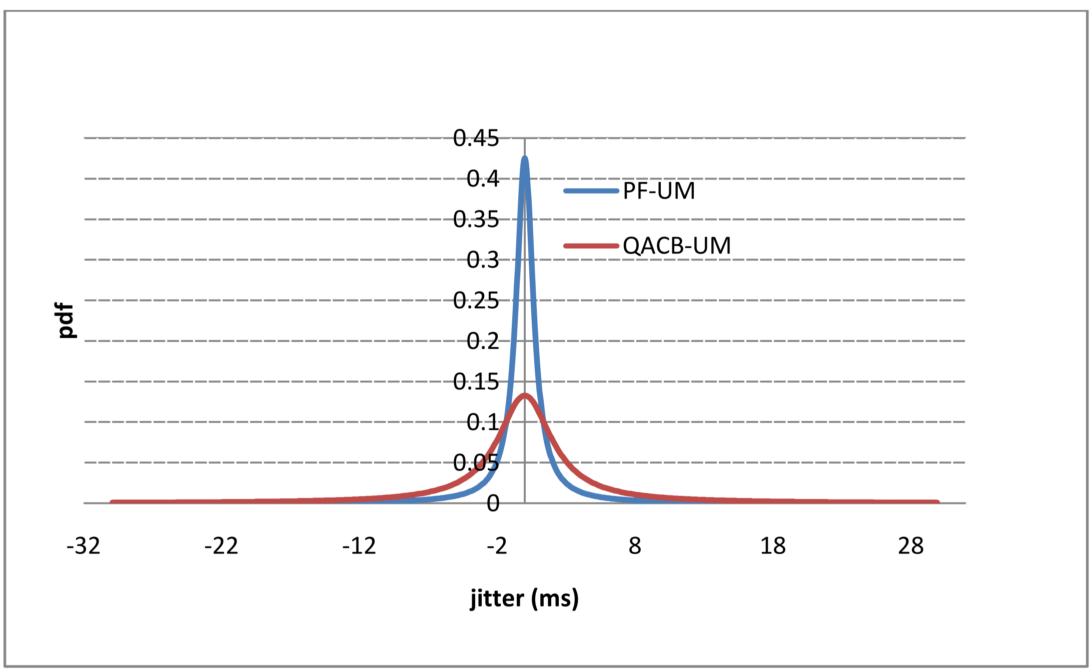

Figure 5.10 Average jitter of voice users for PF and QACB schedulers in UM mode.

5.6.3 Throughput

Though, throughput is not of great importance in terms of performance metric to real time traffic, the essence of presenting the average throughput performance results for both the video and voice traffic is to observe the difference in AM and UM RLC modes. The average throughput for video and voice users transmitting in UM mode for both schedulers is observed to be close to half of the average throughput in the AM mode as shown in Fig. 5.11 and Fig. 5.12. It is further observed that the numbers of packets sent for both AM and UM RLC modes are close to each other as shown in Fig. 5.15 but the throughputs differ. Since the numbers of packets sent for the two modes are almost the same, then it can be said that the different throughputs experienced is due to the retransmissions that occur in AM RLC transmission mode. However, it is worthy to note that retransmission is not preferred in real time traffic like video and voice, such throughput advantage experienced in AM mode is really of priority for these traffic types. The average delay and jitter is of great importance to real time traffic compared to throughput. The two schedulers produce close throughput performance in the two RLC modes.

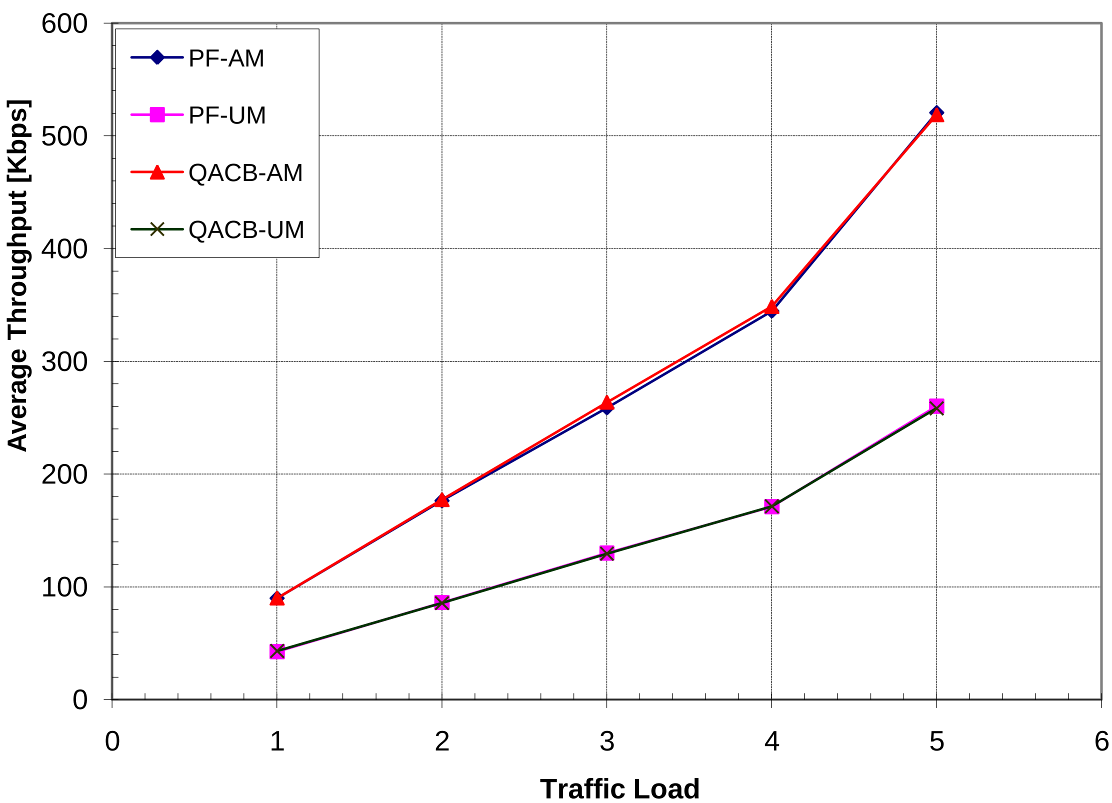

Figure 5.11 Average throughput of video users for PF and QACB schedulers

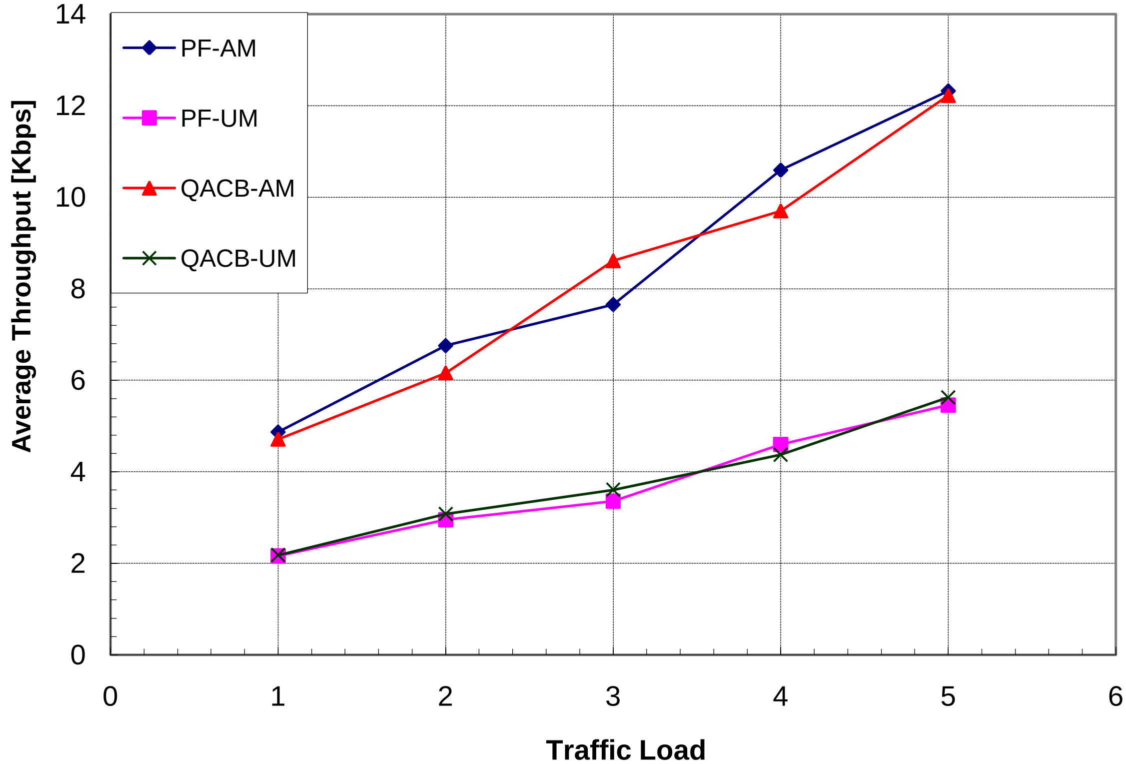

Figure 5.12 Average throughput of voice users for PF and QACB schedulers

Fig. 5.13 and Fig. 5.14 shows that the web and FTP traffic produces an average throughput of close to zero for both schedulers in UM RLC mode. This tallies with the average delay result presented in Fig. 5.5 and 5.6 respectively where both video and voice traffic experience a close to zero delay in UM mode. The reasons behind this result is further clarified in Fig 5.17 and Fig. 5.18 in section 5.5.4, where negligible numbers of packets are sent for both web and FTP traffic in UM RLC modes. The average throughput performances produced by the two schedulers are very close for web traffic.

The two schedulers also produces a close average throughput performance for FTP traffic, except at high traffic load where PF scheduler edges QACB scheduler. This shows that despite the QACB schedulers producing a better delay performance compared to PF scheduler for all traffic types, it is still able to match up with the throughput performances of PF scheduler for all the traffic types at most traffic loads. Much of the differences between the two schedulers is experienced in the delay and jitter performances. These set of throughput results conforms with the instantaneous throughput results obtained in [108] for the terrestrial counterparts.

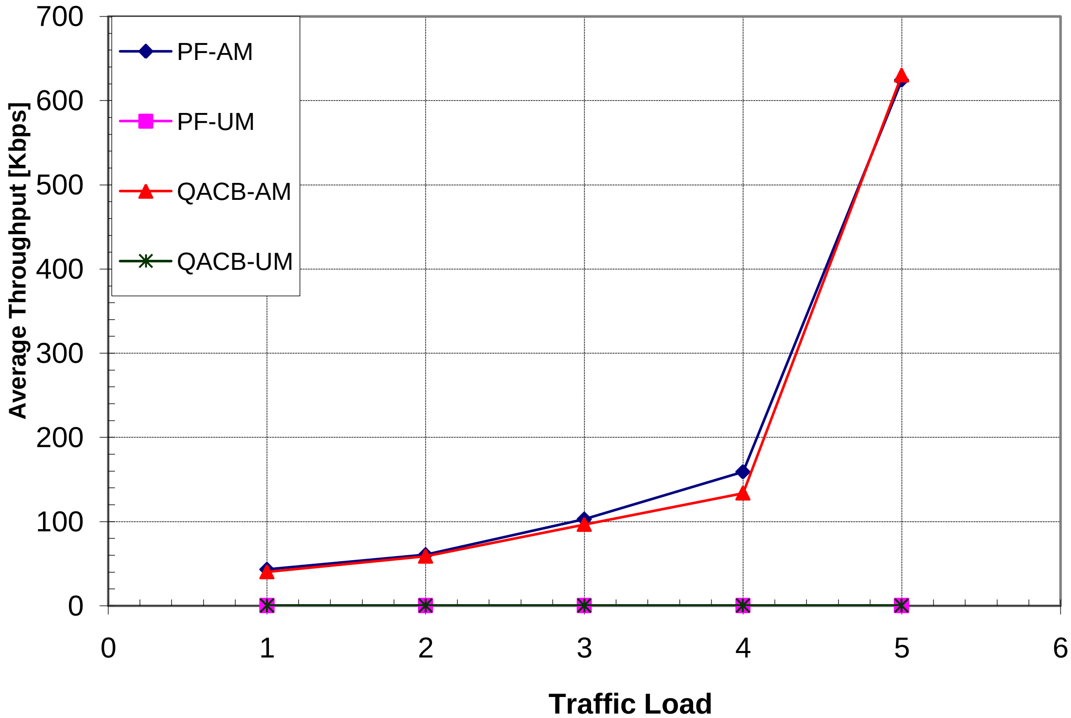

Figure 5.13 Average throughput of web users for PF and QACB schedulers

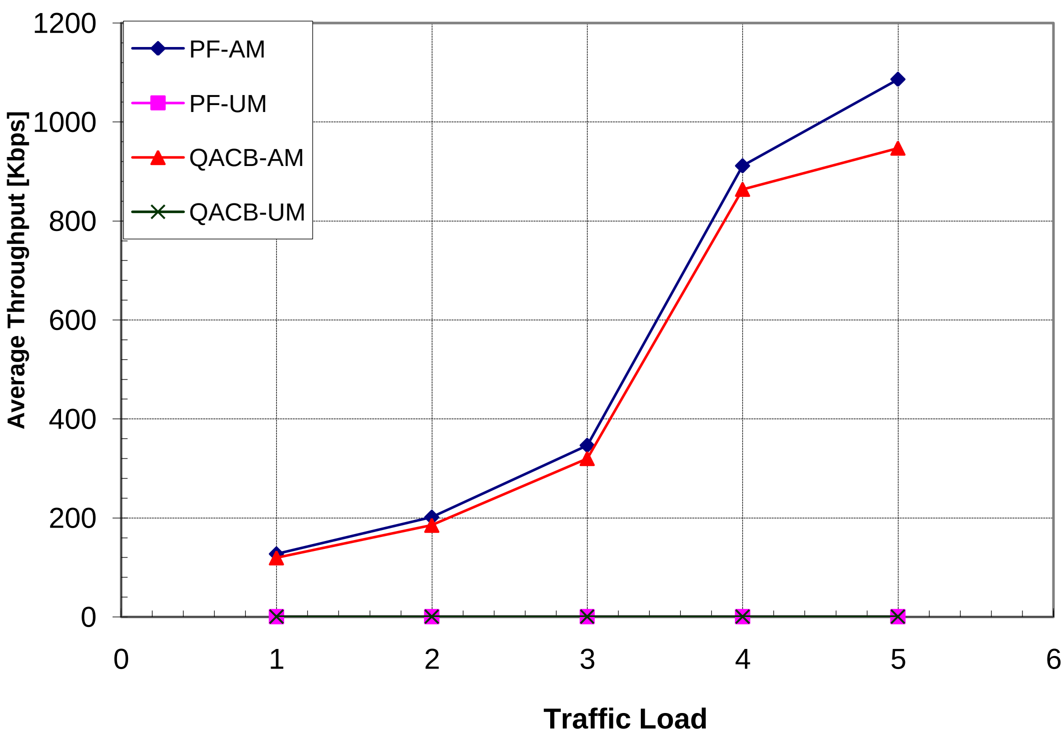

Figure 5.14 Average throughput of FTP users for PF and QACB schedulers

5.6.4 Number of Packets Sent

The packets sent in the two modes are depicted in Fig. 5.15 – 5.18 for all the traffic types. These graphs provide some clarification to some of the results experienced in section 5.5.1 and 5.5.3 where the average delay and throughput results are presented respectively. The numbers of video and voice packets sent as shown in Fig. 5.15 and Fig. 5.16 respectively for both AM and UM RLC mode are close to each other but due to the fact that AM RLC mode allows retransmissions, the average throughput performance is higher. While Fig. 5.17 and Fig. 5.18 shows that very negligible number of web and FTP packets are sent in UM mode, this explains why the average delay results presented in Fig 5.5 and Fig 5.6 are close to zero delay and the average throughput results presented in Fig. 5.13 and Fig. 5.14 are close to zero throughput as well. This is due to the fact that the web and FTP traffic are TCP based applications and therefore needs reliable transmissions. This also conforms to the number of packets results presented in [108].

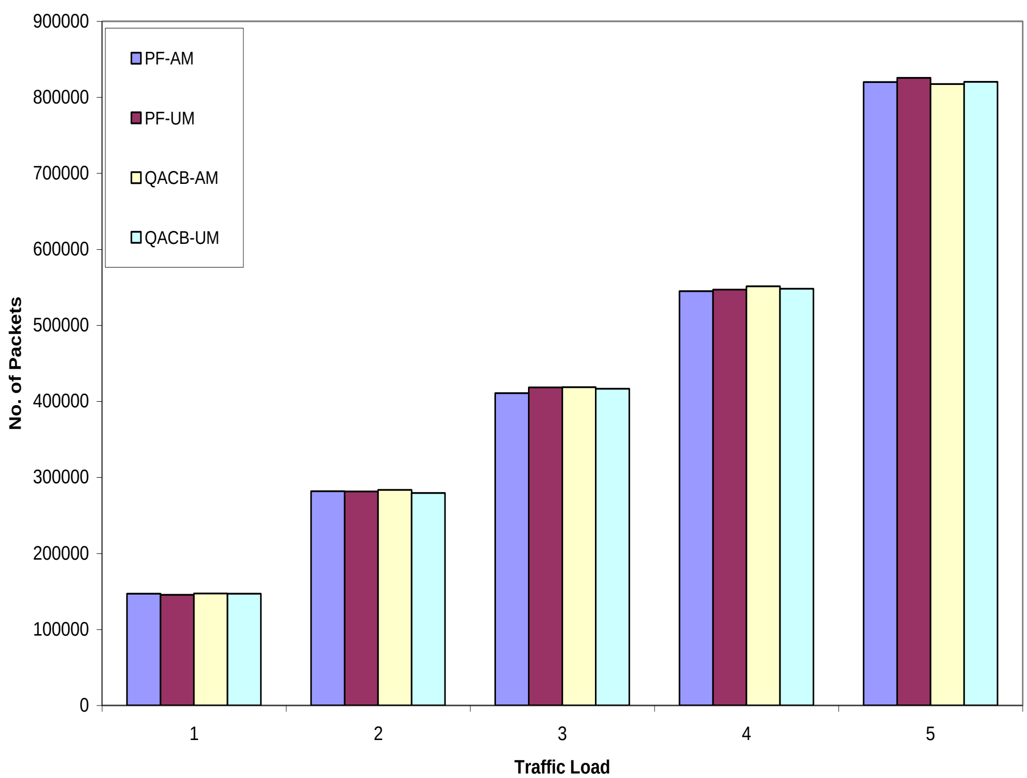

Figure 5.15 Packets sents by video users for PF and QACB scheduler

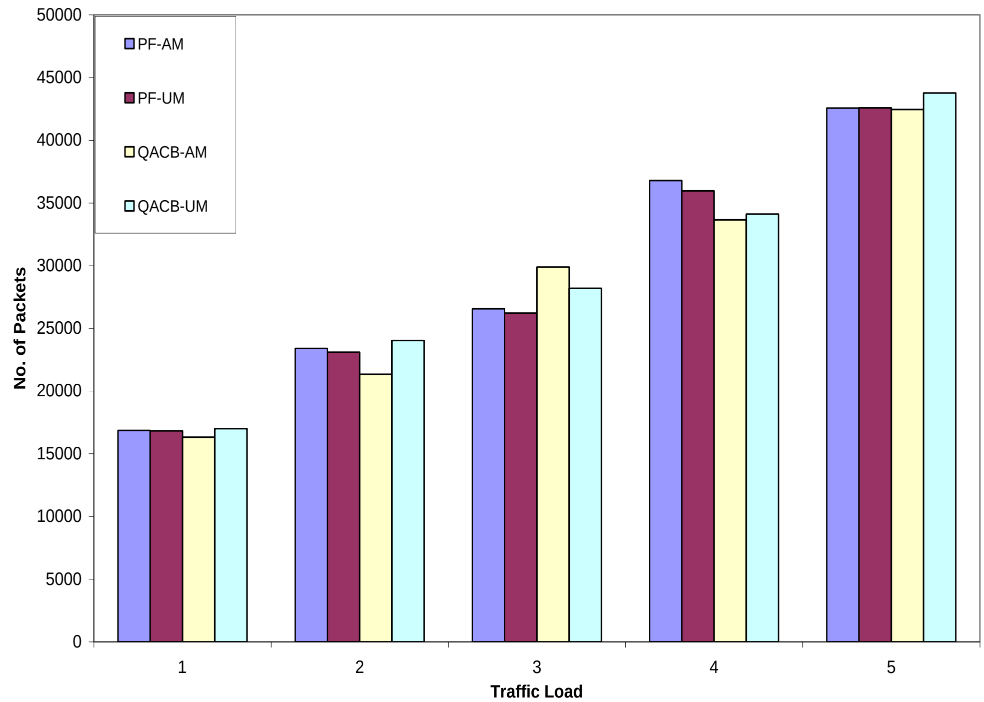

Figure 5.16 Packets sents by voice users for PF and QACB scheduler

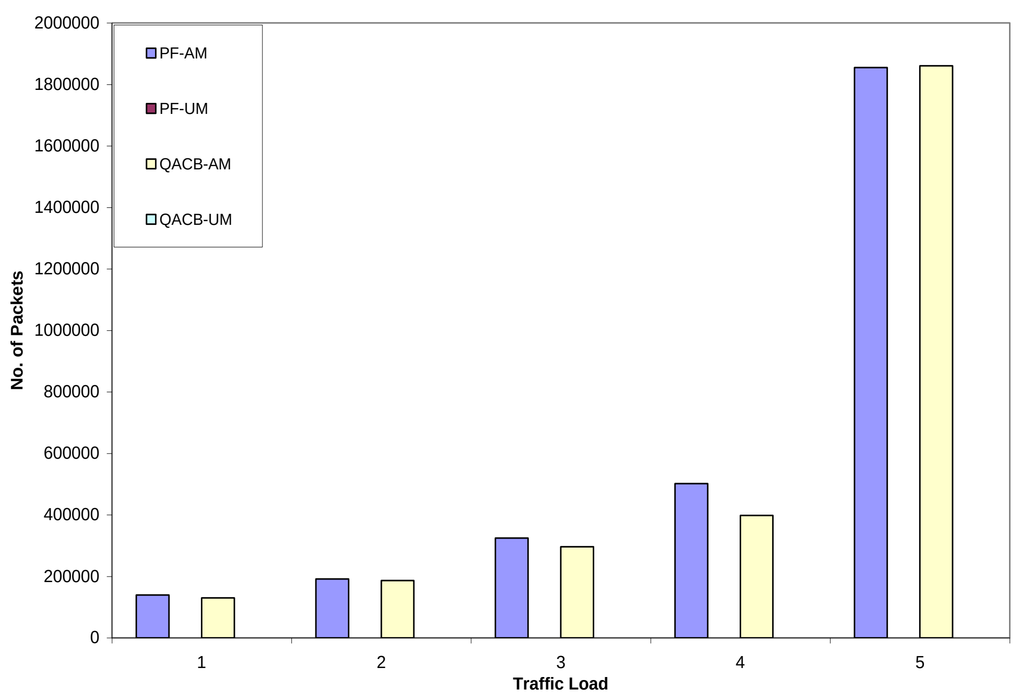

Figure 5.17 Packets sents by web users for PF and QACB scheduler

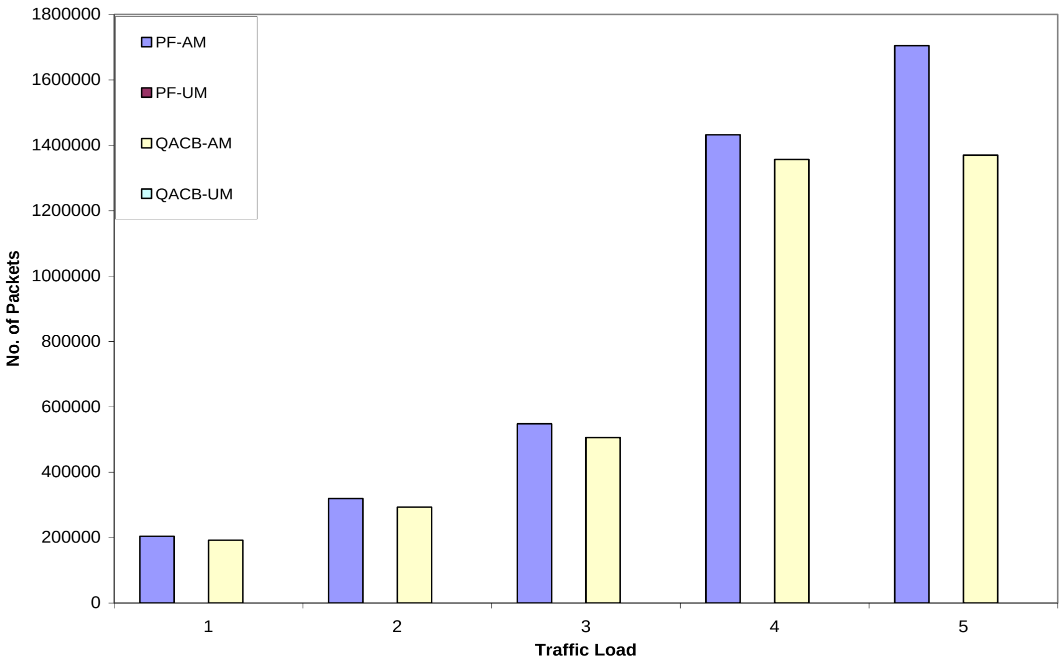

Figure 5.18 Packets sents by FTP users for PF and QACB scheduler

5.7 Summary

This chapter presented an overview of RLC transmission modes in S-HSDPA and the three RLC transmission modes which are AM, UM and TM are also discussed. The simulation that is conducted to investigate the effect of these RLC transmission modes for differentiated traffic types in S-HSDPA is also presented. The four UMTS traffic types used for the simulation are conversational (VOIP), streaming (MPEG-4), interactive (web) and background (FTP). The setup for the simulation is then presented. The performances of each RLC mode considered for each of the two schedulers are presented and discussed. The performances considered are throughput, delay and jitter.

The results obtained shows that the delay sensitive traffic like VOIP and video should be transmitted in RLC UM mode since lower delay and a more stable lower jitter is experienced, this will allow the utilization of the uplink resources that are used for sending acknowledgements and will also ensure retransmission is not applied to delay sensitive traffic. For non-delay sensitive traffic like web and FTP, UM mode is not an option for transmission since they are TCP based applications and needs reliable connection. They should only operate in RLC AM mode.

The result also shows that QACB scheduler provides better performance for delay sensitive traffic like video and voice in the two modes compared to PF scheduler, while QACB scheduler produces close performances for non delay sensitive traffic like web and FTP in terms of throughput.

Also, the result follows the same trend with similar work done for the terrestrial counterpart in [108]. An average delay of below 300 ms is experienced by delay sensitive traffic like video and voice at all traffic loads except for PF scheduler in AM mode, this shows that using the appropriate scheduler and transmitting in the appropriate RLC mode, the satellite HSDPA produces a good delay performance. This shows that satellite systems are valid alternatives for the provision of access network to mobile users.

Chapter 6

CONCLUSION

In an attempt to achieve convergence by providing the same level of service to everyone anytime and anywhere, the satellite networks remain a vital component towards Next Generation Networks (NGN). The ability of the HSDPA to provide higher data rates due to its fast scheduling schemes as compared to UMTS have made it an important area of research at both terrestrial and satellite levels. It is expected that Radio Resource Management (RRM) will play a key role towards ensuring that the limited resources available are utilized and varying QoS of different users are met. The designing of schedulers for S-HSDPA that will put into considerations important factors like channel conditions and queuing factors, in making decisions is therefore a necessity towards achieving an acceptable QoS and high system throughput.

6.1 Dissertation Conclusion

The introductory chapter gives an overview of the evolution of wireless communications and this includes the third and fourth generation networks. The overview of the three different satellite types and mobile satellite communication systems are presented as well. A motivation for the research and an overview of this dissertation are further presented. Lastly, the original contributions and publications from the research work are stated.

In chapter 2, an overview of Radio Resource Management (RRM) with its functions is given. The RRM functions discussed are Admission Control, Handover Control, Packet Scheduling and Power Control. The different interactions using cross-layer concept in RRM are presented. Detailed discussions on packet scheduling schemes are presented. The scheduling schemes for wired networks which include FIFO, FIFO+, Round Robin, DRR and WFQ are first presented for wired networks, followed by scheduling schemes for wireless networks. Finally, the chapter is concluded with the discussion of specific scheduling schemes used in S-HSDPA networks, which are Proportional Fair, Earliest Deadline First and Modified Largest Weighted Delay First.

Chapter 3 presents the overview of channel modeling with a detailed discussion of a two state Markov channel model and the three state Markov channel model that is used for simulation in this dissertation. After the investigation of the effects of Signal to Noise Ratio (SNR) margin introduced to address the problem of long propagation delay experienced in HSDPA via GEO satellite by evaluating the performance of each SNR margin considered, it is concluded that the SNR margin 1.5 dB produces the best performance compared to other SNR margins considered. This is confirmed by evaluating the goodput and delay performance from the simulation and the throughput performance from the semi-analytical work. It is worthy to note that from the simulation the optimal SNR margin varies with the traffic load.

In chapter 4, a new packet scheduling scheme known as Queue Aware Channel Based (QACB) is introduced and its performance is compared with three other schedulers. It is concluded from the simulation results that the PF produces a good throughput performance and EDF gives a better delay performance for the real time traffic at the expense of non-real time traffic. It is also concluded that the PF scheduler can‘t differentiate between real time and non real time traffic, this is confirmed from the delay results where the PF scheduler produces a good delay performance for web traffic at the expense of the video traffic. The most relevant conclusion of this chapter is that although the M-LWDF gives a better delay performance for real time traffic compared to QACB, the QACB produces better performance in terms of system throughput and fairness. Also, worth mentioning is that despite the fact that non real time traffic isn‘t much concerned about delay, M-LWDF has a poor delay performance especially at high traffic load for web traffic. Hence, the newly proposed scheduler is able to improve on throughput and fairness to non real time traffic as compared to M-LWDF scheduler without any serious compromise to real time traffic.

Chapter 5 investigates the effects of the different RLC transmission modes in HSDPA via GEO satellites as it affects the different UMTS traffic types and scheduling schemes. It is concluded that the real time traffic are best for UM RLC transmission mode since the better performance are produced in this mode, while non real time traffic of TCP based can only be operated in AM RLC transmission mode. It is also concluded that the QACB scheduler provides better performance for real time traffic like video and voice in the two modes compared to PF scheduler, while both schedulers‘ produce close performances for non real time traffic like web and FTP in terms of throughput. Another important conclusion from the results obtained is that the QACB scheduler produces an average delay performance close to 300 ms for both RLC modes. This shows that with the proper scheduling scheme and by transmitting at the appropriate RLC mode, satellite access networks remain a valid option to its terrestrial counterparts.

6.2 Future Research Works

There are research areas that can be done to build on what have been presented in this dissertation. The research works are presented as follows;

- Optimization of SNR margin to suit varying traffic loads, so as to effectively address the long propagation delay for Satellite networks and ensure the adequate channel information is used for scheduling and resource allocation.
- An analytical performance analysis can also be used for comparison of the various scheduling schemes and the proposed scheduler can also be extended to terrestrial access network scenario.
- Proposing a joint admission control and packet scheduling for HSDPA via GEO satellite can also be explored in order to achieve a better resource utilization and acceptable QoS.

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APPENDIX A

A. TRAFFIC MODEL

This section describes the modeling of the traffic used for the set of simulation carried out in this dissertation.

A.1 Video Traffic Model

The Transform Expand Sample (TES) model is used to model the Variable Bit Rate (VBR) MPEG-4 video traffic. The MPEG-4 traffic is then modeled by using a trace which consists of set of frames of an encoded video at the rate of 25 frames per second. The trace is then separated into three set of frame types which are the I frames, B frames and P frames. This separation makes it easy to model. A separate TES model is then use to model for each set of trace using the method explained below [103];

The TES is a useful method that is used to generate data that tightly match any set of time series observations. The model is derived in two phases which are the background and foreground phases. The background process which is used to generate a time series of correlated random variables with uniform marginal in [0,1] can be derived as follows;

$$U_n^+ = \begin{cases} U_0 & n = 0 \\ \langle U_{n-1}^+ + V_n \rangle & n > 0 \end{cases}$$

$$U_n^- = \begin{cases} U_n^+ & n\ even \\ \langle 1 - U_n^+ \rangle & n\ odd \end{cases}$$

Where U_0 is the random variable uniformly distributed in [0,1], V_n is a sequence of iid random variable independent of U_0, U_n^+ and U_n^- is used to generate lag-1 autocorrelations in the range [0,1] and [1,0] respectively and the operator < > represents the modulo-1 operation.

The second phase involves using inversion method to generate a synthetic data that resembles the real sample from the background sequence obtained in the first phase. It is assumed that the histogram of the video sequence has been built consisting of *j* cells. The *j th* cell is characterized by probability P_j and positioned on the interval $[l_j, r_j]$. The histogram is thus inverted as follows;

$$\mathbf{H^{-1}(x)} = \sum_{\mathbf{j=1}}^{\mathbf{J}} \mathbf{I}_{[\mathbf{C_j - C_{j-1}}]}(\mathbf{x}) \left[\mathbf{l_j} + (\mathbf{x} - \mathbf{C_{j-1}}) \frac{\mathbf{w}}{\mathbf{P_j}}\right]$$

Where I_A is the indicator function of set A, $C_j = \sum_{i=1}^{J} P_i$ and $w_j = r_j - l_j$ is the width of cell j. The TEStool is used to generate the histograms and the innovation density for the three frame types. This is inputted into C++ code that is developed to frame size of each type [103].

A.2 Web Traffic Model

The web traffic model is modeled using M/Pareto ON/OFF model. It uses the Pareto distribution for the packet size distribution. Each web browsing download has Pareto distributed packet size. The packet size is thus determined as follows;

$$PacketSize = \min(P, m)$$

Where m is the maximum allowed packet size (m = 66666 bytes) and P is the normal Pareto distributed random variable with parameter α = 1.1 and k = 81.5 bytes. The probability density function of the packetsize is stated below [92];

$$f_n(x) = \begin{cases} \frac{\alpha . K^{\alpha}}{x^{\alpha+1}}, k \leq x < m \\ \beta, x = m \end{cases}$$

Where β is the probability that $x > m$ and can be determined as follows;

$$\beta = \int_{m}^{\infty} f_x(x) dx = \left(\frac{k}{m}\right)^{\propto}, \propto > 1$$