QUALITY OF SERVICE FOR BROADBAND SATELLITE INTERNET - ATM AND IP SERVICES

SASTRI L. KOTA

Department of Electrical and Information Engineering, Telecommunication Laboratory, University of Oulu

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Supervised by
Professor Pentti Leppänen
Professor Kaveh Pahlavan

Reviewed by
Professor Raymond L. Pickholtz
Professor C. K. Toh

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Department of Electrical and Information Engineering; Telecommunication Laboratory, University of Oulu, P.O.Box 4500, FIN-90014 University of Oulu, Finland
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Abstract

The current Internet infrastructure must be architected to handle future media-rich, and content rich applications. The success of applications such as video-on-demand, multicast and content distribution depends on Quality of Service and bandwidth guarantees. Over the years, the Internet has encompassed many changes in traffic profiles and applications, in bandwidths and utilization, but the future Internet infrastructure necessitates a very different architecture supporting Quality of Service (QoS). A satellite, distinguished by features such as global coverage, bandwidth flexibility, broadcast, multicast, and reliability, is an excellent candidate to provide broadband integrated Internet access.

The aim of this thesis is to explore suitability of satellite technologies for broadband Internet services with significant emphasis on the question of defining, assessing, and developing QoS models for satellite ATM and IP broadband networks with and without onboard processing. For the satellite Internet, Transmission Control Protocol (TCP) performance is degraded due to long propagation delays, link errors, and bandwidth asymmetry. In this thesis, for satellite ATM, fundamental questions such as buffer requirements, TCP/ATM efficiency, fairness, and multiple access are addressed through extensive simulations in a quantitative way. Buffer designs for TCP over satellite ATM Unspecified Bit Rate (UBR) service are performed. A buffer size equal to half the round trip delay-bandwidth product of the TCP connections provides high efficiency for TCP over satellite UBR. An extensive TCP analysis via simulation study for various TCP mechanisms and end system policies show that for satellite environment end system policies are more important than switch drop policies in terms of efficiency and fairness for World Wide Web traffic. A bandwidth allocation scheme is proposed and analytical model for supporting voice and video service over a broadband satellite network is developed. The study results demonstrate that non-contiguous allocation can afford higher gain in uplink utilizations.

In this thesis, for the first time, Integrated Services and Differentiated Services based QoS architectures for broadband satellite IP networks are proposed and analyzed. In multimedia applications where User Datagram Protocol (UDP) is used along with TCP, a fair excess bandwidth allocation is not possible because TCP is congestion sensitive whereas UDP is congestion insensitive. An extensive simulation model is developed to study the effect of precedence levels for reserved rate utilization and fairness with different buffer management policies. The simulation results indicate that three levels of precedence are required for better utilization. Multiprotocol Label Switching (MPLS) over Satellite network has been proposed and a simulation model developed to study the throughput performance impacts for TCP and UDP. The traffic engineering of MPLS facilitates efficient and reliable network design to optimize the utilization of network resources and enhance the network QoS.

A novel Code Division Multiple Access based Spread ALOHA single code multiple access scheme for broadband satellite return channel is proposed as an alternative to Multifrequency-Time Division Multiple Access based Digital Video Broadcasting-Return Channel via Satellite protocol. It is shown through Monte Carlo simulations that throughput for Spread ALOHA One Long Code equivalent to packet length, is better than Spread ALOHA One Code in which spreading sequence repeats every symbol. The reduction of throughput due to multi-user interference for different number of users is shown. Further research on QoS architectures, performance models for TCP enhancements, interworking functions, interoperability, and standardization efforts is included.

Keywords: DAMA, MPLS, QoS, satellite ATM, satellite IP, Spread ALOHA, TCP, UDP

To Master C.V.V.

Preface

Since the onset of the Internet, it has been increasingly clear to users, technologists, and service providers that maintaining a certain level of Quality of Service (QoS) is imperative to the growth and use of broadband applications. Satellite communication, distinguished by several characteristics such as global coverage, bandwidth on demand, flexibility, multicast, and broadband capability, is an excellent candidate to provide broadband integrated Internet access.

For satellite multimedia networks, both *connectivity network* with full end-to-end user connectivity and *access network* providing regional Internet access, QoS provisioning is a very critical element. Recently, the Internet Engineering Task Force (IETF) proposed some QoS for terrestrial networks, however, the QoS research for satellite and hybrid satellite/terrestrial networks for ATM and IP transport is almost in nascent stage.

The aim of this thesis is to provide QoS architectures, models and simulation results for satellites and investigate the possibilities for satellite use in global Internet and multimedia services. The research has been based on my experience in design and analysis of various satellite communication systems for commercial and military applications. Chapter 1 provides an introduction to satellite Internet, review of earlier work, thesis methodology and scope, contributions, and outline of the thesis. In chapter 2, a brief history and satellite technologies description and their suitability for Internet services is described. The use of satellite for ATM transmission including the analysis necessary answering fundamental questions such as buffer requirements, TCP/ATM, efficiency and fairness and multiple access modes with simulation results are provided in chapter 3. Several options for QoS provisioning over satellite links with semi-analytic and simulation models are compared.

Chapter 4 is devoted to the QoS model for satellite IP network. QoS architectures for satellite IP are proposed and extensive simulation for determining packet loss rate, efficiency, and rate utilization, fairness with and without reservation and throughput for TCP and UDP are provided. Traffic Engineering of MultiProtocol Label Switching for satellite network has been proposed and a simulation model has been developed to study the throughput performance impacts for TCP and UDP traffic flows. The International Standards (e.g. ITU, IETF, The ATM Forum, ETSI) definitions, processes and agreed objectives are included appropriately. Chapter 5 proposes a QoS model for an Internet access network where a satellite up/down link is part of the core network. Both the forward and return links are addressed. For the return link, CDMA based spread ALOH and variants are proposed and performance results are presented.

This thesis is expected to be useful to graduate students, research scientists, and practicing engineers interested in framework (concepts, architecture), models, and standards objectives for Quality of Service for satellite network with extensive references to the literature.

I am extremely thankful to Prof. Kaveh Pahlavan, Professor at the University of Oulu, and Professor of Department of Electrical and Computer Engineering and Director of Center for Wireless Information Network Systems, Worcester Polytechnic Institute, Worcester, my advisor of this thesis for initiating the process of working on the thesis at the Personal Mobile Indoor Radio Conference (PMIRC) 2000, London. I was fortunate to receive inspiring guidance to focus and frequent discussions on the research and details of the thesis. I sincerely appreciate the valuable time and close attention he has given me during my visits to the University of Oulu and throughout the process.

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I would like to appreciate Prof. John G Proakis of Northeastern University, Boston, my advisor during Electrical Engineer's Degree for his guidance when my initial research on Packet Broadcasting over Satellite Channels was performed. I also wish to thank Dr. Robert Metcalfe, the inventor of Ethernet, who served as a mentor and a source of inspiration to me when I started work on satellite networking as an alternative to Ethernet cable for communications.

I am grateful to the reviewers of this thesis, Prof. Raymond L. Pickholtz of George Washington University, Washington D.C, and Dr. C.K.Toh, Director of Research, TRW for taking time to carefully review and provide valuable suggestions, which are included in this thesis.

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I dedicate this thesis to my Guru (Spiritual Teacher) Master CVV for His invisible guidance and making this a reality!

Sastri Kota

Sunnyvale, California, U.S.A

1 November 2002

List of Contributions

This thesis is developed on the research articles published or submitted to journals, book chapter or international conferences/workshops and international standards organizations. Some of the publications have been prepared in co-operation with colleagues at Ohio State University where the author provided technical guidance and supervision to graduate students on satellite network research. The research builds over the author's contributions to real world Ka-band global satellite network and broadband access satellite network. The current thesis research is based on the following publications in the following areas:

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I Kota S (2002) Trends in Broadband Satellite Communication Networks. International Journal of Wireless Information Networks (submitted)

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Contents

1 Introduction

Emerging applications such as Internet Protocol (IP) multicast media streaming, content delivery distribution and broadband access are fueling Internet growth projections. These and other media rich applications require a network infrastructure that offers greater bandwidth and service level guarantees. Residential, small business and enterprise Internet users are already demanding high data rates and high quality services. An outcome of this demand were several access technologies varying from leased line to cable, Digital Subscriber Line (DSL), wireless, and satellite. As the demand for new applications increases, "best effort" service of the Internet will become inadequate and will result in lack of user satisfaction. Over the years, the Internet has encompassed many changes in traffic profiles and applications, in bandwidths and utilization in the number of domains and their degree of connections, but the future Internet infrastructure necessitates a very different architecture supporting Quality of Service (QoS).

A satellite communication network plays a significant role in supporting access to the Internet through a hybrid, satellite/terrestrial, network infrastructure. A satellite communications network is distinguished by several characteristics such as global coverage, scalability, broadcast capability, bandwidth-on-demand flexibility, multicast capability, and reliability; is an excellent candidate to provide broadband integrated Internet access. The current satellite systems operate in C and Ku frequency bands. Most of the proposed satellite network architectures use geostationary orbit (GSO), non-geostationary orbits (NGSO) and multi-spot beams at Ka-band frequencies.

The next generation satellite multimedia networks can be divided into two classes. The first is broadband satellite c*onnectivity network* in which full end-to-end user connectivity is established. The proposed global satellite connectivity networks such as Astrolink, Spaceway, and EuroSkyway have onboard processing and switching capabilities. On the other hand, regional a*ccess networks* like StarBand, IPStar, and WildBlue are intended to provide Internet access. These access systems employ non-regenerative payloads. The critical requirement is to provide high data rate Internet access, global connectivity, and provisioning of QoS within these next generation satellite network systems. For example, a user initiates a voice-over-IP call and expects the call to be intelligible. From a human point of view, call quality is subjective but objective measures of packet rate, delay, jitter, and loss are required for an intelligible call and must be supplied by the network.

Until recently, high speed Internet access has been limited to enterprises, using technologies such as leased T-1, frame relay or Asynchronous Transfer Mode (ATM). But with the growth of the Internet access for residential users, service providers have recognized the great opportunity in the residential, small and home office as well as enterprise broadband markets. Satellite ATM systems play a significant role in achieving global connectivity for both enterprise and residential services for future Ka-band systems. These systems will be able to achieve statistical multiplexing gains while maintaining QoS requirements. However, certain design challenges must be addressed before these systems can be deployed. For example, traffic management implementation in space versus ground segment meeting the weight and power requirements, is an important technical challenge. The International Telecommunications Union (ITU) and the ATM Forum are involved in developing satellite ATM Quality of Service (QoS) models.

Currently, most of the Internet applications use a Transmission Control Protocol (TCP)/Internet Protocol (IP) protocol suite. Although the TCP protocol was developed for terrestrial networks, a number of TCP enhancements have been proposed by the Internet Engineering Task Force (IETF) to accommodate the satellite specific link characteristics such as propagation delay, bandwidth asymmetry, channel impairments, and congestion.

The IP based broadband satellite network for both global connectivity and access must support the user QoS. There is a major effort by IETF in proposing QoS architectures to provide guaranteed service level to different applications. These architectures include, Integrated Services (IntServ), Differentiated Services (DiffServ) and Multiprotocol Label Switching (MPLS). The IETF proposed architectures mainly address terrestrial networks. There is an urgent need for developing Quality of Service architectures for broadband satellite network and identifying the challenges for realizing Ka-band satellite systems.

QoS approaches have been proposed to leverage the congestion controls of TCP currently used by all Internet traffic. Unfortunately, not all applications can reasonably make use of TCP with its elastic response to congestion. These are not particularly suited for real-time applications, which are built around User Datagram Protocol (UDP), Real-time Protocol (RTP) or recently Stream Control Transmission Protocol (SCTP). Specifically, QoS approaches must be studied for real time applications such as streaming video and audio over broadband satellite network.

Several technical challenges exist for broadband satellite network to support high quality, high-speed Internet access. Solutions for various system architecture options range from a simple repeater to a sophisticated satellite with on-board processing and switching (ATM or fast packet) with multiple spot beams and intersatellite links. QoS architectures for those satellite networks supporting multimedia services must be developed.

In this thesis, some of the technical challenges for broadband satellite networks supporting ATM and IP technologies are analyzed. Specifically, technical issue analysis, buffer models for ATM networks, analytical Demand Assignment Multiple Access (DAMA) model for bandwidth allocation are studied. Moreover, extensive simulation studies for TCP/UDP traffic for satellite IP with differentiated services are performed.

This thesis also proposes IntServ, DiffServ, and MPLS based QoS architectures for satellite based IP networks. A new Spread ALOHA based protocol is defined and analyzed for return channel of the broadband satellite Internet access network.

1.1 Satellite Internet Access

Satellite communication networks have evolved from broadcast systems, beam switched transponder systems and to fast packet or ATM switched onboard processing systems. The traditional broadcast architecture of most commercial satellites yield bandwidths of 1 to 2 GHz or less. The new applications demand an effective bandwidth increase by an order of magnitude within the next few years. The mass of satellites has been growing at the rate of 8% per year and communication payloads have been getting more efficient capacity compounded over 35 years, to well over two thousand [1]. The future satellite system designs employ multiple spot beams achieving maximum frequency reuse. Satellite systems can become the imbedded infrastructure in regions with less developed communication infrastructure and complement the Internet service over other technologies such as cable, DSL, fiber, and wireless. It is expected that the market for Content Delivery Distribution via satellite network will grow up to $4 billion by 2006 [2].

The satellite based Internet has several architectural options due to their design alternatives. In a broad sense these are classified into two major categories: connectivity and access networks.

Connectivity network

In connectivity satellite networks, user-to-user connectivity is established via satellite onboard routing. This type of broadband service can be used to provide Internet access avoiding some degree of ground infrastructure. Bandwidth utilization efficiency can be gained through the use of onboard processing and ATM or fast packet switching or even optical switching in the distant future. As a consequence, the complexity and demand for satellite resources can be much higher than that needed for an access satellite. Examples of broadband satellite connectivity networks such as Spaceway, Astrolink, EuroSkyWay, Teledesic, Intelsat, Eutelsat are described in Section 2.4.7.

Access network

Access networks on the other hand, have been evolved with the new application of broadband interactive connectivity to the Internet. The network requires a forward link from the network gateway to the user and the return link from the user to the network gateway. These two links are highly asymmetric and the links may have totally different characteristics. The bandwidth allocation must accommodate two links per user. Frequency Reuse is employed and different frequency bands can be used for the user and gateway links. Examples of satellite access networks include StarBand, WildBlue, iPStar, Astra-BBI, Cyberstar as discussed in Section 2.4.8.

To meet the future application requirements delivering guaranteed Quality of Service levels, research is required to develop new satellite network infrastructure, and investigate Quality of Service mechanisms for broadband satellite networks supporting ATM and IP services, which forms the main focus of this thesis.

1.2 Review of Earlier Work

The Internet which is a *network of networks* incorporates concepts of packet switching employed in the ARPANET [3]. The Internet has received tremendous success providing services ranging from shopping, finding local entertainment to children's homework. The growth of the Internet and the new applications result in significantly higher data rate requirements and new global network infrastructure. The estimated bandwidth in 2005 ranges from 155 Gb – 2.4 Tb for metro networks and 2.4 Tb – 10 Tb for core, as opposed to the current requirement of 2.4 Gb – 10 Gb [4]. Many of the observations show that 75% of traffic on the Internet is web based and there are 3.6 million websites with 300-700 million web pages; the traffic consists of 80% data and 20% voice with a traffic growth of 100-1000% per year [5]. To meet such bandwidth demands the networking technologies range from ATM, frame relay, IP satellite to optical backbones. The access technologies can be dial up, cable, DSL, wireless, and satellite [6].

Due to the several advantages of satellite communications, satellite Internet has attracted several researchers. The Internet 2 project with a partnership of over 130 U.S. Universities, 40 corporations and 30 other organizations has been initiated [7]. The objectives of such a network are to develop scalable, interoperable network to provide QoS support to evolving applications such as distance learning, network collaboration and scientific visualization.

Satellites are used to interconnect heterogeneous network segments and to provide ubiquitous direct Internet access to home and businesses. A satellite Internet architecture is presented and multiple access control, routing, and satellite transport for a global Internet is discussed [8]. [9] analyzes the performance of VSATs for Internet access under a variety of multiple access architectures SCPC/DAMA, CDMA, ALOHA, MAMA, and SAMA. Architecture for interactive service delivery to the home via a satellite based digital broadcast system, has been developed [10]. [11] from NASA describes the status of the Internet for near earth applications and the potential extension of the Internet for use in deep space planetary exploration. The overall challenges of implementing the space Internet and the integration into the terrestrial today's Internet are described.

Current state of the art of broad range of wireless technologies including fixed and mobile, terrestrial and satellite systems and projection in the next decade are discussed in [13]. [14] provides a discussion on multimedia technologies for wireless networks in the third millennium, including systems, standards, and technologies. A qualitative definition of Internet Quality of Service (QoS) and framework methods of QoS delivery and QoS classes are discussed in [15, 16, 17, 18].

Satellite communications historical perspective [20, 21, 22], and background [23, 24, 25, 26], are well documented in the literature. An overview of mobile satellite communications systems covering orbits, multiple access techniques [27], and brief characteristics summary of the current mobile systems, Iridium [28], Globalstar [29], ICO [30], and Inmarsat [31] are described. [32] provides a good description of the use of satellites for personal communication systems. Iridium and Globalstar systems use LEO constellations. ICO uses a MEO constellation.

The growing congestion in C and Ku bands, and the success of the ACTS program increased the interest of satellite system developers in the Ka-band broadband satellite networks for growing Internet access applications [33, 34, 35, 37, 38, 39]. LEO satellite systems have been proposed for personal communication [40, 41, 42]. However, several networking challenges such as MAC protocol, handover, QoS, and interoperability issues need further research.

Regarding the broadband satellite network architecture, selection of onboard processing and switching or non-regenerative payload is very critical for system cost and performance. Most of the global Ka-band satellite networks connecting end user-to-user directly plan to use onboard processing and ATM or packet switching. Throughput analysis [43], processing payload design for Ka-band system [44, 45], and satellite communication network for Internet access [46, 47] are well discussed in the literature. A GEO-LEO SATCOM system at Ka-band, for maximum data rate of 1.2 Gbps is proposed [48].

A number of Ka-band satellite networks are under development, except Astrolink, which is on hold currently. Most of these systems, Astrolink [49], Spaceway [50], Teledesic [51], EuroSkyway [52, 53], Intelsat [54], and Eutelsat [55] are to be deployed to provide high speed Internet access among the other broadband applications. [56] provides a generic ground segment for such a broadband satellite network.

As opposed to global satellite networks (connectivity networks), the regional broadband satellite networks also known as broadband access networks use non-regenerative payloads and the proposed systems use GEO satellite configuration. The systems include Starband [57], Wildblue [58], iPStar [59], Cyberstar [60], and Astra-BBI [61], for Internet access via satellite.

The regional broadband access systems are designed to provide network connection for interactive terminals via satellite and ground based hub/gateway. The DVB/MPEG-2 format carries up to 45 Mbps in the forward link and multi-frequency time division multiple access allows up to 2 Mbps in the return direction. For the return channel protocol, ETSI has developed a standard [62, 63, 64]. Alternatively, Wildblue is planning to use modified Data over Cable Service Interface Specification (DOCSIS) [65] for the return channel. Most of these protocols have to implement QoS models for providing user service level guarantees.

To mitigate the propagation effects on the Ka-band satellite networks, rain attenuation model [69], propagation measurements [70], ACTS propagation experiments and ITU propagation Data and Prediction methods [71, 72] are well documented. Uplink power control in Ka-band satellite links [73] is described. NGSO interface protection requirements using adaptive coding as fading counter measures are described in [76]. Turbo coding to provide coding gain improvements are discussed in [78, 79, 80].

Media Access Control (MAC) is one of the important issues to be researched for broadband satellite networks. During the past several years, multiple access and demand assignment multiple access and dynamic bandwidth allocation algorithm have been studied extensively and results were presented in the literature. However, for broadband satellite networks, for global and regional systems, MAC protocol has to be standardized. Several proprietary versions of MAC protocol exist for satellite systems under development. A comprehensive performance comparison of various MAC protocols are reported in [81, 82, 83, 84, 85, 86, 87, 88, 89, 90]. A comparison of DAMA techniques for circuit and packet switched systems [91] and a reservation DAMA for a satellite ATM [92] are described.

Transmission Control Protocol (TCP) was originally designed for links with low error rates and the packet losses are assumed to be entirely due to network congestion. As a result, the sender decreases its transmission rate. The throughput decreases if the packet loss occurs due to link errors. This problem of TCP enhancements has been studied extensively. Recent survey articles provide excellent discussions on them [93, 94, 95, 96]. The different TCP enhancements which are well documented include, Initial Window [97, 98], delayed acknowledgments [99], byte counting [98], TCP Vegas [100], TCP New Reno [101], TCP SACK [102, 103, 104], TCP FACK [105], Window Scaling [106], T/TCP [107], and Path MTU directory [108]. [109] provides discussion on TCP slow start, congestion avoidance, fast retransmit, and fast recovery algorithms. The explicit congestion notification is reported in [110, 111].

Due to satellite link impairments such as large propagation delay, link errors, and link asymmetry, TCP throughput performance degrades. To enhance the TCP throughput performance, for large bandwidth-delay product environment such as satellite link, many solutions have been proposed. The TCP performance enhancement techniques for satellite links are extensively studied in [112, 113, 114, 115, 116, 117]. The IETF has proposed standard mechanisms for studying the TCP enhancement over satellite channels [118]. Other studies for TCP performance over satellite network include [119, 120, 121, 122, 123, 124, 125, 126]. Performance Enhancing Proxies to mitigate link related degradations attracted the most interest for satellite network implementation [127, 128, 129, 130, 131]. Research issues for TCP in satellite IP network have been addressed [132]. Satellite Internet via Low Earth Orbit satellite is addressed [133]. Solutions for network path asymmetry just like in satellite access networks with ratios of 10:1 or greater for forward to return links are well documented [134, 135]. Extensions of TCP for space communications have been studied [136, 137]. The TCP performance issue over wireless networks has been reported [138].

With the deployment of ATM technology, due to its characteristics of supporting diverse QoS requirements for a variety of traffic sources and providing flexible transport, attracted satellite network designers. As shown in Table 2.5, some of the future satellite systems propose to support ATM technology, with onboard ATM switching. Currently, an extensive documentation on the ATM technology is available [139, 140]. The principal issue of traffic management and congestion control was addressed well enough in [141, 142, 143, 144, 145, 146, 147] and [148] provided the ATM Forum Traffic Management Specification 4.0. Wireless ATM Working Group of the ATM Forum studied reference architectures for satellite ATM [149, 150, 151, 152]. [155, 156, 157] provide ATM network architectures study for multimedia applications. [158] surveys satellite ATM networks. Traffic and congestion control mechanisms for satellite ATM are addressed in [159, 160, 161]. Adaptive MAC protocol has been studied in [161]. The performance and buffering requirements for TCP over ATM ABR and UBR services have been well addressed in [165, 168]. The performance of TCP over ABR with long-range dependent UBR background traffic over terrestrial and satellite network are studied in [168]. [169] provide a simulation study of Guaranteed Frame Rate for TCP/IP.

The performance objectives for B-ISDN ATM layer recommended by ITU-R are given in [170]. The ITU-R-recommended availability objectives for B-ISDN ATM transmission for FSS GEO satellite system are given in [171, 172].

ITU-T recommends network performance objectives for IP based service, defining six classes of service [174]. Many of the applications for real systems under development have to be mapped into these objectives. Congestion control mechanisms and the best effort model for IP networks are addressed in [175, 176]. Congestion control in computer networks, in particular, end-to-end congestion control mechanisms, are studied in [177, 178, 179, 180]. [181] presents a fundamental binary feedback scheme for congestion avoidance in computer networks.

The issue of QoS has been addressed by the IETF, which proposed several mechanisms e.g., Integrated Services (IntServ), Resource Reservation Protocol (RSVP), Differentiated Services (DiffServ), and Multiprotocol Label Switching (MPLS). [182, 183] describe QoS for IP networks. [184, 185] specify guaranteed Quality of Service. IntServ is described in [186, 187, 188] and Resource Reservation Protocol (RSVP) in [189]. [190, 191, 192] describe the DiffServ architectures, Assured Forwarding per Hop Behavior (PHB) [193], Expedited Forwarding PHB [194] are specified by the IETF. [195] provides an architecture for DiffServ and Traffic Engineering. Queue management issues are studied in [196, 197, 198, 199] and [200, 201] describe the policy based QoS management. The ATM Forum specifies IP Differentiated Services over ATM [202]. [203] describes restorable dynamic QoS routing.

The significant issue of delivering QoS over satellite IP has been addressed by a few [204, 205, 206, 207]. QoS provisioning over mitigated TCP/IP over satellite links has been studied [208, 209]. The IETF recommends several performance issues for links with errors [210]. QoS issue over wireless networks has been addressed in [211, 212]. The results of Multilevel Explicit Congestion Notification in satellite network has been reported in [213].

The IETF proposes MPLS and traffic engineering to achieve guaranteed service levels in IP networks. [214, 215, 216, 217] present MPLS architecture, technology and applications. [218] defines MPLS support for Differentiated Services. MPLS and traffic engineering is documented in [219, 220, 221]. These articles in the literature describe application of MPLS and traffic engineering for terrestrial networks. However, QoS delivery mechanisms applied to satellite networks have to be explored fully realizing QoS for the future Ka-band satellite systems.

1.3 Thesis Methodology and Scope

The growing demand for multimedia applications on the Internet provides a major opportunity for broadband enterprise and residential access network services. Applications which require large bandwidths and guaranteed Quality of Service (QoS). Currently, the core networking solutions provide bandwidth up to OC-192 and future networks will employ OC-768. The broadband core offers more bandwidth than is necessary today, whereas there is a bandwidth bottleneck resulting in congestion on the access end. The future networks would realize simultaneous proliferation of new bandwidth hungry applications. To meet such demands and provide QoS guarantees over a hybrid terrestrial/satellite networks, QoS management is extremely critical. This involves the service providers, satellite operators, and network providers. The ITU is developing standards for satellite with ATM switching. For broadband satellite IP networks, the QoS standardization by ITU-T and ITU-R, European Telecommunications Standards Institute (ETSI), and IETF is still in the planning stage. Hence, a considerable research for satellite IP QoS architectures and models must be undertaken. The aim of this thesis is to develop Quality of Service architectures for broadband satellite ATM and satellite based IP networks to support high bandwidth demanding future applications with differentiated service levels.

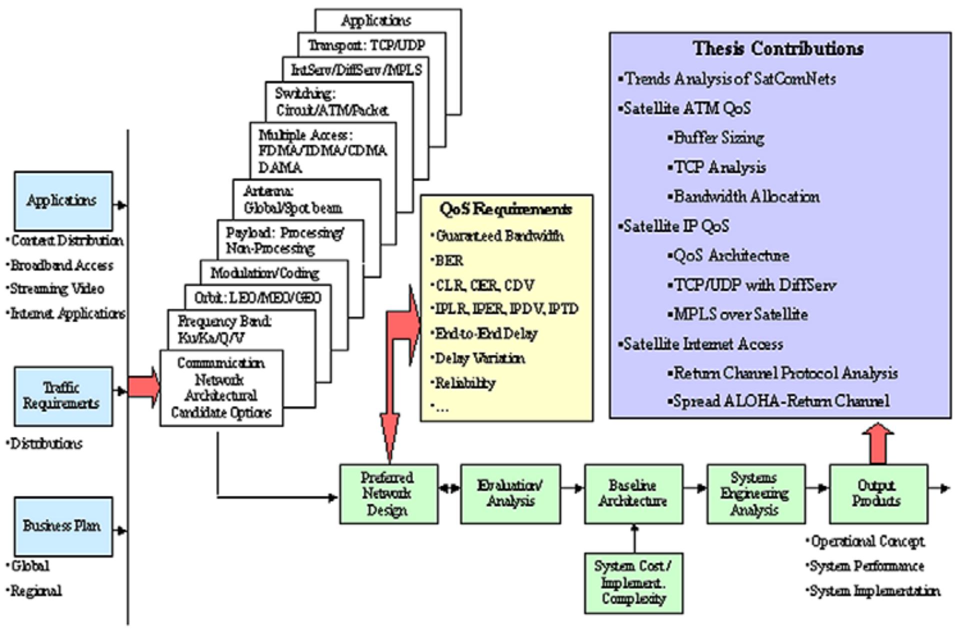


Figure 1.1: Satellite Network System Engineering Methodology - Thesis Contributions

Figure 1.1 shows Broadband Satellite Network design methodology that will satisfy the rapidly evolving application needs, and requires a system engineering and business analysis approach. The process is iterative and begins with application and traffic requirements with business plan either for global or regional service. The capacity plans, and traffic analysis results provide the inputs to decide network architecture.

Architecture Candidates: Several options are available for choosing a baseline satellite network architecture. The architecture is based on the following selection:

- Satellite orbits – Geostationary Earth Orbit (GEO), Medium Earth Orbit (MEO), Low Earth Orbit (LEO)
- Physical layer – Advanced Modulation, Encoding (Concatenated, Turbo)
- Payload – Processing, non-Processing
- Switching – circuit, cell, packet
- Multiple access at Link layer – FDMA, TDMA, CDMA, DAMA
- QoS architecture for Network layer – IntServ, DiffServ, IntServ/DiffServ/aggregate RSVP, MPLS
- Transport layer – TCP/UDP/RTP

The QoS requirements for either ATM or packet switching, provide the inputs to candidate analysis. The design process involving trade studies and analysis, results in systems engineering products, e.g., operational concept, system performance specification and test, and evaluation plans.

The above methodology has been applied by the author during the design process of several real satellite communication networks, for both military and commercial applications. The specific contributions of this thesis are discussed in the next section.

1.4 Contributions of the Thesis

The objectives of this thesis are to develop Quality of Service requirements and Quality of Service architectures for Broadband Satellite based Internet access for global and regional services. Chapters in the rest of the thesis present the specific contributions. The contributions are briefly listed below.

Trends analysis of satellite Internet architectures and technical challenges.

A comprehensive analysis covering different architectural options, technical challenges such as modulation, coding, and multiple access, demand assignment multiple access, Quality of Service models, TCP enhancements, return channel protocols, and traffic management are given. An account of the current mobile satellite communications, future Ka-band global and regional satellite networks is discussed. This study assumed both ATM and IP transport over communication satellites. This section is based on the results of the papers I, II, III, IV, V, VI, VII, VIII, XII, XIII, XVI, XVII, XVIII, XIX, XX, XXV, XXX, XXXI, XXXIV, XXXVII, XL, XLI, XLII, XLIII, XLV, XLVI, XLVII, XLVIII, LVI, LVII, LVIII, LIX, LX and LXI.

Buffer requirements for TCP over satellite ATM supporting UBR service.

A simulation study for the buffer requirements for TCP over ATM based satellite networks is performed. Among the service categories provided by ATM networks, the most commonly used category for data traffic is the Unspecified Bit Rate (UBR) service. UBR allows sources to send data into the network without any network guarantees or control. The results of this section are presented in papers I, IX, XIII, XXI, XXII, XXIII, XXIV, XXV, XLIX and L.

TCP enhancements analysis for satellite ATM UBR+ service.

In recent years, there has been a tremendous growth in the amount of World Wide Web (WWW) traffic over the Internet. WWW traffic is essentially bursty in nature with periods of activity. The traffic pattern for a large number of WWW connections is expected to be different from that generated by the persistent TCP traffic. An extensive full factorial simulation study is performed for WWW traffic model; TCP flavors of Vanilla, Reno, New Reno, and SACK; UBR+ drop policies of Early Packet Drop (EPD) and Selective Drop (SD); buffer sizes of 0.5, 1, and 2 times the round trip bandwidth-delay products; and LEO, MEO and GEO satellite configurations with different propagation delays. The study includes results presented in the papers IX, X, XI, XIV, XV, XXXI and XLIV.

Bandwidth allocation (DAMA) for satellite ATM.

In a broadband satellite ATM network, each user terminal requests for time slots on the uplink channel to utilize bandwidth on Demand Assignment Multiple Access (DAMA) basis. An analytical model is proposed to quantify the gain in uplink utilization due to the non-contiguous allocation for CBR and VBR services. A simulation model is developed for contiguous allocation and the results for blocking probability Vs. offered load showing up to a ten percent increase in channel utilization for non-contiguous case are presented. The study includes the results presented in papers XXXV, XXXVIII and XXXIX.

Quality of Service architectures for satellite IP networks.

Currently, the IETF has proposed different QoS mechanisms including Integrated Services (IntServ) with Resource Reservation Protocol (RSVP), Differentiated Services (DiffServ) and Multiprotocol Label Switching (MPLS). In this section, three new QoS architectures for satellite based IP networks are proposed. These architectures are IntServ, DiffServ, and IntServ/DiffServ and aggregate RSVP based, supporting Internet over satellite. The results of this proposed architectures are presented in papers XXV and LII.

DiffServ based QoS simulations for TCP and UDP over satellite IP.

The TCP and UDP protocols form the basis of the current Internet. Satellite Internet is expected to support applications based on TCP and UDP, e.g. data transfer and audio and video streaming, traffic transport over broadband satellite network. Simulation based study for congestion sensitive TCP and congestion insensitive UDP performance over satellite network are included. Queue management policies such as Random Early Detect (RED) and Selective Drop policies are included in the simulations. The results of these simulations are presented in papers XXVIII, XXIX, XXXII, LIII and LIV.

Multiprotocol Label Switching (MPLS) over Satellite Network.

The IETF is developing MPLS architecture and traffic engineering requirements. For the first time, an MPLS-based satellite IP network is proposed. Using the traffic engineering approach, simulation model for TCP and UDP protocols is developed to show the network throughput improvement with proper traffic engineering. The results are presented in paper XXVI.

MF-TDMA satellite Internet access return channel protocol analysis.

The 2-way interactive satellite Internet access network presents a network asymmetry environment. Two Multi Frequency – Time Division Multiple Access (MF-TDMA) based protocols for return channel standard protocols are proposed by ETSI/DVB-RCS and Cable Labs. However, a proper selection of the return channel protocol for Internet access is extremely critical for Internet access system design. In this contribution, the MF-TDMA based return channel protocols are analyzed. The analysis results are presented in papers I, II and XXXIII.

New CDMA based Spread ALOHA Return Channel protocol.

A new CDMA based Spread ALOHA return channel protocol for satellite Internet is proposed and analyzed. The spread ALOHA with single code multiple access repeats the code or spread sequence every symbol. On the other hand code reuse multiple access or Spread ALOHA One Long Code uses one long code as long as the packet. For the first time the performance analysis of these spread ALOHA protocols for return channel are analyzed using Monte Carlo simulations. The results of this contribution are presented in papers XXVII, XXXIII and XXXIV.

1.5 Outline of the thesis

New Quality of Service architectures for broadband satellite network supporting Internet for various media rich applications have been investigated. The major emphasis has been in developing (a) technical challenges and trends analysis for broadband satellite network design, (b) TCP performance enhancements, with optimal buffer designs, (c) full factorial analysis for satellite networks with ATM switching, and (d) new QoS architectures based on IntServ, DiffServ, and IntServ/DiffServ for the next generation satellite IP networks to support Internet access. A full factorial analysis has been proposed for studying TCP and UDP performance for GEO, MEO and LEO satellite configurations in terms of reserved rate utilization and fairness. A novel MPLS architecture for MPLS over satellite network has been proposed and throughput improvements with traffic engineering for TCP and UDP have been demonstrated.

For broadband access networks, such as Starband, IPStar, MF-TDMA based return channel protocols are analyzed. A new CDMA based Spread ALOHA Multiple Access protocol for broadband Internet access return channel is proposed and throughput performance is analyzed.

Chapter 2, results of which have been presented in I, II, III, IV, V, VII, XII, XIII, XVI, XVII, XVIII, XIX, XX, XXV, XXX, XXXI, XXXIV, XXXVIII, XL, XLI, XLII, XLIII and XLVIII, provide an overview and a comprehensive trends analysis of the architectural issues and technical challenges for broadband satellite networks to support high speed internet access applications. A brief description of the existing, and future global and regional satellite network is presented. The contribution covers both ATM issues and IP QoS challenges for the satellite networks. Some of the discussions of this section serve as background for the rest of the chapters.

Chapter 3, results of which are presented in papers I, IX, X, XI, XIII, XIV, XV, XXIII, XXXI, XLIV, XLIX, L and LI, presents TCP enhancements, buffer designs for traffic management, and end system policies i.e., EPD and SD, for various TCP flavors e.g., Reno, New Reno, and SACK, analysis of variation for GEO, MEO, LEO satellite configurations. The Quality of Service requirements standardization for the satellite supporting ATM transport is presented.

Chapter 4, results of which have been documented in papers I, III, XXV, XXVI, XXVIII, XXIX, XXXII, XXXIV, LII, LIII and LIV, presents Quality of Service Requirements, QoS architecture candidates based on IntServ, DiffServ, IntServ/DiffServ with Aggregate RSVP for IP based satellite networks. Simulation models for QoS for TCP and UDP traffic supporting non-real-time and multimedia application over Internet with Differentiated Services over GEO, MEO, and LEO satellite network configurations are developed. Guaranteed bandwidth allocation and fairness are analyzed in terms of drop precedence levels and congestion management with Random Early Detection (RED). A novel MPLS over satellite network model is proposed and simulation model is developed to demonstrate the throughput improvements with traffic engineering for TCP and UDP traffic.

Chapter 5, results of which are presented in papers I, II, XXV, XXVII and XXXIII, presents various broadband IP satellite return channel protocol analyses for a proper selection of the broadband Internet access architecture. A CDMA based spread ALOHA multiple access protocol using single long code, equivalent to packet length, is proposed and analyzed. Monte Carlo simulations are developed to study the performance and impacts of the interference for broadband access systems.

Chapter 6 concludes the thesis. The main results and contributions of the thesis are summarized. The future Quality of Service research and open problems to support high speed Internet with broadband satellite are discussed.

2 Trends Analysis of Broadband Satellite Networks

Traditionally, satellite communication systems have played a significant role in supporting services such as TV broadcasting, digital messaging, enterprise Virtual Private Networks (VPNs) and point-to-point telecommunications and data services. The recent Internet's growth has resulted in new generation of applications with higher throughput requirements and communication demands. Service providers, network and Internet access providers are faced with a challenge to meet the higher capacity access to the end user and wider service offerings. Satellite communication network systems can be optimized to meet such new service demands. New architectures and networking concepts, designs and implementations based on performance-cost tradeoff studies must be developed. Many issues and technical issues must be resolved for a broadband satellite network design. Some of the questions include:

- What are the driving applications?
- What is the right network architecture based on frequency Ka/Ku or Ku/Ku or Ka/Ka?
- What kind of payload architecture e.g. non-processing or processing should be used?
- What type of onboard switching if used with on-board processing e.g. circuit, ATM or fast packet to be used?
- What type of antenna system, global beam or spot beam, should be used?
- How to provide guaranteed QoS? Do the QoS models developed for terrestrial networks, work for satellite networks?
- What network control and management protocols should be used?
- What TCP/IP enhancements are to be used or whether or not split gateways to be used?
- How do the UDP/IP work for multimedia traffic over satellite?

This chapter addresses some of the above questions reviewing the existing and the future broadband satellite Internet service.

Problem: Broadband Satellite Network architecture: Trends and technical challenges analysis.

> There is no single source in the literature covering an overview of the existing satellite network design problems and technical issues. Therefore, this section provides a comprehensive analysis of the design issues with possible solutions for broadband satellite networks.

Section 2.1 describes a global communication model followed by emerging applications and services in Section 2.2. A Quality of Service (QoS) definition, need for QoS, and possible QoS methods are described in Section 2.3. An overview of satellite communications with characteristics, advantages/disadvantages, frequency (Ka, Ku), orbits (LEO, MEO, GEO), and technology advances is described in Section 2.4. Current mobile communication satellites are described in Section 2.4.5. Section 2.4.6 describes next generation Ka-band satellite network architecture options. Section 2.5 describes technical challenges including physical layer, link layer, and analysis of TCP enhancements for satellite links. Section 2.6 describes the technical issues for satellite ATM and Section 2.7 presents the problems with satellite IP network design. The purpose of this chapter is also to provide sufficient analysis background to the rest of the thesis.

2.1 Global Network Model

The tremendous success of the Internet and the new applications have resulted in significantly higher data rates requirement. One of the key requirements for the emerging "global network" which is a "*network of networks*" is rich connectivity among fixed as well as mobile users. Advances in switching and transport technologies have made increase in transmission bandwidth and switching speeds possible, and still more dramatic increases are achievable via optical switching. The future generation of communications networks provides "multimedia services", "wireless (cellular and satellite) access to broadband networks" and "seamless roaming among different systems".

Figure 2.1 shows a global communication network scenario providing connectivity among corporate networks, Internet, and the ISPs. The networking technologies can vary between ATM, Frame Relay, IP and optical backbones. The access technologies options can be dial up, cable, DSL, and satellite. [12]

Mobile communications are supported by second generation, digital cellular (Global Satellite for Mobile Communications, GSM) and, data service by Generic Packet Radio Services (GPRS). Third generation systems such as IMT-2000 can provide 2 Mbps and 144 Kbps indoor and vehicular environments. Even fourth and fifth generation systems are being studied to provide data rates 2-20 Mbps and 20-100 Mbps respectively.

Several broadband satellite networks at Ka-band are planned and being developed to provide such global connectivity for both fixed satellite service (FSS) and mobile satellite service (MSS) using Geosynchronous (GSO) and Non-Geosynchronous (NGSO) satellites. Currently GSO satellite networks with Very Small Aperture Terminals (VSATs) at Ku-bands are being used for several credit card verifications, rental cars, banking applications. Satellite networks such as StarBand, DIRECWAY, and WildBlue are being developed for high speed Internet access. See Section 2.4.8.

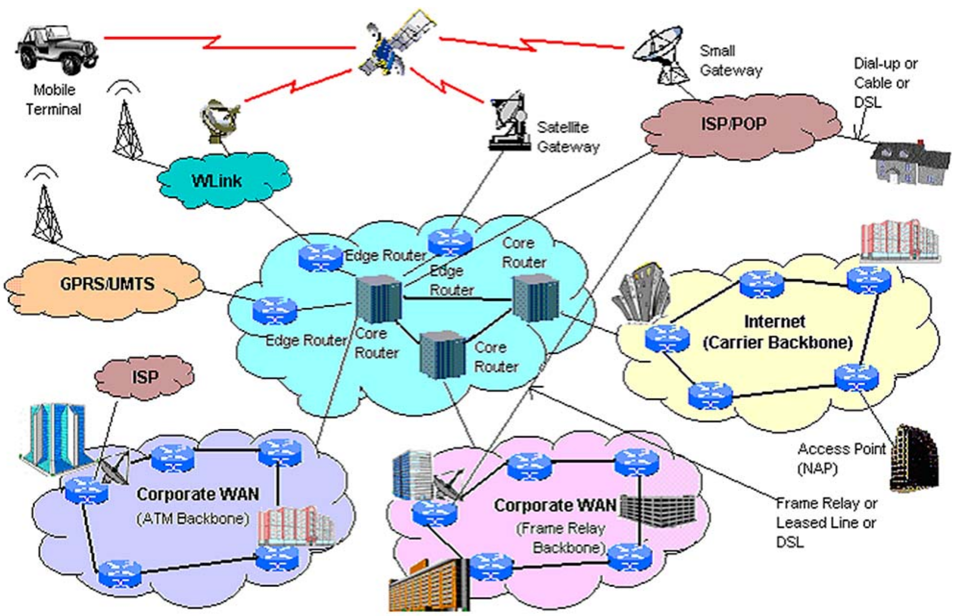
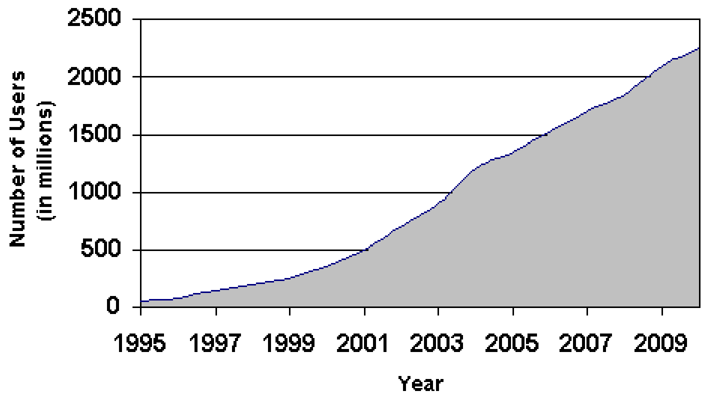


Figure 2.1: Communication Network Scenario

2.2 Emerging Applications and Services

Figure 2.2 shows a projected growth of worldwide Internet users, where most users have demand for broadband services. In the consumer market the growing awareness of the Internet and activities ranging from shopping to finding local entertainment options to children's homework are driving the steep demand for more bandwidth. Education and entertainment content delivery has become one of the prime applications of Internet. During business globalization an increase of virtual business teams, enterprises, increase in competition for highly skilled workers, service providers and equipment vendors, are driving the demand for higher bandwidths or broadband.

Some of the observations are, 75% of traffic on the Internet is web-based; there are 3.6 million websites with 300-700 million web pages; the traffic consists of 80% data and 20% voice with a traffic growth of 100-1000% per year [5].



Source: Nua Internet Surveys + vgc projections

Figure 2.2: Internet Users

2.2.1 Broadband Services

Table 2.1 shows an example of the broadband services and applications. These include entertainment, broadband and business services. A major challenge for these emerging services supported by Internet which is an IP-based network, is to provide adequate QoS. The normal QoS parameters from a user's perspective, both for enterprise as well as residential users include, throughput, packet loss, end-to-end delay, delay jitter and reliability.

Table 2.1: Broadband Services

Entertainment	Broadband	Business	Voice and Data Trunking
Broadcasting (Direct-to-Home, DTH)	High speed Internet access for consumer and enterprise	Telecommuting	IP Transport and ISP
Video on Demand (VOD)	Electronic Messaging	Video Conferencing	Voice over IP
Network or TV distribution	News on Demand	E-finance, B2B	Video, audio and data file transfer
TV-Co-Transmissions	Multimedia	Home Security	
Karaoke on Demand	Distance Learning	Unified Messaging	
Games	MAN and WAN connectivities		
Gambling	Telemedicine		

2.3 Quality of Service Framework

This section describes the need for Quality of Service and QoS framework including definition, parameters, and challenges.

2.3.1 The Need for QoS

The Internet Protocol (IP) has enabled a global network between an endless variety of systems and transmission media. It is being used for email exchange and web browsing as a part of everyday life around the world. These new networks bring in new applications some of which are multimedia and require significant bandwidth and some others have strict timing requirements, or function one-to-many or many-to-many (multicast). These require network services beyond the simple "best-effort" service that IP delivers.

Internet has also evolved through many changes in traffic characteristics and applications and in link bandwidths and utilizations, as opposed to a simple telephone system. Even though the transmission bandwidths increased from speeds of kilobits to several hundreds of megabits and even gigabits, still some of the residential users receive speeds less than 56 Kbps. A proliferation of new bandwidth hungry applications is expected in future. Periods of congestion on Internet will increase more than what is experienced today. A given link or router of the Internet will carry thousands and millions of sessions from a wide variety of applications whose service levels of (satisfaction) requirements are different. Increase in bandwidth will not fully alleviate the problem of congestion. Methods of traffic management, congestion control and Quality of Service approaches become an important issue in future.

IP's best-effort service can satisfy any application requirement is true assuming the network's bandwidth capacity is sufficient to avoid any delays or dropped datagrams. Internet traffic increases in proportion to available bandwidth as fast as it is added, so delays are inescapable. When Internet traffic is very high, congestion occurs thus disabling sections of the Internet. Delays can vary enough to adversely affect applications that have real-time constraints. To provide service guarantees, IP services must be supplemented with some added features to the nature that can differentiate traffic and enable different service levels for different users and applications.

2.3.2 What is QoS?

Quality of Service is the ability of a network element (e.g. an application, host or router) to have some level of assurance that its traffic and service requirements can be satisfied. It requires the cooperation of all network layers from top-to-bottom, as well as every network element from end-to-end. QoS manages bandwidth according to application demands and network management settings. The bandwidth allocated to an application in a resource reservation is no longer available for use by "best-effort" applications. QoS-enabled high-priority applications must not disable the low-priority Internet applications.

According to ISO 8402, QoS is defined as "the totality of characteristics of an entity that bear on its ability to satisfy stated and implied needs." ISO 9000 defines quality as the degree to which a set of inherent characteristics fulfills requirements. ITU-T Recommendation E.800 defines QoS as "the collective effect of service performance which determine the degree of satisfaction of a user of the service" [16]. The end-to-end network performance includes the access network performance and the core network performance. Other ITU-T Recommendations, such as ITU-T I.350 and ITU-T Y.1540 have developed network performance and network interface-network interface QoS. Recently ITU-T Y.1541, has developed 6 classes of applications [174] and provides application QoS objectives as discussed in Section 4.1.

The different viewpoints of QoS are customer's QoS requirements, QoS offered by service provider, QoS achieved by service provider, and QoS perceived by customer. Based on the customer's QoS requirements, QoS offered and achieved by the service provider will be different from the QoS perceived by the customer. A customer's QoS parameters are focused on user perceived effect, and do not depend on the network design. The parameters might be assured to the user by the service providers through contracts.

QoS offered by the service providers is a statement of the level of quality expected to be offered to the customer by the service provider for planning and Service Level Agreements (SLA). Each service would have its own set of QoS parameters. For example, service providers may state that the availability of basic telephony service is planned to be 99.995% in a year with not more than 1 minute break at any one occasion and not more than 3 breaks over the year.

QoS achieved or delivered by the service provider is a statement of the level of quality actually achieved and delivered to the customer. For example, in the previous quarter, availability was 99.97% with 5 breaks of service of which one lasted 55 minutes. QoS perceived by the customer is expressed usually in terms of degrees of satisfaction, through customer surveys. For example, rating of 3 on a 5-point scale indicating excellent service. The service provider may not be in a position to offer customers the level of the required QoS. Ideally all these viewpoints are to be converged for a given service.

An essential aspect of QoS for mobile systems in LEO or MEO satellite networks, is the handover QoS [19]. As the mobile host migrates from one satellite coverage to another, a connection can be forced due to the failure of mobile handover. These failures are characterized by handover blocking probability. This results due to (a) inability to provide the desired QoS in the new satellite (b) failure of a gateway (c) inability to handover in time and (d) failure of satellite. In a satellite ATM architecture these handover QoS parameters include: handover, urgency, probability of handover blocking, probability of new cell blocking, traffic disruption period during handovers, cell loss during handovers, ATM cell sequencing during handovers, and speed of handover operation.

2.3.3 QoS Parameters

There is more than one level of criteria to satisfy the different types of traffic, (e.g., time sensitive financial, still images, large data, video). For example, the time delay in transferring large files and high-resolution images may not be adequate for the voice communication. Quality of service has become an extremely important issue to be addressed. The QoS parameters are discussed in the following paragraphs.

***Delay* is the time for a packet to be transported from the sender to the receiver.**

For the TCP protocol, higher orders of delay result in greater amounts of data held “in transit” in the network. This affects in turn the counters and timers associated with the protocol. TCP is a “self-clocking” protocol, where the sender’s transmission rate is dynamically adjusted to the flow of signal information coming back from the receiver, via the reverse direction acknowledgments. Larger delays between sender and receiver make the feedback loop insensitive, and therefore the protocol becomes more insensitive to short term dynamic changes in network load. For interactive voice and video applications, larger delay are not acceptable.

***Jitter* is the variation in end-to-end transit delay.**

High levels of jitter cause the TCP protocol to make very conservative estimates of round trip time (RTT), causing the protocol to operate inefficiently when it reverts to timeouts to reestablish a data flow. High levels of jitter in UDP-based applications are unacceptable for real-time applications, such as audio or video. Jitter causes the signal to be distorted, which in turn can only be rectified by increasing the receiver’s reassembly playback queue. This increases the delay which is unattractive for interactive applications.

***Bandwidth* is the maximal data transfer rate that can be sustained between two end points.**

This is limited not only by the physical infrastructure of the traffic path within the transit networks, which provides an upper bound to available bandwidth, but is also limited by the number of other flows which share common components of this selected end-to-end path.

***Packet loss* can occur due to a poorly configured or poorly performing switching system which**

delivers packets out-of-order or even drop packets through transient routing loops. Unreliable or error-prone network transit paths can also cause retransmission of the lost packets. For example, TCP cannot distinguish between loss due to the packet corruption and loss due to congestion. Packet loss invokes the same congestion avoidance behavior response from the sender, which reduces the sender’s transmit rate.

***Reliability* is the percentage of network availability depending upon the various environmental**

parameters such as rain and atmospherics. In satellite-based network, the availability depends on the frequency band of operation, power levels, antenna size and the traffic for the service provided. Advanced error control techniques are used to provide good link availability.

Figure 2.3 shows how different applications vary in their QoS requirements.

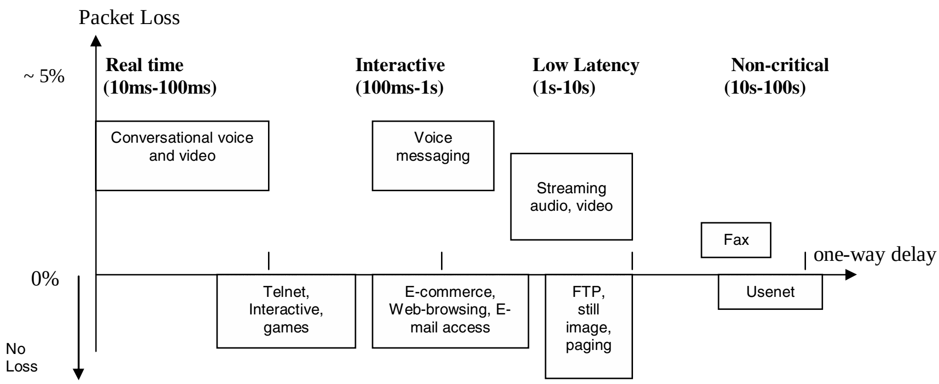


Figure 2.3: Application Specific QoS Requirements Example

2.3.4 QoS Challenges

Some of the challenges for Quality of Service support are as below:

- Network flexibility is becoming central to enterprise strategy.
- Traffic is fractal and bursty.
- Interactive applications such as voice and video have stringent bandwidth and latency demands.
- Multiple application networks are being combined into consolidated corporate utility networks. Web browsing, email, file transfers, or other low-priority or bulk traffic leave little space for critical transaction traffic leading to bandwidth contention and latency problems.
- Decisions have to be made as to who controls and how they control competing demands for network service and backbone connectivity.

Several mechanisms to provide ATM QoS and IP QoS were recommended by the ATM Forum, IETF, and ITU. Applicability of these QoS technologies for broadband satellite network design and implementations which forms the objective of this thesis is described in chapters 3-6.

2.4 Satellite Communications

In 1945, Arthur C. Clarke, then an officer with the Royal Air Force, proposed the use of Geosynchronous Earth Orbit spacecraft operating at an altitude of 22,300 miles or 35,680 kilometers [20]. A decade later, John Pierce, working at Bell Laboratories, examined the use of satellite repeaters for carrying international phone calls. The first trans-Atlantic telephone cable (TAT-1) was designed to carry 36 simultaneous phone calls for nearly $50 million. On October 4, 1957, the Soviet Union launched Sputnik, marking the beginning of the modern space age. The National Aeronautical and Space Administration (NASA) was consequently formed in 1958. However, it was the Army Ballistic Missile Agency that actually launched the first American satellite, "Explorer", on January 31, 1958.

AT&T launched its first Medium Earth Orbit (MEO) spacecraft, Telstar 1 in 1963. In the same year, the U.S. drafted the Communications Satellite Act forming COMSAT with an initial funding of $200 million. Later on, COMSAT became the U.S. signatory to the newly formed Intelsat, an international communication satellite cooperative comprised initially of 11 member countries. In 1965, Intelsat launched EARLY BIRD (Intelsat 1) followed by Soviet Union, USA, UK, Italy, Canada and France. Over the next 25 years, Intelsat grew to over 140 member countries and expanded their fleet of satellites to nearly 20 in-orbit spacecraft. Intelsat became responsible for carrying approximately 90% of international telephone traffic and virtually all international television programming. Beginning in 1972 Canada launched ANIK-1, domestic satellite operators began operating primarily to service the television entertainment industries followed by the United States, Indonesia and Japan in the 1970's; India, Australia, Brazil, Mexico, Israel, China, Sweden, and Luxembourg in the 1980's; Argentina, Pakistan, Chile, Thailand, Korea, Malaysia, and Portugal in the 1990's. In 1984, PanAmSat became the world's first private satellite operator, generating a privatization of Intelsat and Eutelsat, the two largest satellite consortia. Today the satellite industry is dominated by private operators offering global coverage, represented by PanAmSat, Loral Skynet via the Loral Global Alliance, and SES Global formed by GE Americom and SES/Astra.

During the past forty years, satellite communications systems have been developed not only for commercial uses, but also for military applications. These include, Air Force Satellite Communications (AFSATCOM) at Ultra High Frequencies (UHF), Defense Satellite Communication Systems (DSCS) at Super High Frequencies (SHF) and Military Strategic and Tactical Relay (MILSTAR) Satellite System at Extremely High Frequencies (EHF).

Historically, the first satellite network accessing Internet called SATNET was initiated in mid 1975. The network was sponsored by the Advanced Research Projects Agency (ARPA), the Defense Communications Agency, the British Post Office and the Norwegian Telecommunications Administrations [3]. The network consists of four ground stations (two in the Washington, D.C. area at Etam and Clarksburg; one in Goonhilly, England; and one in Tanum, Sweden) interconnected by a simplex, 64 Kbps Intelsat IV-A SPADE channel. The ground station sites are equipped with satellite message processors, called Satellite IMPs (SIMPs) which are extensions of the ARPANET IMPs and which implement channel access and network access protocols.

Gateway computers, implemented with PDP-11 hardware, connect SATNET and ARPANET to permit internetwork communications. One of the goals of the SATNET experiments was to test the feasibility and extensibility of different channel access schemes [22, 23] and to gain experience with the implementation, operation and performance evaluation of packet satellite networks.

Special issues on Satellite Communications, published by IEEE Proceedings [25, 26] provide an excellent research and development on satellite communications. New generation satellite networks are being developed with main requirement of supporting Internet access and multimedia services and applications. These satellite networks are classified as satellite connectivity networks for global coverage and satellite access networks for regional coverage.

The broadband satellite networks, the TCP/IP enhancements, the QoS models, and the design problems which form the main focus of this chapter, are discussed in the following sections.

2.4.1 Satellite Communication Characteristics

The principal attributes of satellites include (a) broadcasting and (b) long-haul communications. For example, Ka-band spot beams generally cover a radius of 200 km (125 miles) or more, while C- and Ku-band beams can cover entire continents. This is in contrast to last mile only technologies that generally range from 2 km to 50 km (or 1-30 miles).

The main advantages of satellite communications are:

– *Ubiquitous coverage*: A single satellite system can reach every potential user across an entire continent regardless of location, particularly in areas with low subscriber density and/or otherwise impossible or difficult to reach. Current satellites have various antenna types that generate different footprint sizes. The sizes range form coverage of the whole earth as viewed from space (about 1/3 of the surface) down to a spot beam that covers much of Europe or North America. All these coverage options are usually available on the same satellite. Selection between coverage is made on transparent satellites by the signal frequencies. It is spot beam coverage that is most relevant for access since they operate to terminal equipment of least size and cost. Future systems will have very narrow spot beams of a few hundred miles across that have a width of a fraction of a degree.
– *Bandwidth flexibility*: Satellite bandwidth can be configured easily to provide capacity to customers in virtually any combination or configuration required. This includes simplex and duplex circuits from narrowband to wideband and symmetric and asymmetric configurations. Future satellite networks with narrow spot beams can deliver rates of up to 100 Mbps with 90 cm antenna and the backplane speed within the satellite switch could be typically in Gbps range. The uplink rate from a 90 cm user terminal is typically 384 Kbps.

– *Cost*: The cost is independent of distance: The wide area coverage from a satellite means that it costs the same to receive the signal from anywhere within the coverage area.
– *Deployment*: Satellites can initiate service to an entire continent immediately after deployment, with short installation times for customer premise equipment. Once the network is in place, more users can be added easily.
– *Reliability and security*: Satellites are amongst the most reliable of all communication technologies, with the exception of SONET fault tolerant designs. Satellite links only require the end stations to be maintained and they are less prone to disabling though accidental or malicious damage.
– *Disaster recovery*: Satellite provides an alternative to damaged fiber-optic networks for disaster recovery options and provide emergency communications.
– *Connectivity*: Satellite networks provide multipoint-to-multipoint communications facilitated by the Internet and broadcasting capability.

2.4.2 Satellite Orbits

A satellite communication network system consists of a space segment and a ground segment. The space segment consists of satellite(s) which are classified into Geostationary orbit (GSO) and Non-GeoStationary orbit (NGSO) satellites. The NGSO satellites are further divided into Medium Earth Orbit (MEO) if their orbit is above 2000 km and Low Earth Orbit (LEO) if their orbit is below 2000 km altitude. Generally altitudes between 2000 and 5000 km are avoided because of the presence of the VanAllen radiation belt which is capable of damaging satellites.

A NGSO satellite appears in constant motion to an observer (or ground station) on the surface. Communications via a NGSO satellite requires the earth station to anticipate, when and in what direction the satellite will rise above the horizon, which is different for every pass. Once acquired, the NGSO satellite must be continually tracked as it moves through the sky until it again disappears below the horizon (or behind a chimney, a neighboring building, or a tree). Most of the first generation satellite systems used GEO where as next generation satellite networks include MEO and LEO constellations in addition to GEOs.

2.4.2.1 GEO Satellite

The altitude of GEO satellites is 35,786 km above the surface of the earth. The orbit period is 24 hours which gives the advantage of continuous visibility and fixed geometry. This has made GEO satellites one of the principal means of distribution of television, telephone and data communications throughout the world. Most of the commercial telecommunications satellites currently in operation are in GEO orbit. A GEO satellite can provide service to a very large area. If the look angle of the ground station is restricted to a minimum of 20^{o} above the horizon, the surface area capable of being served by one satellite corresponds to about 135 million square kilometers, or 26% of the total surface of the Earth. Only 3 GEO satellites are required to provide service to all the tropical and temperate zones but not the poles. Due to their position above the equator, GEO satellites cannot provide service beyond 81 degrees North or South latitudes (76 degrees latitude if the minimum ground station elevation is restricted to 5^{o} above the horizon). The significant problem with GEO is the propagation delay. The typical round trip propagation delay is about 250-280 ms, which makes GEO satellites not applicable to real-time applications.

2.4.2.2 MEO Satellite

The altitude of MEO satellites has to be selected between the inner and outer Van Allen Radiation belts, typically around 10,355 km above the surface of the earth. The orbit period will be 6 hours. The world can be covered by 10-12 satellites in 2-3 planes e.g., 5 satellites in each of the 2 planes or 4 satellites in each of the 3 planes. The typical propagation delay for a MEO satellite is 110-130 ms. ICO is a MEO satellite system.

The LEO and MEO based architectures decrease delay and total weight of each satellite, yet require global constellations. For example, Teledesic, previously had a constellation of 288 LEO satellites. Recently, it has initiated a new system design with 30 MEO satellites. Hughes Spaceway System uses 8 GEO satellites. Latency is a key difference, making LEO/MEO satellites more attractive for interactive applications like voice and video. However, demonstrations show that GEO satellite systems are capable of high-speed Internet access. MEO systems are proposed in a few systems to strike a balance between the complexity of a MEO system constellation and low latency requirements in data services. Table 2.2 shows a comparison of LEO, MEO, GEO system characteristics.

2.4.2.3 LEO Satellite

The altitude of LEO satellites is typically between 700 and 2000 km. The orbit period will be somewhere between 100 to 120 minutes. Providing continuous communications coverage to a given point on Earth using LEO satellites requires the use of 6-8 planes with 6 satellites per plane. To avoid interruptions at the end of each pass, data buffering must be employed at the ground station or the next visible satellite must be acquired and tracked before communications are handed over from the setting satellite. Due to lower altitude, LEO satellites can provide service to a rather small area at any given time. The round trip propagation delay is about 20-25 ms. LEO satellites operating in high inclination orbits can provide service to the near-polar regions. Motorola's Iridium [28] and GlobalStar [29] belong to this category. Iridium uses 66 LEO satellite constellation and GlobalStar uses 48 LEO satellites. Both these systems were designed to provide Telephony services. GlobalStar currently is being used for emerging communications as well.

Table 2.2: Comparison between LEO, MEO and GEO

Orbit type →	Low Earth Orbit (LEO)	Medium Earth Orbit (MEO)	Geostationary Earth Orbit (GEO)
Altitude (km)	700 to 2000	10,000 to 15,000	36,000
Satellites Needed For Global Coverage	> 32	10 to 15	3 to 4 [(1)]
Delay	5-20 ms	100-130 ms	250-280 ms
Elevation Angle	Low to Medium	Medium to High	Low to Medium
Capacity per Satellite	128 Mbps – 10 Gbps	128 Mbps – 10 Gbps	1 – 50 Gbps
Handover	Frequent	Infrequent	Never
Onboard Processing	Possible	Possible	Possible
Operations	Complex	Medium	Simple
Space Segment Cost	High	Low	Medium
Satellite Lifetime (years)	3 to 7	5 to 10	10 to 15
Terrestrial Gateway Costs	High	Medium	Low
Hand Held Terminal	Yes	Yes	Yes
Store and Forward	Yes	Yes	Not Required
Point to Point Connections	No	No	Yes
VSATs	Yes [(2)]	Yes [(2)]	Yes [(3)]
Broadcast TV	No	No	Yes
SNG (Satellite News Gathering)	Yes	Yes	Yes

Notes:
(1) Coverage only extends to latitudes of 70° N and 70° S.
(2) Via a regional gateway.
(3) Private and shared hubs possible.

2.4.3 Frequency Bands

Table 2.3 provides the frequency bands for fixed satellite services and broadcast services. The current operate at C- and Ku-band whereas the next generation satellite networks are being planned at Ka-band.

Table 2.3: Frequency Bands

Band	Earth-to-Space Frequency	Space-to-Earth Frequency
C	5.925-6.425 GHz*	3.7-4.2 GHz
Ku	12.75-13.25, 13.75-14.5 GHz	10.7-12.75 GHz
Ka	27.5-30.0 GHz	17.7-20.2 GHz
Q/V	47.2-50.2 GHz	39.5-42.5 GHz

* Extended C - uplink 6.425-6.725 GHz, downlink 3.4-3.7 GHz

2.4.4 Network Topologies

Satellite networks can be grouped into one of four network topologies: Point-to-Point, Point-to-Multipoint, Multipoint-to-Point, and Multipoint-to-Multipoint. Each network is characterized by the utilization of different technologies to enable the specific topology to operate in the most cost efficient manner. Many satellite networks employ a combination of topologies in order to handle specific types of applications and traffic demands.

2.4.4.1 Point-to-Point Networks

Point-to-point networks involve the establishment of a single communication link (in either simplex or duplex mode) between two locations. The data range for point-to-point circuits varies extensively from 64 Kbps narrowband circuits up to 155 Mbps high data rate Multiple Channel per Carrier (MCPC) circuits.

2.4.4.2 Point-to-Multipoint Networks

Point-to-Multipoint networks involve a single broadcast uplink and an unlimited number of receive sites. In each instance, all receivers in such a network are tuned to the same broadcast frequency, while routing/switching algorithms determine whether a specific receiver is able to listen to the broadcast. Data rates generally range from 1.5 Mbps to 45 Mbps. These networks are used to provide broadcast, multicast and more commonly unicast services (usually for Internet access services).

2.4.4.3 Multipoint-to-Point Networks

Multipoint-to-Point satellite networks are used for collecting or aggregating information at a central location from a large number of remote locations. Receiving incoming data from a large number of remote locations requires careful coordination of network bandwidth which is accomplished through the use of Frequency Division Multiple Access (FDMA), Time Division Multiple Access (TDMA), and Code Division Multiple Access (CDMA). Multipoint-to-point networks are used in applications such as Point-of-Sale terminals, remote sensoring, and specialty corporate WANs.

2.4.4.4 Multipoint-to-Multipoint Networks

Multipoint-to-multipoint networks are designed to operate in a “mesh” configuration where every location has the ability to communicate directly with every other location on a single hop basis which is necessary to minimize latency between sender and receiver.

2.4.5 Mobile Satellite Communications

Table 2.4 provides major characteristics of three deployed non-geostationary satellite systems, ICO, Globalstar and Iridium. Of these three, ICO operates in MEO system while Globalstar and Iridium use LEO satellites. Globalstar uses a CDMA for multiple access whereas ICO and Iridium are TDMA based systems. [27.]

Inmarsat has provided satellite communications services to mobile users since 1981. It currently operates a global satellite system used by independent service providers offering voice and data communications. The combination of 16-QAM and turbo coding is well-suited Inmarsat’s emerging multimedia services. The Inmarsat mobile ISDN service supports high-speed data services at 64 Kbps, voice service using 4.8 Kbps advanced multiband excitation coding algorithm, analogue voice-band modem services (fax, data, encryption, using pulse code modulation encoding), and subscriber interface module card compatibility with Inmarsat-phone readers and cards. Inmarsat is currently serving the aeronautical, maritime, and land mobile community with nine geostationary satellites. Service range form traditional voice to high-speed multimedia applications. In future, it will contribute to the IMT-2000 vision of integrated universal mobile multimedia services.[31.]

Section 2.4.6 describes the future Ka-band systems and some of the technical challenges.

Table 2.4: Mobile Satellite Communications

Services and Cost	ICO	Globalstar	Iridium
Service types	voice, data, fax, short message service	voice, data, fax, paging, position location	voice, data, fax, paging, messaging, position location
Voice (kbps)	4.8	adaptive 2.4 / 4.8 / 9.6	2.4 / 4.8
Data (kbps)	2.4	7.2 sustained throughput	2.4
Modulation	QPSK	QPSK	QPSK
Voice circuits / satellite	4500	2000 - 3000	1100 (power limited) 3840 (max available)
Altitude (km)	10355 (changed to 10390, late 1998)	1410	780
Number of satellites	10 active 2 in-orbit spares	48 active 8 in-orbit spares	66 active 6 in-orbit spares
Number of planes	2	8	11
Inclination (°)	45	52	86.4
Satellite visibility time (minutes)	115.6	16.4	11.1
Minimum mobile terminal elevation angle (°)	10	10	8.2
Multiple access method	TDMA / FDMA / FDD	CDMA / FDMA / FDD	TDMA / FDMA / TDD
Beams per satellite	163	16	48
Mobile downlink frequencies (MHz)	1980 - 2010	2483.5 - 2500.0 (S-band)	1616.0 - 1626.5 (L-band)
Mobile uplink frequencies (MHz)	2170 - 2200	1610.0 - 1626.5 (L-band)	1616.0 - 1626.5 (L-band)
On-board processing (regeneration)?	--	no	yes
Inter-satellite Link (ISL) frequencies (GHz)	N/A	N/A	22.55 -23.550
Handover performed?	yes	yes, seamless	yes

2.4.5.1 Current Status of Some Satellite Systems

Iridium is now (as of writing of this thesis) owned by Iridium Satellite Inc. and it is still operating. The old Iridium company has been brought out of bankruptcy for $25 million. They have a multi-year US Department of Defense (DoD) contract to provide services.

GlobalStar is still operating (as of writing of this thesis). There have been multiple restructuring approaches considered. They have bought back gateways and have changed the billing plans

ICO: The company entered chapter 11 in August 1999 and came out of it in May 2000, after Craig McCaw led a group of International investors to provide 1.2 billion to acquire the company. The first satellite was launched in 2001. In 2002 the company signed an agreement to acquire Constellation Communication Holdings, Inc. and Mobile Communications Holdings, Inc., and is awaiting Federal Communications Commission (FCC) approval and the satisfaction of certain other closing conditions to close their transactions.

2.4.6 Next Generation Ka-band

Many current satellite systems use C- and Ku-band frequencies and bent pipe payload architectures. C band requires large antenna and individual FCC frequency coordinates and licensing. Ku-band systems require small dish antenna and is more attractive but these bands are generally congested. That is how, Ka-band employing spot beam technology became an attractive solution for future broadband satellite networks. As shown in table 2.5, most of the global broadband satellite networks e.g., Spaceway, Astrolink, EuroSkyWay, Teledesic are designed to operate at Ka-band. Intelsat systems use C and Ku while Eutelsat systems use Ka and Ku. Table 2.6 shows that broadband access systems like Starband operates at Ku; Wildblue is being designed at Ka and others e.g., iPStar, Astra-BBI and Cybestar employ Ku, Ka-bands.

Until recently, Ka-band was used for experimental satellite programs in the U.S., Japan, Italy, and Germany. In the U.S, the Advanced Communications Technology Satellite (ACTS) is being used to demonstrate advanced technologies such as onboard processing and scanning spot beams. A number of applications were tested including: distance learning, telemedicine, credit card financial transactions, high data rate computer interconnections, video conferencing and HDTV. The growing congestion of the C and Ku bands and the success of the ACTS program increased the interest of satellite system developers in the Ka-band satellite communications network for exponentially growing Internet access applications. A rapid convergence of technical, regulatory, and business factors has increased the interest of system developers in Ka-band frequencies. [66, 67, 68, 74, 75, 77.]

Several factors influence the development of multimedia satellite networks at Ka-band frequencies: [222, 36]

– *Adaptive Power Control and Adaptive Coding:* Adaptive power control and adaptive coding technologies have been developed for improved performance, mitigating propagation error impacts on system performance at Ka-band.
– *High Data Rata:* A large bandwidth allocation to geosynchronous fixed satellite services (GSO FSS) and non-geosynchronous fixed satellite services (NGSO FSS) makes high data rate services feasible over Ka-band systems.
– *Advanced Technology:* Development of low noise transistors operating in the 20 GHz band and high power transistors operating in the 30 GHz band have influenced the development of low cost earth terminals. Space qualified higher efficiency traveling-wave tubes (TWTAs) and ASICs development have improved the processing power. Improved satellite bus designs with efficient solar arrays and higher efficiency electric propulsion methods resulted in cost effective launch vehicles.

- *Global Connectivity:* Advanced network protocols and interfaces are being developed for seamless connectivity with terrestrial infrastructure.
- *Efficient Routing:* Onboard processing and fast packet or cell switching (e.g., ATM, IP) makes multimedia services possible.
- *Resource Allocation:* Demand Assignment Multiple Access (DAMA) algorithms along with traffic management schemes provide capacity allocation on a demand basis.
- *Small Terminals:* Multimedia systems will use small and high gain antenna on the ground and on the satellites to overcome path loss and gain fades.
- *Broadband Applications:* Ka-band systems, combining traditional satellite strengths of geographic reach and high bandwidth, provide the operators a large subscriber base with scale of economics to develop consumer products.

2.4.6.1 Satellite Network Architecture

The satellite network architecture consists of space and ground segments. The space segments consists of one (GEO) or more (GEO, MEO, LEO) satellites. The ground segment consists of user terminals (UTs), gateway terminals (GWs), network control centres (NCCs) and satellite control centre (SCC). This section provides an analysis of the network architecture with special references to the onboard processing with some examples of global broadband satellite networks. A brief status of the modulation and encoding summary is provided in Section 2.5.1.1. A rain attenuation model for Ka-band systems is described in Section 2.5.1.3. Section 2.5.1.4 describes an adaptive coding approach that can be used for Ka-band systems. Adaptive power control is discussed in Section 2.5.1.5.

Architectural Candidates

Candidate 1: Wide Area Bent Pipe Architecture

The current satellite systems operating in C- and Ku-band use this architecture. Bent Pipe systems simply receive and retransmit signals from the same beam's coverage area. Due to the wide coverage, all the UTs, GWs and NCC are within the coverage of a single beam and can communicate via satellite. The drawback of this Wide Area Bent Pipe architecture is that it cannot take advantage of frequency reuse.

Candidate 2: Spot Beam Bent Pipe (non-regenerative) Architecture

The spot beam bent pipe architecture uses frequency reuse at Ka-band. Each UT, GW and the NCC communicate through one of the satellite's spot beams. The ground element i.e., ET, GW or NCC within the same spot can communicate with all other ground elements via the satellite.

Candidate 3: On-Board Processing and Switching.

An on-board processing and switching architecture provides interbeam connectivity allowing user terminals to access other user terminals, gateways and the NCC in other beams. The capacity of the resulting network is larger than the bent pipe architecture. As a result frequency reuse is possible without a terrestrial infrastructure. This applies to both ATM and packet switching.

On-Board Processing

Many of the next generation broadband satellite systems use on-board processing and fast packet/cell switching. Onboard processing involves demodulation and demultiplexing the received signal. The payload performs decoding and encoding, processing the header information, and routing the data, pointing the antennas, buffering, multiplexing, and retransmitting the data on downlink or inter-satellite link. The major reasons for onboard processing include separation of the uplink from the downlink, a gain of approximately 3 dB in performance, and provision of resources on demand using uplink Demand Assignment Multiple Access (DAMA). The advantages of onboard processing and switching over bent pipe architecture include:

– Improved error rates by using effective encoding techniques
– Separation of uplink and downlink
– System efficiency improvements
– Better delay performance
– Routing decisions onboard or via intersatellite links
– No end-to-end retransmissions
– Capacity improvements

The first generation services that are now in place use existing Ku-band Fixed Satellite Service (FSS) for two-way connections. Using FSS a large geographical area is covered by a single broadcast beam. The new Ka-band systems use spot beams that cover a much smaller area e.g., hundreds of miles across. Adjacent cells can use different frequency range but a given frequency range can be reused many times over a wide geographical area. The frequency reuse in the spot beam technology increases the capacity. In general, Ka spot beams can provide 30-60 times the system capacity of the first generation networks.

In a non-regenerative architecture, the satellite receives the uplink and retransmits it on the downlink without on-board demodulation or processing. In a processing architecture with cell-switching or layer-3 package, the satellite receives the uplink, demodulates, decodes, switches and buffers the data to the appropriate beam after encoding and remodulating the data, on the downlink. In a processing architecture, switching and buffering are performed on the satellite and in a non-processing architecture, switching/routing and buffering are performed within a gateway. The selection of the satellite network architecture is strictly dependent on the target customer applications and performance/cost tradeoffs.

The proposed global satellite connectivity networks such as Astrolink, Spaceway, and EuroSkyway have onboard processing and switching capabilities. On the other hand, regional a*ccess networks* like StarBand, IPStar, and WildBlue are intended to provide an Internet access. These access systems employ non-regenerative payloads.

On-Board Switching

On-board processing satellites with high-gain multiple spot-beams and on-board switching capabilities have been considered as key elements of new-generation satellite communications systems [43]. These satellites support small, cost effective terminals and provide the required flexibility and increased utilization of resources in a bursty multimedia traffic environment.

Although employing an on-board switch function results in more complexity on-board the satellite, the following are the advantages of on-board switches.

- Lowering the ground station costs.
- Providing bandwidth on demand with half the delay.
- Improving interconnectivity.
- Offering added flexibility and improvement in ground link performance, i.e., this allows earth stations in any uplink beam to communicate with earth stations in any downlink beam while transmitting and receiving only a single carrier.
- In-band signalling for combined traffic and TT & C operation

One of the most critical design issues for on-board processing satellites is the selection of an on-board baseband switching architecture. Four types of on-board switches are proposed:

- Circuit switch
- ATM Switch
- Hybrid Switch
- Fast Packet Switch

Inter-Satellite Links (ISLs)

The use of inter-satellite links (ISL) for traffic routing has to be considered. It has to be justified that this technology will bring a benefit which would make its inclusion worthwhile or to what extent on-board switching, or some other form of packet switching, can be incorporated into their use. The issues which need to be discussed when deciding the use of ISLs include –

- Networking considerations (coverage, delay, handover)
- The feasibility of the physical link (inter-satellite dynamics)
- The mass, power & cost restrictions (link budget)

The mass and power consumption of ISO payloads are factors in the choice of whether to include them in the system, in addition to the possible benefits and drawbacks. Also the choice between RF and optical payloads is now possible as optical payloads have become more realizable and offer higher link capacity. The tracking capability of the payloads must also be considered, especially if the inter-satellite dynamics are high. This may be an advantage for RF ISL payloads.

Advantage of ISLs

- Less ground-based control may be achieved with on-board baseband switching reducing delay (autonomous operation).
- Increased global coverage due to oceans and areas without ground stations.
- Single network control centre.

Disadvantages of ISLs

- Complexity and cost of the satellites will be increased.
- Power available for the satellite/user link may be reduced.
- Handover between satellites due to inter-satellite dynamics will have to be incorporated.
- Replenishment strategy.
- Frequency co-ordination.

Spot Beam Technology

Traditional satellite technology utilized broad single beams covering entire continents and regions. Newer Ku- and Ka-band spot beams provide coverage over a much smaller region than previously which is advantageous in providing more bandwidth. By shaping the antenna on the spacecraft into a tighter focus, the size of the footprint on the ground is reduced. Two benefits are created by this modification: the signal strength as seen from ground terminals increases allowing for smaller ground antennas, and the same frequency range can be utilized multiple times in different beams yielding greater total bandwidth. Using frequency reuse through multiple spot beams, Ku- and Ka-band satellites can be configured in a similar fashion to terrestrial cellular networks. Particularly with Ka-band, the limiting factor no longer becomes available spectrum, but the amount of transponder power available and the weight of the entire payload to be launched. Most broadband satellites plan to employ tighter spot beams than predecessors. Specific applications well suited for spot beam technology include local television broadcasting and high-speed Internet access.

2.4.7 Global Broadband Satellite Network (Connectivity Network)

The emerging applications such as streaming audio/video, broadcast Internet access and multicast dictate future broadband satellite network designs. The design parameters include traffic throughput, system delay, flexibility, performance and QoS, and complexity. Traditional bent pipe payloads perform frequency translation and power amplification, but require complex ground infrastructure supporting routing functions in a hub and spoke topology. In the advanced architectures with processing payloads and spot beam antenna technology, complete mesh topologies in addition to, hub and spoke can be supported. If these technologies happen to be economical, the newer architectures use inexpensive, small, low power terminals with onboard demodulation/remodulation, encoding, and routing functions. The broadband user QoS levels can be well supported by onboard traffic management and traffic monitoring, which currently is being supported by the terrestrial infrastructure. The tradeoffs of these architectures and the QoS architecture form the main focus of this thesis as discussed in section 3 and 4.

Figure 2.4 illustrates a broadband satellite network architecture represented by a ground segment, a space segment, and a network control segment. The ground segment consists of terminals and gateways (GWs), which may be further, connected to other legacy public and/or private networks. The Network Control Station (NCS) performs various management and resource allocation functions for the satellite media. Inter-satellite crosslinks in the space segment to provide seamless global connectivity via the satellite constellation is optional. Hybrid network architecture allows the transmission of packets over satellite, multiplexes and demultiplexes datagram streams for uplinks, downlinks, and interfaces to interconnect terrestrial networks. The satellite network configuration also illustrates the signaling protocol e.g. SS7, UNI and the satellite interface unit. The architectural options could vary from ATM switching, IP transport or MPLS over satellite.

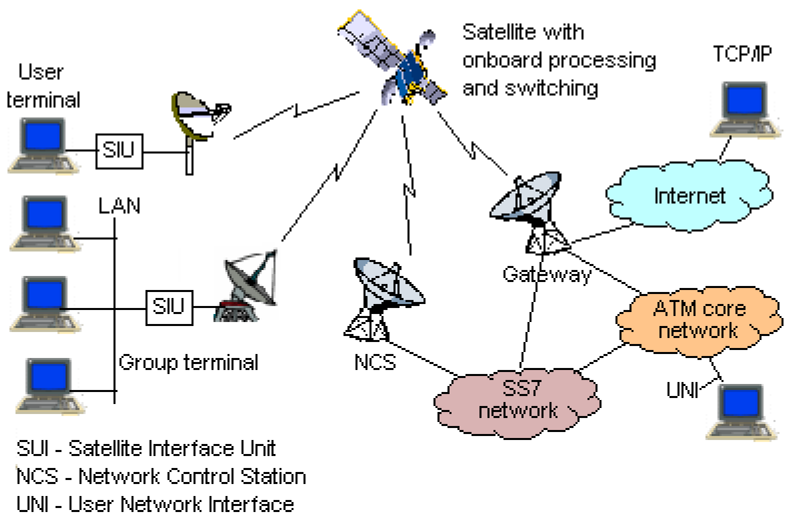


Figure 2.4: Broadband Satellite Network Configuration

Table 2.5 provides some of the new generation Ka-/Ku-band satellite systems comparison. These systems provide global coverage and high bandwidth. As shown in the table, most of the systems employ onboard processing and switching. The data throughput ranges to about 9 Gbps. Except Teledesic which is a MEO based satellite network, the others use GEO satellite configuration. [49, 50, 51, 52, 53, 54, 55.]

Table 2.5: Global Broadband Satellite Networks

Services	Spaceway	Astrolink*	EuroSky Way	Teledesic	Intelsat	Eutelsat
Data uplink	384 Kbps -6 Mbps	384 Kbps-2 Mbps	160 Kbps-2 Mbps	16 Kbps- 2 Mbps	--	Max. 2 Mbps
Data downlink	384 Kbps-20 Mbps	384 Kbps-155 Mbps	128 Kbps-640 Kbps	16 Kbps-64 Mbps	Max. 45 Mbps	55 Mbps
Number of Satellites	8	9 (4, initially)	5	30	--	--
Satellite	GEO	GEO	GEO	MEO	GEO	GEO (Hotbird 3-6)
Frequency Band	Ka	Ka	Ka	Ka	C, Ku	Ku, Ka
Onboard processing and switching	Yes	Yes	Yes	Yes	No	--
Multiple Access	MF-TDMA	MF-TDMA	MF-TDMA	MF-TDMA	--	--
Operation Scheduled	2003	2003	2004	2004/5	--	2001

* Program on hold

2.4.7.1 Current Status of Some Satellite Systems

Spaceway: Hughes Network Systems is expected to provide private line, frame relay services, IP Virtual Private Network services, broadband service to small, medium and Small Office/Home Office (SOHO), teleworkers and for other ‘extranet’ applications. They are planning first spacecraft launch in mid-2003, projected in service by Q2, 2004. The second and third spacecrafts are expected to be launched by min 2004 and in 2005 respectively.

EuroSkyWay: The company has been field testing satellite broadband services on customers. Sales of these services in 2001 are expected to exceed USD 15 million, and by 2002 EuroSkyWay expects to break even. However, the development of a dedicated Ka-band satellite cannot begin until the equipment has been field tested and approved by the customers. It is thought that the satellite could be in place by 2004.

Teledesic: Teledesic’s board has halted work (at the time of writing of this thesis) by a contract building two satellites for the constellation envisioned by Craig McCaw. Twenty-five employees will be let go and 10-12 will stay on to evaluate possible alternative approaches.

2.4.8 Regional Satellite Network (Access Network)

The access systems provide regional coverage and are less complicated than their global or connectivity counterparts. They are more cost-effective, have less associated technical risk and have less regulatory issues. Table 2.6 provides a partial list of access satellite systems for regional coverage. [57, 58, 59, 60, 61.]

Table 2.6: Broadband Access Systems

Services	StarBand	WildBlue	iPStar	Astra-BBI	Cyberstar
Data uplink (Kbps/Mbps)	38-153K	384K-6M	2M	2M	0.5-6M
Data downlink (Kbps/Mbps)	40M	384K-20M	10M	38M	Max. 27M
Coverage Area	US	Americas	Asia	Europe	Multiregional
Market	Consumer	Business/ SME	Consumer & Business	Business	ISPs, Multicast
Terminal cost (US$)	<$350	<$1000	<$1000	~$1800 <$450 (2001)	
Access fee/mo (US$)	$60	$45	--	--	--
Antenna Size (M)	1.2	0.8-1.2	0.8-1.2	0.5	--
Frequency Band	Ku	Ka	Ku & Ka	Ku/Ka	Ku, Ka
Satellite	GEO	GEO	GEO	GEO	GEO
Operation scheduled	Nov 2000	Mid 2002	Late 2002	Late 2000	1999-2001

2.4.8.1 DVB Satellite Access

In this section Internet access via satellite using Digital Video Broadcasting-Return Channel by Satellite (DVB-RCS) is discussed [62]. The DVB network elements consist of an Enterprise Model, SLA, and TCP Protocol Enhancement Proxy (PEP), the Hub station and the Satellite Interactive Terminal (SIT). The target applications of the DVB network could be Small and Medium Enterprises and residential users. One of the major advantages of DVB-RCS is that multicasting is possible at a low cost using the existing Internet standards. The multicast data is tunneled over the Internet via a multicast streaming feeder link from a streaming source to a centralized multicast streaming server and is then broadcasted over the satellite medium to the intended target destination group. In the DVB network, *a satellite* forward and return links typically use frequency bands in Ku (12-18 GHz) and/or Ka (18-30 GHz). The return links use spot beams and the forward link global beams are used for broadcasting and Ku-band. Depending on the frequency bands (Tx/Rx), three popular versions are available: (a)Ku/Ku (14/12 GHz) (b)Ka/Ku (30/12 GHz) and (c) Ka/Ka (30/20 GHz). In business-to-business applications, the SIT is connected to several user PCs via a LAN and a Point-of Presence (POP) Router. The Hub station implements the forward link via a conventional DVB-S chain (similar to Digital TV broadcasting) whereby the IP packet is encapsulated into DVB-streams, IP over DVB. The return link is implemented using the DVB-RCS standard "MF-TDMA Burst

Demodulator bank", IP over ATM like. The HUB station is connected to the routers of several ISP's via a Broadband Access Server. The HUB maps the traffic of all SITs belonging to each ISP in an efficient way over the satellite. The selection of a suitable residential access technology depends on the type of application, site location, required speed, and affordable cost.

The DVB Return Channel System via Satellite (DVB-RCS) was specified by an ad-hoc ETSI technical group founded in 1999. The DVB-RCS system specification in ETSI EN 301 790, v1.2.2 (2000-12) specifies a satellite terminal (sometimes known as a Satellite Interactive Terminal (SIT) or Return Channel Satellite Terminal (RCST) supporting a two-way DVB satellite system (30, 31). Another CDMA based spread ALOHA has been proposed for return channel access (32). This section describes DVB-RCS protocol. The use of standard system components provides a simple approach and should reduce time to market. See Section 5.1.1 for details.

2.5 Technical Challenges

2.5.1 Physical Layer

Data traffic is growing at a much faster rate than voice traffic. Due to this increase, there is a great interest among satellite network designers to apply advanced technologies to increase the data handling capacity of the existing and planned satellite transponders. The advanced technologies are essentially characterized by the way they improve either power performance (E_b/N_o) or bandwidth efficiency (bps/Hz) or both.

2.5.1.1 Higher Order Modulation

Higher order modulation is being studied to increase the achievable bps/Hz compared to the more common QPSK. The two most common higher-order waveforms available in off the shelf products are 8-PSK (Phase Shift Keying) and 16-QAM (Quadrature Amplitude Modulation). Depending on the implementation, 8-PSK may be operated with a non-linear (saturated) power amplifier at the earth station and/or satellite transponder. However, the 16-QAM signal must be used [223] with linear channels – which can significantly reduce the link C/N_o compared to a saturated transponder.

2.5.1.2 Turbo Code

The main advantage of turbo coding is that it reduces the E_b/N_o needed to close a link at a given code rate. Known turbo codes at reasonable block size and the complexity can come quite close to the Shannon channel capacity limit (within about 1 to 2 dB) [78, 79]. For instance, an R=1/3 turbo code can achieve a BER of about 10^{-7} at an E_b/N_o = 1 dB with a block size of less than 2000 bits. Reducing the code rate to R=1/4 would reduce the E_b/N_o required to about 0.7 dB.

2.5.1.3 Propagation Effects

Future Ka-band satellite communications networks are subject to degradation produced by the troposphere which is much more sever than those found at lower frequency bands. These impairments include signal absorption by rain, clouds and gases, and amplitude scintillations arising from refractive index irregularities. For example, rain attenuation at 20 GHz is almost three times that at 11 GHz. Although some of these impairments can be overcome by oversizing the ground station antennas and high power amplifiers, the current trend is using small, low-cost ground stations that can be easily deployed at user premises. As a consequence, most Ka-band systems are expected to employ different forms of fade mitigation that can be implemented relatively easily and at modest cost. The rain fade mitigation approaches are defined by three types of Ka-band communications systems – a low service rate (< 1.5 Mbps), a moderate service rate (1.5 to 6 Mbps) system and a high service rate (> 43 Mbps) system.[69, 70, 71, 72.]

Typically the control signal for the compensation system is derived from direct measurements on the downlink signal (20 GHz) that will allow estimation of the uplink fade (30 GHz). Propagation factors that affect Ka-band satellite links include:

- rain attenuation
- antenna wetting
- depolarization due to rain and ice
- gaseous absorption
- cloud attenuation
- melting layer attenuation
- troposphere scintillation

In general rain fade compensation approaches must consider at least three characteristics of the above propagation factors: range of signal fading, rate of signal fluctuations, and frequency dependence (scaling). Propagation measurements conducted with the ACTS beacons can provide most of the required information. [76.]

2.5.1.4 Adaptive Coding Techniques

The use of adaptive coding allows a satellite system to be throughput efficient while at the same time conserving its most valuable resource - satellite electrical power. The principle relies on the fact that powerful modern coding techniques allow constant user data throughput by employing heavy code in a link-by-link basis depending on the link conditions at any point in time. Algorithms can be used which allow automated sensing of degrading and improving link conditions. One way of adapting throughput is to control the coding rate (always less than 1), which is defined as the ratio of the number of input bits to the encoder to the number of output bits. The channel symbol can be a QPSK symbol or a generalized time and/or frequency slice of a waveform.

There are other possible adaptation schemes, some involving direct adaptation of the bandwidth. All involve effective changes in coding rate, but clearly other factors can also be varied. The focus is exclusively on those systems which only vary the throughput by adapting the coding rate. Thus, in particular, the duration of a channel symbol is assumed constant, or stated another way, the bandwidth of the transmission is constant.

Adaptation of Coding Rates

The coding rate can usually be expressed as the product of two rates $r_s r$. This applies, for instance, in the case of concatenated coding systems. If the effect of the rate adaptation is to vary only one of these rates, say r, then the net effect, given that the bandwidth is constant, is that the time required to transmit a given number of bits changes with r. Without loss of generality, it may be assumed that r is 1 when the throughput is at a maximum, and this is referred to below as the *lightly coded* case. A case of particular interest to system designers is when $1/r$ is an integer in all adapted cases. Then it takes 2, 3, 4, … , etc. times as long to transmit a given set of data in adapted, or *heavily-coded*, cases relative to the lightly-coded case where $r = 1$.

The simplest case, and the one that is focused on, is that in which there is only one adapted case, and $r = ½$ when adapted. It can be said that the heavy code is of rate ½ relative to the light code in this case. Of course, the light code could be no code at all, in which case $r_s = 1$

Adaptive Coding Protocols

In this section, centrally controlled adaptive coding systems are addressed. The central control entity for the adaptive coding system is assumed to reside either in the satellite payload or in a ground facility. In the latter case, communication between all ground terminals and the control entity is assumed to go through the satellite which forms the analysis. The central ground control entity is called the network control center.

The delay before the implementation of a different down link coding rate by a ground terminal is defined as the time interval between a change in fading state up until the time that a new code rate is actually implemented. Note that this time allowance includes the time required for the terminal to discover a change in link conditions. The target delay for control by a central network control center is 1.0 second. This defines the level of rain activity for which the system offers service, as discussed next.

Now consider the rate at which rain attenuation changes, referred to below as the fade slope. This is a key determining factor in the design of an adaptive coding protocol. As shown in Figure 2.5, the dynamics of rain are correlated with the availability goals of a system. As shown, the rain attenuation design goal and availability can be related to the fade slope. The data for these experimental results were collected by COMSAT Laboratories at Clarksburg, Maryland from an ACTS Satellite down link beacon signal in 1995. They show that the higher the design goal is, in terms of the fade depth, with consequently higher terminal availability, the faster the received power decreases, for approaching rain, or the faster the received power increases, for receding rain.

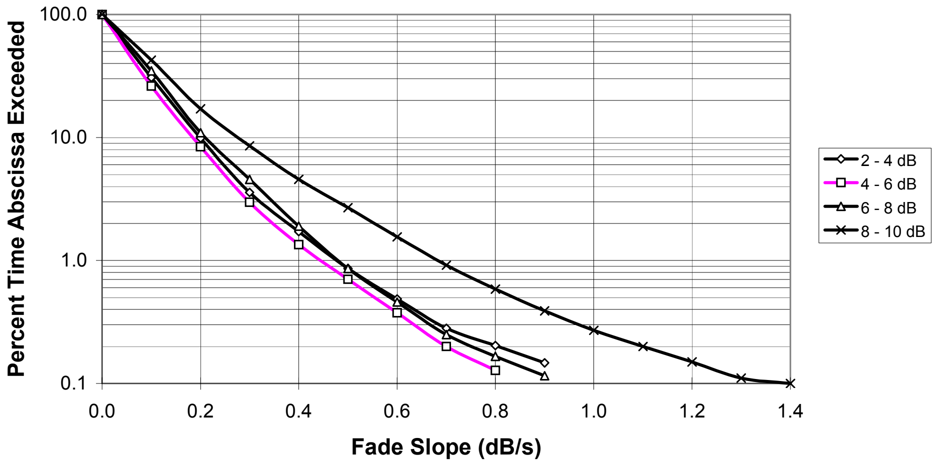


Figure 2.5: Experimental Complementary Distribution Function of Fade Slopes at 20.2 GHz.

Efficient adaptive coding systems allocate as little system capacity as possible to link margin. Typically, the most critical link margin is calculated at the edge-of-beam where, for a fixed size receive earth station, it is at a minimum. Under the assumption that the GSO satellite EIRP is backed off at higher terminal elevation angles, it may be assumed that the worst-case, edge-of-beam link margin is constant for all beams in the GSO system.

Assuming a 6-8 dB rain attenuation design goal, Figure 2.5 shows that a 0.9 dB/s fade slope can be accommodated 99.9% of the time. Given this fade slope, a prudent system design with adaptive coding control at a network control center requires 1 dB of clear-sky margin so that the system bit-error rate design goal can be maintained.

Heavy Code Capacity Resource in an Adaptively-Coded Down Link Beam

The previous two sections indicate how an adaptively coded system can adapt its link robustness to match the particular link propagation characteristics that exist on a link-by-link and second-by-second basis. The scheme that is focused here allows an end-to-end service to maintain a constant data rate within the availability envelope regardless of the conditions of the down link. In this scheme the bandwidth of a transmission remains constant. All that really changes in response to a code rate change is the length of time a terminal must receive symbols over each down link frame in order to maintain a given data rate. To maintain a data rate in the heavy coding state, in each frame a terminal must receive data over twice the duration of time required to receive the same data rate in the light coding state. The concept of heavy code capacity (i.e., “spare capacity”) in this context therefore reduces to allocating twice the time for a given transmission data rate to a user employing heavy coding than to one employing light coding. If a down link frame is made up of equal duration time slots, then links operating with heavy code use a minimum of two time slots.

The Role of Link Margin in an Adaptively-Coded Down Link Beam

Link margin in an adaptively coded system is required for one purpose only: to allow enough time to enact a coding state change. Theoretically, if the distributed control system allowed instantaneous code rate changes with a minimum of detection time, then very little margin would be required. Practical implementations employing a central control authority clearly require margin to allow enough time for detection, the coding state change protocol between terminals and the network control center, and also for processing and state transitions within the satellite transmitter, terminal receivers and resource allocation maps.

In order to protect service availability from rain events, excess down link margin in an adaptive coding system allows each terminal enough time to: (1) determine that a rain event is going to cause, with high probability, performance incompatible with the present coding state, (2) request a new coding state via a centralized protocol, and (3) respond to a command from the regional network control center to change the coding state, given that the terminal's request is granted.

Rain events which terminals can mitigate by this mechanism to maintain service availability with rain attenuation levels below about 6-8 dB were characterized in Section 2.5.1.3. The target for the system described here is 1 dB/s.

2.5.1.5 Adaptive Power Control

This section provides an important issue of compensation of rain attenuation – the power control for uplink and downlink. [73.]

Uplink Power Control – Bent Pipe Satellite

The uplink power control approach for dynamic allocation of additional power to the transmit carrier(s) at an earth station is used in order to compensate for rain attenuation. Three types of power control techniques can be considered:

– Open-loop: One station receives its own transmit carrier and must rely on its measurement of beacon fading in the downlink in order to perform uplink power control.
– Closed-loop: Two earth stations are in the same beam coverage and an earth station can receive its own transmit carrier. Uplink power control based on this carrier is erroneous due to changes in input and output backoffs under uplink and downlinks fading. It needs to be on the reception of a distinct carrier transmitted from another station.
– Feedback loop: A central control station monitors the levels of all carriers it receives, and commands the affected earth stations to adjust their uplink powers accordingly. This technique has inherent control delays, and requires more earth segment and space segment resources.

Downlink Power Control – Bent Pipe Satellite

This technique allocates additional power to the transmit carrier(s) at the satellite in order to compensate for rain attenuation. As the downlink fading occurs, downlink carrier power degrades and sky noise temperature seen by the earth station increases. Power control correction of approximately 1.5 times fade is required to maintain carrier to noise ratio. If correction is applied at the satellite, the effective isotropic radiated power for the entire beam is increased, raising the signal level at both faded and non-faded earth stations. For quasi-linear transponder operation, correction can be applied at the transmitting earth station. The Traveling Wave Tube Amplifiers (TWTAs) with variable output power levels can be commanded into high-power modes to counteract downlink fades. For beam diameters greater than rain cell diameter, correction should be applied only when a certain percentage of terminals within the beam exceed an attenuation threshold. This technique must ensure that power flux density limits are not exceeded at non-faded terminals. The system design must ensure that the communications channel is not interrupted during power level changes.

Uplink Power Control – Onboard Processing Satellite

With an onboard processing satellite, for a link transmitting from point A and receiving at point B, the received information bit error rate at point B is given by:

$$\mathrm{BER}_{\mathrm{B}} \equiv \mathrm{BER}_{\mathrm{Uplink}} + \mathrm{BER}_{\mathrm{Downlink}}$$

where

$$\mathrm{BER}_{\mathrm{Uplink}} = \text{uplink information bit error rate} = \mathrm{FU}(\mathrm{BER}_{\mathrm{Uplink}})$$

$BER_{Downlink}$ = uplink information bit error rate = $FU(BER_{Downlink})$

FU( ) or FD( ) = function describing relationship between uplink (or downlink) information BER and uplink (or downlink) channel BER.

Site Diversity – Bent Pipe or Onboard Processing Satellite

This technique involves tandem operation of two earth stations, and exploits the finite size of rain cells (5 km to 10 km). Fading at sites separated by distances exceeding the average rain cell size are expected to be correlated. Diversity gain or the reduction in link margin depends on site separation, frequency, elevation angle, and baseline orientation angle. Diversity gain calculation assumes ideal switching in which the least affected size is always selected. Site diversity is very effective in combating rain fades. On the other hand, the cost of the second site, dedicated terrestrial interconnection, service interruptions, required data buffering and site synchronization are considered significant system overload or major disadvantages. In general this technique is best suitable for network control centers and major gateways, and not applicable for low-cost earth stations unless applied with public switched networks to interconnect the terminals.

2.5.2 Link Layer: Media Access Control

Media Access Control protocols are classified into fixed assignment, random access, and Demand Assignment Multiple Access (DAMA) [81, 82]. Fixed Assignment may be made on a frequency, time, or code basis. Major techniques include frequency-division multiple access (FDMA), time-division multiple access (TDMA), and code-division multiple access (CDMA). In FDMA and TDMA systems, each station utilizes its own dedicated channel. They are contention-free, and can provide QoS guarantees at the expense of efficient utilization of resources. FDMA was the first fixed assignment multiple access method used in satellite systems. TDMA is popular mainly because of its compatibility with the nonlinear nature of transponders and is used in the majority of current satellite systems. In a CDMA system, each user is assigned a unique code sequence which is used to spread the data signal over a wider bandwidth than that required to transmit the data.

2.5.2.1 Random access

Due to technological advances, small and inexpensive terminals with lower data rates are now widely available, thus stimulating home or personal use of satellite access service. The number of stations within a satellite footprint increases from a few to several hundreds or thousands. In addition, traffic generated is very bursty. For such applications random access schemes such as slotted ALOHA provide a throughput of 37%. However, improvements were reported in the literature. Random access protocols are used in hybrid architectures and for control channels.

2.5.2.2 Demand Assignment

Although random access may better accommodate a large number of terminals with bursty traffic, it provides no QoS guarantees. DAMA protocols attempt to solve this problem by dynamically allocating system bandwidth in response to user requests. A resource request must be granted before actual data transmission. The transmission of requests is itself, a multiple access problem. After a successful reservation, bandwidth is allocated on an overall FDMA or TDMA architecture, and data transmission is guaranteed to be collision-free. Resource reservation can be made either explicitly or implicitly. Explicit reservation is on a per-transmission basis, and usually a dedicated reservation channel is shared among all stations. [86, 87.]

Priority-Oriented Demand Assignment (PODA) and first-in first-out (FIFO) Ordered Demand Assignment (FODA) combine implicit and explicit requests. Each PODA TDMA frame consists of a control part and a data part with an adjustable boundary. Explicit requests contend in the control part by slotted ALOHA, while implicit requests are piggybacked on data packets.

Combined free/demand assignment multiple access (CFDAMA) freely assigns remaining channels according to some strategy (e.g., round-robin). In combined random access and TDMA, reservation multiple access (CRRMA) remaining resources are open for random access. A hybrid scheme called Round-Robin Reservation (RRR) is based on fixed TDMA. Similar hybrid methods may combine the advantages of different schemes.[88, 89, 90]

A standard for MAC protocols is yet to be developed for broadband satellite either global or regional networks providing differentiated services resulting in guaranteed Quality of Service to high bandwidth applications. [261]

2.5.3 Satellite Network: Transport Issues

Many enhancements for TCP have been proposed by the IETF considering large bandwidth-delay product, round trip time (RTT), non-congestion losses and bandwidth asymmetry. This section discusses the satellite link characteristics affecting TCP and a comparative analysis is provided in Section 2.5.3.3. Appendix 2A describes various TCP enhancements.

2.5.3.1 Satellite Link Characteristics Affecting TCP

Most of the global and regional satellite network architectures address supporting Internet applications. TCP/IP is the most widely used protocol suite accessing the Internet, and is one reason for its tremendous success. The performance of TCP and UDP in a satellite environment will be affected by large and changing latencies, bandwidth and path asymmetries, and occasionally high error rate on satellite links. The main characteristics of the end-to-end path that affects transport protocols are latency, bandwidth, packet loss due to congestion, and losses due to transmission error links. Recently, number of researchers have addressed this issue of TCP enhancements for satellite or large bandwidth delay environment.[112-122] In this section, an overview of the link characteristics affecting TCP performance under satellite environment is provided.

Latency

Among the three components of latency, propagation delay, transmission delay and queuing delay, propagation delay is the dominant part in broadband satellite links. In case of LEO satellites the propagation delay will vary, the connection path may change, and the queuing delay may be significant. Large variations of RTT may lead to false timeouts and transmissions. A long round trip time (RTT), in excess of 500 ms for GEO satellite networks, does not allow for quick feedback to the sender. This can reduce the rate of increase of the transmission rate, as well as the maximum transmission rate at the sender since TCP is self-clocking, based on acknowledgements. Furthermore, the sender's retransmission timeout value must reflect the long RTT. As a result, retransmission delays can be longer and reduces throughput.

Link Impairments

Satellite systems are subject to various impairments including multipath, interference, fading, rain attenuation and shadowing. Even though advanced modulation and adaptive coding techniques help to improve the normal BER of the order of 10^{-10}, normal satellite networks experience higher bit-error rate than terrestrial networks. TCP cannot distinguish between a segment loss due to congestion and a loss due to bit errors. Consequently, the sender will reduce its transmission rate even when segments are lost because of bit errors. This is an erroneous course of action since congestion may not exist in the network. Satellite networks can at times have a higher BER than terrestrial networks and so the assumptions of the congestion control algorithm can further reduce the throughput over satellites.

Bandwidth Asymmetry

In broadband satellite access networks bandwidth asymmetry in terms of forward to the return channels exist anywhere in the order of 10:1 or more. Network asymmetry affects the performance of TCP because the protocol relies on feedback in the form of cumulative acknowledgments from the receiver to ensure reliability. TCP uses the arrival rate of ACKs on the reverse time to control the flow of packets in the forward direction. For example a low bandwidth acknowledgment path can significantly slow the growth of the TCP sender window during slow start independent of forward link bandwidth. Thus bottlenecks or congestion in the reverse direction can lead to poor channel utilization in the forward direction. Bi-directional traffic flow may also exaggerate the problems of asymmetric channel bandwidth since ACKs must compete for bandwidth with traffic flows in the return direction.

Multiple Segment Loss

The bandwidth*delay product defines the maximum amount of data that can be in-flight (transmitted but unacknowledged). In connections with a large bandwidth*delay product, such as geostationary satellite networks, TCP senders and receivers with limited congestion/receive windows will not be able to take advantage of the bandwidth that is available. Those that can take advantage also increase their probability of a multiple segment loss within a single window. A multiple segment loss is a fundamental issue with TCP performance.

RTT Fairness

Because the TCP algorithm is self-clocking, based on received ACKs, several connections sharing the same bottleneck may see their clocks running at different speeds. A long RTT connection will not be able to increase its congestion window as quickly as a short RTT connection. The short RTT connection will unfairly capture a larger portion of the network bandwidth as a result. This is particularly true in the presence of congestion and subsequent loss, where the congestion window is opened linearly based on the RTT of the connection. Moreover, the congestion window of long RTT connections has “farther to go” before the optimal value is reached due to the large bandwidth*delay product. This further limits the throughput of long RTT connections through the bottleneck, as compared to shorter RTT connections.

2.5.3.2 Enhancements

The IETF TCP over Satellite working group has recently made a number of recommendations to enhance the performance of TCP over satellite links in its RFCs. The last two schemes listed below are non-TCP techniques:

– TCP selective acknowledgment (SACK) options [102, 103, 104] allow the receive to specify the correctly received segments. Thus, the sender needs to retransmit only the lost packets. TCP SACK can recover multiple losses in a transmission window within one RTT.

- TCP over transaction (T/TCP) [107] attempts to reduce the connection handshaking latency from two RTTs to one RTT, which is a significant improvement for short transmissions.
- Persistent TCP connection, supported in HTTP 1.1, allows multiple small transfers to download in a single persistent TCP connection. It is more efficient.
- The Path maximum transfer unit (MTU) discovery mechanism allows TCP to use the largest possible packet size, thus avoiding IP segmentation. It reduces the overhead and eliminates fragmentation and defragmentation.
- Forward Error Correction (FEC) is employed in link layer protocols to improve the quality of satellite links, but it should not be expected to fix all problem associated with manmade nose, such as military jamming, and some natural noise such as that caused by rain attenuation. Besides FEC, some other link layer approaches (e.g., bit interleaving, link layer automatic repeat request schemes) can also be used to improve packet error rate in transmission over satellite links. [24, 80]

TCP extensions can solve some of the limitations of standard TCP over satellite links, but other problems such as long end-to-end latency and asymmetry are not effectively addressed. One way to alleviate the effects of large end-to-end latency is to split the TCP connection into two or more parts at the ground stations connecting the satellite network and terrestrial networks. There are three approaches to splitting TCP connections over satellite links: [118, 132]

- TCP spoofing: The divided connections are isolated by the ground stations, which prematurely send spoofing acknowledgments upon receiving packets. The ground stations at split points are also responsible for retransmitting any missing data.
- TCP splitting: Instead of spoofing, the connection is fully split. A proprietary transport protocol can be used in a satellite network without interference to standard TCP in terrestrial networks. It is more flexible, and some kind of protocol converter should be implemented at the splitting points.
- Web caching: In contrast to the above two schemes, the TCP connection is split by a Web cache in the satellite network. Users in the satellite network connected to this Web cache need not set up TCP connections all the way to servers outside if the required contents are available from the cache. Web caching effectively reduces connection latency and bandwidth consumption.

2.5.3.3 Performance Comparison

The extent to which TCP protocol enhancements are able to improve the performance of TCP over a satellite system is shown in Table 2.7. [114]

Table 2.7: TCP Enhancements Comparison

TPCSat Limitations	Latency	Large Bandwidth Delay Product (BDP)	Impairments and Disconnections	Asymmetry	Implementation changes
T/TCP	X				Sender/receiver stack
Large Initial Window (IW)	X				Sender stack
Byte Counting	X			X	Research
Duplicate ACKs (DACKs)	X				Sender stack
TCP NewReno	X	X	X		
TCP SACK	X	X	X		Sender
TCP FACK	X	X	X		Sender stack
Explicit Congestion Notification (ECN)	X	X			Sender/receiver
TCP Vegas	X	X	X		
Windows Scaling	X	X			
PMTU Discovery	X				
Header Compression			X		Sender/receiver

2.5.3.4 Performance Enhancing Proxies (PEP) Mechanisms

To mitigate the disadvantages of TCP over long-latency links, researchers have been introducing performance-enhancing proxies (PEPs) into networks. [127-131] PEPs can be classified by layer (Transport Vs Application), by implementation distribution (single node Vs several nodes e.g., Two PEPs in a satellite link), by treatment of connections (splitting) in a satellite link or by degree of transparency (network layer, transport layer, application layer).

PEP mechanisms include ACK spacing, ACK regeneration (not in the draft yet), local acknowledgements, local transmissions, tunnels to control routing of packets, header compression, payload compression, and priority based multiplexing.

Some of the environments where PEPs are used include Satellite VSAT networks, (Mobile) Wireless WAN (W-WAN) networks, Wireless LAN (W-LAN) networks, Wireless Application Protocol (WAP) networks etc.

PEP implications are listed below.

End-to-end security

Since PEPs need to see inside IP packets and, in some implementations, generate IP packets on behalf of an end system, PEPs cannot be used with end-to-end IP Sec. Tunneled IPSec should be used with PEPs as the tunnel end points. This requires the PEPs to be trusted by the user. In general, security mechanisms at or above the transport layer (e.g. TLS or SSL) can be used with PEPs.

End-to-end fate sharing

Most PEP implementations keep state. A failure of a PEP implementation which keeps "soft" state may support fail over to alternate paths. A failure of a PEP implementation which keeps "hard" state (e.g. state required to support split connections) will generally cause a connection to fail even though an alternate end-to-end path exists for the connection.

End-to-end reliability

PEP Implementations may affect the end-to-end reliability of a connection, especially if the PEP interferes with application layer acknowledgements. Applications should not rely on lower level (e.g. TCP) acknowledgements to guarantee end-to-end delivery. TCP PEPs generally do not interfere with application layer acknowledgements.

End-to-end failure diagnostics

Using PEPs can replace certain constraints on the routing topology. Suboptimal routing might be required to force traffic to go through a PEP. Tunnels might be required to force traffic to go through a PEP, especially in an asymmetric routing environment. Using PEPs with mobile hosts might require PEP state to be handed off as the hosts move. PEP potentially interferes with the use of end-to-end failure diagnostics tools.

Use of PEPs and IPSec is generally mutually exclusive unless the PEP is also both capable and trusted to be the end point of an IPSec tunnel. Use of an IPSec tunnel is deemed good enough security for the applicable thread model. User/network administrator must choose between improved performance and network layer security. In some cases, transport of higher layer security can be used in conjunction with a PEP to mitigate the impact of not having network layer security. PEP itself must be protected from attack, represent a network point where the traffic is exposed. This makes it an ideal platform for launching denial of service or man in the middle attack. Taking the PEP out of action is a potential denial of service attack. PEP must be protected (e.g. by a firewall) or must protect itself from improper access by an attacker just like any other device which resides in the network.

TCP Splitting

Spoofing involves the transparent splitting of a TCP connection between the source and destination by some entity within the network path. The objective of spoofing involves isolating the long-latency link by introducing a middle agent which splits the TCP connection. However, unlike a proxy cache, spoofing is transparent to both sender and receiver. The responsibility of the spoofer is to intercept, cache, and acknowledge data received by the sender and then forward that data to the receiver. As a result, spoofing breaks the end-to-end semantic of TCP. In the following paragraphs, examples of two segment and three segment TCP splitting over a satellite are discussed.[112]

Two segment splitting scheme

The two segment splitting technique is to divide end-to-end TCP connections into two segments by inserting a gateway at a sender side as shown in Figure 2.6. This technique is of great use under the following scenarios:

– Networks with star topology consisting of a hub (central) station and many remote stations are widely used for distributing contents data from the hub station;
– TCP enhancement is usually needed in the outbound direction (hub to remotes) only;
– It may not be economically feasible either to introduce the gateway into every earth station or to install new TCP software that supports window scaling option etc. into all PC terminals.

In this scheme, transmit data speed from the TCP sender is enhanced by separating a satellite segment from the original TCP connection. It also appears for the sender that RTT is reduced compared with the standard TCP.

– In the split-TCP 2, the gateway can send data packets regardless of the TCP window size advertised from the receiver. It also employs the optimized congestion control for this segment to improve TCP performance.

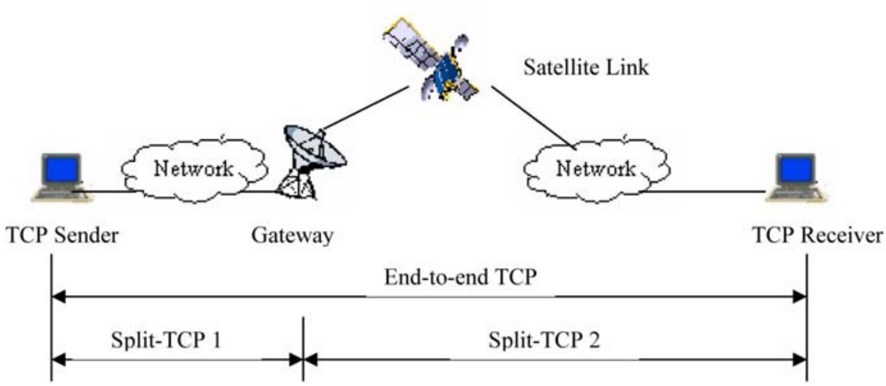


Figure 2.6: Two segment splitting scheme

Figure 2.7 illustrates the protocol model for the two segment split TCP. In this case a standard TCP implementation at the gateway/PEP which would allow a single PEP to be used in cases where one end system TCP at the terminal can be configured for satellite operation. Between the end user and the gateway the TCP connection benefits from throughput ramp-up during slow start transmission phase due to early local acknowledgements during data transfers. TCP enhancements such as large initial window sizes and window scaling improves performance over the long propagation delay TCP connection (Split-TCP 2).

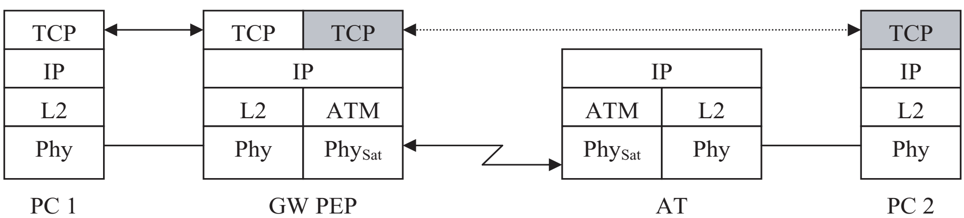


Figure 2.7: Protocol model for two segment TCP

Three segment splitting scheme

The three segment splitting scheme is to divide end-to-end TCP connections into three segments by two gateways. The concept is to separate the satellite segment from the terrestrial segments and use an optimized TCP protocol over the satellite link between gateways.

As illustrated in Figure 2.8, gateways are located at each earth station and an end-to-end TCP connection is divided into three segments. In the terrestrial segments, a standard TCP is used for communicating between TCP sender/receiver and gateway. The gateway converts TCP to a protocol optimized for satellite links. The main advantage of this method is that TCP enhancement is achieved without any modification to the PCs located at the sender and receiver side while other techniques require installing new TCP software. This scheme is especially suitable for point-to-point networks where enhancement for both-way is needed such as ISP backbone.

Since the gateway analyses a TCP header, it should be noted that the enhancement becomes unavailable when an encryption is made over TCP segment. This problem also occurs in two segment splitting technique.

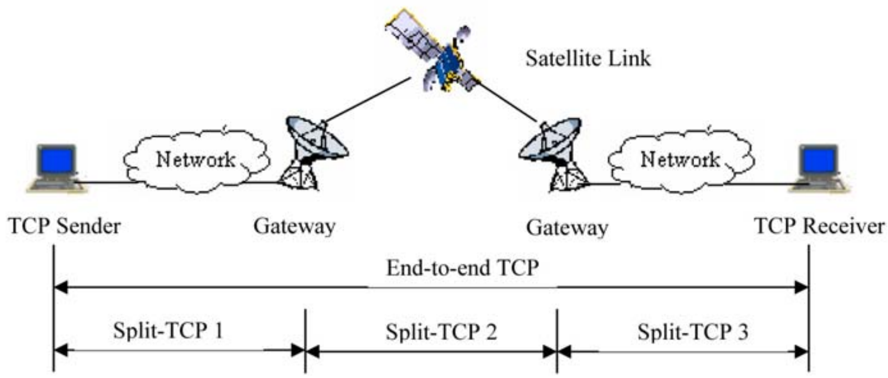


Figure 2.8: Three segment splitting scheme

The "segment splitting techniques" improve TCP performance over satellite links. In particular, two segment splitting technique results show that this method can improve TCP throughput to a great extent. Another advantage is found when using satellite networks with star topology by only having to install a gateway at the hub earth station. This enables improvement of all TCP throughputs for downloading files from hub to remote stations.

Figure 2.9 shows the protocol model for three segment TCP splitting. The use of a TCP over a satellite link provides a number of implementation advantages including the maintenance of a single transport protocol at the PEP. The window scaling and SACK TCP enhancements would provide good throughput performance for high bandwidth-delay products and/or impaired channels.

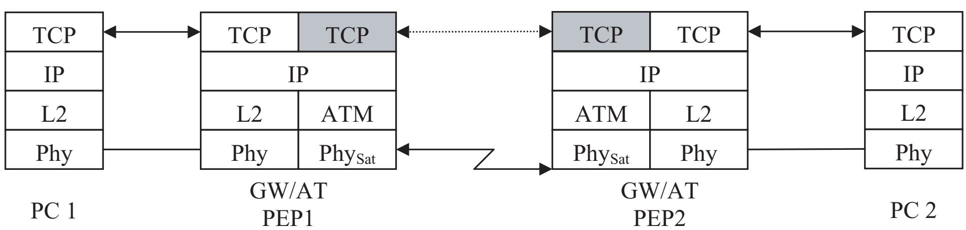


Figure 2.9: Protocol model for three segment TCP

2.5.3.5 Network Path Asymmetry

Asymmetric channel capabilities are exhibited by several network technologies, including:

- Direct broadcast satellite (e.g. an IP service using digital video broadcast (DVB) with an interactive return channel)
- Very small aperture satellite terminals (VSAT)
- Cable networks (e.g. DOCSIS cable TV networks)
- Asymmetric digital subscriber line (ADSL)
- Several packet radio networks

These networks are increasingly being deployed as high-speed Internet access networks, and it is therefore highly desirable to achieve good TCP performance over them. However, the asymmetry of the network paths often makes this challenging.

Several mitigations have been proposed and some evaluated in practice, to improve the efficiency of data transfer over connections with asymmetrical capability in each direction. These solutions use a combination of mechanisms, consisting of:

- Header compression of various kinds
- Reducing the frequency of TCP ACKs
- Techniques to handle this reduced ACK frequency to retain the TCP sender's acknowledgment-triggered self-clocking
- Scheduling data and ACK packets in the reverse direction to improve performance, in this presence of two-way traffic.

Tables 2.8 and 2.9 show the recommendations for possible mitigation measures concerning host and transparent modifications.[134, 135]

Table 2.8: Recommendations concerning host modifications

Modified Delayed ACKs	NOT REC
Large MSS & NO FRAG	REC
Large MSS & IP FRAG	NOT REC
ACK Congestion Control	EXPERIMENTAL
Window Predict. Mech (WPM)	NOT REC
Window Congestion window Est.	NOT REC
TCP Sender Pacing	EXPERIMENTAL
Byte Counting	NOT REC (*2)
Backpressure	EXP (*1)

*1) Implementation of this technique may require changes to the internal implementation of the protocol stack in end hosts.

*2) Dependent on a scheme for preventing excessive TCP transmission bursts in to the Internet.

Table 2.9: Recommendations concerning transparent modifications

Header Compression (V-J)	REC
Header Compression (ROHC)	REC *1
ACK Filtering (AF)	EXPERIMENTAL *2
ACK Decimation	EXPERIMENTAL *2
ACK Reconstruction (AR)	NOT REC
ACK Compaction/Compand	EXPERIMENTAL
Gen. Traffic Shaping (GTS)	REC
Fair Queuing (FQ)	REC
ACKs-First Scheduling	NOT REC

2.5.3.6 Stream Controlled Transmission protocol (SCTP)

The new transport protocol called Stream Controlled Transmission Protocol (SCTP)[224] was developed between the IETF and ITU-T SG-11 for the transport of PSTN signaling traffic over IP networks. The applicability and adaptation to satellite IP networks is further to be investigated.

2.6 Satellite ATM: Technical Issues

A brief discussion about the ATM technology and its applicability to satellites, technical challenges in terms of traffic management and congestion control for satellite network is provided in this section.

2.6.1 Satellite ATM Architectures

The interconnection of satellites with ATM technology has several advantages such as:

- Providing ATM services over a wide geographical area
- Providing system efficient communication using bandwidth on demand or DAMA capabilities for multimedia services
- Allowing good Quality of Service due to the traffic management function of ATM technology
- Easier addition of new lines to the network

There are two principal architectural options. One is to use ATM switching and processing with the onboard payload as in EuroSkyWay[52] and Spaceway [50]. Secondly, ATM switching can be used in the gateway on the ground, using non-processing or bent pipe payloads. The different advantages and disadvantages are discussed in the following sections.

2.6.1.1 Satellite ATM: Bent Pipe

The current GEO satellite networks at C- and Ku-band use bent pipe architecture. The bent pipe payloads serve as repeaters between two communication points on the ground, by receiving and retransmitting the signals. Because of the wide coverage of the beam, all the earth terminals, gateways and network control stations are within the coverage of a single beam.

On the other hand, satellite network with bent pipe architecture but with spot beams, takes advantage of frequency reuse. Also, each ground element can communicate with all other within the same beam via the satellite. Ground elements, which are not located within the same beam, cannot communicate via the satellite because the payload does not support ATM switching necessary for interbeam connectivity. This architecture requires a sophisticated terrestrial network infrastructure to provide interterminal communication. Alternatively, hub-spoke architecture can be designed with terminals being connected through a gateway and the gateways share routing information. This architecture requires two satellite hops as opposed to onboard ATM switching as discussed in 2.6.1.2.

2.6.1.2 Satellite ATM: On-Board Processing and Switching

A satellite ATM network with onboard processing and ATM switching allows terminals to communicate with other terminals, gateways and network control station in other beams. The capacity of the resulting network is larger than the bent pipe architecture as a result of frequency reuse. The onboard ATM switch does not require a centralized uplink for efficient downlink bandwidth utilization. An uplink Demand Assignment Multiple Access (DAMA) algorithm provides a flexible operation of the network. The traffic management functions [148] if performed on board the switching payloads, increases the complexity and weight and power requirements. Alternatively, the congestion control function and monitoring can be accomplished at the ground.

User applications, throughput, flexibility, performance, and QoS implementation complexity are some of the factors which determine the selection of onboard processing and switching versus bent pipe architectures. The selection is mainly driven by the driving application and the economics of the system and the comparative study is the subject of ongoing research.

2.6.2 Satellite ATM Technical Challenges

Satellite-ATM systems will play a significant role in achieving global connectivity by interconnecting geographically dispersed ATM networks. These systems will be able to achieve statistical multiplexing gains while maintaining Quality of Service (QoS) requirements. The ATM paradigm is aimed at supporting the diverse requirements of a variety of traffic sources, and providing flexible transport and switching services in an efficient and cost-effective manner. Hence, there is a growing interest in satellite-ATM networks [225, 226] and proposed broadband satellite systems plan to use onboard ATM or fast packet switching. However, certain design challenges must be addressed before these systems can be deployed. [153, 154.]

Table 2.10 provides some of these technical challenges which have to be addressed. For example, ATM has been developed for fiber networks that have low error rate with random packet loss. However, satellite links are characterized by much higher error rates with packet loss occurring in a burst. Given the limited error correction capability of ATM cell headers, for satellite ATM links, effective encoding techniques are required to reduce link error rate and randomize the packet loss. Similarly, limited bandwidth available on satellite links make it necessary to use DAMA techniques in order to support multimedia applications.[85]

Table 2.10: Satellite-ATM Technical Challenges

Attributes/Issues	Terrestrial ATM	Satellite ATM
Encoding for error control	(HEC) only header error control	Link encoding powerful, adaptive coding
Signaling	Standards Q.2931, SS7	Requires modifications
DAMA	No	For efficient resource utilization
Traffic management	Standard ATMF V.4.0	Requires modifications for onboard switching for ABR and UBR services
Switching	VP and VPI/VCI	VPI/VCI
QoS Propagation delays	Less impact, but number of hops during the path	IETF RFC 2488, ITU-R S.1420
User protocol interface	UNI, NNI, PNNI, B-ICI standards	Terminal and gateway implementation
Standards	Specification completed	ITU-R S.1420, TIA-TR34.1

2.6.3 Traffic Management and Congestion Control

Traffic management tries to maximize traffic revenue subject to constraints of traffic contracts, QoS, and “fairness.” The traffic management problem is especially difficult during periods of heavy load particularly if traffic demands cannot be predicted in advance. For this reason, congestion control is an essential part of traffic management.

Congestion control is critical to both ATM and non-ATM networks. Several congestion control schemes provide feedback to the hosts to adjust their input rates to match the available link capacity. One way to classify congestion control schemes is by the layer of the ISO/OSI reference model at which the scheme operates. For example, there are datalink, network, and transport layer congestion control schemes. Typically, a combination of such schemes is used both by networks and end systems. The effectiveness of a scheme depends heavily upon factors like the severity, duration, and location of the congestion.

2.6.3.1 Functional Allocation

The satellite ATM network has to implement the Traffic Management 4.0 [148] with necessary modifications to suit the satellite links. [227, 228] Table 2.9 provides the major functions of traffic management. The real challenge to the satellite ATM system designer is to decide where to implement the traffic management functions, e.g., power hungry functions like Usage Parameter Control (UPC) or Network Parameter Control (NPC) and resource management, whether in payload or at the Network Control Center (NCC). The satellite weight and power budgets dictate the functional allocations to the payload versus the NCC. The traffic management functional allocation for a broadband satellite network are shown in Figure 2.10.

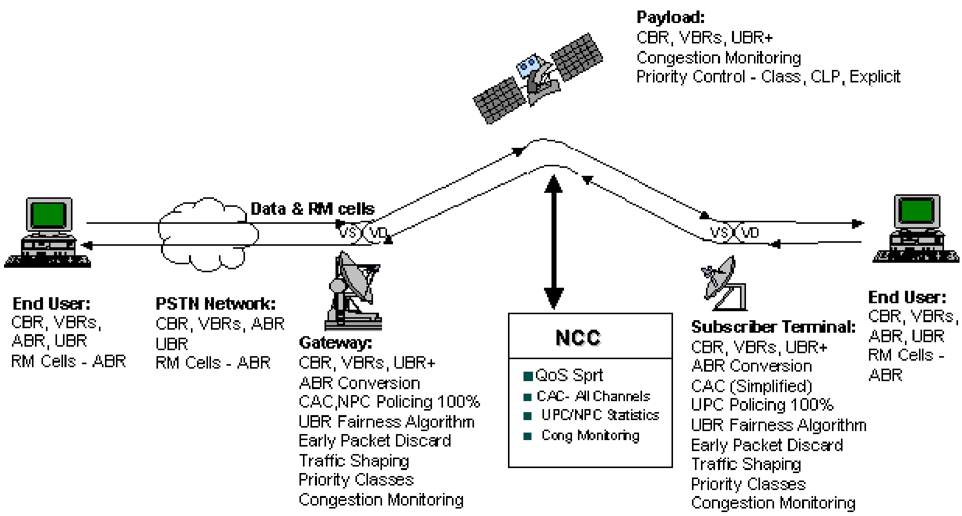


Figure 2.10: Traffic Management and Functional Allocation

Table 2.9: Traffic Management Functional Allocation for Satellite ATM Network

Function	Gateway	Payload	NCC	Terminal
Connection Admission Control (CAC)	X		X	X
Network Parameter Control (NPC)	X		X	
User Parameter Control (UPC)			X	X
Fairness Algorithm	X			X
Early Packet Discard	X			X
Traffic Shaping	X			X
Priority Control	X	X		X
Congestion Monitoring	X	X	X	X

The resource management cells are processed by the end users in the case of ABR service support [148].

The technical challenge to realize satellite ATM is to devise congestion control algorithms and implementation tradeoffs between the payload and the ground segment to optimize the space, weight and power requirements.

2.6.3.2 Traffic Management and QoS

One of the significant advantages of ATM technology is providing QoS guarantees as described in the ATM Forum's Traffic Management Specification [148]. The framework supports five service categories, namely constant bit rate (CBR), real-time variable bit rate (rt-VBR), non-real-time VBR (nrt-VBR), unspecified bit rate (UBR), and available bit rate (ABR), with an additional one, guaranteed frame rate (GFR). With the exception of UBR, all ATM service categories require incoming traffic regulation to control network congestion and ensure QoS guarantees. This function is done by access policing devices to determine whether the traffic conforms to certain traffic characterizations. The conformant cells are allowed to enter the ATM network and receive QoS guarantees, where as the non-conformant cells will be either dropped or tagged. Tagged cells may be allowed into the network but will not receive any QoS guarantees. The other traffic management functions include Connection Admission Control (CAC), traffic shaping, Usage Parameter Control (UPC), resource management, priority control, cell discarding, and feedback controls. Table 2.10 provides the various traffic parameters and the QoS parameters.

Table 2.10: ATM Service Category Attributes

Service categories	Traffic parameters			QoS Parameters			
	PCR, $CDVT_{PCR}$	SCR, MBS, $CDVT_{SCR}$	MCR	Pk-to-pk CDV	Max CTD	CLR	Others
CBR	Yes	No	No	Yes	Yes	Yes	No
rt-VBR	Yes	Yes	No	Yes	Yes	Yes	No
nrt-VBR	Yes	Yes	No	No	No	Yes	No
UBR	Yes	No	No	No	No	No	No
ABR	Yes	No	Yes	No	No	No	Feedback
GFR	Yes	MFS, MBS	Yes	No	No	No	No

PCR – Peak Cell Rate
SCR – Sustainable Cell Rate
MCR – Minimum Cell Rate
MBS – Maximum Burst Size
CDVT – Cell Delay Variation Tolerance
CDV – Cell Delay Variation
CTD – Cell Transfer Delay
CLR – Cell Loss Ratio

2.6.4 Explicit Rate Congestion Control for ABR

The explicit rate congestion control for ABR service has been reported widely in the literature and in Traffic Management specification 4.0 [163, 168]. However, this scheme needs to be analyzed in terms of the end-to-end delay requirements for satellite-ATM networks [166, 167]. The ABR service periodically advises sources about the rate at which they should be transmitting. The switches compute the available bandwidth and divide it fairly among active VCs. The feedback from the switches to the sources is indicated in Resource Management (RM) cells. The RM cells are periodically sent by the sources and turned around by the destinations.

The RM cells contain the current cell rate (CCR) of the source, in addition to several fields that can be used by the switches to provide feedback to the sources. One of those fields, the Explicit Rate (ER) field, indicates the rate that the network can support at that time. Each switch on the path of the VC reduces the ER field to the maximum rate it can support. The sources examine the returning RM cells and adjust their transmission rates. The Explicit Rate Indicator Algorithm (ERICA) and ERICA^{+} algorithms have been proposed and well documented for terrestrial ATM switches.

2.6.5 Virtual Source/Virtual Destination (VS/VD) algorithm for ABR

In the long propagation delay satellite configurations, the feedback delay is the dominant factor (over round trip time) in determining the maximum queue length. A feedback delay of 10 ms corresponds to about 3670 cells of queue for TCP over ERICA, while a feedback delay of 550 ms corresponds to 201,850 cells. This indicates that satellite switches need to provide at least one feedback delay worth of buffering to avoid loss on these high delay paths. Satellite switches can isolate downstream switches from such large queues by implementing Virtual Source/Virtual Destination (VS/VD) options. [147].

VS/VD option allows a switch to divide an ABR connection into separately controlled ABR segments. On one segment, the switch behaves as a destination end system, i.e., it receives data and turns around resource management (RM) cells (which carry rate feedback) to the source end system. On the other segment the switch behaves as a source end system, i.e., it controls the transmission rate of every virtual circuit (VC) and schedules the sending of data and RM cells. Such a switch is called "VS/VD switch". In effect, the end-to-end control is replaced by segment-by-segment control as shown in Figure 2.11.

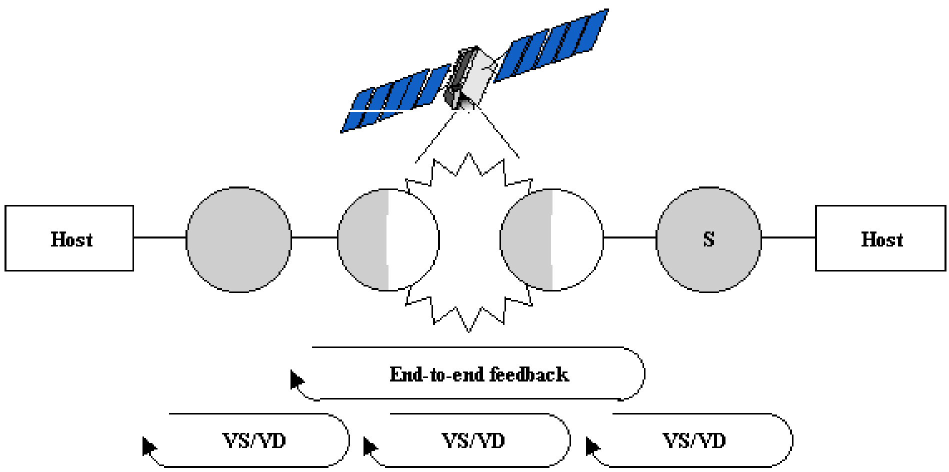


Figure 2.11 Virtual Source/Virtual Destination (VS/VD) Option for Satellite ATM-ABR

One advantage of the segment-by-segment control is that it isolates different networks from each other. An example is the interface point between a satellite network and a LAN. The gateway switches at the edge of a satellite network can implement VS/VD to isolate downstream workgroup switches from the effects of the long delay satellite paths (like long queues).

A second advantage of segment-by-segment control is that the segments have shorter feedback loops which can potentially improve performance because feedback is given faster to the sources whenever new traffic bursts are seen. The VS/VD option requires the implementation of per-VC queuing and scheduling at the switch. But if per-VC queuing and scheduling are already in place, the incremental cost to implement VS/VD is small.

Chapter 3 provides buffer requirements for handling TCP over satellite network supporting UBR. Several TCP mechanisms e.g., Vanilla, Reno, and SACK and cell discard policies e.g., EPD, SD are used to study the parameter impact and parameter interactions to achieve rate utilization and fairness in the satellite network. The results are discussed in Chapter 3.

2.7 IP based Satellite Network – Challenges and Issues

2.7.1 Satellite IP Routing

Satellite networks with onboard processing and switching capabilities allow direct interconnection between satellite terminals located in any satellite beam. Within a designated service coverage region, network management, onboard switch control, service access, routing and IP/ATM address translation function are managed by a network control center (NCC). The NCC hosts a network management system (NMS), a Switching Control System (SCS), and a Satellite Router System (SRS). The SRS is responsible for service management routing enforcing routing policies and connectivity constraints and IP/ATM address translation. Further details on SRS can be found in [229, 230].

2.7.2 Satellite IP QoS Challenges

QoS mechanisms provide service differentiation and performance measures for Internet applications. Service differentiation provides different services to different applications according to their requirements. Performance assurance addresses bandwidth, loss, delay, and delay variation. Bandwidth is a fundamental resource for satellite communication and its proper allocation determines the system throughput. End-to-end delay is also important for several applications. There are a variety of choices to provide the Internet QoS. These are Integrated Services (IntServ) [186, 187, 188], Differentiated Services (DiffServ) [190, 191, 192], and Multiprotocol Label Switching (MPLS) [214, 215, 216, 217]. However, the research of application of these QoS framework to broadband satellite network is required. Research must also be done to support IP QoS in a dynamic demand assignment capacity environment.

There are many issues for IP based networks and services and a lack of proven robust and scalable standard mechanisms for terrestrial and more so for satellite networks.

– Dynamic allocation of resource optimized for packet loss and delay.
– Assuring that the required end-to-end network performance objectives are achieved.
– Seamless signaling of the desired end-to-end QoS across both the network and interfaces.
– Performance monitoring of IP based networks and services consistent with planning method.
– Rapid and complete restoration of connectivity following severe outages or heavy congestion levels.

The QoS mechanisms recommended by the IETF, i.e, IntServ, DiffServ, Resource Reservation Protocol (RSVP)[189] and aggregate RSVP, and MPLS are described in chapter 4. This thesis focuses on the TCP performance study of satellite, buffer designs and performance analysis of TCP flavors and end system policies for ATM transport over satellite in chapter 3. IntServ, DiffServ based QoS architectures are proposed for satellite IP networks in chapter 4. Also, TCP, UDP performance over satellite network with differentiated services is analyzed in chapter 4.

2.7.3 Satellite Network Security

GEO satellite environment has some security challenges. Eavesdropping and active intrusion is much easier than in terrestrial fixed or mobile networks because of the broadcast nature of satellites. This can weaken the concept of firewalls for isolating company private data from intruders. Satellite channels experience high bit error rates, which may cause the loss of security synchronization. This demands a careful evaluation of encryption systems to prevent QoS degradation because of security processing. The security requirement for end-to-end communications depends on the applications and there is no single security solution for applications with varying requirements.

Satellite ATM Security: For satellite ATM transport ATM Forum has defined four security services. [173.]

User plane security: The user plane security defines the mechanisms to allow for secure communication between nodes in an ATM network, which can be subdivided into access control, authentication, data confidentiality, and data integrity.

Control plane security: The control plane defines the call control signaling needed to establish, maintain and close a certain Virtual Connection (VC). Thus, authentic signaling has been defined as the main target of control plane security for any endpoint-to-endpoint, switch-to-switch, or endpoint-to-switch signaling communication.

Support services: The support services define the certification infrastructures, the key exchange mechanisms, and the basic negotiation of security requirements and capabilities.

Management plane security: The management plane is responsible for both performing management functions for the system as a whole (plane management), and for performing network and system management functions such as resource management (layer management).

ATM Forum specifications address the security issues in terrestrial fixed network only. Considerable amount of research has to be done addressing security in a satellite environment which has high bursty error rates. It is important to investigate the encryption algorithms for high link data rates.

Internet Security (IPSec): The IETF has provided security standards for the Internet known as IP Security (IPSec). The IPSec protocol suite is used to provide interoperable cryptographically based security services (i.e. confidentiality, authentication and integrity) at the IP layer. It is composed of an authentication protocol - Authentication Header (AH), a confidentiality protocol - Encapsulated Security Payload {ESP), and an Internet Security Association Establishment and Key Management Protocol (ISAKMP).

Although these protocols satisfy to a large extent the needs of secure communications in the Internet, they are aimed mainly at unicast transmissions between one sender and one receiver.

IP AH and IP ESP may be applied alone or in combination with each other. Each protocol can operate in one of two modes: Transport mode or tunnel mode. In transport mode, the security mechanisms of the protocol are applied only to the upper layer data and the information pertaining to IP layer operation as contained in the IP header is left unprotected. In tunnel mode, both the upper layer protocol data and the IP header of the IP packet are protected or “tunneled” through encapsulation. The transport mode is intended for end-to-end protection that can be implemented only by the source and destination hosts of the original IP datagram. Tunnel mode can be used between firewalls.

PEP Security

Some of the current satellite networks employ PEP to enhance the TCP performance as discussed section 2.5.3.4. The PEPs do not attempt to replace any application level end-to-end function, but only optimize the performance to a subpath of the end-to-end path between the application endpoints. The potential end-to-end security implications in PEP implementations are discussed in the following paragraphs.

In most cases, security applied above the transport layer can be used with PEPs, especially transport layer PEPs but today only a limited number of applications include support for the use of transport (or higher) layer security. On the other hand, IPSec can be used by any application, transparently to the application.

A user or network administrator must choose between using PEPs and using IPSec. If IPSec is employed end-to-end, PEPs that are implemented on intermediate nodes in the network cannot examine the transport or application headers of IP packets because encryption of IP packets via IPSec’s ESP header (in either transport or tunnel mode) renders the TCP header and payload unintelligible to the PEPs.

If a PEP implementation is non-transparent to the users and they trust the PEP in the middle, IPSec can be used separately between each end system and PEP. This is not an acceptable alternative because, end systems cannot trust PEPs in general, this is less secure than end-to-end security, and it can lead to potentially misleading security level assumptions by the end systems. To prevent this, PEP could force the same level of security to each end system which again increases the complexity.

With a transparent PEP implementation, it is difficult for the end systems to trust the PEP because they may not be aware of its existence. Even if the user is aware of the PEP, setting up acceptable security associations with the PEP while maintaining the PEP’s transparent nature is problematic. Even when a PEP implementation does not break the end-to-end semantics of a connection, the PEP implementation may not be able to function in the presence of IPSec.

[127] has reported security implication mitigation methods such as end user using IPSec for some traffic and not for other, and implementing IPSec between two PEPs of a distributed PEP implementation. In both the cases significant complexity being added to the end system implementation has been reported. A further research has to be undertaken for developing alternative approaches for achieving end-to-end secured satellite networks.

2.8 Technical Challenges Summary

Future satellite communication networks are intended to provide global connectivity and regional access. One of the main applications driving the system designs is high speed Internet access for bandwidth rich streaming applications. Many of these systems are being designed at Ka-band, employ onboard processing and either ATM or fast packet switching. These new system architectures enhance flexibility and efficiency of operation but increase the implementation complexity. The new system architectures must support the application QoS. To achieve this goal a number of technical challenges must be addressed.[284] These include:

- ***Architecture selection***: The first fundamental decision is the baseline architecture to be selected. Questions such as (a) GSO system vs. NGSO (MEO or LEO) systems vs. hybrid systems (employing both GSO and NGSO), (b) bent pipe vs. on-board processing, (c) global vs. regional coverage, and (d) operating frequency bands (Ka, Ku, C, X, and/or Q/V), must be decided.
- ***Coverage***: Tradeoffs must be made between a wide area coverage and narrow spot beams. Wide coverage satellite antennas have lower antenna gain, and therefore require larger user terminals with perhaps higher terminal RF power. Narrow spot beam satellite antennas have higher gain, and thus require smaller user terminals, but are more complex.
- ***User Terminals***: An important system consideration is the user terminal antenna size. Large antennas are expensive, undesirable for consumer applications and in some cases are prohibited by local building codes or other regulations. Antenna for high frequencies such as Ka-band are more expensive due to the need for greater precision in manufacturing. The narrow beam widths of Ka-band antennas requires accurate pointing to the satellite and maintaining pointing alignment in the presence of wind, snow, and settling of the building where it is mounted. The larger the antenna the smaller the beamwidth and thus the more sever the pointing problem.
- ***Availability***: Satellite communication systems operating at Ka-band are susceptible to RF link degradation due to attenuation by rain. The magnitude of the loss increases with frequency and challenges Ka-band system designers in several ways.
- ***Applications and Services***: Business applications must be developed for satellite networks. Though the satellite systems provide more power and bandwidth through the use of spot beam technology, new "killer" applications drive the architectures.
- ***Traffic Management***: Traffic Management functions especially, congestion control algorithms to be developed and performance over satellite networks with TCP/UDP must be developed.
- ***TCP Enhancements***: Several TCP enhancements proposed require to be tested and performance evaluated over satellite IP networks, under lossy conditions and asymmetry.
- ***QoS for satellite ATM and satellite IP***: New QoS architectures for satellite IP networks, DVB networks, and access networks must be developed. Application QoS should be simulated and mapped to classes of service.

- ***Security***: Security methods proposed by IETF are being with satellite IP network with PEPs. Since PEPs need to see inside IP packets and, in some implementations, generate IP packets on behalf of an end system, PEPs cannot be used with end-to-end IP Sec. Tunneled IPSec should be used with PEPs as the tunnel end points. This requires the PEPs to be trusted by the user. A further research has to be undertaken for developing alternative approaches for achieving end-to-end secured satellite networks.
- ***Standards Development***: Standards for the satellite IP QoS and access control protocols for the return channel must be developed. The standard development must be coordinated by the IETF, ITU, TIA, and ETSI.

3 Quality of Service for Satellite ATM Connectivity Networks

Many of the proposed Ka-band satellite systems will use ATM technology to seamlessly support Internet traffic. ATM was originally designed for fiber-based terrestrial networks that exhibit low latencies and error rates. With the increasing demand for electronic connectivity across the world, satellite networks play an indispensable role in the deployment of global networks. Ka-band satellites using the GHz frequency spectrum can reach user terminals across most of the populated world. ATM-based satellite networks can effectively provide real-time as well as non-real-time communications services to remove areas. [222, 279, 280]

The large delays in geostationary Earth orbit (GEO) systems and delay variations in low Earth orbit (LEO) systems affect both real-time and non-real-time applications. In an acknowledgment- and time-out-based congestion control mechanism (like TCP), performance is inherently related to the delay-bandwidth product of the connection. Moreover, TCP round-trip time (RTT) measurements are sensitive to delay variations that may cause false timeouts and retransmissions. As a result, the congestion control issues for broadband satellite networks are somewhat different from those of low-latency terrestrial networks.

Several design options exist for transporting TCP over satellite with onboard ATM switch. These design options include: (a) end system policies – Vanilla TCP, Selective ACK, Fast Retransmit, and Recovery (b) buffer management – Early Packet Discard (EPD), Tail Drop, per-VC accounting (c) feedback control – explicit rate, end-to-end, hop-by-hop and (d) queue management – per-class queuing, per-VC queuing. During recent years traffic management for terrestrial networks has received a great attention [148]. Buffer design of the onboard ATM switch for a GEO satellite network providing end-to-end connectivity is discussed.

In this chapter, QoS models and simulation results for broadband satellite network with onboard ATM switch are presented. The areas of contribution include, buffer requirements, TCP enhancements analysis, and bandwidth allocation via Demand Assignment Multiple Access for GEO, LEO and MEO networks. Specifically, the following three problems are addressed in this chapter:

Problem 1: Buffer requirements for TCP over satellite ATM supporting UBR service.

A simulation study for the buffer requirements for TCP over ATM based satellite networks is performed. Among the service categories provided by ATM networks, the most commonly used category for data traffic is the Unspecified Bit Rate (UBR) service. UBR allows sources to send data into the network without any network guarantees or control. Simulation results for buffer requirements for TCP over UBR transport over a GEO satellite network with onboard ATM switch are given in Section 3.4.

Problem 2: TCP enhancements analysis for satellite ATM UBR+ service.

Analysis of variation technique has been applied to study the influence of end system policies and the switch parameters on the efficiency and fairness in a large delay x bandwidth environment. Simulation results on the performance of TCP enhancements for GEO satellite ATM network for UBR+ service, with limited switch buffers for WWW traffic are given in Section 3.5. The end system policies considered include Fast Retransmit and Recovery, New Reno, and Selective acknowledgement (SACK) and the drop policies, Early Packet Discard (EPD) and Selective Drop (SD) are considered.

Problem 3: Bandwidth allocation via DAMA for satellite ATM for multimedia.

Section 3.6 presents a Demand Assignment Multiple Access (DAMA) analytical model for multimedia satellite network. Analysis and simulation results for a non-contiguous and contiguous resource allocation for CBR and VBR service requests are given. Results are summarized in Section 3.7.

3.1 ATM Quality of Service Model

ATM networks carry traffic from multiple categories, and support QoS requirements for each service category. [148] defines five service categories for ATM networks. Each service category is defined using a traffic contract and a set of QoS parameters. The *traffic contract* is a set of parameters that specify the characteristics of the source traffic. This defines the requirements for compliant cells of the connection.

The traffic contract consists of the *source traffic descriptors.* These are used to specify the characteristics of the traffic from a source end system. The Peak Cell Rate (PCR) specifies the maximum rate at which a source can send at any time. The Sustained Cell Rate (SCR) specifies the average rate maintained by the source. The Maximum Burst Size (MBS) specifies the maximum number of back-to-back cells at PCR that can be sent by the source without violation of the SCR. The Cell Delay Variation Tolerance (CDVT) and Burst Tolerance (BT) parameters are used to specify a tolerance for PCR and SCR respectively. The Generic Cell Rate Algorithm (GCRA) specified in (a version of the leaky bucket algorithm) uses the PCR/SCR and the respective tolerance parameters to ensure that the incoming cells are compliant with the traffic contract.

BT = (MBS – 1)(1/SCR – 1/PCR).

Figure 3.1 shows the traffic parameters measured at the User Network Interface (UNI). The QoS parameters are negotiated by the source with the network, and are used to define the expected QoS provided by the network.

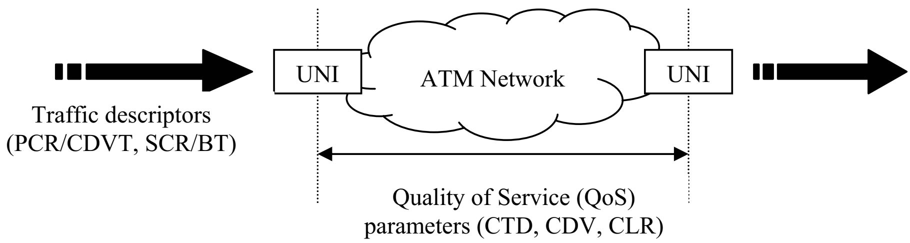


Figure 3.1: ATM QoS Model

The parameters measured from UNI to UNI are:

Cell Transfer Delay (CTD)

CTD is defined as the elapsed time between a cell exit at the measuring point MP1 e.g., source UNI, and the corresponding cell entry event at measurement point MP2 e.g., the destination UNI, for a particular connection. This delay consists of both a cell processing delay and a variable queuing delay at the switch.

Cell Delay Variation (CDV)

Two performance parameters associated with CDV are the One-Point CDV and the Two-Point CDV which are defined below:

1-point CDV at a Measurement Point: *The 1-point CDV (y_k) for cell* k *at an MP is the difference between the cell's reference arrival time (c_k) and actual arrival time (a_k) at the MP:*

The reference arrival time pattern (c_k) is defined as follows:

$$c_0 = a_0 = 0$$

$$c_k + 1 = \begin{cases} c_k + T \text{ when } c_k \geq a_k \text{ or when cell k does not arrive} \\ a_k + T \text{ otherwise} \end{cases}$$

Positive values of 1-point CDV (early cell arrivals) correspond to cell clumping; negative values of 1-point CDV (late cell arrivals) correspond to gaps in the cell stream.

Cell Delay Variation between two MPs (2-point CDV): The 2-point CDV (v_k) for cell k between MP1 and MP2 is the difference between the absolute cell transfer delay (x_k) of cell k between the two MPs and a defined reference cell transfer delay ($d_{1,2}$) between the same two MPs:

$$v_k = x_k - d_{1,2}$$

The absolute cell transfer delay (x_k) of cell k between MP1 and MP2 is the difference between the cell's actual arrival time at MP2 (a_{2k}) and the cell's actual arrival time at MP1 (a_{1k}) as in Figure 3.2:

$$x_k = a_{2k} - a_{1k}$$ [1]

The reference cell transfer delay ($d_{1,2}$) between MP1 and MP2 is the absolute cell transfer delay experienced by cell 0 between the two MPs.

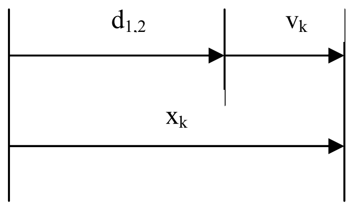


Figure 3.2: Cell Delay Variation 2-point definition

where

$d_{1,2}$ Absolute cell transfer delay between MP1 and MP2

x_k Absolute cell k transfer time between MP1 and MP2

v_k 2-point CDV value between MP1 and MP2

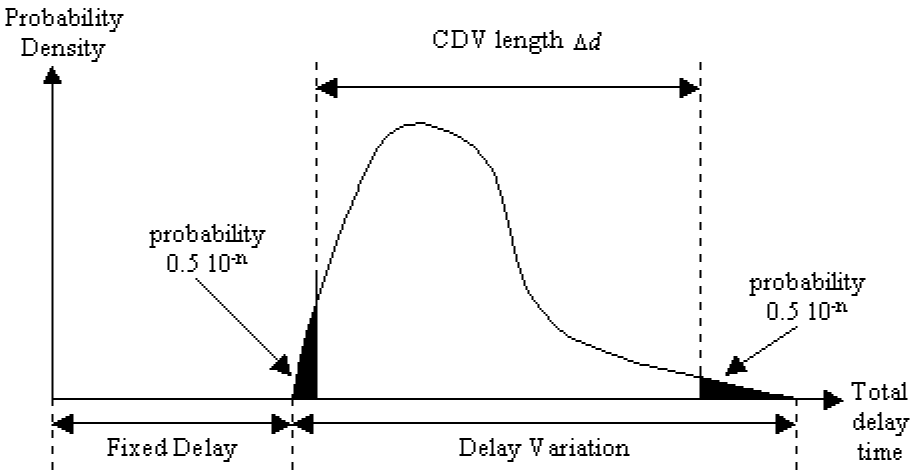


Figure 3.3: Definition of CDV length

Figure 3.3 illustrates the quality specification for the width of the CDV distribution. The delay time contains the fixed part composed of the propagation delay and the processing delay as well as the varying part due to the fluctuations in various delays (such as waiting time). The width of the CDV distribution is specified using the pair (n, Δd) where Δd is the width of the value between the point where tail of the varying part is 10^{-n}. The 2-point CDV is defined [231] as the upper bound on the difference between upper and lower 10^{-8} quantiles of CTD. Hence n should be taken as 8 and Δd for real-time services has been recommended as 3ms [231]. For non-real-time services this value is expected to be between 600ms and several seconds. The magnitude of the delay variation produced in the satellite link is larger than that of the terrestrial network hence CDV in the satellite link is a serious problem. The most widely used method to compensate for CDV is the use of a shaper buffer at the receiver terminal.

Cell Loss Ratio (CLR)

The CLR is the ratio of the total lost cells to total transmitted cells in a population of interest. There are three different CLR definitions depending on the priority of the traffic. These are CLR_0, CLR_{0+1}, CLR_1. The requested CLR will be an upper bound on the cell loss probability.

CLR_0 is the ratio of total lost cell with high priority and the number of corresponding tagged cells to the number of Cell Loss Priority (CLP) = 0 transmitted cells.

CLR_{0+1} is the ratio of lost cells to the total number of generated cells.

CLR_1 is the ratio of lost cells with CLP = 1 to the number of transmitted cells with CLP = 1.

Cell Error Ratio (CER)

The CER is defined as the ratio of the errored cells to the successfully transferred cells plus the errored cells.

Severely Errored Cell Block Ratio (SECBR)

The SECBR is defined as the ratio of the severely errored cell blocks to the total transmitted cell blocks.

Cell Misinsertion Rate (CMR)

The CMR is defined for a connection as the ratio of misinserted cells to the time interval.

The Constant Bit Rate (CBR) class defined for traffic that requires a constant amount of bandwidth, specified by PCR, to be continuously available. The network guarantees that all cells emitted by the source that conform to this PCR will be transferred by the network at PCR.

The real-time Variable Bit Rate (rt-VBR) class is characterized by PCR, SCR and MBS that controls the bursty nature of VBR traffic. The network attempts to deliver cells of these classes within fixed bounds of cell delay (max-CTD) and delay variation (peak-to-peak CDV). Non-real-time Variable Bit Rate (nrt-VBR) sources are also specified by PCR, SCR and MBS, but are less sensitive to delay and delay variation than the real time sources. The network does not guarantee the CTD and CDV parameters for nrt-VBR.

The Available Bit Rate (ABR) service category is specified by a PCR as well as an MCR which is guaranteed by the network. The bandwidth allocated by the network at an ABR connection may vary during the life of a connection, but may not be less than MCR. ABR connections use a rate-based closed-look feedback-control mechanism for congestion control. The network tries to maintain a low CLR by changing the allowed cell rates (ACR) at which a source can send.

The Unspecified Bit Rate (UBR) class is intended for best effort applications, and this category does not support any service guarantees. UBR has no built in congestion control mechanisms. The UBR service manages congestion by efficient buffer management policies in the switch.

3.2 Satellite ATM Network Model

As discussed in Section 2.4.6, most of the next generation multimedia satellite systems have in common, features like onboard processing, ATM or fast packet switching. These systems include user terminals, gateways and employ common protocol standards. Some of them might use inter-satellite links. Figure 3.4 illustrates a typical broadband satellite network architecture for the discussion purposes in this chapter. The network consists of a ground segment, a space segment, and a network control segment [232, 233]. The ground segment consists of terminals and gateways (GWs) which may be further connected to other legacy public and/or private networks. The Network Control Station (NCS) performs various management and resource allocation functions for the satellite media. Inter-satellite crosslinks in the space segment provide seamless global connectivity via the satellite constellation. The network allows the transmission of ATM cells over satellite, multiplexes and demultiplexes ATM cell streams for uplinks, downlinks, and interfaces ATM networks as well as legacy LANs.

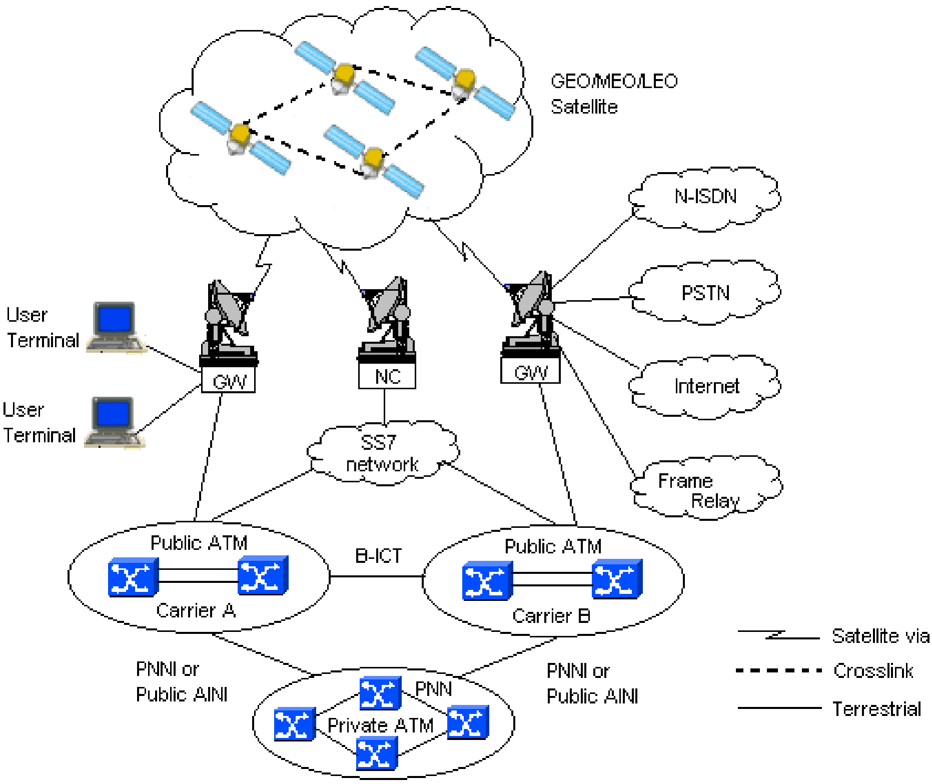


Figure 3.4: Broadband Satellite Network Architecture

3.2.1 Ground Segment

The ground segment consists of gateways, user terminals and control segment.

Gateways (GWs): The gateways support several protocol standards such as ATM User Network Interface (ATM-UNI), Frame Relay UNI (FR-UNI), Narrow-band Integrated Service Digital Network (N-ISDN), and Transmission Control Protocol/Internet Protocol (TCP/IP). The gateways interface unit provides external network connectivity. The number and placement of these gateways in both GEO and LEO systems depend on the traffic demand, performance requirements, and other international regulatory issues.

User Terminals (UTs): The user Terminals Interface Unit (TIU) supports several protocol standards adapting to the satellite network interface. It includes the physical layer functionalities such as channel coding, modulation/demodulation, and other RF functions. Different types of terminals might support transmission rates starting from 16 kbps, 144 kbps, 384 kbps, or even 2.048 Mbps.

Control Segment: A Network Control Station (NCS) performs various control and management functions, e.g., configuration management, resource allocation, performance management, and traffic management. The number and location of these NCSs depend on the size of the network, coverage, and other international standards and regulatory issues.

3.2.2 Space Segment

The space segment consists of either a GEO, MEO or LEO constellation depending on the system design as discussed in Chapter 2. In the multimedia systems, within payloads full onboard processing and ATM or fast packet switching is assumed. The onboard functions include multiplexing/demultiplexing, channel encoding/decoding, packet modulation/demodulation, and formatting. Many of the switching units are being developed for satellite networks.

3.2.3 Interfaces

Interconnectivity to the external private or public networks is possible with the support of the standard protocol. For the satellite ATM case, the signaling protocols based on ITU-T Q.2931 can be used when necessary. For other networks, the common channel signaling protocol, e.g., Signaling System No. 7 (SS7), can be used. The other interconnection interfaces between public and private ATM networks are the ATM Inter-Network Interface (AINI), the Public User Network Interface (PUNI) or the Private Network-Network Interface (PNNI), and the default interface between two public ATM networks, namely, the B-ISDN Inter Carrier Interface (B-ICI). However, these interfaces require further modifications to suit the satellite interface unit development. There is a definite need for an integrated satellite-ATM network infrastructure and standards for interfaces and protocols are in development process.

3.3 Satellite ATM QoS Requirements

This section identifies the QoS requirements that will be impacted by the Time Division Multiple Access (TDMA) frame structures and time slot allocations. The objectives for the satellite ATM network are also described.

3.3.1 ATM QoS objectives

Table 3.1 summarizes the QoS class definition and the network performance objectives established for end-to-end ATM connections. ITU-T Rec G.114 specifies 400 ms as the delay limit for voice. This is an end-to-end application delay (which can include coder delay, packetization delay, and AAL delay). This recommendation implies that if the 400 ms delay value is exceeded, the network planners should be aware that a degradation in quality can occur. It is the difficult to meet the CDV objectives on low-rate ATM links.

To accommodate the characteristics and the requirements of various traffic types, the following classes of service have been defined. Class 1 (stringent class) is a delay-sensitive class and it is intended to support constant bit rate (CBR) and real-time variable bit rate (VBR) services such as telephony and videoconference. Class 2 (tolerant class) is a delay-tolerant class and supports available bit rate (ABR) and non-real-time VBR services such as video and data. Class 3 (bi-level class) supports VBR and ABR services such as high-speed data. Class 4 (unspecified class) supports unspecified bit rate (UBR) services such as file transfers and email.

Table 3.1: QoS class definitions and network performance parameters

Default objectives	CTD	2-pt. CDV	CLR_{0+1}	CLR_0	CER	CMR	SECBR
	No default	No default	No default	No default	4×10^{-6}	1/day	1×10^{-4}
QoS classes:							
Class 1 (stringent class)	400 ms	3 ms	3×10^{-7}	None	Default	Default	Default
Class 2 (tolerant class)	Unspecified	Unspecified	1×10^{-5}	None	Default	Default	Default
Class 3 (bi-level class)	Unspecified	Unspecified	Unspecified	1×10^{-5}	Default	Default	Default
Unspecified class	Unspecified	Unspecified	Unspecified	Unspecified	Unspecified	Unspecified	Unspecified

3.3.2 Performance Objectives for Satellites (Class 1 Service)

The QoS class required by each application is part of the contract negotiation procedure between the user and the network. If the network can provide the requested service level, the connection will be established. If there is any performance objective that cannot be met, the connection will be denied. Once a connection is established, the network must ensure that the performance objectives of the QoS class are met during the connection. The performance objectives for satellite ATM are given in Table 3.2.[170]

Table 3.2: ATM performance objectives for satellites (Class 1 services)

Performance parameters	ITU objective end-to-end	ITU objective satellite
CLR	3×10^{-7}	7.5×10^{-8}
CER	4×10^{-6}	1.4×10^{-6}
SECBR	1×10^{-4}	3.5×10^{-5}
CTD	400 ms	320 ms (maximum)
CDV	3 ms	Negligible
CMR	1 per day	1 per 72 hours

3.3.3 CTD and CDV Contributions

The sources of CTD and CDV in a broadband satellite network consist of the following items:

- Transmit Terminal Queuing Delay
- Transmission Delay
- Propagation Delay
- MAC Layer Delays
- Onboard Processing Delays
- Receive Terminal Smoothing Buffer
- Timing Synchronization

The impact of each of these sources is addressed below.

3.3.3.1 Transmit Terminal Queuing Delay

Transmit terminal queuing delay is associated with variations in the cell arrival rate which would cause the arrival at a particular queue to exceed the queue service rate. This delay is variable in nature and therefore contributes to the CDV QoS performance. For the CBR and rt-VBR services, it is expected that the allocated ATM bandwidths for such connections will correspond to the connection PCRs so as to minimize the queuing delay.

3.3.3.2 Transmission Delay

The transmission delay is a fixed component of the overall CTD budget for a connection. It corresponds to the time required to transmit cells at a given carrier rate. In general, the transmission delay is small compared to the propagation delay.

3.3.3.3 Propagation Delay

The propagation delay from a transmit terminal to a receive terminal (i.e., round trip delay) has a fixed and time-varying components. The fixed component of the delay budget for a geostationary satellite is about 264 ms for links among terminals with minimum elevation angles of 20^{o}. The variable component is due to satellite station keeping. The selection of satellite network architecture depends on the choice of the satellite orbit which is dependent up on the application it is intended for. For example, the propagation delay is larger in GEO as compared to MEO or LEO satellites. However, the delay variation due to routing has to be addressed in the case of LEO and MEO networks. An end-to-end delay example for GEO and LEO is presented in Appendix 3A. The buffering delay analysis forms the main focus of Sections 3.4 and 3.5.

3.3.3.4 MAC Layer Delays

MAC delays are introduced at the ATM layer by the underlying MAC layer. In the broadband satellite network, the time slot allocation within a frame can cause cells arriving during a TDMA frame to wait for an available time slot to be transmitted. This could introduce cell delay variations, which may be as large as one TDMA frame duration. In addition, some delay variations may be introduced when cells from different Virtual Connections (VCs) arriving on a high-speed ATM interface are transmitted on lower speed links which causes cell-clumping. The CDV due to cell clumping is not expected to be a major contributor to the CDV budget.

3.3.3.5 Onboard Processing Delays

The onboard switch contributes to CTD of about 20 ms in addition to the transfer delays in the priority queues. It also yields a CDV of up to 2.5 ms in addition to the delay variation in the priority queue buffer, where the CDV is defined as the standard deviation of CTD. Both the CTD and CDV are variable and depend on the traffic loading of the switch. The 2-point CDV is defined as the upper bound on the difference between upper and lower 10^{-8} quantiles of CTD. The onboard buffer sizing is critical for the QoS.

3.3.3.6 Receive Terminal Smoothing Buffer

The traffic shaping buffer is typically at least one TDMA frame long and could introduce an additional delay and delay variation. Proper buffer design is critical to minimize the impact on the QoS performance.

3.3.3.7 Timing Synchronization

At the transmit terminal, the receive clock from the terrestrial equipment may be slightly different from the TDMA transmit frame clock. This will cause either the cell arrival rate to exceed the transmit rate or the cell transmit rate to exceed the arrival rate. In the former case extra cells may be delayed for one frame or may be discarded. The latter case may require either transmission of a lesser number of cells in a frame or an insertion of unwanted cells. The timing accuracies (earth terminal clock, onboard clock and terrestrial clock), and the interface timing techniques impact the QoS performance.

3.4 Buffer Requirements for Satellite ATM – UBR Service

Traffic management tries to minimize traffic revenue subject to constraints of traffic contracts, QoS and fairness. The traffic management problem is especially difficult during periods of heavy load particularly if traffic demands cannot be predicted in advance. For this reason, congestion control [234, 281, 282] is an essential part of traffic management.

Congestion control is critical to both ATM and non-ATM networks. Several congestion control schemes provide feedback to the hosts to adjust their input rates to match the available link capacity. One way to classify congestion control schemes by the layer of the ISO/OSI reference model at which the scheme operates. For example, there are datalink, network and transport layer congestion control schemes. Typically, a combination of such schemes is used both by networks and end systems. The effectiveness of a scheme depends heavily upon factors like the severity, duration and location of the congestion. [235, 236]

For the congestion control schemes to function properly, the onboard payload buffer requirements have to be developed. This section develops a buffering model and simulation results are provided in Section 3.4 for TCP traffic over Satellite ATM providing UBR service.

3.4.1 Buffering Delay

Data buffering is required for following reasons:

- To temporarily store cells at the payload until resources are available to process and forward them to the downlink
- To allocate the available capacity of the communications link among various flows in a “fair” manner, in accordance with the transmission policies
- To temporarily store cells until their recipient is acknowledged by the destination
- To allow retransmission of any cells which are not received by the destination (if needed)

The buffering delay at any buffer is a function of the buffer size divided by the rate at which cells are serviced from the buffer. To minimize buffering delay, buffer size must be kept as small as possible. Buffer size, however, must be large enough to avoid frequent loss of cells due to buffer overflow, which would require retransmission of the lost cells with consequently increased transmission delay for cell transfer and loss of network efficiency.

Buffers are necessary at any network congestion point and at any node where several data flows are aggregated into a single data stream. In a satellite network, buffers are usually placed at the input to the satellite network (the source earth station) and, if the satellites aggregate multiple data streams, also at the input of each satellite with onboard ATM switch. Clearly, since a typical transmission via a LEO network involves relaying by a greater number of satellites, transmission via a LEO network will generally involve a much larger number of buffers than transmissions via an equivalent GEO network.

3.4.2 Buffering Requirements

In order to assess the minimum buffering requirements, the behavior of the selected data transmission protocol under the various impairments of the space communications path must be analyzed and/or simulated. Since all Internet traffic is carried using the TCP/IP protocol suite, considerable resources in both industry and standard organizations have been dedicated to enhance the performance of the TCP protocol over satellite networks. Many approaches incorporate special protocols only within the space segment of the network and provide gateways at the interface to make the satellite network appear more or less "transparent" to the terrestrial network. In order to determine the minimum buffering requirements, the following parameters must be considered.

Propagation delay: In any network, the functions of flow control, error detection and recovery must be performed at some protocol layer. For Internet traffic, this function is performed by the TCP protocol. TCP performs its control functions by the return transmission of acknowledgment messages denoting the receipt of specific message segments. When a segment has not been acknowledged within certain limits, the segment is assumed lost and is retransmitted by the source. If several segments in sequence are lost, TCP assumes congestion in the network and reduces the rate of data transmission. The round trip propagation time through the network will affect the time for recovery of lost data segments and regaining optimum transmission speed and therefore directly affects the overall transmission efficiency.

Delay variation: As discussed earlier, delay variation, including discontinuities due to handovers, particularly in LEO satellite networks, must be taken into account in the determination of buffer sizes.

Link error characteristics: In space communications, bit-errors caused by noise are not uncommon. To some degree, forward error correction can compensate for such errors, but the space link is rarely as clean as those of modern terrestrial networks. The effects of transmission efficiency of the loss of packets due to noise and the consequent retransmission of data segments must be considered when determining buffer sizes.

Transmission protocol characteristics: ATM provides different quality of services to applications such as real-time voice traffic and e-mail transfers which requires assigning different priorities. Generally, different priority traffic is queued in separate buffers. The Quality of Service categories to be provided by the network and the prioritization policies of the network will also affect the number and size of buffers that must be provided. Network design must be provided for scalability as the number of independent data flows is increased and fairness in the assignment of resources among data flows.

3.4.3 TCP Transport over Satellite ATM – UBR

TCP/IP is the most popular network protocol suite and hence it is important to study how well these protocols perform on long delay satellite links. The main issue affecting the performance of TCP/IP over satellite links is very large feedback delay compared to terrestrial links. The inherent congestion control mechanism of TCP causes source data rate to reduce rapidly to very low levels with even a few packet loss in a window of data. The increase in data rate is controlled by ACKs received by the source. Large feedback delay implies a proportional delay in using the satellite link efficiently again. Consequently, a number of TCP enhancements (NewReno[101], SACK[102, 103]) have been proposed that avoid multiple reductions in source data rate when only a few packets are lost. These enhancements also avoid resending packets already received at the destination. It is important to study the effectiveness of these enhancements in achieving better performance on satellite links [162, 283]. The enhancements in end-to-end TCP protocol are called End System Policies. Satellite ATM link performance can also be improved by using intelligent switch policies associated with multiple hop satellite links. Since UBR is the cheapest service provided by ATM networks, majority of Internet Service Providers may use ATM-UBR service for their TCP/IP traffic. [164].

The Early Packet Discard (EPD) policy [237] maintains a threshold R in the switch buffer. When the buffer occupancy exceeds R, all new incoming packets are dropped. Partially received packets are accepted if possible. In terrestrial networks, EPD improves the efficiency of TCP over UBR but does not improve fairness. The effect of EPD on satellite latencies has not been exhaustively studied.

The Selective Drop (SD) policy [169] uses per-VC accounting to keep track of current buffer utilization of each UBR VC. A fair allocation is calculated for each VC and if the VC's buffer occupancy exceeds its fair allocation, its subsequent incoming packet is dropped. The scheme maintains a threshold R, as a fraction of the buffer capacity K. When the total buffer occupancy (X) exceeds $R \times K$, new packets are dropped depending on the VC_i's buffer occupancy (Y_i). In SD, a VC's entire packet is dropped if

$$(X > R \times K) \text{ AND } (Y_i \times N_a/X > Z)$$

where N_a is the number of active VCs (VCs with at least one cell in the buffer) and Z is another threshold parameter ($0 < Z <= 1$) used to scale the effective drop threshold. In terrestrial networks, SD has been shown to improve the fairness TCP connections running over UBR.

UBR with Guaranteed Rate Allocation

A multiserver satellite network will transport higher priority variable bit rate traffic along with UBR traffic. The effect of higher priority traffic on TCP over UBR has not been studied before.

3.4.3.1 Simulation Model

Figure 3.5 shows the basic network simulation configuration. In the figure, the switches represent the earth stations that connect to the satellite constellation. The entire satellite network is assumed to be of 155 Mbps ATM link without any onboard processing or queuing. All processing and queuing are performed at the earth stations. All sources are identical, infinite and unidirectional TCP sources. Two different configurations are simulated that represent multiple LEO hops and a single GEO hop. The link delays between the switches and the end systems are 5 ms in all configurations. The inter-switch (earth station to earth station) propagation delays are 100 ms, and 275 ms for LEO and GEO configurations respectively. The maximum value of the TCP receiver window is 2,500,000 bytes and 8,704,000 bytes for LEO and GEO. These window sizes are sufficient to fill the155.52 Mbps links. The TCP maximum segment size is 9180 bytes. The duration of simulation is 100 seconds. All link bandwidths are 155.52 Mbps, and peak cell rate at the ATM layer is 149.7 Mbps after the SONET overhead. The buffer sizes (in cells) used in the switch are:

LEO: 780, 1500, 3125, 6250, 12.5K, 50K (=1 RTT), and 100K.

GEO: 3375, 6750, 12500, 25K, 50K, 100K, 200K (=1 RTT), and 400K.

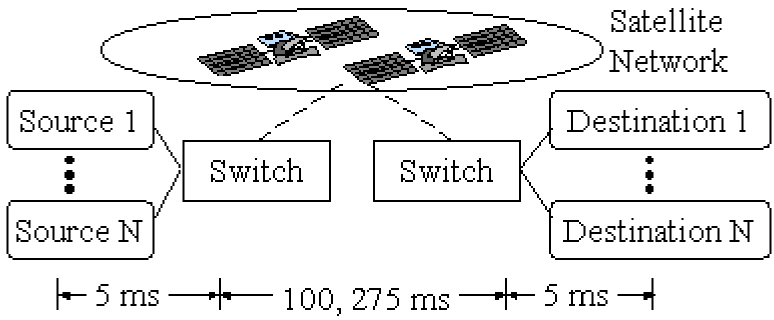


Figure 3.5: The N source TCP configuration

- N identical infinite TCP sources, SACK TCP
- Link Capacity = PCR = 155.52 Mbps
- Per-VC buffer management in switches (sel. drop)
- Simulation time = 100 s

3.4.3.2 Performance Metrics

When ATM networks carry TCP data, the end-to-end performance is measured at the TCP layer in the form of TCP throughput. To measure network performance, the throughputs of all TCPs passing through the bottleneck link are added and expressed as a fraction of the total capacity of the bottleneck link. This is called the *efficiency* of the network.

Let N TCP source-destination pairs send data over a network with bottleneck link capacity R bits/s. Let x_i be the observed throughput of the ith TCP source ($0 < i < N$). Let C be the maximum TCP throughput achievable on the link.

Definition 1 (Efficiency, E)

The efficiency of the network is the ratio of the sum of the actual TCP throughputs to the maximum possible throughput achievable at the TCP layer:

$$E(x_1, \ldots, x_N, C) = \sum_{i=1}^{i=N} x_i$$

The TCP throughputs x_i's are measured at the destination TCP layers. Throughput is defined as the total number of bytes delivered to the destination application (excluding retransmission and losses) divided by the total connection time.

The maximum possible TCP throughput C is the throughput attainable by the TCP layer running over an ATM network with link capacity R. For example, consider TCP over UBR on a 155.52 Mbps link with a 9180 byte TCP MSS. For 9180 bytes of data, the ATM layer receives 9180 bytes of data with 20 bytes of TCP header, 20 bytes of IP header, 8 bytes of Logical Link Control (LLC) header and 8 bytes of AAL5 trailer. These are padded to produce 193 ATM cells. Thus, each TCP segment results in 10229 bytes at the ATM layer. From this, the maximum possible throughput ≈ 9180/10299 ≈ 89.7 per cent ≈ 135 Mbps. It should be noted that ATM layer throughput does not necessarily correspond to TCP level throughput, because some bandwidth may be wasted during TCP retransmissions.

In addition to providing high overall throughput, the network must also allocate throughput fairly among competing connections. The definitions of fairness is determined by the particular service guarantees. For example, although UBR makes no service guarantees, fairness for TCP over UBR can be defined as the ability for UBR to provide equal throughput to all greedy TCP connections. Fairness for TCP over a best effort service is measured using the fairness index F.

Definition 2 (Fairness Index, F)

The fairness index is a function of the variability of the throughput across the TCP connections defined as

$$F((x_1, e_1), \ldots, (x_n, e_n)) = \frac{\left(\sum_{i=1}^{i=N} x_i/e_i\right)^2}{N \times \sum_{i=1}^{i=N} (x_i/e_i)^2}$$

where x_i is the observed throughput of the ith TCP connection ($0 < i < N$), and e_i the expected throughput or fair share for the ith TCP connection.

For a symmetrical configuration using TCP over UBR, e_i can be defined as an equal share of the bottleneck link capacity ($e_i = C/N$). Thus, the fairness index metric applies well to N-source symmetrical configurations. In this case, it is to be noted that when $x_1 = x_2 = \ldots = x_n$ then fairness index = 1. Also, low values of fairness index represent poor fairness among the connections. The desired values of the fairness index must be close to 1. A fairness index of 0.99 is considered to be near perfect. A fairness index of 0.9 may not be acceptable depending on the application and the number of sources involved. Also it should be noted that the fairness index may not be a good metric for a small number of connections. Details on the fairness metric can be found in [238]. In general, for a more complex configuration, the value of e_i can be derived from a rigorous formulation of a fairness definition that provides max-min fairness to the connections.

3.4.3.3 Simulation Parameters

The effects of the following parameters is studied:

Latency. The primary aim is to study the performance of large latency connections. The typical one-way latency from earth station to earth station for a single LEO 9700 km altitude, 60 degrees elevation angle) hop is about 5 ms. The one-way latencies for multiple LEO hops can easily be up to 50 ms from earth station to earth station. GEO one-way latencies are typically 275 ms from earth station to earth station. These three latencies (5 ms, 50 ms, and 275 ms) with various number of sources and buffer sizes are studied.

Number of sources. To ensure that the recommendations are scalable and general with respect to the number of connections, configurations are used with 5, 15 and 50 TCP connections on a single bottleneck link. For single hop LEO configurations, 15, 50 and 100 sources are used.

Buffer size. This is the most important parameter of this study. The set of values chosen are 2^{-k} × Round Trip Time (RTT), k = -1..6, (i.e., 2, 1, 0.5, 0.25, 0.125, 0.0625, 0.031, 0.016 multiples of the round trip delay-bandwidth product of the TCP connections.) The buffer size is plotted against the achieved TCP throughput for different delay-bandwidth products and number of sources.

Switch drop policy. Selective Drop Policy is used.

End system policies. An enhanced version of TCP called SACK TCP has been used for this study. SACK TCP improves performance by using selective acknowledgments for retransmission [103].

3.4.3.4 Simulation Results

Figures 3.6 and 3.7 show the resulting TCP efficiencies of the two different latencies. Each point in the figure shows the efficiency (total achieved TCP throughput divided by maximum possible throughput) against the buffer size used. Each figure plots a different latency and each set of points (connected by a line) in a figure represents a particular value of *N* (number of sources).

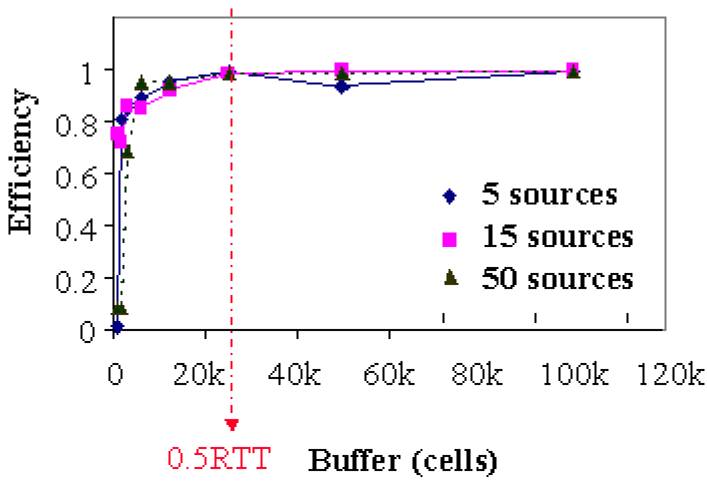


Figure 3.6: Multiple hop LEO

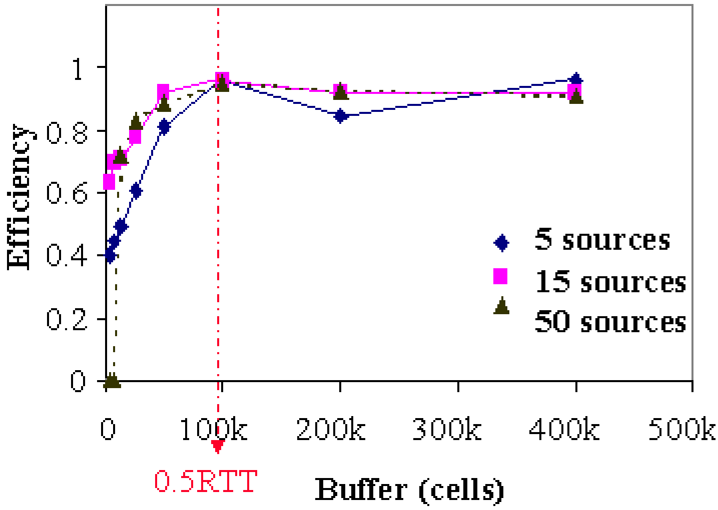


Figure 3.7: Single hop GEO

For very small buffer sizes, (0.031 × RTT, 0.0625 × RTT), the resulting TCP throughput is very low. In fact, for a large number of sources ($N = 50$), the throughput is sometimes close to zero. For moderate buffer sizes (less than 1 round trip delay-bandwidth), TCP throughput increases with increasing buffer sizes. TCP throughput asymptotically approaches the maximal value with further increase in buffer sizes. TCP performance over UBR for sufficiently large buffer sizes is scalable with respect to the number of TCP sources. The throughput is never 100%, but for buffers greater than 0.5 × RTT, the average TCP throughput is over 98% irrespective of the number of sources. Fairness is high for a large number of sources. This shows that TCP sources with a good per-VC buffer allocation policy like selective drop, can effectively share the link bandwidth. [239]

Example: Consider a GEO satellite ATM network in the real world, with 64 downlink beams, 8 gateway beams and 2 InterSatellite Links supporting ATM classes of service. Assume downlink data rate of 100Mbps per downlink beam, data rates of 155 Mbps per gateway beam, and 340 Mbps per InterSatellite Link. Normally CBR, rt-VBR, and nrt-VBR may not require buffering if peak rate is reserved. Whereas, the buffering required can be calculated for UBR or ABR service to be approximately about 6500 KCells. The buffer size required is way within the optimal size as shown in figure 3.7.

3.5 TCP Performance Analysis for Satellite ATM

In this section simulation results on the performance of TCP enhancements over ATM-UBR+ for satellite latencies with limited switch buffer for WWW traffic are presented. Analysis results of the impact of TCP flavors such as Fast Retransmit and Recovery, New

Reno, SACK and drop policies such as Early Packet Discard (EPD) and Selective Drop (SD) on LEO, MEO, GEO satellite configurations for several buffer sizes are discussed. [240]

3.5.1 TCP Enhancements

TCP uses a window-based flow control and uses it also to limit the number of segments in the network. “Vanilla TCP” consists of the slow start and congestion avoidance phases for congestion control. It detects segment losses by the retransmission timeout. Coarse granularity of timeouts are the primary cause of low TCP throughput over the UBR service. TCP Reno implements the fast retransmit and recovery algorithms that enable the connection to quickly recover from isolated segment losses [109]. TCP Reno can efficiently recover from isolated segment losses, but not from bursty losses. As a result, TCP Reno results in poor efficiency for long latency configurations especially for low buffer sizes and persistent traffic.

3.5.1.1 TCP New Reno: A Modification to Fast Retransmit and Recovery

As indicated above, TCP Reno cannot recover effectively from multiple packet drops. A modification to Reno, popularly known as NewReno was proposed to overcome this shortcoming. [101] introduced a “fast-retransmit phase”, in which the sender remembers the highest sequence number sent (RECOVER) when the fast retransmit is first triggered. After the first unacknowledged packet is retransmitted (when three duplicate ACKs are received), the sender follows the usual fast recovery algorithm and increases the CWND by one segment for each duplicate ACK it receives. When the sender receives an acknowledgment for the retransmit packet, it checks if the ACK acknowledges all segments including RECOVER. If so, the sender exits the fast retransmit phase, sets its CWND to SSTHRESH and starts a linear increase (congestion avoidance phase). On the other hand, if the ACK is a partial ACK, i.e., it acknowledges the retransmitted segment and only some of the segments before RECOVER, then the sender immediately retransmits the next expected segment as indicated by the ACK. A partial ACK also resets the CWND to SSTHRESH. This continues until all segments including RECOVER are acknowledged. The NewReno retransmits one segment every round trip time until an ACK is received for RECOVER. This mechanism ensures that the sender will recover from N segment losses in N round trips.

NewReno algorithm has been proposed in [101]. This description recommends a modification in which on receiving a partial ACK the congestion window is reduced by amount of new data acknowledged and then incremented by 1 MSS. Another modification is suggested to avoid multiple fast retransmits.

Another issue raised in [101] is whether the retransmit timer should be reset after each partial ACK or only after the first partial ACK. For satellite links, where retransmission timeout value is not much larger than the round trip time (RTT), the first option is better. If the retransmit timer is reset only after the first partial ACK, a retransmission timeout will be caused even for a small number of packets lost in a window. For satellite links with their long delays, a timeout is very costly. However, for WAN links, the retransmission timeout value is much larger than the RTT. For WAN links, if there are a number of packets lost in a window, it is better to timeout and retransmit all the packets using slow-start than to retransmit just 1 packet every RTT. In such a case, the second option is better.

In this implementation, the initial SSTHRESH value is set to RTT-bandwidth product and one new packet beyond RECOVER is sent upon receiving 2 duplicate ACKs while in the fast-retransmit phase (to keep the "flywheel" going). Since the TCP delay ACK timer is NOT set, all segments are ACKed as soon as they are received.

3.5.1.2 SACK TCP: Selective Acknowledgments

TCP with Selective Acknowledgments (SACK TCP) has been proposed to efficiently recover from multiple segment loss [103]. In SACK TCP, acknowledgments contain additional information about the segments that have been received by the destination. When the destination receives out-of-order segments, it sends duplicate ACKs (SACKs) acknowledging the out-of-order segments it has received. From these SACKs, the sending TCP can reconstruct information about the segments not received at the destination. On receiving three duplicate ACKs, the sender retransmits the first lost segment and enters "fast-retransmit" phase as in NewReno. The CWND is set to half its current value. SSTHRESH is set to the new value of CWND and the highest sequence number sent so far is recorded in RECOVER. As in NewReno, the sender does not come out of the fast-retransmit phase until it has received the ACK for RECOVER. However, in the fast-retransmit phase if allowed by the window, the sending TCP uses the SACK information to retransmit lost segments before sending any new data. A sender implementing NewReno can retransmit only one lost segment every RTT. Thus it recovers from N segment losses in N RTTs. However, a sender implementing SACK can recover from N segment losses much faster.

In the implementation, SACK is based on the description in [102, 103]. The SACK is sent whenever out-of-sequence data is received. All duplicate ACKs contain the SACK option. The receiver keeps track of all the out-of-sequence data blocks received. When the receiver generates a SACK, the first SACK block specifies the block of data formed by the most recently received data segment. This ensures that the receiver provides the most up to date information to the sender. After the first SACK block, the remaining blocks can be filled in any order.

The sender keeps a table of all the segments sent but not ACKed. When a segment is sent, it is entered into the table. When the sender receives an ACK with the SACK option, it marks all the segments specified in the sack option blocks as SACKed. The entries for each segment remain in the table until the segment is ACKed. When the sender receives three duplicate ACKs, it retransmits the first unacknowledged packet and enters fast-retransmit phase. During the fast retransmit phase, when the sender is allowed to send, it first tries to retransmit the holes in the SACK blocks before sending any new segments. When the sender retransmits a segment, it marks the segment as retransmitted in the table. If a retransmitted segment is lost, the sender times out and performs slow start. When a timeout occurs, the sender resets the SACK table.

During the fast retransmit phase, the sender maintains a variable called “PIPE” that indicates the number of bytes currently in the network. When the third duplicate ACK is received, PIPE is set to the value of CWND – 3 segments and CWND is reduced by half. For every subsequent duplicate ACK received, PIPE is decremented by one segment because the ACK denotes a packet leaving the network. The sender sends data (new or retransmitted) only when PIPE is less than CWND value. This implementation is equivalent to inflating the CWND by one segment for every duplicate ACK and sending segments if the number of unacknowledged bytes is less than the congestion window value.

When a segment is sent, PIPE is incremented by one segment. When a partial ACK is received, PIEP is decremented by two. The first decrement is because the partial ACK represents a retransmitted segment leaving the pipe. The second decrement is done because the original segment that was lost, and had not been accounted for, is now actually considered to be lost.

3.5.2 UBR+ Drop Policies

The basic UBR service can be enhanced by implementing intelligent drop policies at the switches. In this study, Early Packet Discard (EPD) and Selective Drop (SD) as described in Section 3.5 are used. [241, 242]

3.5.3 WWW Traffic Model

The WWW uses Hypertext Transfer Protocol (HTTP). HTTP uses TCP/IP for communication between WWW client and WWW servers [243]. Modeling of the WWW traffic is difficult because of the changing nature of web traffic. In this section, the model used here and the inherent assumptions are outlined.

3.5.3.1 Implications of the HTTP/1.1 standard

The main difference between version 1.1 of the Hypertext Transfer Protocol, HTTP/1.1 [244], and earlier versions is the use of persistent TCP connections as the default behavior for all HTTP connections. In other words, a new TCP connection is not set up for each HTTP/1.1 request. The HTTP client and the HTTP server assume that the TCP connection is persistent until a *Close* request is sent in the HTTP Connection header field. Another important difference between HTTP/1.1 and earlier versions is that the HTTP client can make multiple requests without waiting for responses from the server (called *pipelining*). The earlier models were *closed-loop* in the sense that each request needed a response before the next request could be sent.

3.5.3.2 WWW Server Model

The WWW servers are modeled as infinitely fast servers getting file requests from WWW clients. The model is an extension of that specified in SPECweb96 benchmark [245]. In this model, a WWW server, on receiving a request from a WWW client sends some data back. The amount of data to be sent (the requested file size) is determined by classifying the client request into one of five classes (Class 0 through Class 4), shown in Table 3.3. As shown in the table, 20% of the requests are classified as Class 0 requests, i.e., less than 1 KB of data is sent in response. Similarly 28% of the file requests are classified as Class 1 requests and so on. The file size range of each class and the percentage of client requests falling in that class are also shown.

There are nine discrete sizes in each class (e.g. Class 1 has 1 KB, 2 KB, up to 9 KB and Class 2 has 10 KB, 20 KB, through 90 KB and so on.). Within a class, one of these nine file sizes is selected according to a Poisson distribution with mean 5. The model of discrete sizes in each class is based on the SPECweb96 benchmark [245]. There are three key differences from the SPEC model. First, an infinite server is assumed, i.e. no processing time taken by server for a client request. Secondly, a new class of file sizes (1 MB - 10 MB), which allows to model file sizes larger than those in the SPEC benchmark is created. Finally, the percentage distribution of client requests had to be changed into server file size classes to accommodate the new class.

The reason for a new class of file sizes is to model the downloading of large software and offline browsing of search results. The percentages of requests falling into each of file size classes have been changed so that average requested file size is around 120 KB, as opposed to 15 KB in SPECweb96 model. The new figure better represents the current WWW traffic scenario. The reason for having 20% of the requests classified as Class 0 requests is explained in next sub-section.

Table 3.3: WWW Server File Size Classes

Class	File Size Range	Frequency of Access
Class 0	0 - 1 KB	20 %
Class 1	1 KB - 10 KB	28 %
Class 2	10 KB – 100 KB	40 %
Class 3	100 KB - 1 MB	11.2 %
Class 4	1 MB - 10 MB	0.8 %

3.5.3.3 WWW Client Model

The HTTP-model in [245] describes an empirical model of WWW clients based on observations in a LAN environment. Specifically, a typical client is observed to make, on the average, four HTTP GET requests for a single document. Multiple requests are needed to fetch inline images, if any. With the introduction of JAVA scripts in web pages, additional accesses maybe required to fetch the scripts. Therefore, five is used as the average number of HTTP GET requests. In the model, a WWW client makes 1 to 9 requests for a single document, Poisson distributed around a mean of 5. These requests are separated by a random time interval between 100ms to 500 ms. Caching effects at the clients are ignored.

Typically, the first request from an HTTP client accesses the index page (plain text), which is of size 1 KB or less. Since every fifth request is expected to be an index page access, WWW server classifies 20% (= 1/5) of the client requests as Class 0 requests and sends 1 KB or less data in response.

A time lag between batches of requests (presumably for the same document) that corresponds to the time taken by the user to request a new document, as a constant, 10 seconds is also modeled. While this may be too short a time for a human user to make decisions, it also weights the possibility of offline browsing where the inter-batch time is much shorter.

The user behavior is not modeled across different servers. The main purpose of using this simplistic model is to approximate the small loads offered by individual web connections, and to study the effects of aggregation of such small loads on the network.

3.5.4 Simulation Configuration And Experiments

The configuration shown in Figure 3.8 consists of 100 WWW clients being served by 100 WWW servers, one server for each client. Both WWW clients and servers use underlying TCP connections for data transfer. The switches implement the UBR+ service with optional drop policies described before. The following subsections describe various configuration, TCP and switch parameters used in the simulations. A client makes 1 to 9 requests (Poisson distributed around a mean of 5) every 10 seconds. Server classifies each client request in one of 5 classes. *Frequency of Access* of a class, indicates how often a client request is classified as belonging to that class. *File Size Range* associated with each class consists of 9 equally spaced discreet sizes. A file size among 9 in the range is chosen according to a Poisson distribution with mean 5. Finally, server sends the file as and when allowed by underlying TCP connection.

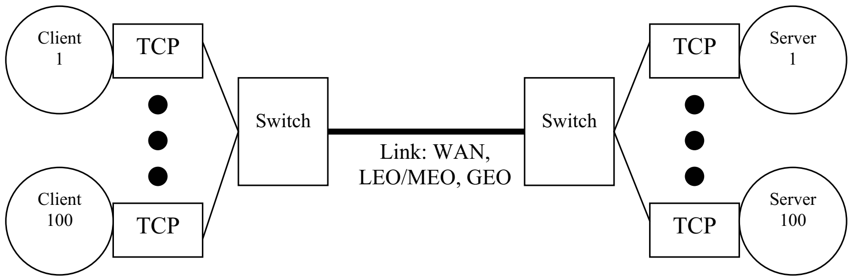


Figure 3.8: Simulation Configuration

3.5.4.1 Configuration Parameters

The configuration consists of 100 WWW client-server connections using TCP for data transfer. Links connecting server/client TCPs to switches have a bandwidth of 155.52 Mbps (149.76 Mbps after SONET overhead), and a one way delay of 5 microseconds. The link connecting the two switches simulates the desired (WAN/LEO/MEO/GEO) link respectively and has a bandwidth of 45Mbps (T3). The corresponding one-way link delays are 5ms (WAN), 100ms (multi hop LEO/single hop MEO) and 275ms (GEO) respectively. Since the propagation delay on the links connecting client/server TCPs to switches is negligible compared to the delay on the inter-switch link, the round trip times (RTTs) due to propagation delay are 10ms, 200ms and 550ms respectively. All simulations run for 100 seconds. Since every client makes a new set of requests every 10 secs, the simulations run for 10 cycles of client requests.

3.5.4.2 TCP Parameters

Underlying TCP connections send data as specified by the client/server applications. A WWW client asks its TCP to send a 128 byte packet as a request to the WWW server TCP. TCP maximum segment size (MSS) is set to 1024 bytes for WAN simulations and 9180 bytes otherwise. TCP timer granularity is set to 100 ms. TCP maximum receiver window size is chosen so that it is always greater than RTT-bandwidth product of the path. Such a value of receiver window ensures that receiver window does not prevent sending TCPs from filling up the network pipe. For WAN links (10 ms RTT due to propagation delay), the default receiver window size of 64K is sufficient. For MEO links (200 ms RTT), RTT-bandwidth product in bytes is 1,125,000 bytes. By using the TCP window scaling option and having a window scale factor of 5, a maximum window size of 2,097,120 bytes is achieved. Similarly, for GEO links (550 ms RTT), the RTT-bandwidth product in bytes is 3,093,750 bytes. A window scale factor of 6 is used to achieve maximum window size of 4,194,240 bytes. TCP "Silly Window Syndrome Avoidance" is disabled because in WWW traffic many small segments (due to small request sizes, small file sizes or last segment of a file) have to be sent immediately. Instead of having a fixed initial SSTHRESH of 64 KB, the RTT-bandwidth product of the path should be used as initial SSTHRESH. In these simulations, the round trip propagation delay - bandwidth product has been used as the initial SSTHRESH value. Hence the initial SSTHRESH values for WAN, MEO and GEO links are 56,250, 1,125,000 and 3,093,750 bytes respectively. The TCP delay ACK timer is NOT set. Segments are ACKed as soon as they are received.

3.5.4.3 Switch Parameters

The drop threshold *R* is 0.8 for both drop policies – Early Packet Discard (EPD) and Selective Drop (SD). For SD simulations, threshold *Z* also has a value 0.8. Three different values of buffer sizes are used in this experiments. These buffer sizes approximately correspond to 0.5 RTT, 1 RTT and 2 RTT - bandwidth products of the end-to-end TCP connections for each of the three propagation delays respectively. For LEO delays, the largest buffer size is 2300 cells. This is a little more than the 2 RTT - bandwidth product. The reason for selecting 2300 is that this is the smallest buffer size that can hold one complete packet (MSS=1024 bytes) for each of the 100 TCP connections. For LEO, 0.5 RTT and 1 RTT buffers are not sufficient to hold a single packet from all TCPs. This problem will also occur in MEO and GEO TCPs if the number of flows is increased. Some preliminary analysis has shown that the buffer size required for good performance may be related to the number of active TCP connections as well as the RTT-bandwidth product. Further research needs to be performed to provide conclusive results of this effect. Table 3.4 shows the switch buffer sizes used in the simulations.

Table 3.4: Switch Buffer Sizes Used for Simulations

Link Type (RTT)	RTT-bandwidth product (cells)	Switch Buffer Sizes (cells)
LEO (10ms)	1062	531, 1062, 2300
MEO (200 ms)	21230	10615, 21230, 42460
Single-Hop GEO (550 ms)	58380	29190, 58380, 116760

3.5.5 Performance Metrics

The performance of TCP is measured by the efficiency and fairness index which are defined as follows. Let x_i be the throughput of the ith WWW server $(1 \le i \le 100)$. Let C be the maximum TCP throughput achievable on the link. Let E be the efficiency of the network, E is defined as in Section 3.4.3.2 as

$$E = \frac{\sum_{i=1}^{i=N} x_i}{C}$$

where $N = 100$ and $\sum x_i$ is sum of all 100 server throughputs.

The server TCP throughput values are measured at the client TCP layers. Throughput is defined as the highest sequence number in bytes received at the client from the server divided by the total simulation time. The results are reported in Mbps.

Due to overheads imposed by TCP, IP, LLC and AAL5 layers, the maximum possible TCP over UBR throughput over a 45Mbps link is much less and depends on the TCP maximum segment size (MSS). For MSS = 1024 bytes (on WAN links), the ATM layer receives 1024 bytes of data + 20 bytes of TCP header + 20 bytes of IP header + 8 bytes of LLC header + 8 bytes of AAL5 trailer. These are padded to produce 23 ATM cells. Thus each TCP segment of 1024 bytes results in 1219 bytes at the ATM layer. Thus, the maximum possible TCP throughput C is 1024/1219 = 84% = 37.80Mbps approximately on a 45Mbps link. Similarly, for MSS = 9180 bytes (on MEO,GEO links), C is 40.39Mbps approximately. Since, the "Silly Window Syndrome Avoidance" is disabled (because of WWW traffic), some of the packets have less than 1 MSS of data. This decreases the value of C a little. However, the resulting decrease in the value of C has an insignificant effect on the overall efficiency metric.

In all simulations, the 45Mbps(T3) link between the two switches is the bottleneck. The maximum possible throughput C on this link is 37.80 Mbps (for WAN) and 40.39 Mbps (for MEO/GEO). The average total load generated by 100 WWW servers is 48 Mbps[1].

1. A WWW server gets on average 5 client requests every 10s and sends on average 120 KB of data for each request. This means that on average a WWW server schedules 60KBps i.e. 480Kbps of data. Hence average total load generated by 100 WWW servers is 48Mbps.

Fairness is measured by calculating the Fairness Index F defined as in Section 3.4.3.2 by:

$$F = \frac{\left(\sum_{i=1}^{i=N} x_i/e_i\right)^2}{N \times \sum_{i=1}^{i=N} (x_i/e_i)^2}$$

where $N = 100$ and e_i is the expected throughput for connection i. In the simulations, e_i is the max-min fair share that should be allocated to server i. On a link with maximum possible throughput C, the fair share of each of the 100 servers is $C/100$. Let S_i be the maximum possible throughput that a server can achieve, calculated as the total data scheduled by the server for the client divided by simulation time.

For all i for which $S_i < C/100$, $e_i = S_i$, i.e., servers that schedule less than their fair share are allocated their scheduled rates. This determines the first iteration of the max-min fairness calculation. These e_i's are subtracted from C, and the remaining capacity is again divided in a max-min manner among the remaining connections. This process is continued until all remaining servers schedule more than the fair share in that iteration, in this case e_i = the fairshare.

3.5.6 Simulation Analysis

In this section, a statistical analysis of simulation results for LEO, MEO and GEO links and draw conclusions about optimal choices for TCP flavor, switch buffer sizes and drop policy for these links is presented. The analysis technique used here is described in detail in [238]. A brief description of these techniques is given in this section. The following subsections present simulation results for LEO, MEO and GEO links respectively. [233]

3.5.6.1 Analysis Technique

The purpose of analyzing results of a number of experiments is to calculate the individual effects of contributing factors and their interaction. These effects can also help us in drawing meaningful conclusions about the optimum values for different factors. In the present case, the effects of the TCP flavor, buffer size and drop policy in determining the efficiency and fairness for LEO, MEO and GEO links have to be analyzed. Thus, TCP flavor, switch buffer size and drop policy are the 3 factors. The values a factor can take are called ‘levels’ of the factor. For example, EPD and SD are two levels of the factor 'Drop Policy'. Table 3.5 lists the factors and their levels used in simulations.

Table 3.5: Factors and Levels in Simulations

Factor	Level			
TCP flavor	Vanilla	Reno	NewReno	SACK
Switch drop policy	EPD		SD	
Switch buffer size	0.5 RTT	1 RTT	2 RTT	

The analysis is done separately for efficiency and fairness, and consists of the calculating the following terms:

- *Overall Mean*: This consists of the calculation of the overall mean '*Y*' of the result (efficiency or fairness).
- Total Variation: This represents the variation in the result values (efficiency or fairness) around the overall mean '*Y*'. The goal of the analysis to calculate, how much of this variation can be explained by each factor and the interactions between factors.
- *Main Effects*: These are the individual contributions of a level of a factor to the overall result. A particular main effect is associated with a level of a factor, and indicates how much variation around the overall mean is caused by the level. The main effects of 4 TCP flavors, 3 buffer sizes, and 2 drop policies are calculated.
- *First Order Interactions*: These are the interaction between levels of two factors. In the experiments, there are first order interactions between each TCP flavor and buffer size, between each drop policy and TCP flavor, and between each buffer size and drop policy.
- *Allocation of Variation*: This is used to explain how much each factor contributes to the total variation.
- *Overall Standard Error*: This represents the experimental error associated with each result value. The overall standard error is also used in the calculation of the confidence intervals for each main effect.
- *Confidence Intervals for Main Effects*: The 90% confidence intervals for each main effect are calculated. If a confidence interval contains 0, then the corresponding level of the factor has no significant effect (with 90% confidence) towards the result value (efficiency or fairness). If confidence intervals of two levels overlap, then the effects of both levels are assumed to be similar.

The first step of the analysis is the calculation of the overall mean '*Y*' of all the values. The next step is the calculation of the individual contributions of each level '*a*' of factor '*A*', called the 'Main Effect'. The 'Main Effect' of '*a*' is calculated by subtracting the overall mean '*Y*' from the mean of all results with '*a*' as the value for factor '*A*'. The 'Main Effects' are calculated in this way for all the levels of each factor.

The interactions between levels of two factors is then calculated. The interaction between levels of two factors is called 'First-order interaction'. For calculating the interaction between level '*a*' of factor '*A*' and level '*b*' of factor '*B*', an estimate is calculated for all results with '*a*' and '*b*' as values for factors '*A*' and '*B*'. This estimate is the sum of the overall mean '*Y*' and the 'Main Effects' of levels '*a*' and '*b*'. This estimate is subtracted from the mean of all results with '*a*' and '*b*' as values for factors '*A*' and '*B*' to get the 'Interaction' between levels 'a' and 'b'. In the present analysis, only up to 'First-order interactions' are calculated. Generally, to get an accurate model of simulation results, 'Main Effects' and 'First-order interactions' are sufficient.

The calculation of the 'Total Variation' and 'Allocation of Variation' are then performed. First, the value of the square of the overall mean 'Y' is multiplied by the total number of results. This value is subtracted from the sum of squares of individual results to get the 'Total Variation' among the results. The next step is the 'Allocation of Total Variation' to individual 'Main Effects' and 'First-order interactions'. To calculate the variation caused by a factor 'A', the sum of squares of the main effects of all levels of 'A' is taken and multiplied with the number of experiments conducted with each level of 'A'. For example, to calculate the variation caused by TCP flavor, the sum of squares of the main effects of all its levels (Vanilla, Reno, NewReno and SACK) is taken and multiplied by 6 (with each TCP flavor 6 different simulations involving 3 buffer sizes and 2 drop policies are conducted). In this way, the variation caused by all factors is calculated. To calculate the variation caused by first-order interaction between two factors 'A' and 'B', the sum of squares of all the first-order interactions between levels of 'A' and 'B' is taken and multiplied with the number of experiments conducted with each combination of levels of 'A' and 'B'.

The next step of the analysis is to calculate the overall standard error for the results. This value requires calculation of individual errors in results and the degrees of freedom for the errors. For each result value, an estimate is calculated by summing up the overall mean 'Y', main effects of the parameter levels for the result and their interactions. This estimate is subtracted from the actual result to get the error 'e_i' for the result.

If a factor 'A' has 'N_A' levels, then the total number of degrees of freedom is $\Pi(N_A)$. Thus, for the analysis, the total number of degrees of freedom is $4 \times 2 \times 3 = 24$. The degrees of freedom associated with the overall mean 'Y' is 1. The degrees of freedom associated with 'main effects' of a factor 'A' are '$N_A - 1$'. Thus, degrees of freedom associated with all 'main effects' are $(N_A - 1)$. Similarly, the degrees of freedom associated with first-order interaction between 2 factors 'A' and 'B' are $(N_A - 1) \times (N_B - 1)$. Thus, degrees of freedom associated with all first-order interactions are $(N_A - 1) \times (N_B - 1)$, with the summation extending over all factors. In the analysis, the degrees of freedom associated with all 'main effects' are $3 + 1 + 2 = 6$ and the degrees of freedom associated with all first-order interactions are $(3 \times 1) + (3 \times 2) + (1 \times 2) = 11$.

Since the overall mean 'Y', the main effects of individual levels and their first-order interactions to calculate the estimate are used, the value of the degrees of freedom for errors 'd_e' is calculated as follows:

$$d_e = \Pi(N_A) - 1 - \Sigma(N_A - 1) - \Sigma(N_A - 1) \times \Sigma(N_B - 1)$$

In the present case, $d_e = 24 - 1 - 6 - 11 = 6$.

To calculate the overall standard error 's_e', the sum of squares of all individual errors 'e_i' is divided by the number of degrees of freedom for errors 'd_e' (6 in the present case). The square root of the resulting value is the overall standard error.

$$S_e = \sqrt{(\Sigma e_i^2)/d_e}$$

Finally, based on the overall standard error, the 90% confidence intervals for all 'main effects' of each factor are calculated. For this purpose, the standard deviation 's_A' associated with each factor 'A' is calculated as follows:

$$S_A = S_e \times \sqrt{(N_A - 1)/\Pi N_A}$$

Here, 'N_A' is the number of levels for factor 'A' and $\Pi(N_A)$ is the total number of degrees of freedom.

The variation around the 'main effect' of all levels of a factor 'A' to get a 90% confidence level is given by the standard deviation 's_A' multiplied by $t[0.95, d_e]$, where $t[0.95, d_e]$ values are quantiles of the t distribution. Hence, if 'ME_a' is the value of the main effect of level 'a' of factor 'A', then the 90% confidence interval for 'ME_a' is $\{ME_a \pm s_A \times t[0.95, d_e]\}$. The main effect value is significant only if the confidence interval does not include 0.

3.5.7 Simulation Results for LEO links

Table 3.6 presents the individual efficiency and fairness results for LEO links. Table 3.7 shows the calculation of 'Total Variation' in LEO results and 'Allocation of Variation' to main effects and first-order interactions. Table 3.8 shows the 90% confidence intervals for the main effects. A negative value of main effect implies that the corresponding level of the factor decreases the overall efficiency and vice versa. If a confidence interval encloses 0, the corresponding level of the factor is assumed to be not significant in determining performance.

Table 3.6: Simulation Results for LEO links

Drop Policy	TCP Flavor	Buffer = 0.5 RTT		Buffer = 1 RTT		Buffer = 2 RTT	
		Efficiency	Fairness	Efficiency	Fairness	Efficiency	Fairness
EPD	Vanilla	0.4245	0.5993	0.5741	0.9171	0.7234	0.9516
	Reno	0.6056	0.8031	0.7337	0.9373	0.8373	0.9666
	NewReno	0.8488	0.8928	0.8866	0.9323	0.8932	0.9720
	SACK	0.8144	0.7937	0.8948	0.8760	0.9080	0.8238
SD	Vanilla	0.4719	0.6996	0.6380	0.9296	0.8125	0.9688
	Reno	0.6474	0.8230	0.8043	0.9462	0.8674	0.9698
	NewReno	0.8101	0.9089	0.8645	0.9181	0.8808	0.9709
	SACK	0.7384	0.6536	0.8951	0.8508	0.9075	0.8989

Table 3.7: Allocation of Variation for LEO Efficiency and Fairness Values

Component	Sum of Squares		% of Variation	
	Efficiency	Fairness	Efficiency	Fairness
Individual Values	14.6897	18.6266		
Overall Mean	14.2331	18.3816		
Total Variation	0.4565	0.2450	100	100
Main Effects:				
TCP Flavor	0.2625	0.0526	57.50	21.49
Buffer Size	0.1381	0.1312	30.24	53.55
Drop Policy	0.0016	0.0002	0.34	0.09
First-order Interactions:				
TCP Flavor-Buffer Size	0.0411	0.0424	8.99	17.32
TCP Flavor-Drop Policy	0.0104	0.0041	2.27	1.68
Buffer Size-Drop Policy	0.0015	0.0009	0.33	0.38

Standard Error, s_e = 0.0156(For Efficiency), 0.0472(For Fairness)

Table 3.8: Main Effects and their Confidence Intervals for LEO

Factor	Main Effect		Confidence Interval	
	Efficiency	Fairness	Efficiency	Fairness
TCP Flavor:				
Vanilla	-0.1627	-0.0308	(-0.1734,-0.1520)	(-0.0632,0.0016)
Reno	-0.0208	0.0325	(-0.0315,-0.0101)	(0.0000, 0.0649)
NewReno	0.0939	0.0573	(0.0832,0.1046)	(0.0248, 0.0898)
SACK	0.0896	-0.0590	(0.0789,0.1003)	(-0.0914, -0.0265)
Buffer Size:				
0.5 RTT	-0.1000	-0.1034	(-0.1087,-0.0912)	(-0.1299,-0.0769)
1 RTT	0.0163	0.0382	(0.0076,0.0250)	(0.0117, 0.0647)
2 RTT cells	0.0837	0.0651	(0.0749,0.0924)	(0.0386, 0.0916)
Drop Policy:				
EPD	-0.0081	-0.0030	(-0.0142, -0.0019)	(-0.0217,0.0157)
SD	0.0081	0.0030	(0.0019,0.0142)	(-0.0157, 0.0217)

3.5.7.1 Analysis of Efficiency Values: Results and Observations

The following conclusions can be made from the above tables.

TCP type explains 57.5% of the variation and hence is the major factor in determining efficiency. It can be established from confidence intervals of effects of different TCP types that NewReno and SACK have better efficiency performance than Vanilla and

Reno. Since the confidence intervals of effects of SACK and NewReno overlap, it cannot be said that one performs better than the other. Confidence intervals for the effects of Vanilla and Reno suggest that Reno performs better than Vanilla.

Buffer size explains 30.24% of the variation and hence is the next major determinant of efficiency. Confidence intervals for effects of different buffer sizes clearly indicate that efficiency increases substantially as buffer size is increased. However, on looking at the individual efficiency values, it can be noticed that only Vanilla and Reno get substantial increase in efficiency as buffer size is increased from 1 RTT to 2 RTT.

The interaction between buffer size and TCP type explains 8.99% of the variation. The large interaction is because of the fact that only Vanilla and Reno show substantial gains in efficiency as the buffer size is increased from 1 RTT to 2 RTT. For SACK and NewReno, increasing buffer sizes from 1 RTT to 2 RTT does not bring much increase in efficiency. This indicates that SACK and NewReno can tolerate the level of packet loss caused by a buffer size of 1 RTT.

Though the variation explained by drop policy is negligible, it can be seen that for Vanilla and Reno, SD results in better efficiency than EPD for the same buffer size. This is because for EPD, after crossing the threshold R, all new packets are dropped and buffer occupancy does not increase much beyond R. However for SD, packets of VCs with low buffer occupancy are still accepted. This allows the buffer to be utilized more efficiently and fairly and to better efficiency as well as fairness.

However, for NewReno and SACK, the efficiency values are similar for EPD and SD for same buffer size. This is because NewReno and SACK are much more tolerant of packet loss than Vanilla and Reno. Thus the small decrease in number of packets dropped due to increased buffer utilization does not cause a significant increase in efficiency.

It can be noticed from individual efficiency values that SACK generally performs a little better than NewReno except when buffer size is very low (0.5 RTT). Better performance of NewReno for very low buffer size can be explained as follows. Low buffer size means that a large number of packets are dropped. When in fast retransmit phase, NewReno retransmits a packet for every partial ACK received. However, SACK does not retransmit any packet till *pipe* goes below *CWND* value. A large number of dropped packets mean that not many duplicate or partial ACKs are forthcoming. Hence *pipe* may not reduce sufficiently to allow SACK to retransmit all the lost packets quickly. Thus, SACK's performance may perform worse than NewReno under extreme congestion.

It can be concluded that SACK and NewReno give best performance in terms of efficiency for LEO links. For NewReno and SACK, a buffer size of 1 RTT seems to be sufficient for getting close to best efficiency with either EPD or SD as the switch drop policy. As discussed before, buffer requirements need to be verified for situations where number of flows is much larger.

3.5.7.2 Analysis of Fairness values: Results and Observations

Buffer size largely determines fairness as 53.55 % of the variation is explained by the buffer size. Confidence intervals for effects of buffer sizes suggest that fairness increases substantially as buffer size is increased from 0.5 RTT to 1 RTT. Since confidence intervals for buffers of 1 RTT and 2 RTTs overlap, it cannot be concluded that 2 RTT buffers result in better performance than 1 RTT buffers.

TCP type is the next major factor in determining fairness as it explains 21.49 % of the variation. Confidence intervals for effects of TCP type on fairness, clearly suggest that NewReno results in best fairness and SACK results in the worst fairness.

SD only increases fairness for low buffer sizes. Overall, both the allocation of variation to drop policy, and confidence intervals for effects of SD and EPD suggest that SD *does not* result in higher fairness when compared to EPD for bursty traffic in LEO links unless buffer sizes are small. This result is interesting and means that *per-flow accounting* to improve fairness will be successful only in presence of sufficiently large buffers.

3.5.8 Simulation Results for MEO links

Table 3.9 presents the individual efficiency and fairness results for MEO links. Table 3.10 shows the calculation of 'Total Variation' in MEO results and 'Allocation of Variation' to main effects and first-order interactions. Table 3.11 shows the 90% confidence intervals for main effects. [246]

Table 3.9: Simulation Results for MEO Links

Drop Policy	TCP Flavor	Buffer = 0.5 RTT		Buffer = 1 RTT		Buffer = 2 RTT	
		Efficiency	Fairness	Efficiency	Fairness	Efficiency	Fairness
EPD	Vanilla	0.8476	0.9656	0.8788	0.9646	0.8995	0.9594
	Reno	0.8937	0.9659	0.9032	0.9518	0.9091	0.9634
	NewReno	0.9028	0.9658	0.9105	0.9625	0.9122	0.9616
	SACK	0.9080	0.9517	0.9123	0.9429	0.9165	0.9487
SD	Vanilla	0.8358	0.9649	0.8719	0.9684	0.9009	0.9615
	Reno	0.8760	0.9688	0.8979	0.9686	0.9020	0.9580
	NewReno	0.8923	0.9665	0.8923	0.9504	0.8976	0.9560
	SACK	0.9167	0.9552	0.9258	0.9674	0.9373	0.9594

Table 3.10: Allocation of Variation for MEO Efficiency and Fairness Values

Component	Sum of Squares		%age of Variation	
	Efficiency	Fairness	Efficiency	Fairness
Individual Values	19.3453	22.1369		
Overall Mean	19.3334	22.1357		
Total Variation	0.0119	0.0012	100	100
Main Effects:				
TCP Flavor	0.0067	0.0003	56.75	29.20
Buffer Size	0.0026	0.0001	21.73	7.70
Drop Policy	0.0001	0.0001	0.80	6.02
First-order Interactions:				
TCP Flavor-Buffer Size	0.0016	0.0001	13.42	10.16
TCP Flavor-Drop Policy	0.0007	0.0003	6.11	22.60
Buffer Size-Drop Policy	0.0001	0.0001	0.53	6.03

Standard Error, s_e = 0.0036(For Efficiency), 0.0060(For Fairness)

Table 3.11: Main Effects and Their Confidence Intervals for MEO

Factor	Mean Effect		Confidence Interval	
	Efficiency	Fairness	Efficiency	Fairness
TCP Flavor:				
Vanilla	-0.0251	0.0037	(-0.0276,-0.0226)	(-0.0004,0.0078)
Reno	-0.0005	0.0024	(-0.0030,0.0019)	(-0.0017,0.0065)
NewReno	0.0038	0.0001	(0.0013,0.0062)	(-0.0040,0.0042)
SACK	0.0219	-0.0062	(0.0194,0.0244)	(-0.0103,-0.0020)
Buffer Size:				
0.5 RTT	-0.0134	0.0027	(-0.0154,-0.0114)	(-0.0007,0.0060)
1 RTT	0.0016	-0.0008	(-0.0005,0.0036)	(-0.0042,0.0026)
2 RTT	0.0119	-0.0019	(0.0098,0.0139)	(-0.0052,0.0015)
Drop Policy:				
EPD	0.0020	-0.0017	(0.0006,0.0034)	(-0.0041,0.0007)
SD	-0.0020	0.0017	(-0.0034,-0.0006)	(-0.0007,0.0041)

3.5.8.1 Analysis of Efficiency values: Results and Observations

TCP flavor explains 56.75% of the variation and hence is the major factor in deciding efficiency value. Non-overlapping confidence intervals for effects of TCP flavors clearly indicate that SACK results in best efficiency followed by NewReno, Reno and Vanilla. However, it should be noticed that difference in performance for different TCP flavors is not very large.

Buffer size explains 21.73% of the variation and hence is the next major determinant of efficiency. Confidence intervals for effects of different buffer sizes indicate that efficiency does increase but only slightly as buffer size is increased. However, Vanilla's efficiency increases by about 5% with increase in buffer size from 0.5 RTT to 2 RTT. The corresponding increase in efficiency for other TCP flavors is around 2% or less. This also explains the large interaction between buffer sizes and TCP flavors (explaining 13.42% of the total variation).

Drop policy does not cause any significant difference in efficiency values.

Thus the simulation results indicate that SACK gives best performance in terms of efficiency for MEO links. However, difference in performance for SACK and other TCP flavors is not substantial. For SACK, NewReno and Fast Retransmit and Recovery (FRR), the increase in efficiency with increasing buffer size is very small. For MEO links, 0.5 RTT seems to be the optimal buffer size for all non-Vanilla TCP flavors with either EPD or SD as drop policy[2].

3.5.8.2 Analysis of Fairness values: Results and Observations

As can be seen from individual fairness values, there is not much difference between fairness values for different TCP types, buffer sizes or drop policies. This claim is also supported by the fact that all 9 main effects have very small values, and for 8 of them, their confidence interval encloses 0. Thus, these simulations do not give us any information regarding fairness performance of different options.

3.5.9 Simulation Results for GEO links

Table 3.12 presents the individual efficiency and fairness results for GEO links. Table 3.13 shows the calculation of 'Total Variation' in GEO results and 'Allocation of Variation' to main effects and first-order interactions. Table 3.14 shows the 90% confidence intervals for main effects. [247]

2. Again this result needs to be verified in presence of much larger number of flows.

Table 3.12: Simulation Results for GEO Links

Drop Policy	TCP Flavor	Buffer = 0.5 RTT		Buffer = 1 RTT		Buffer = 2 RTT	
		Efficiency	Fairness	Efficiency	Fairness	Efficiency	Fairness
EPD	Vanilla	0.7908	0.9518	0.7924	0.9365	0.8478	0.9496
	Reno	0.8050	0.9581	0.8172	0.9495	0.8736	0.9305
	NewReno	0.8663	0.9613	0.8587	0.9566	0.8455	0.9598
	SACK	0.9021	0.9192	0.9086	0.9514	0.9210	0.9032
SD	Vanilla	0.8080	0.9593	0.8161	0.9542	0.8685	0.9484
	Reno	0.8104	0.9671	0.7806	0.9488	0.8626	0.9398
	NewReno	0.7902	0.9257	0.8325	0.9477	0.8506	0.9464
	SACK	0.9177	0.9670	0.9161	0.9411	0.9207	0.9365

Table 3.13: Allocation of Variation for GEO Efficiency and Fairness Values

Component	Sum of Squares		%age of Variation	
	Efficiency	Fairness	Efficiency	Fairness
Individual Values	17.3948	21.4938		
Overall Mean	17.3451	21.4884		
Total Variation	0.0497	0.0054	100	100
Main Effects:				
TCP Flavor	0.0344	0.0008	69.16	14.47
Buffer Size	0.0068	0.0006	13.65	11.48
Drop Policy	0.0001	0.0001	0.25	2.31
First-order Interactions:				
TCP Flavor-Buffer Size	0.0037	0.0012	7.54	22.16
TCP Flavor-Drop Policy	0.0025	0.0014	4.96	26.44
Buffer Size-Drop Policy	0.0002	0.0001	0.41	1.45

Standard Error, s_e = 0.0182(For Efficiency), 0.0139(For Fairness)

Table 3.14: Main Effects and Their Confidence Intervals for GEO

Factor	Mean Effect		Confidence Interval	
	Efficiency	Fairness	Efficiency	Fairness
TCP Flavor:				
Vanilla	-0.0295	0.0037	(-0.0420,-0.0170)	(-0.0058,0.0133)
Reno	-0.0252	0.0027	(-0.0377,-0.0127)	(-0.0068,0.0123)
NewReno	-0.0095	0.0034	(-0.0220,0.0030)	(-0.0062,0.0129)
SACK	0.0642	-0.0098	(0.0517,0.0768)	(-0.0194,-0.0003)
Buffer Size:				
0.5 RTT	-0.0138	0.0050	(-0.0240,-0.0036)	(-0.0029,0.0128)
1 RTT	-0.0099	0.0020	(-0.0201,0.0004)	(-0.0058,0.0098)
2 RTT	0.0237	-0.0070	(0.0134,0.0339)	(-0.0148,0.0009)
Drop Policy:				
EPD	0.0023	-0.0023	(-0.0049,0.0095)	(-0.0078,0.0033)
SD	-0.0023	0.0023	(-0.0095,0.0049)	(-0.0033,0.0078)

3.5.9.1 Analysis of Efficiency values: Results and Observations

TCP flavor explains 69.16% of the variation and hence is the major factor in deciding efficiency value. Confidence intervals for effects of TCP flavors clearly indicate that SACK results in substantially better efficiency than other TCP flavors. Since confidence intervals overlap for NewReno, Reno and Vanilla, one cannot be said to be better than other in terms of efficiency.

Buffer size explains 13.65% of the variation and interaction between buffer size and TCP flavors explains 7.54% of the variation. Confidence intervals for 0.5 RTT and 1 RTT buffer overlap, thus indicating similar performance. There is a marginal improvement in performance as buffer size is increased to 2 RTT. Vanilla and Reno show substantial efficiency gains as buffer size is increased from 1 RTT to 2 RTT. There is not much improvement for Vanilla and FRR when buffer is increased from 0.5 RTT to 1 RTT. Hence, in this case, 1 RTT buffer does not sufficiently reduce number of packets dropped to cause an increase in efficiency. However, for a buffer of 2 RTT, the reduction in number of dropped packets is enough to improve Vanilla and Reno's performance.

Drop policy does not have an impact in terms of efficiency as indicated by negligible allocation of variation to drop policy.

From the observations above, it can be concluded that SACK with 0.5 RTT buffer is the optimal choice for GEO links with either of EPD and SD as switch drop policy.

3.5.9.2 Analysis of Fairness values: Results and Observations

The conclusion here is similar to MEO delays. As can be seen from individual fairness values, there is not much difference between fairness values for different TCP types, buffer sizes or drop policies. All 9 main effects have very small values, and for 8 of them, their confidence intervals enclose 0. Thus, these simulations do not give us any information regarding relative fairness performance of different options.

3.5.10 Discussion

It is interesting to notice how the relative behavior of different TCP types change as link delay increases. As link delay increases, SACK clearly comes out to be superior than NewReno in terms of efficiency. For LEO delays, SACK and NewReno have similar efficiency values. For MEO delays, SACK performs a little better than NewReno and for GEO delays, SACK clearly outperforms NewReno. The reason for this behavior is that NewReno needs N RTTs to recover from N packet losses in a window whereas SACK can recover much faster and start increasing CWND again. This effect becomes more and more pronounced as RTT increases.

Per-flow accounting scheme SD does not always lead to increase in fairness when compared to EPD. This result can partly be attributed to nature of WWW traffic. SD accepts packets of only under-represented VCs after crossing the threshold R. For sufficient buffer size, many of these VCs are under represented in switch buffer because they do not have a lot of data to send. Thus, SD fails to cause significant increase in fairness. Also, it seems that per-flow accounting is useful only in presence of sufficiently large buffers. It is intuition that required buffer size for a link is mainly determined by its bandwidth-delay product as well as number of flows. Finding optimal buffer size for given link and traffic conditions remains a research problem.

In summary, this simulation study indicates that as delay increases, the marginal gains of end system policies become more important compared to the marginal gains of drop policies and larger buffers.

3.6 Bandwidth Allocation for Satellite ATM

A number of TDMA based multiple access schemes have been proposed and evaluated over the years [87, 91]. These schemes span from fully random access schemes such as ALOHA to hybrid reservation DAMA schemes at the other end of the spectrum. Table 3.15 provides a comparative evaluation of these schemes. However, they are not adequate enough to support the future broadband applications over satellite ATM networks.

Table 3.15: MAC Protocol Candidates

Protocol	Throughput	Delay	Scalability	Stability	Reconfig-urability	Complexity	Comments
Pure ALOHA	0.18	Low	Very Low	Low	Medium	Very Low	Simplest at the expense of throughput. Variable length message.
Slotted ALOHA	0.37	Low	Low	Low	Medium	Medium	Simplest
Tree-CRA	0.43-0.49	Medium	Medium	Medium	Medium	Medium	Stable
Reservation	0.6-0.8	High	Medium-High	Medium-High	Medium-High	Medium-High	High delay
DAMA (R-ALOHA)	0.6-0.8	Medium	High	High	High	Medium-High	Low delay
Hybrid Random/ Reservation	0.6-0.8	Variable	High	High	High	High	Low delay for short messages

Many different traffic types, such as data, voice and video, need to be supported simultaneously over the satellite system. Each traffic type has specific quality of service (QoS) performance parameters. Traditionally, slot based uplink systems, such as TDMA, have only had to dynamically allocate fixed bandwidth resources, such as voice or trunk traffic, or dealt with traffic of a single type. The uplink channel resources of a network capable, multimedia satellite needs to be managed such that all of these traffic types can be effectively and efficiently transported together in the same uplink beams that are eventually aggregated in the input port of a typical on-board processing satellite and switched to the appropriate downlink beam. If TDMA or MF-TDMA were utilized, then a multimedia capable Demand Assignment Multiple Access (DAMA) approach would have to be developed to respect the diverse QoS needs.

Two key design challenges for a multimedia DAMA are: 1) how efficiently can the system handle multiple channels of CBR and packet types of traffic and 2) how to allocate the slots efficiently. Isochronous, or constant bit rate (CBR), traffic covers most all circuit switched traffic one can find on today's legacy TDM networks, such as voice and video. CBR data has severe constraints placed on the time delay and delay variation that can be experienced before degradation can occur such as loss of synchronization. Normal data transfers do not very stringent timing issues and so there always is an inherent priority given to CBR type of traffic.

In the following sections, a bandwidth allocation scheme analytical model is developed and simulation results are presented in Section 3.6.1.

3.6.1 TDMA/DAMA Analytical Model

Normally, user transmit rates can typically be in the range of a few hundred kilobits per second (kbps) to well in the low Mbps. The terminal uplink uses multiple access techniques that can be TDMA or multi-frequency TDMA. The terminal downlink can be either multiple access or shared time division multiplexing that is usually in excess of the uplink rates. The gateway provides higher rate access links that are typically in the tens or to even hundreds of Mbps in both the uplink and downlink.

The satellite spot beam antenna systems that support frequency and spatial diversity to re-use the spectrum many times. Onboard channelization and switching functions support multiple access on each uplink and downlink coverage beam. Most initial onboard processing satellites will deploy switching systems with bandwidth in Gbps range. The uplink will support a DAMA technique that can dynamically assign and re-assign bandwidth to support terminal connectivity needs on an on-demand basis. In general the terminal frequency space is assigned and reconfigured only when needed. However, bandwidth is assigned to each terminal as it requests the time slots either by an onboard controller or alternatively by a ground based controller.

The terminals are capable of transmitting and receiving many different types of traffic to support a multitude of missions. There is an increasing need to simultaneously transport a number of single and multimedia applications across wide areas.

A TDMA frame structure consisting of a number of slots is assumed. Each user terminal requests time slots on the uplink channel to utilize bandwidth through the payload DAMA basis. It is proposed that slots assigned to a user need not be contiguous since a contiguous slot requirement means that some requests could be denied even though sufficient bandwidth exists (just not in a contiguous segment). An analytical model is proposed to quantify the gain in uplink utilization due to the non- contiguous allocation. [248] It is assumed that when the requested number of slots is not available, the entire request is rejected. This assumption will hold for constant-bit-rate (CBR) requests and for variable-bit-rate (VBR) requests of one slot.

The methodologies used are an exact analytic model of the system without a contiguous slot requirement and a simulation for the contiguous case. One advantage of using two approaches is that each model can be used to validate the other.

3.6.1.1 Frame Structure

Consider a superframe structure of N frames per superframe with each frame consisting of n slots as illustrated in Figure 3.9. For the results in this section, it is assumed that $n = 24$; The value of N is irrelevant since the model assumes relatively long holding times. A 16 Kbps call would require one slot per frame as shown by the “x’s” in the figure.

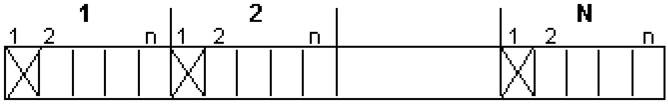


Figure 3.9: Super Frame Structure

3.6.1.2 Non-Contiguous Slot Analytical Model

The following resource sharing model and its analysis is due to J. S. Kaufman [241] and is shown in Figure 3.10. Applying this to the current system, the resource being shared is the collection of DAMA slots in a frame.

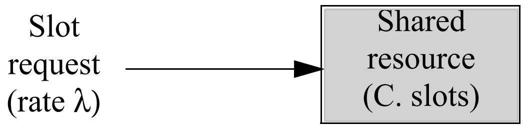


Figure 3.10: Model Overview

Requests for resources arrive at a mean rate λ and have two requirements:

- a spatial requirement - b slots
- a temporal requirement - the slots are required for τ units of time

The slots assigned to a request *need not be contiguous*.

A. General Assumptions

A1. k customer types, each with distinct spatial and/or temporal requirements. k is finite but arbitrary.

A2. A request whose spatial requirement cannot be satisfied is blocked and has no further affect on the system.

B. Stochastic Assumptions

B1. The slot requests arrive according to a stationary Poisson process (with mean rate λ).

B2. The spatial requirement b is an arbitrary discrete random variable with probability density given by

$$P\{b = b_i\} = q_i, \qquad i = 1, \ldots, k\,, \tag{3.1}$$

B3. A customer with spatial requirement b_i has residency time τ_i whose distribution has a rational Laplace transform and whose mean is denoted by $1/\mu_i$.

Assumptions B1 and B2 imply that requests requiring b_i slots arrive according to a Poisson process with mean rate

$$\lambda_i = \lambda q_i \tag{3.2}$$

The offered load to the system is

$$\rho = \sum_{i=1}^{k} \lambda q_i / \mu_i \tag{3.3}$$

Although some of the results in [241] also apply to the case of finite sources, only the case of Poisson arrivals is considered. The Poisson arrival assumption will be accurate for a large number of user terminals sharing the same uplink channel.

C. Sharing Policy

It is assumed that the slots in a frame are completely shared by competing requests. It is possible to extend the results to other sharing policies (e.g., reserving some slots for particular types of sources).

The slots in the frame can be completely shared, or certain customers may have exclusive use of the portions of the resource (slots) while the excess commonly shared. The resource sharing policy has a strong effect on the blocking experienced by different customers. In a complete sharing policy, a customer requiring b units of the resource is blocked if and only if fewer than *b* units of the total resource is available.

State Description

Assume that the residency times are exponentially distributed. The state description is given by

$n = (n_1, \ldots n_k)$

where n_i = number of slots occupied by source type *i*

If each of the C slots of a type *i* source simultaneously require b_i slots, unlike blocking models with batched Poisson arrivals, all b_i slots must be simultaneously relinquished when the services is done. The notation is as follows:

$$n_i^+ = (n_1, \ldots n_{i-1}, n_i + 1, n_{i+1}, \ldots n_k)$$

$$n_i^- = (n_1, \ldots, n_{i-1}, n_i - 1, n_{i+1}, \ldots n_k)$$

Ω = set of allowable slot assignment states

$$\delta_i^+(n) = \begin{cases} 1 & \text{if } n_i^+ \in \Omega \\ 0 & \text{otherwise} \end{cases}$$

$$\delta_i^-(n) = \begin{cases} 1 & \text{if } n_i^- \in \Omega \\ 0 & \text{otherwise} \end{cases}$$

$$n \bullet b = \sum_{i=1}^{k} n_i b_i$$

Because each resource sharing policy, gives rise to a set of Ω of allowable states, any set Ω can be viewed as a resource sharing policy provides that,

$$n \in \Omega \Rightarrow n_i \geq 0 \qquad i = 1, \ldots, k \text{ and } n \bullet b \leq C \tag{3.4}$$

State distribution for any resource sharing policy is described in the following:

a. Partial Sharing Policy

Coordinate convex policies, for which the state distribution holds is given by

$$n \in \Omega \Rightarrow n_i \geq 0 \qquad i = 1, \ldots k$$

$$\text{and } n \in \Omega \text{ and } n_i \geq 0 \Rightarrow n_i^- \in \Omega \tag{3.5}$$

$$\Omega = \left\{ n : 0 \leq n_i b_i \leq C_1 + C_0, \quad i = 1, \ldots, k, \quad 0 \leq n \bullet b \sum_{i=1}^{k} C_i + C_0 \right\}$$

Type i source has C_i units of slots dedicated and all sources contend and share the remaining C_o units.

If $C_i = 0$, $i = 1, \ldots, k$, we have complete sharing policy and if $C_i = 0$ we have, in effect, k non shared slots, with capabilities $C_1, \ldots. C_k$.

b. Complete Sharing Policy with an Ordering Constraint

$$\Omega = \{ n : 0 \leq n \bullet b \leq C, \qquad n_i b_i \leq n_{i+1} b_{i+1} \qquad i = 1, \ldots, k-1 \}$$

Although the resource is completely shared, the policy insists that type i slots do not occupied more of the resource than type $i + 1$ slots.

In the coordinate convex policy, departures are never blocked. The details are found in [249].

3.6.1.3 Solution to the Analytic Model

Performance measure P_{b_i}

The performance measure is the blocking probability.

P_{b_i} , i.e., probability that type i arrival requiring b_i slots of the source, is blocked.

If $P(\bullet)$ denotes the state distribution given that policy Ω is in effect then

$$P_{b_i} = \sum_{n \in B_i^+} P(n) \tag{3.6a}$$

where

$$B_i^+ = \left\{ n \in \Omega : n_i^+ \notin \Omega \right\} \tag{3.6b}$$

The state distribution corresponding to an arbitrary resource sharing policy Ω is given by

$$P(n) = \prod_{i=1}^{k} \frac{a_i^{n_i}}{n_i!} G^{-1}(\Omega) \qquad \text{all } n \in \Omega \tag{3.7}$$

where

$$G(\Omega) = \sum_{n \in \Omega} \prod_{i=1}^{k} \frac{a_i^{n_i}}{n_i!} \tag{3.8}$$

The type i offered load $a_i = \dfrac{\lambda_i}{\mu_i}$

where $\lambda_i = \lambda_{q_1}$

and $\dfrac{1}{\mu_i}$ is the mean type i slots

The blocking experienced by a type i slot $i = 1,\ldots,k$ when policy Ω in effect is given by

$$P_{b_i} = \frac{G(B_i^+)}{G(\Omega)} \tag{3.9}$$

where B_i^+ is defined by Eq. 3.6a.

It is interesting to observe that the resource sharing policy Ω enters the state distribution $P(\bullet)$ only via the normalization constant G(Ω).

The normalization constant G(Ω) is viwed as a set function $G(\bullet)$, completely determines the blocking probabilities. In the special case of complete sharing, the normalization constant is denotd by G(C,K) and Eq. 3.9 can be written as

$$P_{b_i} = 1 - \frac{G(C - b_i, k)}{G(C, k)} \tag{3.10}$$

$$\text{where } G(C, k) = \sum_{n \in \Omega} \prod_{i=1}^{k} \frac{a_i^{n_i}}{n_i!}$$

This is a familiar result from BCMP queuing networks [249] and much applied work has been done in deriving efficient numerical methods for computing the normalization constant above. The “BCMP” queuing networks derive their name from the well-know four-author paper [249]. In the present case, a real contribution of [241] was in developing a simple one-dimensional recursion for computing the blocking probabilities.

3.6.1.4 The One-Dimensional Recursion

As discussed in previous section, $\Omega = \{n{:}0 \le n \bullet b \le c\}$ and from Eq. 3.10 G(C,k) is computed recursively by using

$$\begin{aligned} G(j, i) &= \sum_{l=0}^{[j/b_i]} \frac{a_i^l}{l!} G(j - lb_i, i-1) \qquad i = 2, \ldots, k\,;\, j = 0, 1, \ldots, C \\ G(j, 1) &= \sum_{l-0}^{[j/b_1]} \frac{a_1^l}{l!} \qquad j = 0, 1, \ldots, C \end{aligned} \tag{3.11}$$

and [x] = largest integer $\le$ x

Let the random variable $j = \boldsymbol{b} \bullet \boldsymbol{n}$ be the total number of slots occupied. Then the distribution of j is given by

$$q(j) = \sum_{\{n:n \bullet b = j\}} \prod_{i=1}^{k} \frac{a_i^{n_i}}{n_i!} \bullet G^{-1}(C, k) \tag{3.12}$$

The key result from a computational viewpoint (proved in [241]) is that the distribution satisfies

$$\sum_{i=1}^{k} a_i b_i q(j - b_i) = jq(j) \tag{3.13}$$

for $j = 0, \ldots, C$, where $q(x) = 0$ for $x < 0$ and

$$\sum_{j=0}^{c} q(j) = 1 \tag{3.14}$$

The previous equation defines a one-dimensional recursion (for arbitrary k) which trivially generates $q(j), j = 1, \ldots, C$, starting with $q(0) = 1$. The above equation is then used to normalize the elements. Finally, the desired blocking probabilities are given by

$$P_{b_i} = \sum_{i=0}^{b-1} q(C-1) \tag{3.15}$$

for $i = 1, \ldots, k$. The carried load (or utilization of the uplinks) is given by

$$\text{carried load} = \sum_{i=1}^{k} \lambda(1 - P_{b_i})q_i \tag{3.16}$$

3.6.2 Contiguous Slot Simulation

In order to quantify the advantage (if any) of allowing totally flexible assignments there is need to compute the blocking and carried load achievable when requests for slots are only allowed if the required slots can be assigned contiguously within a frame. Deriving a general analytical model for the non-contiguous case is possible in principle but the explosion in the state space required to keep track of the lengths of available runs of slots (and which calls occupy which slots) is prohibitive and therefore is not practical. Therefore a discrete event simulation for the contiguous case is considered.

3.6.3 Performance Results

Example 1

A case with heterogeneous requests is considered. In particular, it is assumed that 30% of requests are for 16Kbps CBR, 50% are for 64Kbps CBR, and 20% are for 128Kbps CBR. The results are given in Figures 3.11 and 3.12. Note that calls requiring more slots see significantly higher blocking than smaller bandwidth calls. It can be noted that the contiguous requirement increases blocking for some calls and decreases it for others. In particular, the calls requiring 128Kbps worth of bandwidth see higher blocking as would be expected since to be accepted, there always needs to be 8 available slots in a row. Since less of these large bandwidth calls are carried, there is more capacity available for the low bandwidth calls which explains their decreased blocking probability. In this case, the 64Kbps calls see about the same blocking. Although there is a disparate behavior in blocking for different streams, the total carried load is less for the contiguous case as is seen in Figure 3.12. Note that allowing totally flexible slot assignment results in an 8% increase in uplink channel utilization at an offered load of 80%.

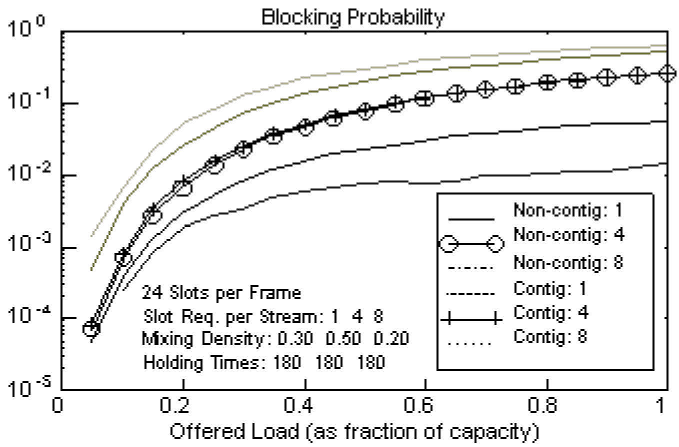


Figure 3.11: Blocking probability for mixture of 16Kbps, 64Kbps, 128Kbps CBR

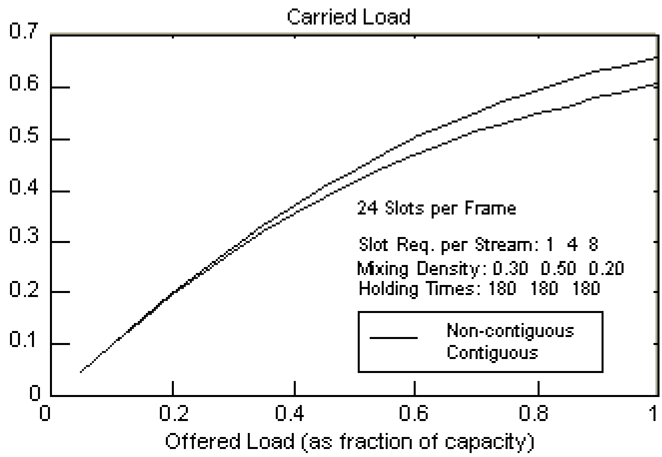


Figure 3.12: Slot utilization for mixture of 16Kbps, 64Kbps, 128Kbps CBR

Example 2

In this example, mixture of two source types is considered. An assumption is made that 95% of requests are for an average of 16Kbps bandwidth (1 slot) and 5% require an average of 192Kbps (12 slots). The results shown in Figures 3.13 and 3.14 show that, once again, the non-contiguous case affords a roughly 10% gain in uplink utilization. [248] Figure 3.13 exhibits an interesting behavior which becomes intuitive with a little thought. As the load increases from zero the blocking for both types of calls increases although there is a good chance that one large call is occupying slots so that the small calls must contend for the remaining slots. As the load increases, however, the blocking of the large calls reaches a point where it is likely that new large bandwidth requests will be blocked and there is a significant chance that no large calls are in the system. Now the small calls have a better chance of contending for all 24 slots so the blocking probability decreases. It continues to decrease as the blocking for large calls increases until the small calls themselves cause congestion at which point the blocking probability increases again. This example serves as another case for which the analytic and simulation models are mutually validated.

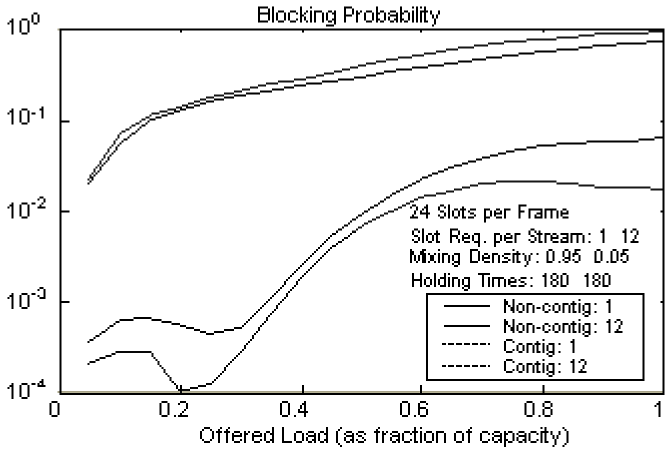


Figure 3.13: Blocking probability for mixture of 16Kbps and 192Kbps CBR

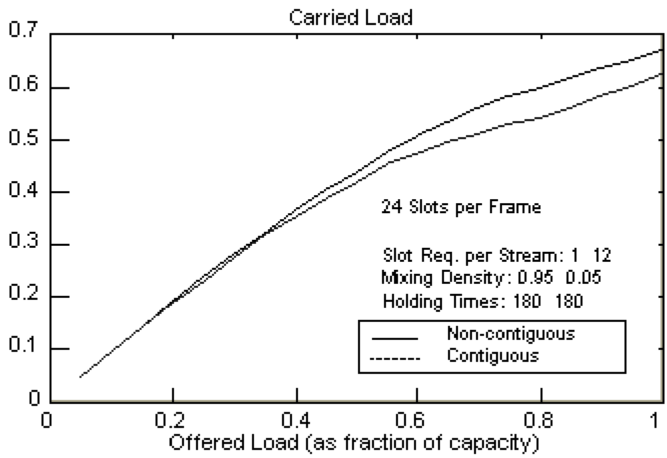


Figure 3.14: Slot utilization for mixture of 16Kbps and 192Kbps CBR

3.6.4 Discussion

To improve system responsiveness and flexibility, many DAMA schemes have been proposed and some of them are under development for satellite communications systems. In this section, a DAMA model for CBR and VBR services was proposed. Two possible resource (slot) allocations, contiguous and non-contiguous, are presented. Analytical and simulation results for blocking probability vs. offered load show that non-contiguous allocation results in up to a ten percent increase in channel utilization.

3.7 Summary

Multimedia satellites networks use onboard processing and ATM or fast packet switching to provide two-way communications. Buffer requirements, TCP enhancements analysis by full factorial simulations, and bandwidth allocation for a satellite ATM network supporting UBR service was studied. The simulation study for buffer requirements for TCP over ATM UBR for LEO and GEO configurations was performed. The end system policy of TCP SACK and selective drop policy was assumed. The simulation results indicate that a buffer size of about 0.5 × RTT to 1 × RTT is sufficient to provide 98% of throughput to infinite TCP traffic for long latency networks and large number of sources. Fairness is high for a large number of sources because of per-VC buffer management performed at the switches.

The TCP analysis study in Section 3.5 assumed several TCP flavors, e.g., Vanilla, Fast Retransmit Recovery (Reno), New Reno, and SACK; drop polices of Early Packet Drop (EPD) and Selective Discard (SD); buffers sizes of 0.5, 1, and 2 times the round trip delay-bandwidth product and propagation delays for GEO, LEO and MEO configuration.

It is interesting to notice how the relative behavior of different TCP types change as link delay increases.

As link delay increases, SACK clearly comes out to be superior than NewReno in terms of efficiency. For LEO, SACK and NewReno have similar efficiency values. For MEO, SACK performs a little better than NewReno and for GEO, SACK clearly outperforms NewReno. The reason for this behavior is that NewReno needs N RTTs to recover from N packet losses in a window whereas SACK can recover faster, and start increasing CWND again. This effect becomes more and more pronounced as RTT increases.

SD does not always lead to increase in fairness as compared to EPD. This result can again be attributed to nature of WWW traffic. SD accepts packets of only under-represented VCs after crossing the threshold R. For sufficient buffer size, many of these VCs are under represented in switch buffer because they do not have a lot of data to send. Thus, SD fails to cause significant increase in fairness.

It is concluded that for long delay links, end system policies are more important than switch drop policies in terms of efficiency and fairness for WWW traffic.

Section 3.6 proposes a bandwidth allocation scheme for satellite ATM based on DAMA concept. A DAMA model for supporting CBR and VBR services is developed. Two possible resource (slot) allocations, contiguous and non-contiguous, for a TDMA based satellite ATM network are presented. Analytical and simulation results for block probability vs. offered load for non-contiguous and contiguous cases are presented. An example with 30% request for 16 Kbps CBR, 50% requests for 64 Kbps CBR and 20% for 128 Kbps CBR, non-contiguous i.e., totally flexible slot assignment results in an 8% increase in uplink channel utilization at an offered load of 80%. Another example with 95% requests for an average of 16 Kbps and 5% requests for an average of 192 Kbps show, that non-contiguous case affords roughly a 10% gain in uplink utilization.

4 Quality of Service for Satellite IP Connectivity Networks

Quality of Service describes the guaranteed delivery of applications based on the low delay and packet loss requirements. A set of Service Level Agreements (SLAs) normally specify the QoS requirements as described in Section 2.3. QoS is delivered by restricting the total amount of traffic competing for the same amount of resources if the ingress traffic exceeds the minimum capacity of the network. The network capacity is effectively proportioned by the packet prioritization, so that higher priority traffic is unaffected by the lower-priority traffic. QoS guarantees can be made either over an aggregate of communication associations or individual flows. QoS is assured by reserving resources such as bandwidth and buffer space. [250]

Applications differ in their QoS requirements [112]. Most of the applications are loss sensitive, while data applications can recover from packet loss through retransmissions. Packet losses of about 5% generally lead to very poor effective throughput. Data applications such as file transfer are not generally delay sensitive. Applications such as streaming audio and video require fixed bandwidth.

Traditionally, the current Internet supports only “best effort” i.e., same service to all the applications. The emerging applications which require differential service guarantees, cannot be supported by the current Internet. Considerable research efforts have been undertaken by several institutions and the IETF to develop new QoS mechanisms for the Internet [251].

Satellite communications system is an excellent candidate to provide global broadband Internet service [112]. TCP and UDP protocols form the basis of the Internet. The Satellite Internet is expected to continue to support applications based on TCP and UDP. However, the performance of these protocols will be affected by the long delay and error prone characteristics of satellite links. The TCP over satellite working group and Performance Implications of Link Characteristics (PILC) Working Groups of the IETF have proposed several recommendations on the TCP enhancements for the large delay bandwidth environments such as satellite links. [252] provides a study of QoS for Radio access networks.

Recently, QoS over satellite IP networks has attracted many researchers [252, 254, 255]. Currently, the proposed QoS mechanisms include IntServ [184, 186] with RSVP [189], DiffServ, and MPLS [214]. These QoS mechanisms are addressed for terrestrial networks.

It is very timely and crucial to investigate and to assess whether these QoS mechanisms can be applied to satellite networks.[285] If so,

– What are the QoS architectures for satellite based Internet?
– How do the TCP and UDP protocols behave under satellite networks supporting differentiated services?
– How does the traffic engineering of MPLS perform over a satellite network?

In this chapter, the above questions are addressed and the answers are presented. Specifically, the following problems are addressed in this chapter:

Problem 1: QoS Architectures for Satellite IP Network

Three QoS architectures for satellite IP are proposed Section 4.4. These architectures are (a) IntServ based, (b) DiffServ based, and (c) IntServ/DiffServ with aggregate RSVP. It is concluded that the architectures 1 and 2, based on IntServ and DiffServ respectively, are the simplest to implement at the terminals and the gateways. They provide end-to-end QoS. Architecture 3 is a scalable architecture. It provides an end-to-end QoS with dynamic aggregate RSVP reservations. However, this aggregate RSVP is not widely deployed in terrestrial environments. Considering the advantages of the aggregate RSVP as opposed to individual flow RSVP, this architecture 3 is a compromise between 1 and 2.

Problem 2: DiffServ QoS Simulation for TCP and UDP over Satellite IP Network.

In Section 4.5 a simulation model with a wide range of simulations varying several factors to identify the significant ones influencing fair allocation of excess satellite network resources among congestion sensitive TCP and congestion insensitive UDP flows are developed for GEO, MEO and LEO networks (Appendix 4A.1 has LEO simulation results). One of the goals of deploying multiple drop precedence levels in an Assured Forwarding traffic class on a satellite network in GEO network, is to ensure that all customers achieve their reserved rate and a fair share of excess bandwidth. The study is extended for network access satellite architecture, without on-board processing and switching and the impact analysis of BER and other factors on the performance of assured forwarding (given in Appendix 4A.2).

Problem 3: MPLS over Satellite Network

MPLS traffic engineering is applied to study TCP and UDP throughput improvements over a satellite network in Section 4.6. Simulation results show that the total network throughput improves significantly with proper traffic engineering. Congestion-insensitive (UDP) flows affect the throughput of congestion-sensitive (TCP) flows. Therefore, different types of flows should be isolated in different trunks in order to guarantee quality of service. For the above scheme to be effective the trunks need to be end-to-end, otherwise the advantages of isolation in other parts of the network are eliminated or reduced significantly.

Section 4.1 presents an ITU-T developed set of QoS classes and application specific QoS requirements are provided in Section 4.2. Section 4.3 provides a discussion on IntServ, DiffServ and MPLS QoS mechanisms. Section 4.4 proposes IntServ and DiffServ based QoS architectural candidates for broadband satellite IP networks emphasizing the QoS functionality at the gateway and the edge router. Section 4.5 describes a simulation model for transporting TCP and UDP dependent applications over a satellite IP network. The effect of number of drop precedences is studied. The Analysis of Variation (ANOVA) technique is used to study the parameter sensitivity for the TCP and UDP and the results are presented for GEO, MEO and LEO satellite networks in Section 4.5.5 [256, 257]. Section 4.6 [258] proposes a simulation model for traffic engineering of MPLS applied to the broadband satellite IP network. The throughput performance improvement results for TCP and UDP are presented. Section 4.7 summarizes the study results.

4.1 IP QoS Classes

ITU-T Study Group 13 developed an ITU-T Recommendation Y.1541 for developing the IP end-to-end performance objectives within a defined set of six QoS classes for IP-based networks in January 2002. The QoS classes are specifically intended for data and for real-time applications including voice. Table 4.1 presents the IP QoS classes.

Table 4.1: IP QoS Classes

QoS Class	Applications (Example)	Node Mechanisms
0	Real-Time, Jitter sensitive, high interaction (VoIP, VTC)	Separate Queue with preferential servicing, Traffic grooming
1	Real-Time, Jitter sensitive, interactive (VoIP, VTC)	
2	Transaction Data, Highly Interactive (Signaling)	Separate Queue, Drop priority
3	Transaction Data, Interactive	
4	Low Loss Only (Short Transactions, Bulk Data, Video Streaming)	Long Queue, Drop priority
5	Traditional Applications of Default IP Networks	Separate Queue (lowest priority)

4.2 IP QoS Performance Objectives

ITU-T Recommendation Y.1541 dated 18 Feb, 2002 defines six QoS classes which are intended to be basis of agreement between end users and network service providers and between service providers. [174] These QoS objectives are applicable when access link speeds are at T1 or E1 rate or higher. Table 4.2 provides the provisional IP QoS class definitions and network performance objectives. These require to be evaluated for satellite transport. The network performance parameters include:

- IP Packet Transfer Delay (IPTD)
- IP Packet Delay Variation (IPDV)
- IP Packet Loss Ratio (IPLR)
- IP Packet Error Ratio (IPER)

Table 4.2: Provisional IP QoS class definitions and network performance objectives

Network Performance Parameter	Nature of Network Performance Objectives	QoS Classes					
		Class 0	Class 1	Class 2	Class 3	Class 4	Class 5 Unspecified
IPTD	Upper bound on the mean IPTD	100ms	400ms	100ms	400ms	1 s	Unspecified
IPDV	Upper bound on the $1\text{-}10^{-3}$ quantile of IPTD minus the minimum IPTD	50ms	50ms	Unspecified	Unspecified	Unspecified	Unspecified
IPLR	Upper bound on the packet loss probability	1×10^{-3}	1×10^{-3}	1×10^{-3}	1×10^{-3}	1×10^{-3}	Unspecified
IPER	Upper bound			1×10^{-4}			

4.2.1 Satellite IP QoS Objectives

The use of geostationary satellites was considered during the study of the Hypothetical Reference Paths (HRPs). A single geostationary satellite can be used within the HRPs and still achieve end-to-end objectives on the assumption that it replaces significant terrestrial distance, multiple IP nodes, and/or transit network sections. The use of low- and medium-Earth orbit satellites was not considered in connection with these HRPs.

When a path contains a satellite hop, this portion will require an IPTD of 320 ms, to account for low earth station viewing angle, low rate TDMA systems, or both. In the case of a satellite possessing onboard processing capabilities, 330 ms of IPTD is needed, to account for onboard processing and packet queuing delays.

It is expected that most HRPs which include a geostationary satellite will achieve IPTD below 400 ms. However, in some cases the value of 400 ms may be exceeded. For very long paths to remote areas, network providers may need to make additional bilateral agreements to improve the probability of achieving the 400 ms objective.

4.3 IP QoS Mechanisms Background

The current Internet infrastructure treats all IP packets equally on a best effort basis. There is also no possibility of controlling packet delays; neither between packets nor end-to-end. Forwarding decisions are made hop by hop. There is no guarantee that packets belonging to the same application get routed across the same path.

This approach has worked well, so far, for most data applications. The performance of delay and delay variation (jitter) sensitive applications such as voice and video over the Internet has been poor. However, the performance of the same applications in a controlled environment such as test beds or Internet, has been acceptable. This indicates the ability of IP to support mulitmedia applications, provided that sufficient network resources are available. The use of ATM as the transport network infrastructure allows for the control of end-to-end jitter and delay variation [170]. Standards have been in place to transport IP using ATM. However supporting different application segments will involve more than just mapping IP services to UBR connections [202].

4.3.1 Integrated Services (IntServ)

IntServ is a per-flow based QoS framework with dynamic resource reservation. In this, routers need to reserve resources in order to provide quantifiable QoS for specific traffic flows. Resource Reservation Protocol (RSVP) is the signaling protocol for application to reserve network resources. It adopts a receiver-initiated reservation style which is designed for a multicast environment and accommodates heterogeneous receiver service needs.

In RSVP, the flow source sends a PATH message to the intended flow receiver(s), specifying the characteristic of the traffic. Each network router along the way records path characteristics such as available bandwidth, as the PATH message propagates towards the receiver(s). Upon receiving a PATH message, the receiver responds with a RESV message to request resources along the path recorded in the PATH message in reverse order from the sender to the receiver. If the request is accepted by all intermediate routers, link bandwidth and buffer space are allocated for the flow. The flow-specific state information is stored in the routers. Reservation can be shared along branches of the multicast delivery trees.

The IntServ architecture adds two service classes to the existing best-effort model – guaranteed service and controlled load service. Guaranteed service [184] provides an upper bound on end-to-end queuing delay and is aimed to support applications with hard real-time requirements. Controlled-load service [189] provides a quality of service similar to best-effort service in an underutilized network, with almost no loss and delay and is aimed to share the aggregate bandwidth among multiple traffic streams in a controlled way under overload conditions.

IntServ can deliver fine-grained QoS guarantees by using per-flow resource reservation. Introducing flow-specific states to routers means a fundamental change to the current Internet architecture, especially in the Internet backbone where it will be difficult for the router to maintain a separate queue for each of the hundred thousand flows which may be present.

Many people in the Internet community believe that IntServ is more suitable for intra-domain QOS or for specialized applications such as high-bandwidth flows. IntServ is difficult to be realized across the network because incremental deployment is only possible for controlled-load service, while ubiquitous deployment is required for guaranteed services.

4.3.2 Differentiated Services (DiffServ)

DiffServ [190, 191] has been proposed by IETF with scalability as the main goal. It is a per-aggregate-class based service discrimination framework. using packet tagging. Packet tagging uses bits in the packet header to mark a packet for preferential treatment. The type-of-service (TOS) byte is used to mark packets in IPv4. The TOS byte consists of a 3-bit precedence field, 4-bit field indicating requests for minimum delay, maximum throughput, maximum reliability, and minimum cost, and one unused bit. DiffServ redefines this byte as the DS field. Six bits of the DS fields form the DiffServ Code Point (DSCP) and the remaining two bits are unused.

DiffServ uses DSCP to select per-hop behavior (PHB) a packet experiences at each node. A PHB is packet forwarding treatment specified in a relative format compared to other PHBs, such as relative weight for sharing bandwidth or relative priority for dropping. Before a packet enters a DiffServ domain, its DSCP field is marked by the end-host or the first-hop router according to the service quality the packet requires. Within the IntServ domain, each router only looks at DSCP and decides, without needing complex classification or per-flow state, how the packet is to be treated.

DiffServ has two important design principles - pushing complexity to the network boundary and the separation of policy and supporting mechanisms.

Pushing Complexity to network boundary: This is important for the scalability of DiffServ. The network boundary refers to the application hosts, leaf routers and edge routers. Operations can be performed at fine granularity (operations such as complex packet classification and traffic conditioning) at the network boundary as it has relatively small number of flows. Network core routers have larger number of flows and so should perform fast and simple operations.

Separation of control policy and supporting mechanisms: This is important for the flexibility of DiffServ. DiffServ only defines several PHBs as the basis for QoS provisioning. It leaves the control policy as an issue for further work, which can be changed as needed while the PHBs should be kept relatively stable.

DiffServ provides two service models besides best effort model – Premium service and Assured service. Premium service is a guaranteed peak rate service, which is optimized for very regular traffic patterns and offers small or no queuing delay. It provides absolute

QoS assurance. Assured service is based on statistical provisioning. It tags packets as In or Out according to their service profiles. If packets are needed to be dropped, those marked Out are dropped first. This service provides relative QoS assurance and can be used to build “Olympic Service” which as gold, silver, and bronze service levels.

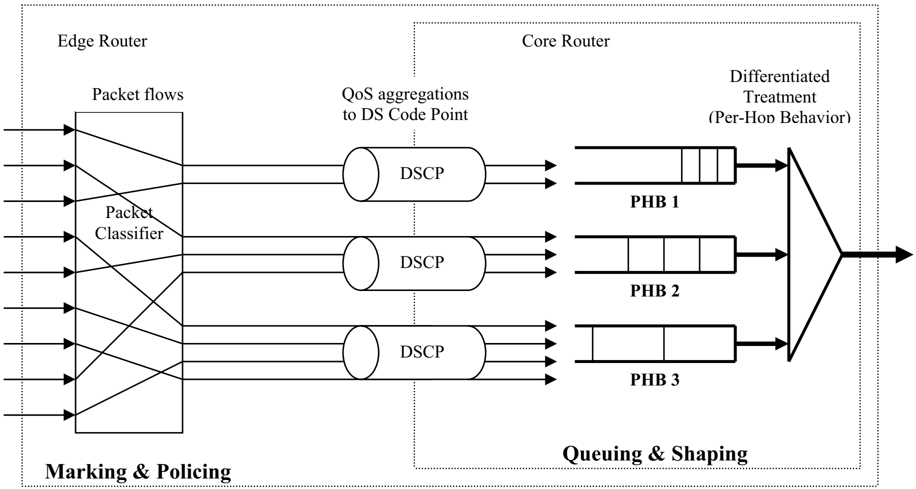


Figure 4.1: Functions of DiffServ

Figure 4.1 shows some of the DiffServ functions to be performed at the routers – marking, classifying, policing, queuing, and shaping.

Service Profiles: A service profile indicates which traffic is to be treated differently and what type of service is requested. The former may be indicated by setting a packet filter based on packet header fields such as IP addresses. The latter can be specified using a token bucket filter (a rate and a burst size), along with some delay requirements. The service profile is likely to be set up administratively, although it is technically possible to use RSVP messages to set up profiles as well.

Packet Classification: The ingress router must check all received packets against the service profiles it has to check if the packets should receive differential treatment. It may use packet filters to match packet headers and token filters to check conformance to the profile. Packets that do not meet profiles can either be discarded or sent into the network with higher drop precedence, depending on the nature of service to be provided.

Packet Marking: The ingress router must also mark the packets as they enter the network with appropriate values so interior routers can handle the packets differentially. The marking can use header fields. For example, for IPv4 packets, the TOS octet can be used. This has 3 bits for IP Precedence and 4 bits for Type of Service.

4.3.2.1 Queue Management

In the interior routers, differentiated packets have to be handled differently. To do so, the router may employ multiple queues, along with some Class Based Queuing (CBQ) service discipline or simple priority queuing. Generally, delay-sensitive traffic will be serviced sooner, and loss-sensitive traffic will be given larger buffers. The loss behavior can also be controlled using various forms of Random Early Detection (RED). These disciplines using probabilistic methods to start dropping packets when certain queue thresholds are crossed, in order to increase the probability that higher-quality packets can be buffered a the expense of more dispensable packets.

RED is a congestion avoidance algorithm that detects congestion and keeps the average queue length in a region of low delay and high throughput. During congestion, when routers drop packets, it is very likely that the dropped packets belong to many different connections. By detecting congestion and notifying only a randomly selected fraction of users, RED avoids congestion and the global synchronization problem. It also avoids bias against bursty connections.[196, 197]

RED utilizes two thresholds, T_{low} and $T_{high.}$ If the average queue length is below the lower threshold, the algorithm is in the normal operation zone and all packets are accepted. If it is above the higher threshold, RED is in the congestion control region and all incoming packets are dropped. If it is between both thresholds, RED is in the congestion avoidance region and the packets are discarded within a certain probability P_a which is increased by two factors – by the counter which maintains number of packets in queue and as the average queue length approaches the higher threshold. The algorithm computes an intermediate probability P_b, whose maximal value P_{max} is reached when the average queue length is equal to T_{high}. For a constant average queue length, all incoming packets have the same probability to get dropped. As a result, RED drops packets in proportion to the connection's share of the bandwidth.

The average queue length Q_{avg} is given by

$$Q_{avg} = (1 - W_q)Q_{avg} + W_q q, \qquad 0 \le W_q \le 1$$

$$P_b = P_{max} \frac{(Q_{avg} - T_{low})}{T_{high} - T_{low}}$$

where,

P_b = dropping probability

W_q = weight

T_{low} = Queue length (threshold)

T_{high} = Queue length (threshold 2)

$$P_a = \frac{P_b}{1 - counter.P_b}$$

From the formulas, W_q can be calculated as

$$W_q = L + 1 + \frac{(1 - W_q)^{L+1} - 1}{W_q} < T_{low}$$

From the above formulas, it can be observed that W_q determines how fast the algorithm will respond to changes in the queue length. If the weight is too high, when the average will not filter out transient congestion

Table 4.3 provides a comparison of the IntServ and DiffServ mechanisms.

Table 4.3: Comparison of IntServ and DiffServ

	Integrated Services	Differentiated Services
QoS Nature	Absolute	Relative
Granularity	Individual Flow	Aggregate Flow
State in Routers	Per flow	Per Aggregate
Traffic Classification	Several header fields	DS field (6 bits) of the IP header
Admission Control	Required	Required for absolute differentiation
Signaling	RSVP	Not Required
Scalability	Limited by the number of flows	Limited by the number of classes of service
Complexity	High	Low
Availability	Yes	Yes

4.3.3 Multi-Protocol Label Switching (MPLS)

The third architecture proposed by the IETF for provisioning QoS is Multi-Protocol Label Switching (MPLS). MPLS attempts to set up paths in a network along which packets that carry appropriate labels can be forwarded very efficiently (i.e., the forwarding engine would not look at the entire packet header, rather only at the label and use that to forward the packet). Not only does this allow packets to be forwarded more quickly, it allows the paths to be set up in a variety of ways: the path could represent the normal destination-based routing path, it could represent a policy-based explicit route, or it could represent a reservation-based flow path. Ingress routers classify incoming packets and wrap them in an MPLS header that carries the appropriate label for forwarding by the interior routers.

In the MPLS model, the labels are distributed by a dynamic Label Distribution Protocol (LDP), which effectively sets up a Label Switched Path (LSP) along the Label Switched Routers (LSR). The LDP could be driven off destination-based routing (e.g., OSPF) or from reservation requests (e.g., RSVP) or some other policy-based explicit route. In some sense, LDP is creating label state in the network, but this is not so different from the normal forwarding tables created by routing protocols. It is important to note that this label state is not per packet or per flow, but usually represents some aggregate (e.g., between some source-destination pair). Therefore, the state produced by MPLS is manageable and scalable. [214, 216]

The MPLS model is compatible with the DiffServ model. The ingress routers can use service profiles to assign labels to packets. The LSPs could represent provisioned paths inside the networks, and the labels carried in MPLS headers can be used to differentiate packets. The drop precedence of the packet can also be indicated in the MPLS header. It has also been proposed to use RSVP as the LDP so that a single protocol can be used for setting up various types of LSPs, including destination-based, policy-based, and reservation-based paths.

4.4 Quality of Service Architectures for Satellite IP

The current Internet treats all IP packets equally on a best effort basis. The data applications which are not delay sensitive, might receive satisfactory performance. But the performance for delay sensitive applications such as voice and video over satellite Internet has been poor. To improve the QoS performance for multimedia applications over the Internet, three different approaches have been proposed [218]. These IP QoS schemes are proposed for terrestrial access of the Internet only. As discussed in Chapter 2, many satellite systems are currently supporting Internet applications and more satellite networks are being planned and developed to support Internet in the future. However, there are no current QoS provisioning architecture standards for the support of multimedia services over satellite. [112, 259]

This section proposes three QoS architectures for satellite IP networks. These architectures are:

Architecture 1: Integrated Services (IntServ) based model.

Architecture 2: Differentiated Services (DiffServ) based model.

Architecture 3: IntServ/DiffServ/Aggregate RSVP based model.

Figure 4.2 shows a generic broadband satellite network supporting IP services.

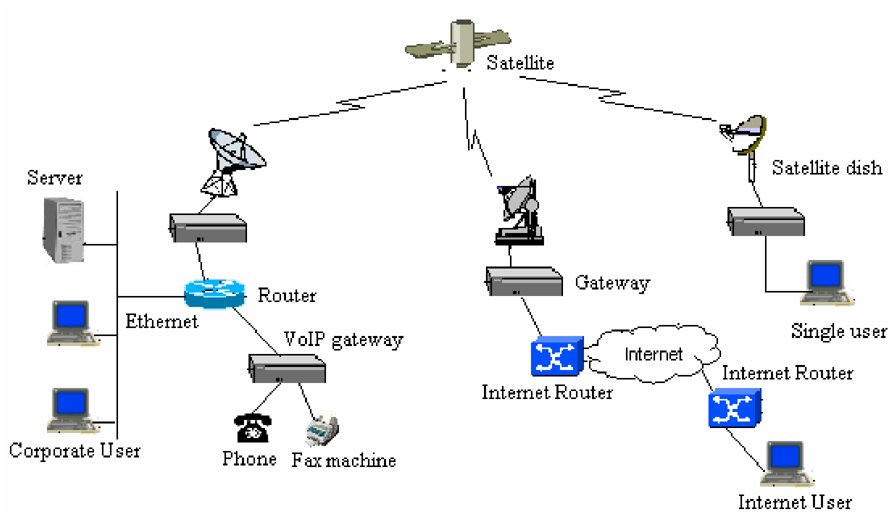


Figure 4.2: Broadband Satellite Network

The architectures describe necessary interworking functions at the terminals, gateways and onboard packet switch.

4.4.1 Satellite IP QoS Architecture 1 – IntServ

Figure 4.3 shows the QoS architecture for satellite IP network, based on IntServ.

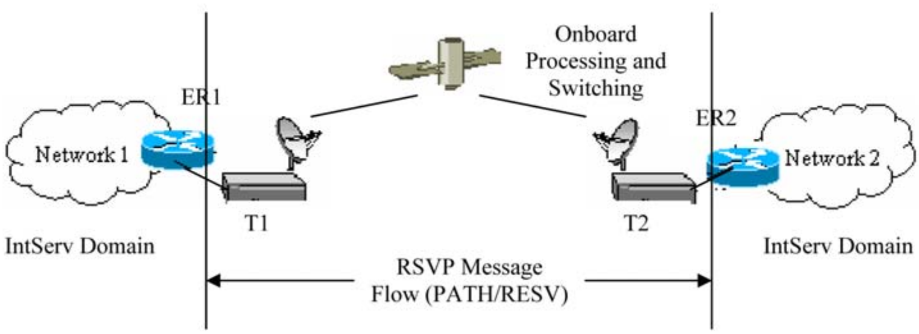


Figure 4.3: IntServ QoS Architecture

The terminal T1 and the hosts connected to it are RSVP capable. An application sends a PATH message requesting Guaranteed Services (GS) for example, from the Network 1. The terminal T1 upon receiving PATH message indicating a GS service request, registers

RSVP flow and modifies ADSPEC. The terminal performs the Call Admission Control and determines whether the call can be accepted as specified. If the call is accepted, the PATH message reaches the destination terminal T2.

Terminal T2 participates in RSVP, registers the flow, modifies ADSPEC and forwards PATH.

The edge router ER2 forwards the PATH message to the destination host. The edge routers have to perform multifield flow classification, and maintain flow state.

PATH message is created and sent back to the sender in the reverse path.

In a terrestrial network, RSVP has to be implemented in all routes along the path to achieve end-to-end QOS, which is difficult from an implementation point of view. In a satellite IP network, end-to-end IntServ QoS architecture is feasible because the packets flow through a router in a bent pipe or regenerative satellite network bypassing several routers in a terrestrial backbone.

4.4.2 Satellite IP QoS Architecture 2: DiffServ

A DiffServ enabled host in the DiffServ domain labels the traffic according to the service needed. Each type of service has an associated Service Level Agreement (SLA) between two entities. In DiffServ the ingress and egress nodes have the function of making sure that the traffic is compliant with SLA.

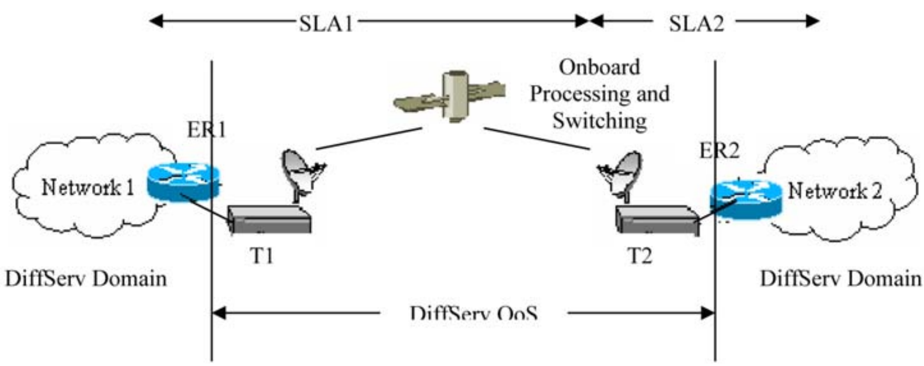


Figure 4.4: DiffServ QoS Architecture

Figure 4.4 shows an end-to-end QoS architecture for satellite IP network. This architecture assumes that the terminals T1 and T2 are DiffServ capable. There is a SLA between Network 1 and Network 2. The SLA allows the user to have its traffic labeled as highest priority within the Assured Forwarding class.

As an example, if the Network 2 is the Internet Service Provider (ISP) and T2 is the gateway, then a SLA exists between Network 1 and the gateway (SLA1) and another SLA exists between the gateway and the ISP (SLA2).

Non-compliant traffic can be treated in many ways. It can be dropped, service type changed, buffered or rate shaped into conformity. These functions are done by the edge router.

4.4.3 Satellite IP QoS Architecture: IntServ/DiffServ

Figure 4.5 shows the QoS architecture for satellite IP network, based on IntServ and DiffServ.

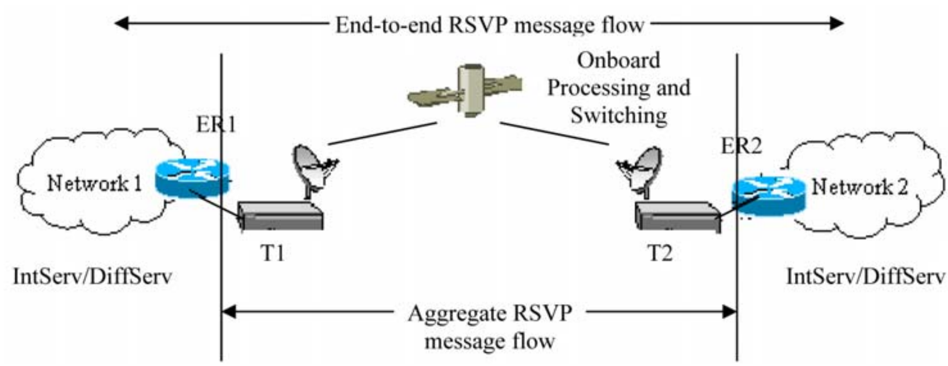


Figure 4.5: IntServ/DiffServ QoS Architecture

The terminals T1 and T2 are DiffServ capable and also Aggregate RSVP capable. IETF recently proposed the Aggregate RSVP. [260].

Aggregate RSVP is an extension to RSVP developed to enable reservation to be made for an aggregation of flows between edges of a network region, rather than for individual flows as supported by the version of RSVP. Aggregate RSVP is a protocol proposed to the aggregation of individual RSVP reservations that cross an "aggregation region" and share common ingress and egress routers with one RSVP reservation from ingress to egress.

The end-to-end RSVP messaging is performed between Network 1 and Network 2. The ingress and egress routers ER1 and ER2 dynamically create the aggregate reservations and classify the traffic to which the aggregate reservation applies. They also determine how much bandwidth is needed to achieve the requirement and recover the bandwidth when the individual reservations are no longer required. The end-to-end RSVP messages indicate what DSCP should be used for each flow.

When an end-to-end RSVP request cannot be met by the existing aggregate allocations between ER1 and ER2, aggregate RSVP is used to request new aggregation allocation or to modify the existing allocations.

Terminals T1 and T2 process the aggregate RSVP messages and perform the admission control.

Figure 4.6(a) shows the message sequence for end-to-end RSVP between ER1 and ER2.

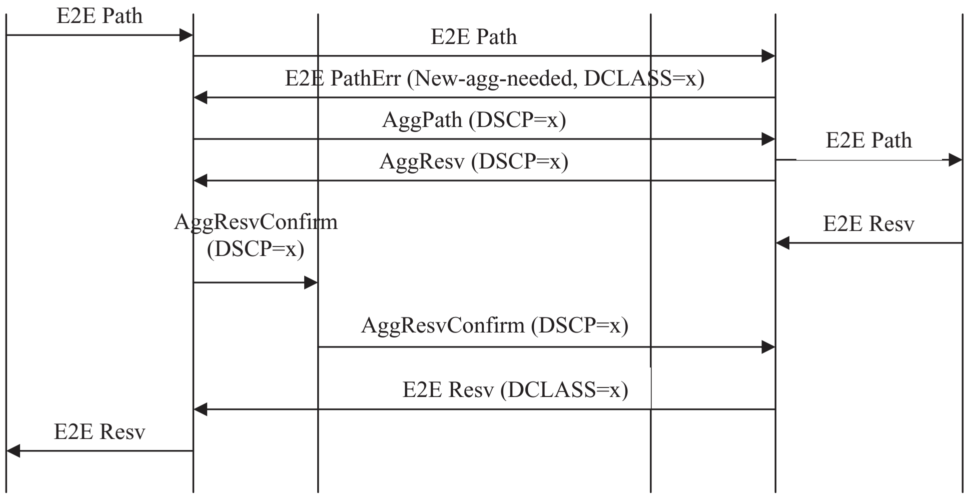


Figure 4.6(a): End-to-End RSVP Messaging

Figure 4.6(b) shows the messaging sequence for the aggregate allocation when sufficient bandwidth is available.

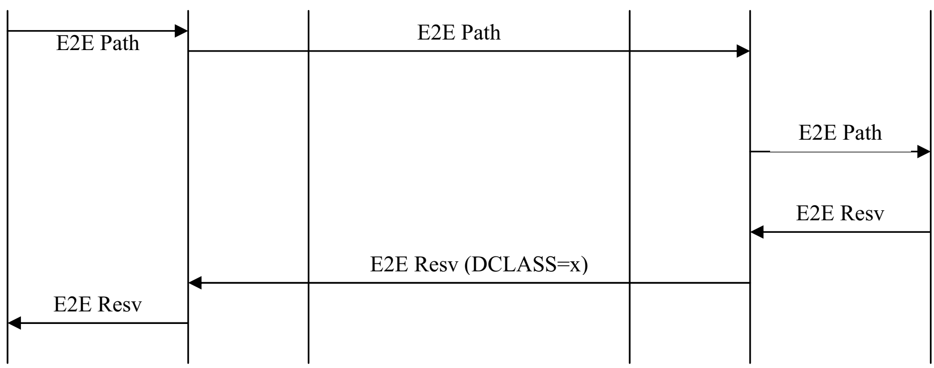


Figure 4.6(b): Aggregate Allocation – Sufficient Bandwidth

Figure 4.6(c) shows the message sequence for aggregate allocation when the bandwidth is insufficient.

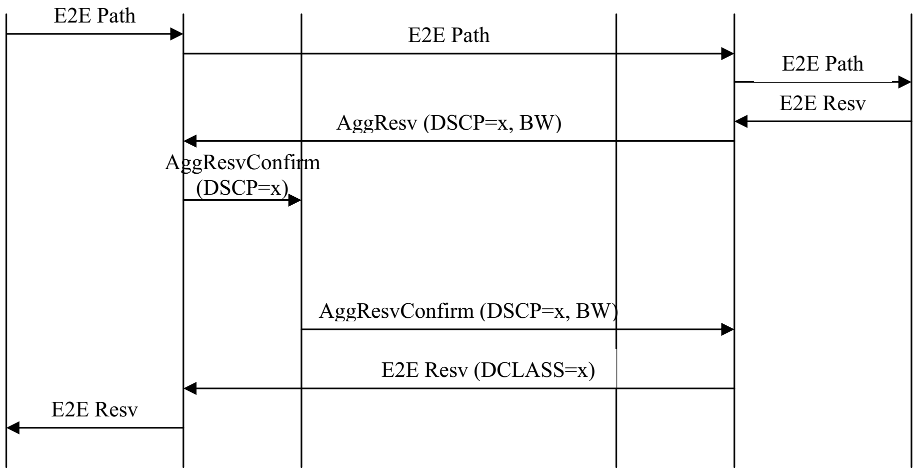


Figure 4.6(c): Aggregate Allocation – Insufficient Bandwidth

4.4.3.1 Access Network Example

In Figure 4.5 T2 is replaced by a gateway and Network 2 by an ISP.

- There exists a DiffServ SLA1 between T1 and gateway, and a different DiffServ SLA2 between gateway and Network 2.
- DiffServ flows arriving at the gateway from multiple networks are further aggregated to DiffServ service classes between the gateway and the ISP.
- Aggregate RSVP messaging between network 1 and the gateway controls bandwidth between T1 and the gateway (GW).
- A separate aggregate flow between GW and the ISP controls bandwidth between the GW and ISP.
- SLA between Network 1 and the GW can be changed dynamically based on RSVP signaling.

4.4.4 Discussion

Architectures 1 and 2 based on IntServ and DiffServ respectively are the simplest to implement at the terminals and gateways (for satellite access network). They provide end-to-end QoS.

Architecture 3 is a scalable architecture. It provides an end-to-end QoS with dynamic aggregate RSVP reservations. However, this aggregate RSVP has not been widely deployed in terrestrial environment. Considering the advantages of the aggregate RSVP as opposed to individual flow RSVP, this architecture 3 could be a compromise between architecture 1 which is entirely IntServ based and architecture 2 which is entirely DiffServ based.

4.5 DiffServ QoS Simulation for TCP and UDP over Satellite IP Network

Satellite based Internet service is receiving great interest by the users and service providers. Dynamic bandwidth allocation to provide application specific QoS levels for the user is an important issue [261]. However, guaranteed QoS support is still a technical challenge to be solved in supporting IP based satellite networks e.g. Spaceway, Teledesic.

There has been an increased interest in developing DiffServ architecture for provisioning IP QoS over satellite networks. DiffServ aims to provide scalable service differentiation in the Internet that can be used to permit differentiated pricing of Internet service [190, 191]. This differentiation may either be quantitative or relative. DiffServ is scalable as traffic classification and conditioning is performed only at network boundary nodes. The service to be received by a traffic is marked as a code point in the DS field in the IPv4 and IPv6 header. The DiffServ Code Point (DSCP) in the header of an IP packet is used to determine the Per-Hop Behavior (PHB), i.e. the forwarding treatment it will receive at a network node. Currently, formal specification is available for two PHBs – Assured Forwarding[193] and Expedited Forwarding [194].

In Expedited Forwarding (EF), a transit node uses policing and shaping mechanisms to ensure that the maximum arrival rate of a traffic aggregate is less than its minimum departure rate. At each transit node, the minimum departure rate of a traffic aggregate should be configurable and independent of other traffic at the node. Such a per-hop behavior results in minimum delay and jitter and can be used to provide an end-to-end ‘Virtual Leased Line’ type of service.

In Assured Forwarding (AF), IP packets are classified as belonging to one of four traffic classes. IP packets assigned to different traffic classes are forwarded independent of each other. Each traffic class is assigned a minimum configurable amount of resources (link bandwidth and buffer space). Resources not being currently used by another PHB or an AF traffic class can optionally be used by remaining classes. Within a traffic class, a packet is assigned one of three levels of drop precedence (green, yellow, red). In case of congestion, an AF compliant DiffServ node drops low precedence (red) packets in preference to higher precedence (green, yellow) packets.

The Differentiated Services approach is selected for satellite IP QoS due to the following:

- Satellite systems are designed to support aggregate traffic; scalable solutions are preferred.
- Simpler to implement DiffServ in an integrated satellite and terrestrial environment.

– DiffServ pushes the complexity to the edge, hence, the satellite networks with limited onboard processing and switching favor DiffServ approach.

In this section, a simulation model with a wide range of simulations, varying several factors to identify the significant ones influencing fair allocation of excess satellite network resources among congestion sensitive e.g. TCP and insensitive e.g. UDP flows are developed for GEO, MEO and LEO satellite networks. The factors that are studied in Section 4.5.1 include (a) number of drop precedences required (one, two or three), (b) percentage of reserved (highest drop precedence) traffic, (c) buffer management (Tail drop or Random Early Drop with different parameters), and (d) traffic types (TCP aggregates, UDP aggregates). Section 4.5.4 describes the simulation configuration and parameters and experimental design techniques. Section 4.5.5 describes Analysis of Variation (ANOVA) technique. Simulation results for TCP and UDP, for reserved rate utilization and fairness are also given. [256, 257, 258].

4.5.1 QoS Framework

The key factors that affect the satellite network performance are those relating to bandwidth management, buffer management, traffic types and their treatment, and network configuration.

Bandwidth Management

Bandwidth management relates to the algorithms and parameters that affect service (PHB) given to a particular aggregate. In particular, the number of drop precedence (one, two, or three) and the level of reserved traffic were identified as the key factors in this analysis.

Buffer Management

Buffer management relates to the method of selecting packets to be dropped when the buffers are full. Two commonly used methods are tail drop and random early drop (RED). Several variations of RED are possible in case of multiple drop precedence.

Traffic Types

Two traffic types that are considered here are TCP and UDP aggregates. TCP and UDP were separated out because of their different response to packet losses. In particular, it was a concern that if excess TCP and excess UDP were both given the same treatment, TCP flows will reduce their rates on packet drops while UDP flows will not change and get the entire excess bandwidth. The analysis shows that this is in fact the case and that it is important to give a better treatment to excess TCP than excess UDP.

Network Configuration

A simple configuration of GEO, MEO, and LEO satellite network configuration is assumed for the simulation study.

The following QoS issues are addressed in the simulation study:

- Three drop precedence (green, yellow, and red) help clearly distinguish between congestion sensitive and insensitive flows.
- The reserved bandwidth should not be overbooked, that is, the sum should be less than the bottleneck link capacity. If the network operates close to its capacity, three levels of drop precedence are redundant as there is not much excess bandwidth to be shared.
- The excess congestion sensitive (TCP) packets should be marked as yellow while the excess congestion insensitive (UDP) packets should be marked red.
- The RED parameters have significant effect on the performance. The optimal setting of RED parameters is an area for further research.

4.5.2 Buffer Management Classifications

Buffer management techniques help identify which packets should be dropped when the queues exceed a certain threshold. It is possible to place packets in one queue or multiple queues depending upon their color or flow type. For the threshold, it is possible to keep a single threshold on packets in all queues or to keep multiple thresholds. Thus, the accounting (queues) could be single or multiple and the threshold could be single or multiple. These choices lead to four classes of buffer management techniques:

- Single Accounting, Single Threshold (SAST)
- Single Accounting, Multiple Threshold (SAMT)
- Multiple Accounting, Single Threshold (MAST)
- Multiple Accounting, Multiple Threshold (MAMT)

Random Early Discard (RED) is a well known and now commonly implemented packet drop policy. It has been shown that RED performs better and provides better fairness than the tail drop policy. In RED, the drop probability of a packet depends on the average queue length which is an exponential average of instantaneous queue length at the time of the packet's arrival [262]. The drop probability increases linearly from 0 to max_p as average queue length increases from min_th to max_th. With packets of multiple colors, one can calculate average queue length in many ways and have multiple sets of drop thresholds for packets of different colors. In general, with multiple colors, RED policy can be implemented as a variant of one of four general categories: SAST, SAMT, MAST, and MAMT.

Single Average Single Threshold RED has a single average queue length and same min_th and max_th thresholds for packets of all colors. Such a policy does not distinguish between packets of different colors and can also be called color blind RED. In Single Average Multiple Thresholds RED, average queue length is based on total number of packets in the queue irrespective of their color. However, packets of different colors have different drop thresholds. For example, if maximum queue size is 60 packets, the drop thresholds for green, yellow and red packets can be {40/60, 20/40, 0/10}. In these simulations, Single Average Multiple Thresholds RED is used.

In Multiple Average Single/Multiple Threshold RED, average queue length for packets of different colors is calculated differently. For example, average queue length for a color can be calculated using number of packets in the queue with same or better color [191]. In such a scheme, average queue length for green, yellow and red packets will be calculated using number of green, yellow + green, red + yellow + green packets in the queue respectively. Another possible scheme is where average queue length for a color is calculated using number of packets of that color in the queue [176]. In such a case, average queue length for green, yellow and red packets will be calculated using number of green, yellow and red packets in the queue respectively. Multiple Average Single Threshold RED will have same drop thresholds for packets of all colors whereas Multiple Average Multiple Threshold RED will have different drop thresholds for packets of different colors.

4.5.3 Simulation Configuration and Parameters

Figure 4.7 shows the network configuration for simulations. The configuration consists of customers 1 through 10 sending data over the link between Routers 1, 2 and using the same AF traffic class. Router 1 is located in a satellite ground station. Router 2 is located in a GEO, MEO or LEO satellite and Router 3 is located in destination ground station. Traffic is one-dimensional with only ACKs coming back from the other side. Customers 1 through 9 carry an aggregated traffic coming from 5 Reno TCP sources each. Customer 10 gets its traffic from a single UDP source sending data at a rate of 1.28 Mbps. Common configuration parameters are detailed in Tables 4.4, 4.5 and 4.6. All TCP and UDP packets are marked green at the source before being 'recolored' by a traffic conditioner at the customer site. The traffic conditioner consists of two 'leaky' buckets (green and yellow) that mark packets according to their token generation rates (called reserved/green and yellow rate). In two-color simulations, yellow rate of all customers is set to zero. Thus, in two-color simulations, both UDP and TCP packets will be colored either green or red. In three-color simulations, customer 10 (the UDP customer) always has a yellow rate of 0. Thus, in three-color simulations, TCP packets coming from customers 1 through 9 can be colored green, yellow or red and UDP packets coming from customer 10 will be colored green or red. All the traffic coming to Router 1 passes through a Random Early Drop (RED) queue. The RED policy implemented at Router 1 can be classified as Single Average Multiple Threshold RED as explained in the following paragraphs.

Many of the commercial simulation tools available do not support the QoS models as of today. Hence, NS simulator version 2.1 has been used for these simulations. The code has been modified to implement the traffic conditioner and multi-color RED (RED_n). [263]

4.5.3.1 GEO Simulation Configuration

In this study, full factorial simulations involving many factors are performed:

– *Green Traffic Rates:* Green traffic rate is the token generation rate of green bucket in the traffic conditioner. Green rates of 12.8, 25.6, 38.4 and 76.8 kbps per customer have been experimented with. These rates correspond to a total of 8.5%, 17.1%, 25.6% and 51.2% of network capacity (1.5 Mbps). In order to understand the effect of green traffic rate, simulations with green rates of 102.4, 128, 153.6 and 179.2 kbps for two color cases have been conducted. These rates correspond to 68.3%, 85.3%, 102.4% and 119.5% of network capacity respectively. Note that in last two cases, the available network bandwidth has been oversubscribed.

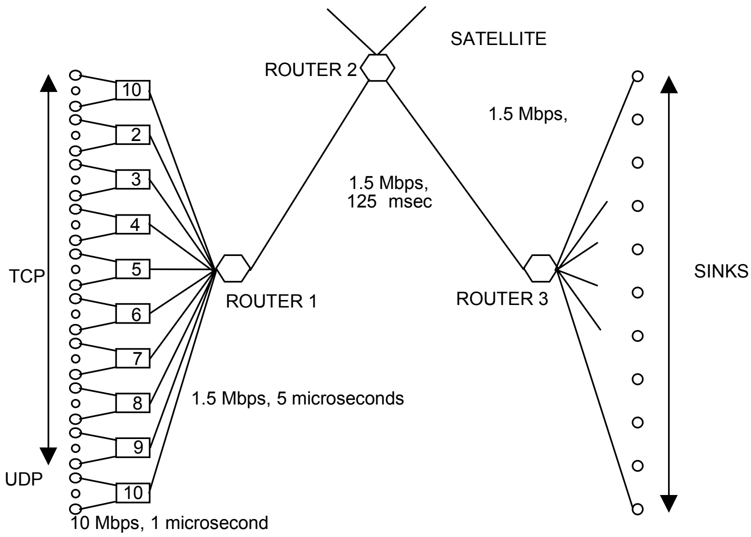


Figure 4.7: Simulation Configuration

– *Green Bucket Size:* 1, 2, 4, 8, 16 and 32 packets of 576 bytes each.
– *Yellow Traffic Rate* (only for three-color simulations): Yellow traffic rate is the token generation rate of yellow bucket in the traffic conditioner. Yellow rates of 12.8 and 128 kbps per customer have been experimented with. These rates correspond to 7.7% and 77% of total capacity (1.5 Mbps) respectively. A high yellow rate of 128 kbps has been used so that all excess (out of green rate) TCP packets are colored yellow and thus can be distinguished from excess UDP packets that are colored red.
– *Yellow Bucket Size* (only for three-color simulations): 1, 2, 4, 8, 16, 32 packets of 576 bytes each.
– *Maximum Drop Probability:* Maximum drop probability values used in the simulations are listed in Tables 4.5 and 4.5.

Table 4.4: Simulation Configuration Parameters

Simulation Time	100 seconds
TCP Window	64 packets
IP Packet Size	576 bytes
UDP Rate	1.28 Mbps
Maximum queue size (for all queues)	60 packets
Link between Router 1 and Router 2:	
Link bandwidth	1.5 Mbps
One way delay	125 milliseconds
Drop policy	From Router 1: RED_n To Router 1: DropTail
Link between Router 2 to Router 3:	
Link bandwidth	1.5 Mbps
One way delay	125 milliseconds
Drop policy	DropTail
Link between Router 3 and Sinks:	
Link bandwidth	1.5 Mbps
One way delay	5 microseconds
Drop policy	DropTail
Link between UDP/TCPs and Customers:	
Link bandwidth	10 Mbps
One way delay	1 microsecond
Drop policy	DropTail
Link between Customers & Router 1:	
Link bandwidth	1.5 Mbps
One way delay	5 microseconds
Drop policy	DropTail

Table 4.5: Two-Color Simulation Parameters

Simulation ID	Green Rate (kbps)	Max Drop Probability {Green, Red}	Drop Thresholds {Green, Red}	Green Bucket (in Packets)
1-144	12.8	{0.1, 0.1}	{40/60, 0/10}	1
201-344	25.6	{0.1, 0.5}	{40/60, 0/20}	16
401-544	38.4	{0.5, 0.5}	{40/60, 0/5}	2
601-744	76.8	{0.5, 1}	{40/60, 20/40}	32
801-944	102.4	{1, 1}		4
1001-1144	128			8
1201-1344	153.6			
1401-1544	179.2			

Table 4.6: Three-Color Simulation Parameters

Simulation ID	Green Rate (kbps)	Maximum Drop Probability {Green, Yellow, Red}	Drop Thresholds {Green, Yellow, Red}	Yellow Rate (kbps)	Bucket Size	
					Green	Yellow
1-720	12.8	{0.1, 0.5, 1}	{40/60, 20/40, 0/10}	128	16	1
1001-1720	25.6	{0.1, 1, 1}	{40/60, 20/40, 0/20}	12.8	1	16
2001-2720	38.4	{0.5, 0.5, 1}			2	2
3001-3720	76.8	{0.5, 1, 1}			32	32
		{1, 1, 1}			4	4
					8	8

– *Drop Thresholds* for red colored packets: The network resources allocated to red colored packets and hence the fairness results depend on the drop thresholds for red packets. Different values of drop thresholds have been experimented with for red colored packets so as to achieve close to best fairness possible. Drop thresholds for green packets have been fixed at {40,60} for both two and three color simulations. For three-color simulations, yellow packet drop thresholds are {20,40}.

In these simulations, size of all queues is 60 packets of 576 bytes each. The queue weight used to calculate RED average queue length is 0.002. For easy reference, an identification number has been given to each simulation as shown in Tables 4.5 and 4.6. The simulation results are analysed using ANOVA techniques [238] briefly described in the following paragraphs.

4.5.3.2 Performance Metrics

Simulation results have been evaluated based on utilization of reserved rates by the customers and the fairness achieved in allocation of excess bandwidth among different customers.

Utilization of reserved rate by a customer is measured as the ratio of green throughput of the customer and the reserved rate. Green throughput of a customer is determined by the number of green colored packets received at the traffic destination(s). Since in these simulations, the drop thresholds for green packets are kept very high in the RED queue at Router 1, chances of a green packet getting dropped are minimal and ideally green throughput of a customer should equal its reserved rate.

The fairness in allocation of excess bandwidth among n customers sharing a link can be computed using the following formula given in Section 3.4.3.2:

$$\mathit{Fairness\ Index} = \frac{\left(\sum x_i\right)^2}{n \times \sum (x_i^2)}$$

Where x_i is the excess throughput of the *i*th customer. Excess throughput of a customer is determined by the number of yellow and red packets received at the traffic destination(s).

4.5.4 GEO Satellite Network Simulation Results

Simulation results of two and three color simulations are shown in Figure 4.8. In this figure, a simulation is identified by its Simulation ID listed in Tables 4.5 and 4.6. Figure 4.8 shows the fairness achieved in allocation of excess bandwidth among ten customers for two-colors in GEO architecture [264, 265]. Figure 4.9 shows simulation results of fairness achieved for GEO networks for three colors with different reserved rates. It is clear from Figures 4.8 and 4.9 that fairness is not good in two-color simulations. With three colors, there is a wide variation in fairness results with best results being close to 1. Note that fairness is zero in some of the two color simulations. In these simulations, total reserved traffic uses all the bandwidth and there is no excess bandwidth available to share. As shown, there is a wide variation in reserved rate utilization by customers in two and three color simulations.

4.5.4.1 Fairness - Two-Color Vs Three-Color

Table 4.7: Main Factors Influencing Fairness Results in Three Color Simulations

Factor/Interaction	Allocation of Variation (in %age)
Yellow Rate	41.36
Yellow Bucket Size	28.95
Interaction between Yellow Rate and Yellow Bucket Size	26.49

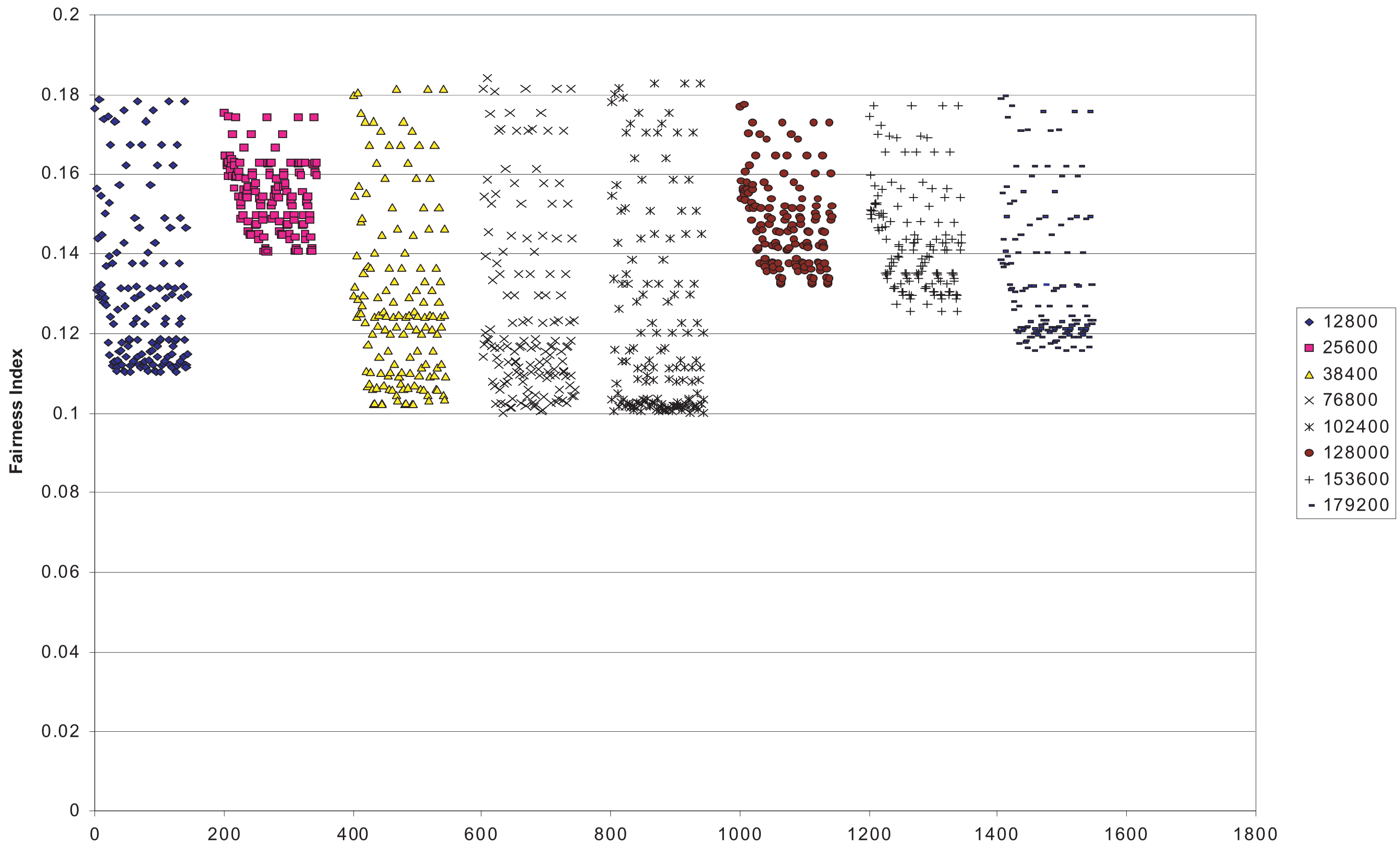


Figure 4.8: GEO Simulation Results: Fairness Achieved in Two-Color Simulations with Different Reserved Rates

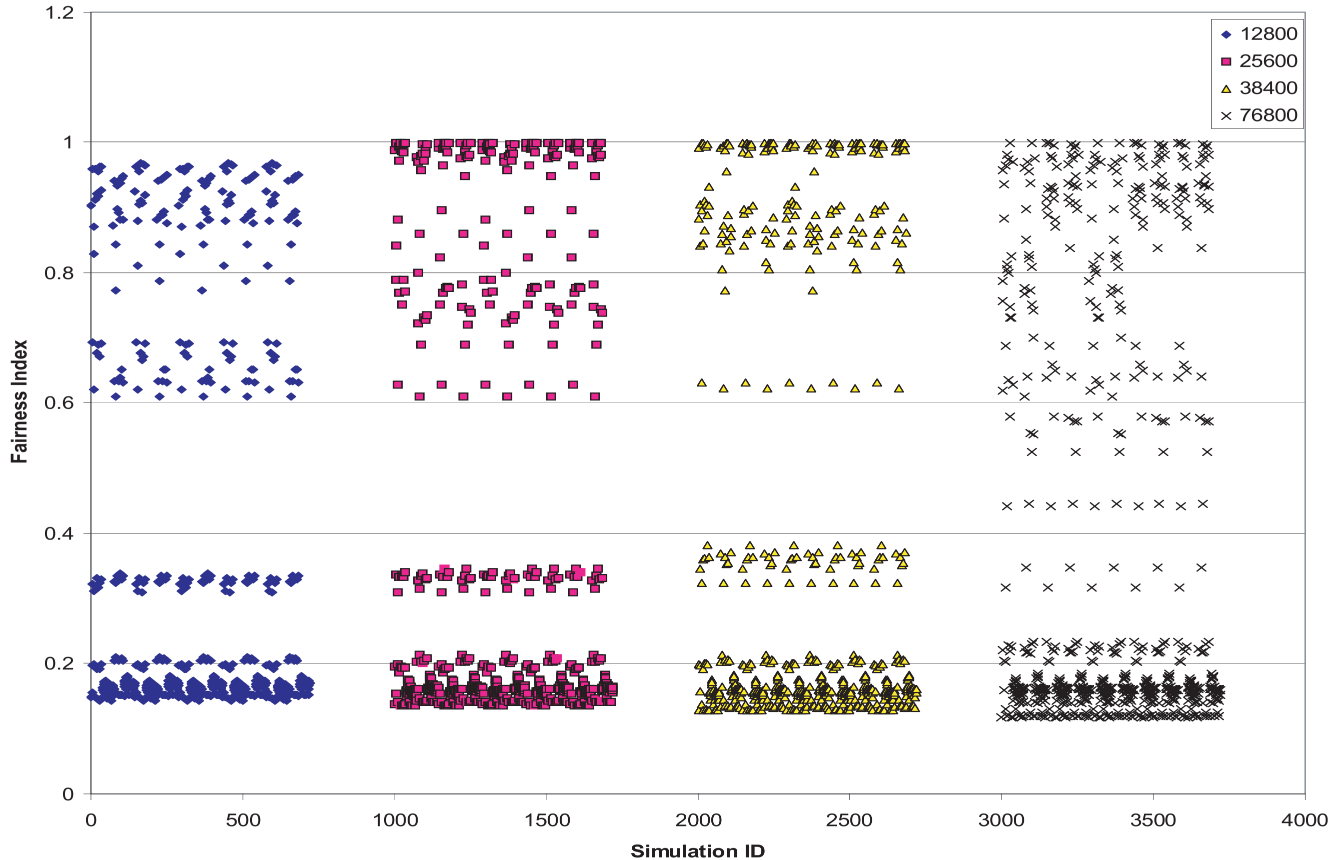


Figure 4.9: GEO Simulation Results: Fairness Achieved in Three-Color Simulations with Different Reserved Rates

4.5.4.2 Reserved Rate – Two-Color Vs Three-Color

Figure 4.10 shows reserved rate utilization by TCP customers in two color for GEO networks. Figure 4.12 shows reserved rate utilization for TCP customers in three colors, for GEO. For TCP customers, the average reserved rate utilization in each simulation has been plotted. Note that in some cases, reserved rate utilization is slightly more than one. This is because token buckets are initially full which results in all packets getting green color in the beginning. Figures 4.11 and 4.13, show that UDP customers have good reserved rate utilization in almost all cases. In contrast, TCP customers show a wide variation in reserved rate utilization.

In order to determine the influence of different simulation factors on the reserved rate utilization and fairness achieved in excess bandwidth distribution, simulation results are analyzed statistically using Analysis of Variation (ANOVA) technique. A brief introduction to ANOVA technique used in the analysis is provided in Section 4.5. In later paragraphs, the results of statistical analysis of two- and three-color simulations are presented.

Table 4.8: Main Factors Influencing Reserved Rate Utilization Results

Factor/Interaction	Allocation of Variation (in %age)			
	2 Colors		3 Colors	
	TCP	UDP	TCP	UDP
Green Rate	1.60	15.65	2.21	20.40
Green Bucket Size	97.51	69.13	95.24	62.45
Green Rate - Green Bucket Size	0.59	13.45	1.96	17.11

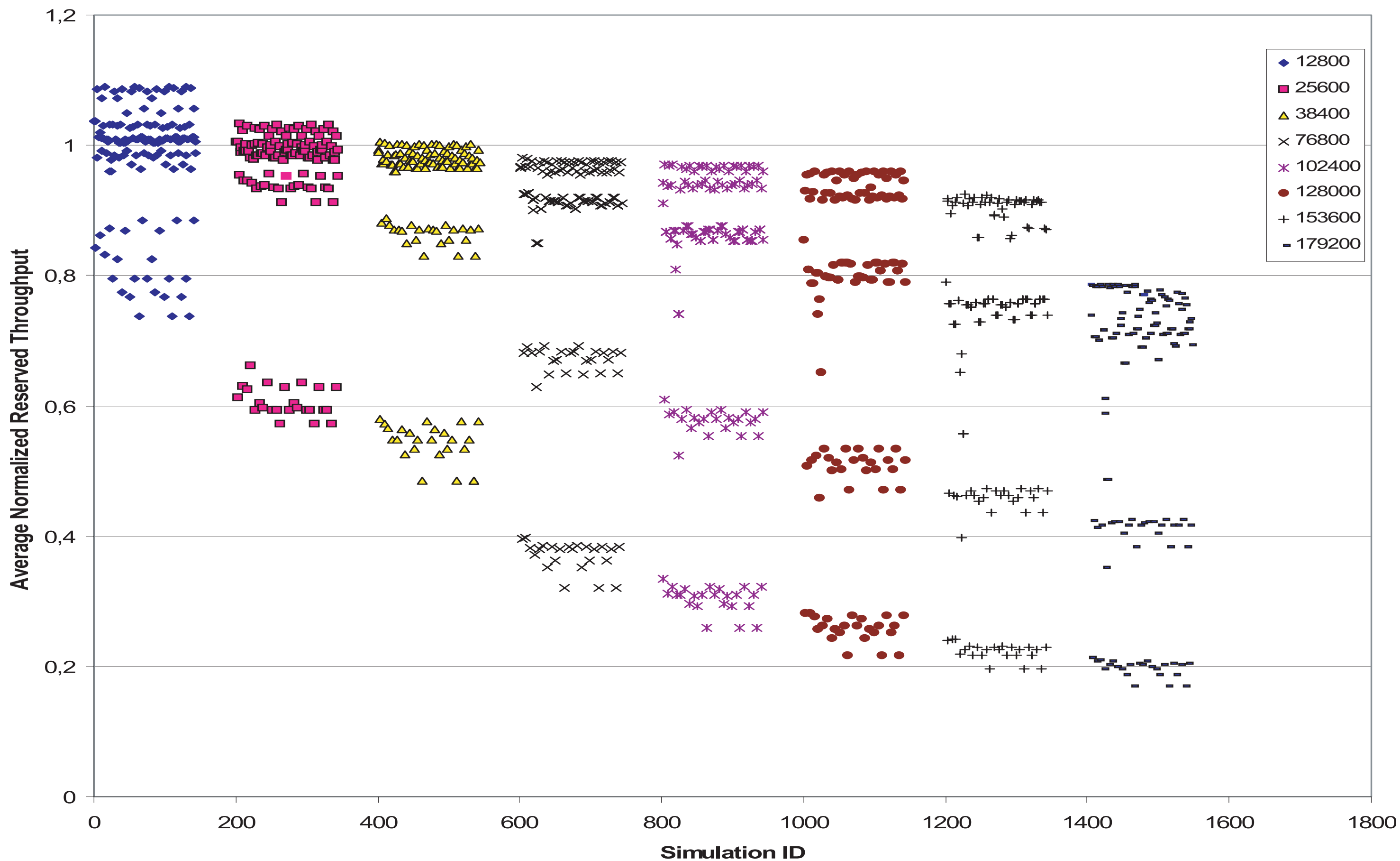


Figure 4.10: GEO Simulation Results: Reserved Rate Utilization by TCP Customers in Two-Color Simulations

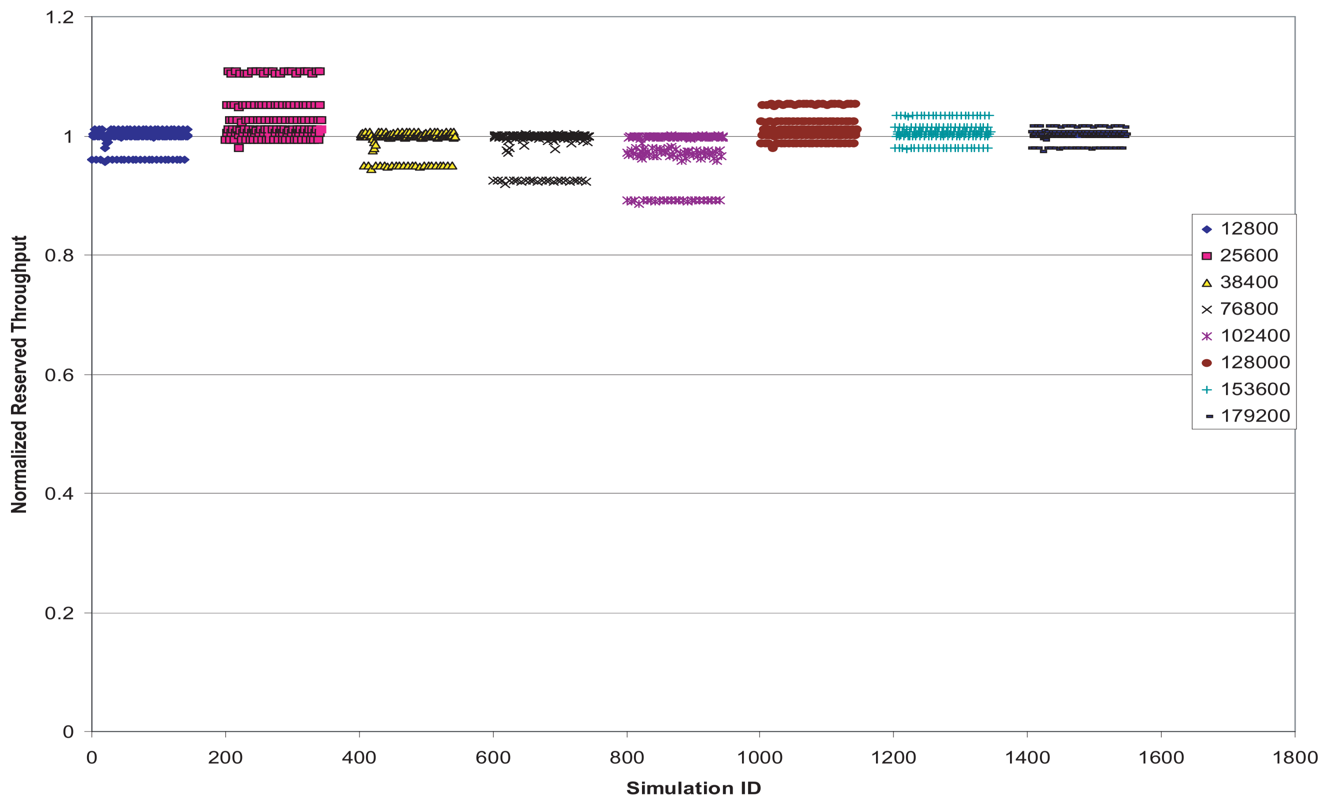


Figure 4.11: GEO Simulation Results: Reserved Rate Utilization by UDP Customers in Two-Color Simulations

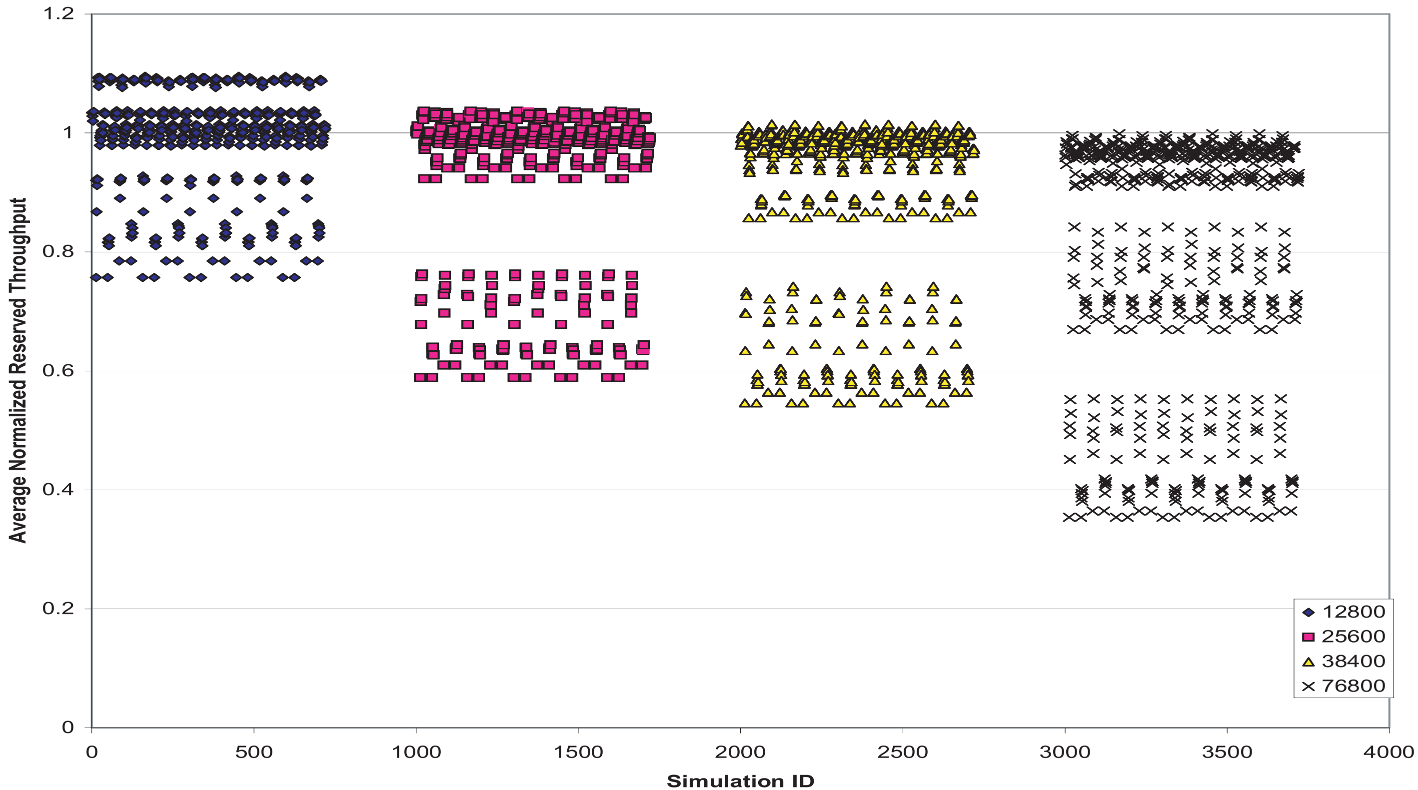


Figure 4.12: GEO Simulation Results: Reserved Rate Utilization by TCP Customers in Three-Color Simulations

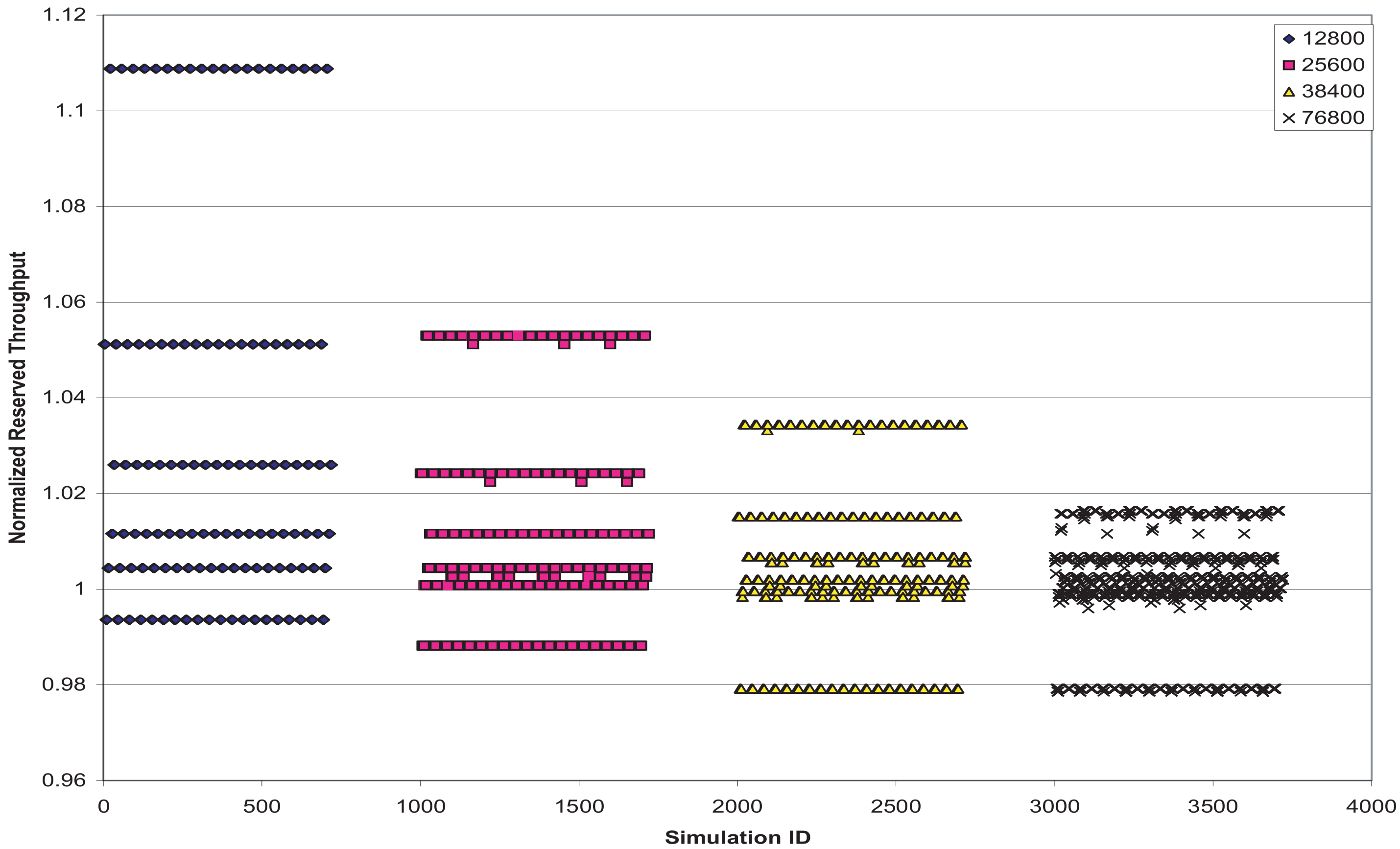


Figure 4.13: GEO Simulation Results: Reserved Rate Utilization by UDP Customers in Three-Color Simulations

4.5.5 Analysis Of Variation (ANOVA) Technique

The results of a simulation are affected by the values (or levels) of simulation factors (e.g. green rate) and the interactions between levels of different factors (e.g. green rate and green bucket size). The simulation factors and their levels used in this simulation study are listed in Tables 4.5 and 4.6. Analysis of Variation of simulation results is a statistical technique used to quantify these effects. In this section, a brief account of Analysis of Variation technique is presented. More details can be found in [238].

Analysis of Variation involves calculating the Total Variation in simulation results around the Overall Mean and doing Allocation of Variation to contributing factors and their interactions. Following steps describe the calculations:

1. Calculate the *Overall Mean* of all the values.
2. Calculate the individual effect of each level a of factor A, called the *Main Effect* of a:

   $Main\ Effect_a = Mean_a - Overall\ Mean$

   where, $Main\ Effect_a$ is the main effect of level a of factor A, $Mean_a$ is the mean of all results with a as the value for factor A.

   The main effects are calculated for each level of each factor.

3. Calculate the *First Order Interaction* between levels a and b of two factors A and B respectively for all such pairs:

   $Interaction_{a,b} = Mean_{a,b} - (Overall\ Mean + Main\ Effect_a + Main\ Effect_b)$

   where, $Interaction_{a,b}$ is the interaction between levels a and b of factors A and B respectively, $Mean_{a,b}$ is mean of all results with a and b as values for factors A and B, $Main\ Effect_a$ and $Main\ Effect_b$ are main effects of levels a and b respectively.

4. Calculate the *Total Variation* as shown below:

   $Total\ Variation = \sum(result^2) - (Num\_Sims) \times (Overall\ Mean^2)$

   where, $\sum(result^2)$ is the sum of squares of all individual results and Num_Sims is total number of simulations.

5. The next step is the *Allocation of Variation* to individual main effects and first order interactions. To calculate the variation caused by a factor A, the sum of squares of the main effects of all levels of A is taken and multiplied with the number of experiments conducted with each level of A. To calculate the variation caused by first order interaction between two factors A and B, the sum of squares of all the first-order interactions between levels of A and B is taken and multiplied with the number of experiments conducted with each combination of levels of A and B. The allocation of variation for each factor and first order interaction between every pair of factors are calculated.

4.5.5.1 ANOVA Analysis for Reserved Rate Utilization

Table 4.8 shows the Allocation of Variation to contributing factors for reserved rate utilization. As shown in Figures 4.11 and 4.13, reserved rate utilization of UDP customers is almost always good for both two and three color simulations. However, in spite of very low probability of a green packet getting dropped in the network, TCP customers are not able to fully utilize their reserved rate in all cases. The little variation in reserved rate utilization for UDP customers is explained largely by bucket size. Large bucket size means that more packets will get green color in the beginning of the simulation when green bucket is full. Green rate and interaction between green rate and bucket size explain a substantial part of the variation. This is because the definition of rate utilization metric has reserved rate in denominator. Thus, the part of the utilization coming from initially full bucket gets more weight for low reserved rate than for high reserved rates. Also, in two color simulations for reserved rates 153.6 kbps and 179.2 kbps, the network is oversubscribed and hence in some cases UDP customer has a reserved rate utilization lower than one. For TCP customers, green bucket size is the main factor in determining reserved rate utilization. TCP traffic because of its bursty nature is not able to fully utilize its reserved rate unless bucket size is sufficiently high. In the current simulations, UDP customer sends data at a uniform rate of 1.28 Mbps and hence is able to fully utilize its reserved rate even when bucket size is low. However, TCP customers can have very poor utilization of reserved rate if bucket size is not sufficient. The minimum size of the leaky bucket required to fully utilize the token generation rate depends on the burstiness of the traffic.

4.5.5.2 ANOVA Analysis for Fairness

Fairness results shown in Figure 4.8 indicate that fairness in allocation of excess network bandwidth is very poor in two color simulations. With two colors, excess traffic of TCP as well as UDP customers is marked red and hence is given same treatment in the network. Congestion sensitive TCP flows reduce their data rate in response to congestion created by UDP flow. However, UDP flow keeps on sending data at the same rate as before. Thus, UDP flow gets most of the excess bandwidth and the fairness is poor. In three-color simulations, fairness results vary widely with fairness being good in many cases. Table 4.7 shows the important factors influencing fairness in three-color simulations as determined by ANOVA analysis. Yellow rate is the most important factor in determining fairness in three-color simulations. With three colors, excess TCP traffic can be colored yellow and thus distinguished from excess UDP traffic which is colored red. Network can protect congestion sensitive TCP traffic from congestion insensitive UDP traffic by giving better treatment to yellow packets than to red packets. Treatment given to yellow and red packets in the RED queues depends on RED parameters (drop thresholds and max drop probability values) for yellow and red packets. Fairness can be achieved by coloring excess TCP packets as yellow and setting the RED parameter values for packets of different colors correctly. In these simulations, yellow rates of 12.8 kbps and 128 kbps are experimented with. With a yellow rate of 12.8 kbps, only a fraction of excess TCP packets can be colored yellow at the traffic conditioner and thus resulting fairness in excess bandwidth distribution is not good. However with a yellow rate of 128 kbps, all excess TCP packets are colored yellow and good fairness is achieved with correct setting of RED parameters. Yellow bucket size also explains a substantial portion of variation in fairness results for three color simulations. This is because bursty TCP traffic can fully utilize its yellow rate only if yellow bucket size is sufficiently high. The interaction between yellow rate and yellow bucket size for three color fairness results is because of the fact that minimum size of the yellow bucket required for fully utilizing the yellow rate increases with yellow rate.

It is evident that three colors are required to enable TCP flows get a fairshare of excess network resources. Excess TCP and UDP packets should be colored differently and network should treat them in such a manner so as to achieve fairness. Also, size of token buckets should be sufficiently high so that bursty TCP traffic can fully utilize the token generation rates.

4.5.6 MEO Satellite IP Network Simulation Results

4.5.6.1 Simulation Configuration

Figure 4.14 shows the simulation configuration for MEO satellite IP QoS model. Table 4.9 provides the configuration parameters. [258]

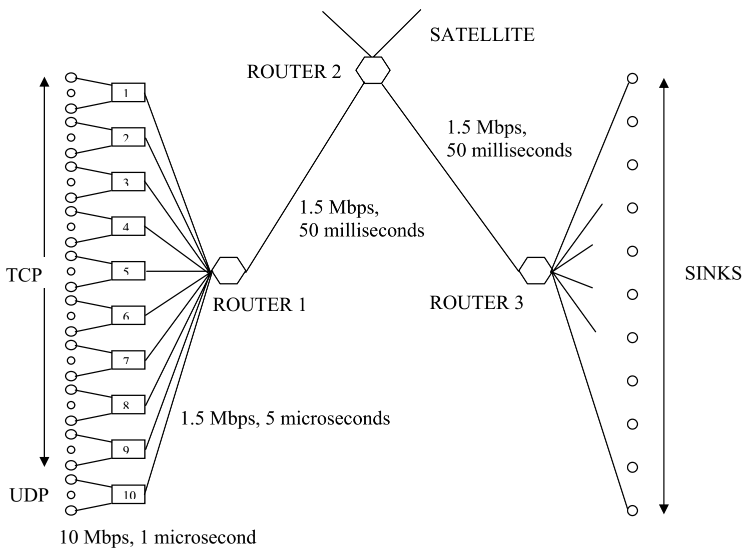


Figure 4.14: Simulation Configuration of MEO Satellite

Table 4.9: MEO: Simulation Configuration Parameters

Simulation Time	100 seconds
TCP Window	64 packets
IP Packet Size	576 bytes
UDP Rate	1.28 Mbps
Maximum queue size (for all queues)	60 packets
Link between Router 1 and Router 2:	
Link bandwidth	1.5 Mbps
One way delay	50 milliseconds
Drop policy	From Router 1: RED_n To Router 1: DropTail
Link between Router 2 to Router 3:	
Link bandwidth	1.5 Mbps
One way delay	50 milliseconds
Drop policy	DropTail
Link between Router 3 and Sinks:	
Link bandwidth	1.5 Mbps
One way delay	5 microseconds
Drop policy	DropTail
Link between UDP/TCPs and Customers:	
Link bandwidth	10 Mbps
One way delay	1 microsecond
Drop policy	DropTail
Link between Customers & Router 1:	
Link bandwidth	1.5 Mbps
One way delay	5 microseconds
Drop policy	DropTail

4.5.6.2 Fairness – Two-Color Vs Three-Color

Figures 4.15 and 4.16 show the fairness results in two colors and three colors for MEO satellite network.

Table 4.10: MEO: Main Factors Influencing Fairness Results

Factor/Interaction	Allocation of Variation (in %age)
Yellow Rate	50.18
Yellow Bucket Size	24.57
Interaction between Yellow Rate and Yellow Bucket Size	21.92

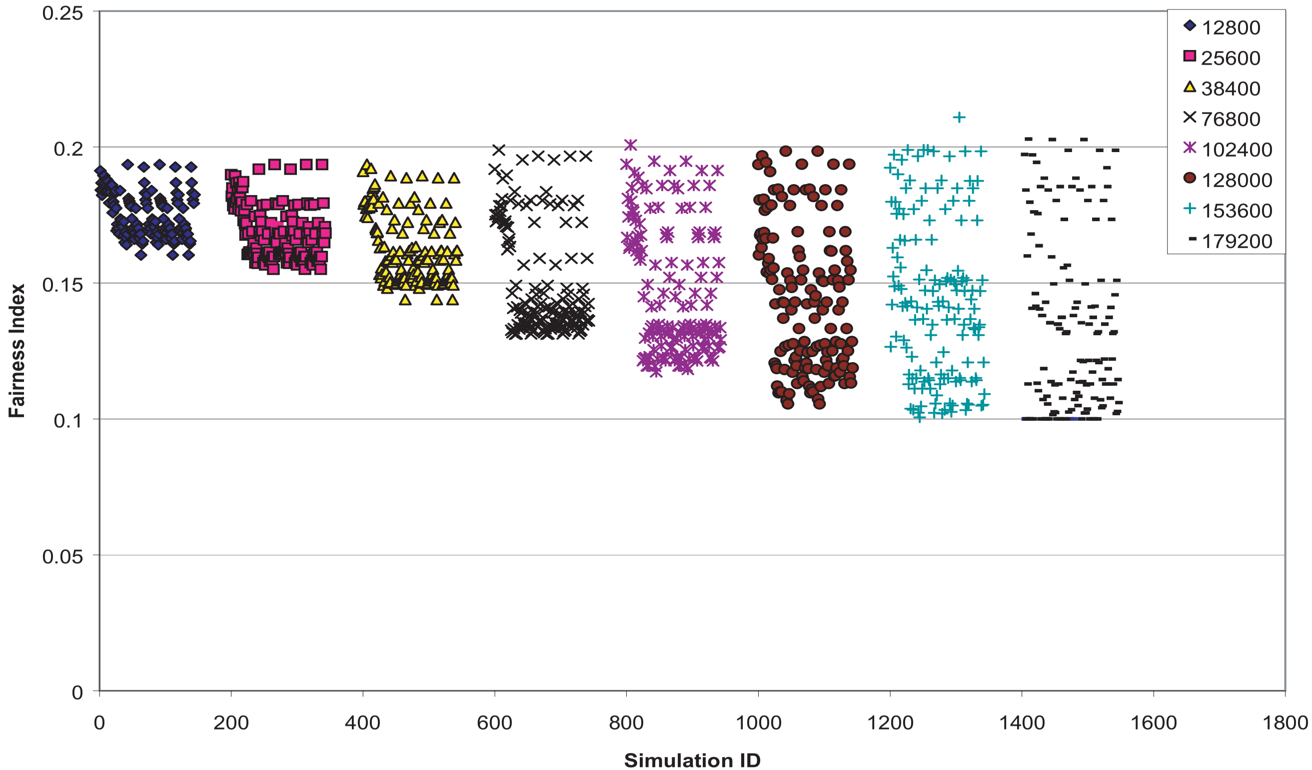


Figure 4.15: MEO Simulation Results: Fairness achieved in Two-Color Simulations with Different Reserved Rates

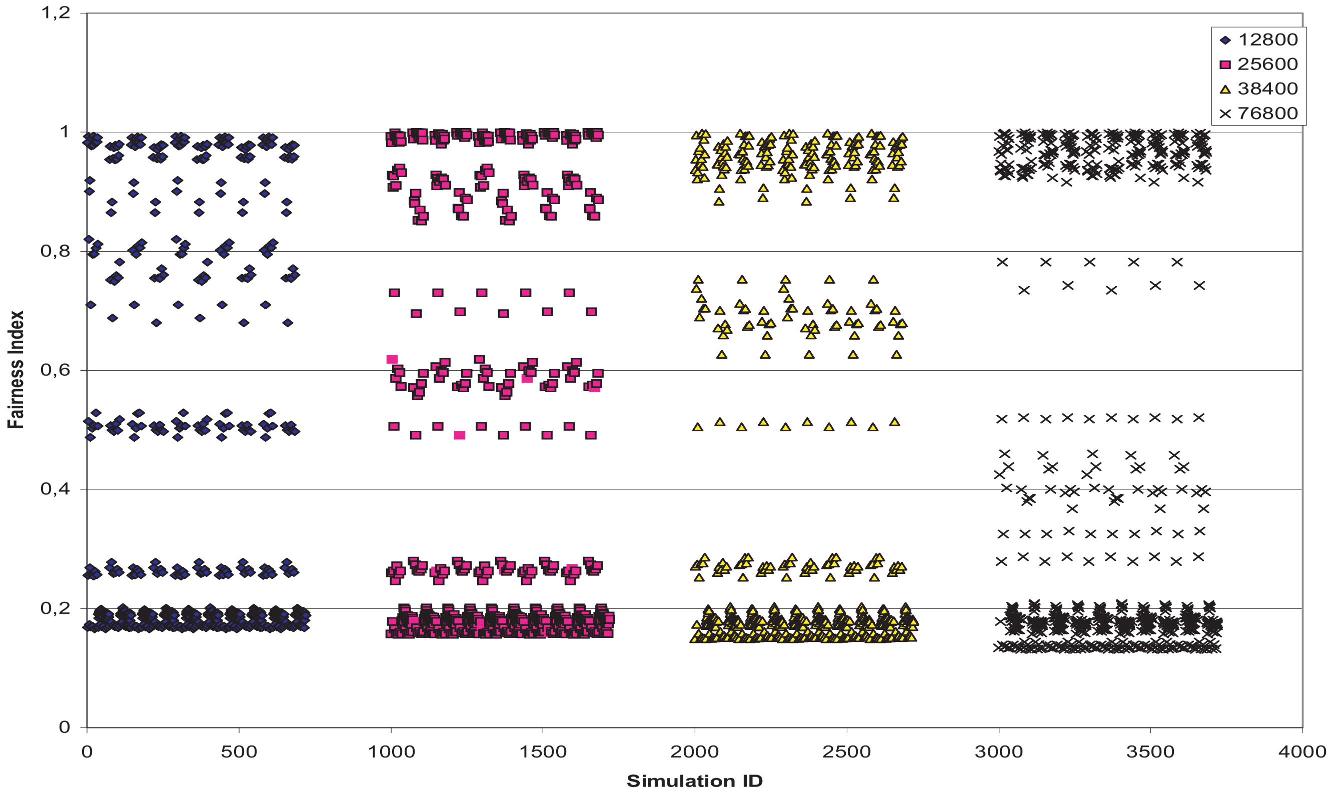


Figure 4.16: MEO Simulation Results: Fairness achieved in Three-Color Simulations with Different Reserved Rates

4.5.6.3 Reserved Rate Utilization – Two-Color Vs Three-Color

Figures 4.17-4.20 show the reserved rate utilization results in two colors and three colors for MEO satellite network.

Table 4.11: MEO: Main Factors Influencing Reserved Rate Utilization Results

Factor/Interaction	Allocation of Variation (in %age)			
	2 Colors		3 Colors	
	TCP	UDP	TCP	UDP
Green Rate	4.25	37.32	1.12	18.99
Green Bucket Size	93.90	35.04	96.72	63.64
Green Rate - Green Bucket Size	1.60	27.10	0.58	17.31

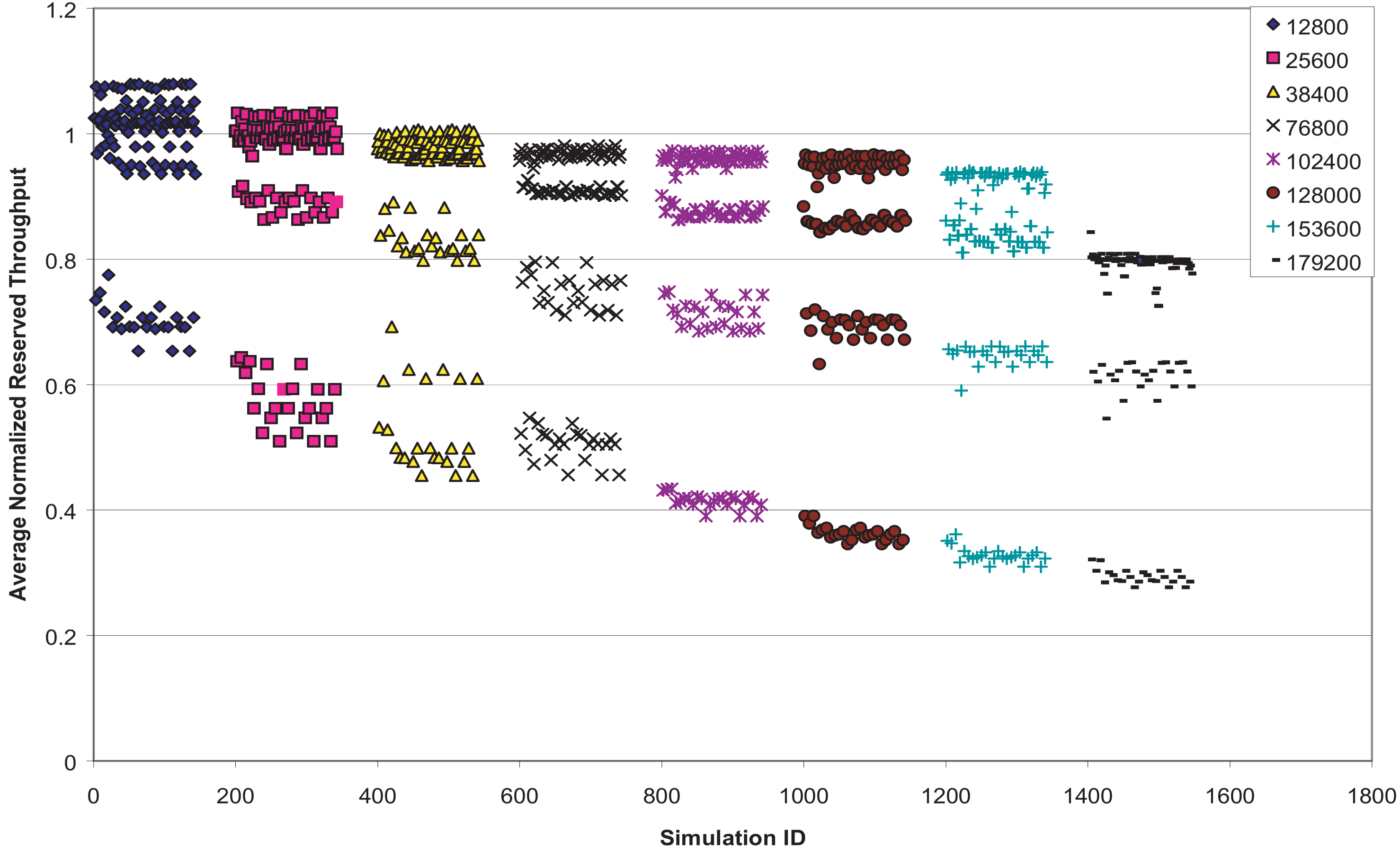


Figure 4.17: MEO Simulation Results: Reserved Rate Utilization by TCP Customers in Two-Color Simulations

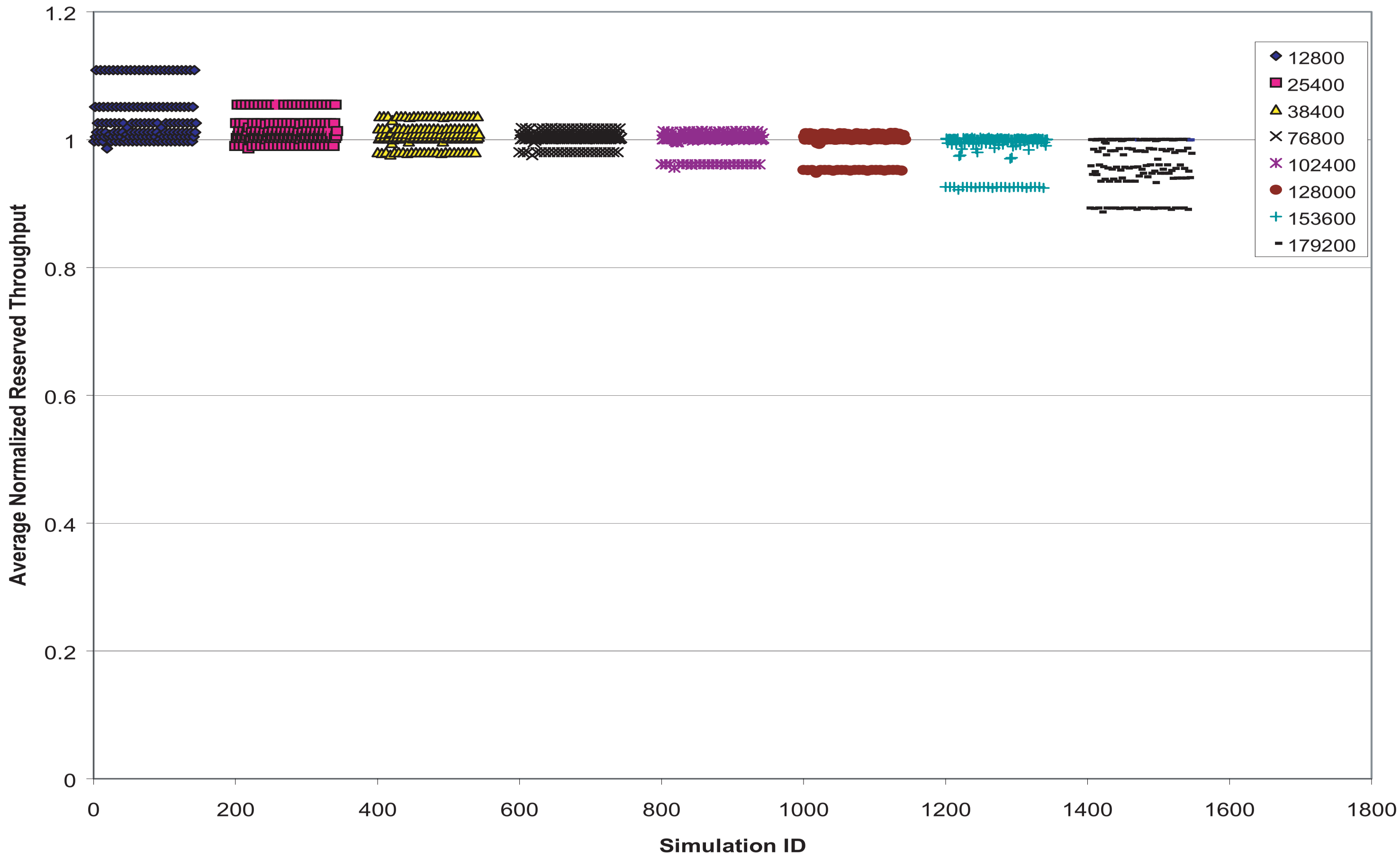


Figure 4.18: MEO Simulation Results: Reserved Rate Utilization by UDP Customers in Two-Color Simulations

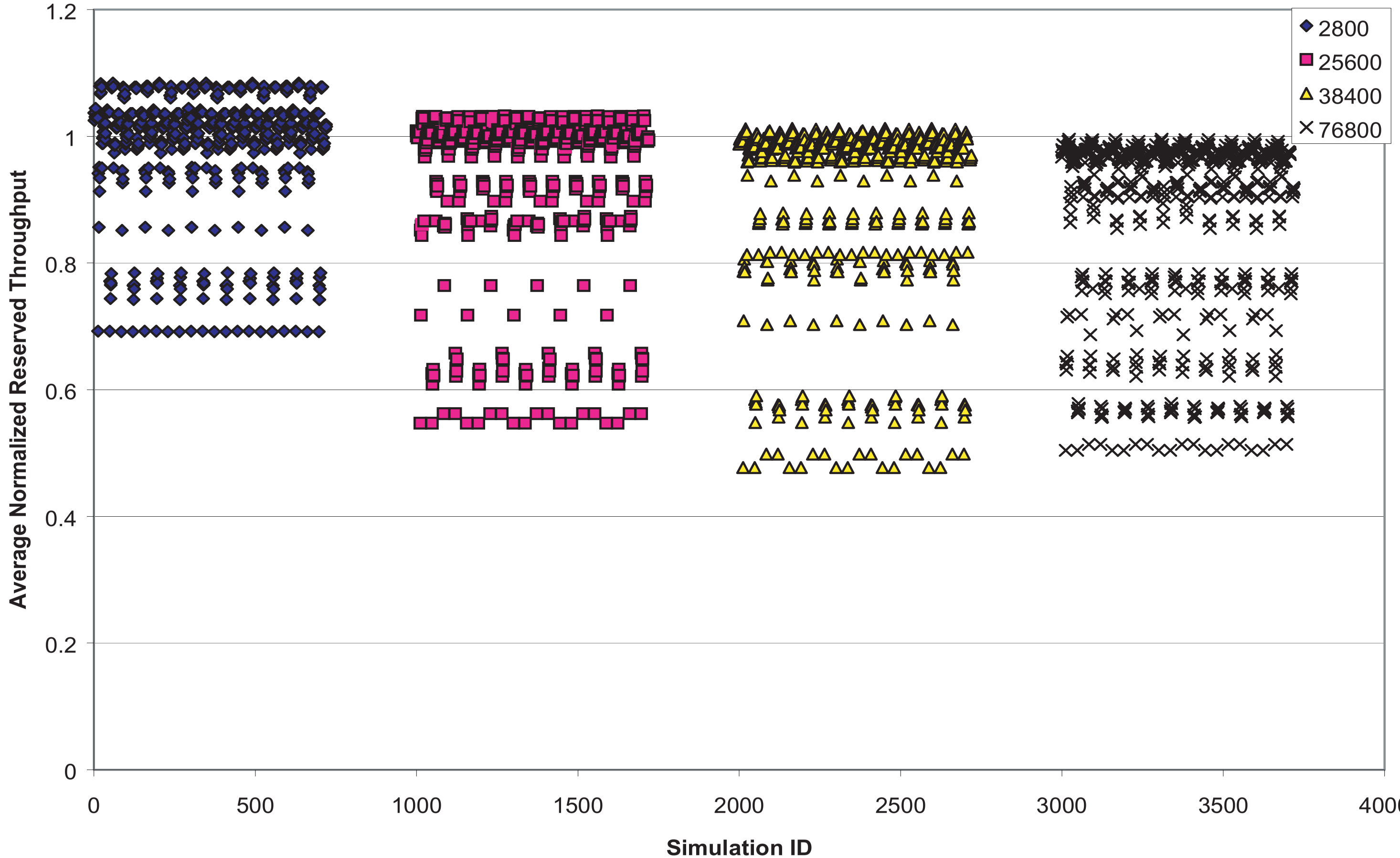


Figure 4.19: MEO Simulation Results: Reserved Rate Utilization by TCP Customers in Three-Color Simulations

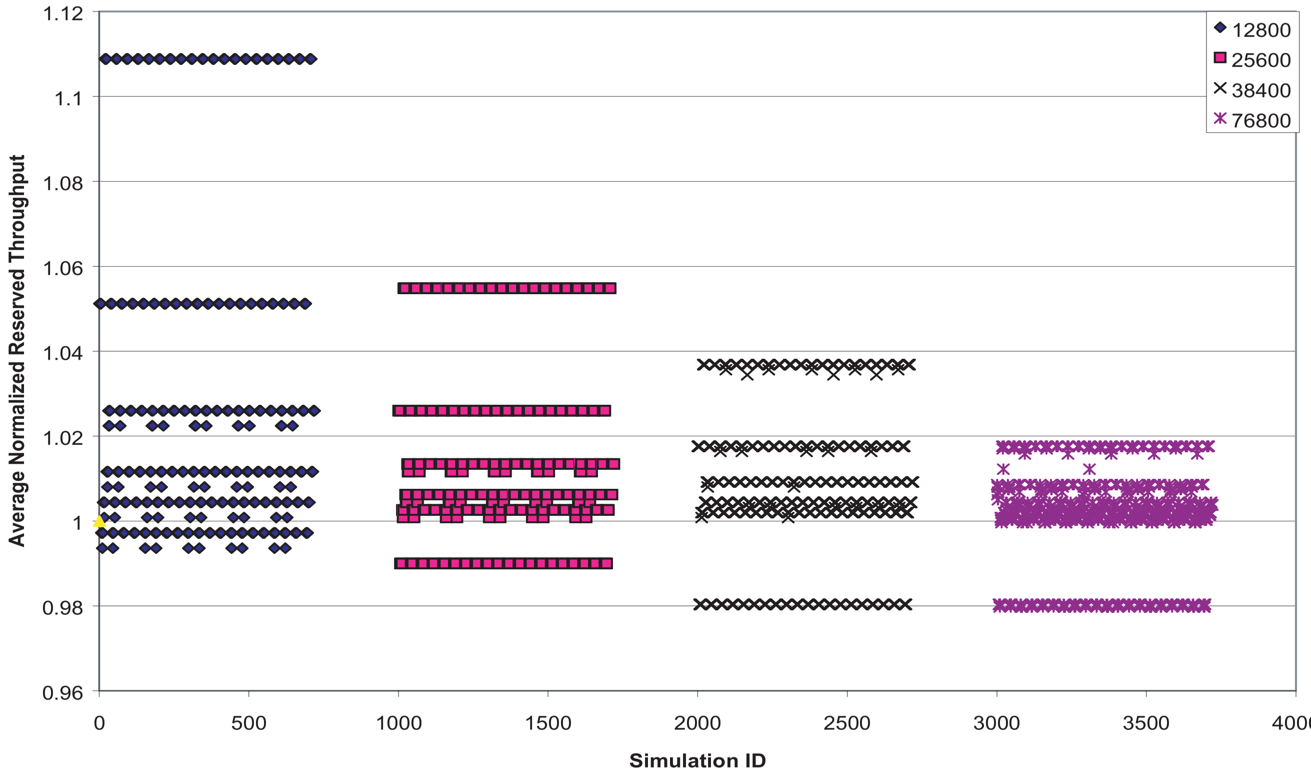


Figure 4.20: MEO Simulation Results: Reserved Rate Utilization by UDP Customers in Three-Color Simulations

4.5.6.4 MEO Simulation Results Discussion

It is clear from figures 4.15 and 4.16 that fairness is not good in two-color simulations. With three colors there is a wide variation in fairness results with the best being close to 1. It can be noted that fairness is 0 in some of the two-color simulations as the total reserved traffic uses all the bandwidth and there is no excess bandwidth available to share.

Figures 4.17 and 4.19 show the reserved rate utilization for TCP traffic and it is obvious that three-color simulations is better than two colors. Figures 4.18 and 4.20 show the reserved rate utilization for two-color UDP and three-color UDP. As discussed in Section 4.5.3, in three-color simulation, the UDP customer always has a yellow rate of 0. However, TCP packets coming from customers 1-9 are colored green, yellow or red (three precedence levels) and UDP packets are colored green or red (two precedence levels). It can be noted that in some cases reserved rate utilization is slightly more than 1. This is because token buckets are initially full which results in all packets getting green color in the beginning.

Clearly, three levels of drop precedence (colors) are required for high reserved rate utilizations. The fair allocation of excess network bandwidth can be achieved only by giving different treatment to congestion-sensitive TCP and congestion-insensitive UDP packets.

4.6 MPLS over Satellite Network

Due to the advantages of MPLS such as simplified forwarding based on exact match of fixed length labels, separation of routing and forwarding networks, integration of ATM and IP technologies, and traffic engineering and QoS routing, it can be applied over satellite to guarantee the QoS service levels. This section provides traffic engineering aspects of MPLS for Satellite Network.

4.6.1 MPLS Overview

MPLS stands for "Multiprotocol Label Switching"[214]. It's a layer 3 switching technology aimed at greatly improving the packet forwarding performance of the backbone routers in the Internet or other large networks. The basic idea is to forward the packets based on a short, fixed length identifier termed as a 'label', instead of the network-layer address with variable length match. The labels are assigned to the packets at the ingress node of an MPLS domain. Inside the MPLS domain, the labels attached to packets are used to make forwarding decisions. Thus, MPLS uses indexing instead of a longest address match as in conventional IP routing. The labels are finally popped out from the packets when they leave the MPLS domain at the egress nodes. By doing this, the efficiency of packet forwarding is greatly improved. Routers which support MPLS are known as "Label Switching Routers", or "LSRs".

Although the original idea behind the development of MPLS was to facilitate fast packet switching, currently its main goal is to support traffic engineering and provide quality of service. The goal of traffic engineering is to facilitate efficient and reliable network operations, and at the same time optimize the utilization of network resources. Most current network routing protocols are based on the shortest path algorithm, which implies that there is only one path between a given source and destination end system.

In contrast, MPLS supports explicit routing, which can be used to optimize the utilization of network resources and enhance traffic oriented performance characteristics. For example, multiple paths can be used simultaneously to improve performance from a given source to a destination. MPLS provides explicit routing without requiring each IP packet to carry the explicit route, which makes traffic engineering easier. Another advantage is that using label switching, packets of different flows can be labeled differently and thus received different forwarding (and hence different quality of service).

A Label Switched Path (LSP) is referred to as a path from the ingress node to the egress node of an MPLS domain followed by packets with the same label. A traffic trunk is an aggregation of traffic flows of the same class, which are placed inside an LSP [216, 219]. Therefore, all packets on a traffic trunk have the same label and the same 3-bit class of service (currently experimental) field in the MPLS header.

Traffic trunks are routable objects like virtual circuits in ATM and Frame Relay networks. These trunks can be established either statically or dynamically (on demand) between any two nodes in an MPLS domain.

A trunk can carry any aggregate of micro-flows, where each micro-flow consists of packets belonging to a single TCP or UDP flow. In general, trunks are expected to carry several such micro-flows of different transport types. However, as shown in this analysis, mixing different transport types can cause performance problems such as starvation and unfairness for certain traffic.

Figure 4.21 illustrates the relationships between the various abstractions that have been described above.

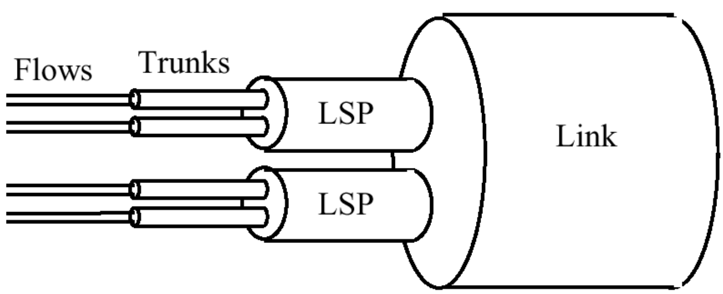


Figure 4.21: Relationships among Flows, Trunks, LSPs and Links

In this section, the performance impact of mixing TCP and UDP traffic is analyzed. The two transports have a very different congestion response. Specifically, TCP has congestion control and reduces its traffic in response to packet loss whereas, UDP has no congestion control and does not respond to losses. While it is possible to design UDP applications that are sensitive to congestion losses, very few such applications exist.

4.6.2 Network Topology

The initial simulation and analysis with label switching and CBQ (Class-based Queuing) for trunk service is described in the following paragraphs.

In the simulations, the network topology shown in Figure 4.22 was used. It consists of fix routers and six end-systems. The routers are MPLS capable. The routers R2, R4, R5 are gateways in a broadband satellite network. There are 3 flows. Source S1 sends UDP traffic to destination D1. Sources S2 and S3 send TCP traffic to destination D2 and D3, respectively (here n=3). The UDP source sends traffic at a given rate. The TCP sources are "infinite ftp" sources and send packets whenever its congestion window allows. The actual throughputs are monitored at the destination nodes.

The two TCP flows are different only in their packet sizes. The first TCP flow (TCP1) between S2 and D2 uses a MSS of 512 bytes. The second flow (TCP2) between S3 and D3 uses an MSS of 1024 bytes. Thus, the two flows are almost identical except for the packet sizes.

As seen in Figure 4.22, there exist two parallel paths between routers R2 and R5, one (R2-R3-R5) of a high bandwidth (45 Mbps) and the other (R2-R4-R5) of comparatively low bandwidth (15Mpbs). Routers R2, R3, R4, R5 belong to satellite networks.

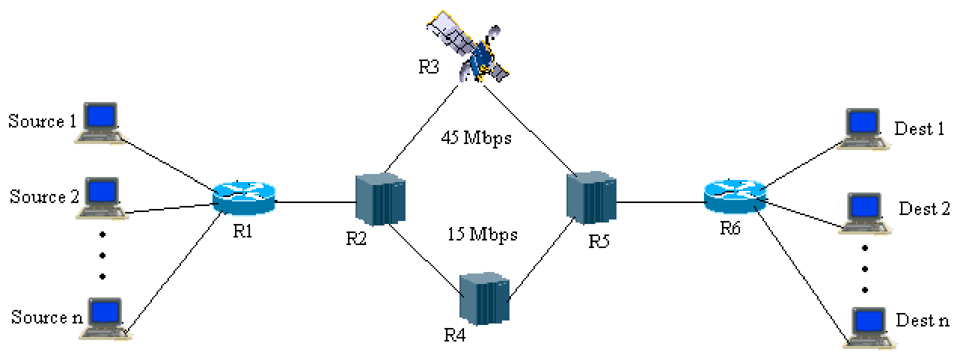


Figure 4.22: Network Topology

4.6.3 Simulation Results

To see the effect of MPLS traffic trunks, four different scenarios with different intermixing of TCP and UDP flows has been analyzed. In each scenario, the UDP rate is varied and measured the throughput of the three flows. These scenarios and the analysis results are as follows.

4.6.3.1 Case 1: No Trunks, No MPLS

The first case current IP with best effort shortest path routing without any trunks is analyzed. All traffic in this case follows the high-speed path R2-R3-R5 and the alternative path R2-R4-R5 is unused. The results are shown in Figure 4.23.

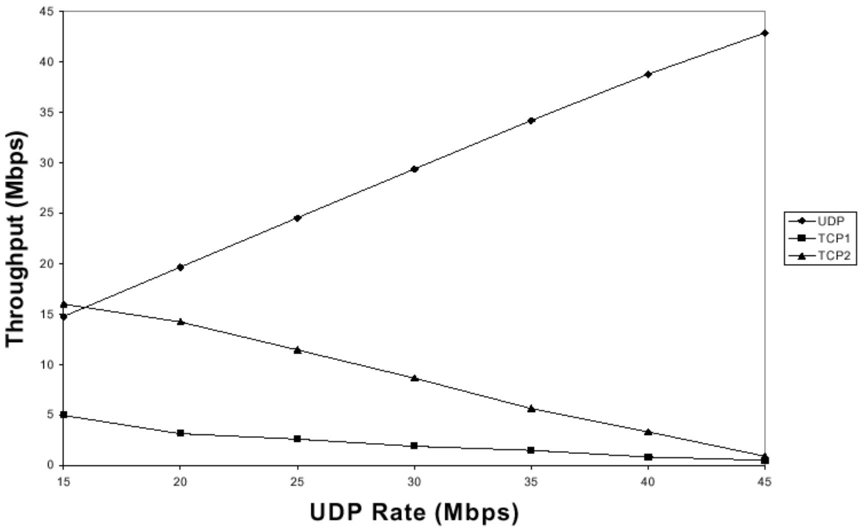


Figure 4.23: Results with Normal IP routing

Notice that as the UDP flow increases its rate, the throughputs of both TCP flows decrease. When the UDP rate reaches 45Mbps, both TCP flows can send almost nothing because all the bandwidth of link R2-R3 is used by UDP. That is, the congestion-sensitive traffic suffers at the hands of congestion-insensitive traffic.

Notice also that the two TCP flows that are different only in their packet sizes get very different throughput. The flow with smaller packet size (TCP1) gets very small throughput. This is a known behavior of TCP congestion mechanism.

4.6.3.2 Case 2: Two Trunks using Label Switched Paths

In this case two trunks were configured over the network in the following way. The first trunk carries UDP and TCP1 and follows the label switched path (LSP) R1-R2-R3-R5-R6 (high bandwidth path). The second trunk carries TCP2 and follows the path R1-R2-R4-R5-R6 (low bandwidth path). The UDP source rate is configured and the throughput observed for the various flows. The results are shown in Figure 4.24.

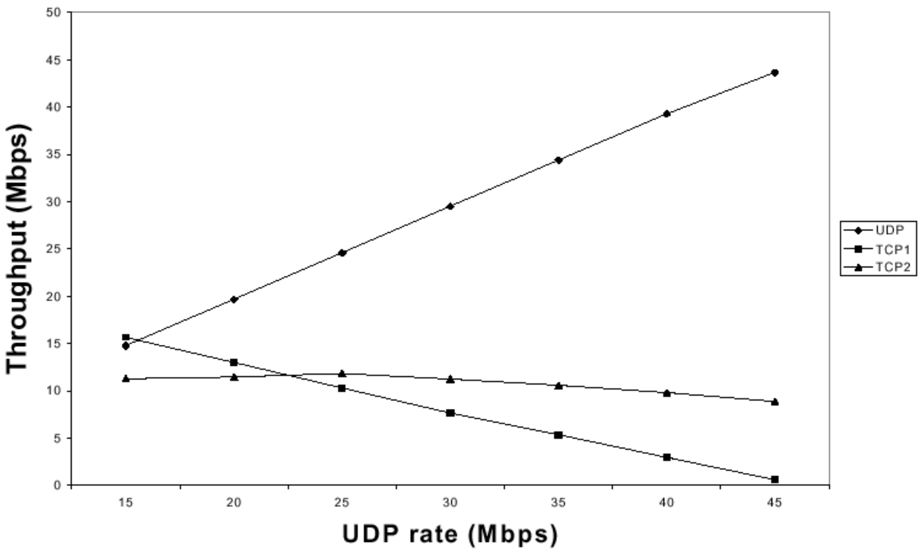


Figure 4.24: Results using two trunks, with TCP and UDP mixed on one Trunk

There is a significant improvement in the throughput of the flows. This is because the low bandwidth link which was not being utilized till now is also being used. Hence, TCP2 which is directed on a separate trunk and follows the low bandwidth link is unaffected by the increase in the UDP source rate and hence shows an almost constant throughput. But, it can also be observed that as the UDP source rate increases, the throughput of the TCP1 flow mixed with it on the same trunk suffers. This is because the TCP flow cuts down its rate in response to network congestion. Thus, it can be said that use of separate trunks definitely improves the network performance but, is not sufficient to provide the Quality of Service which might be desirable for some applications. Thus, in order to guarantee the Quality of Service to a given flow, UDP flows and TCP flows in different trunks need to be isolated so that even if a given UDP flow increases its source rate, TCP flows are unaffected by this.

4.6.3.3 Case 3: Three Trunks with isolated TCP and UDP flows

In this case, three isolated trunks are used, one each for the UDP and TCP flows, respectively. This is done using Class Based Queuing (CBQ) to guarantee bandwidth allocations for all the different trunks at the routers. By doing this, essentially the UDP flows are separated from the TCP flows and generate the results shown in Figure 4.25.

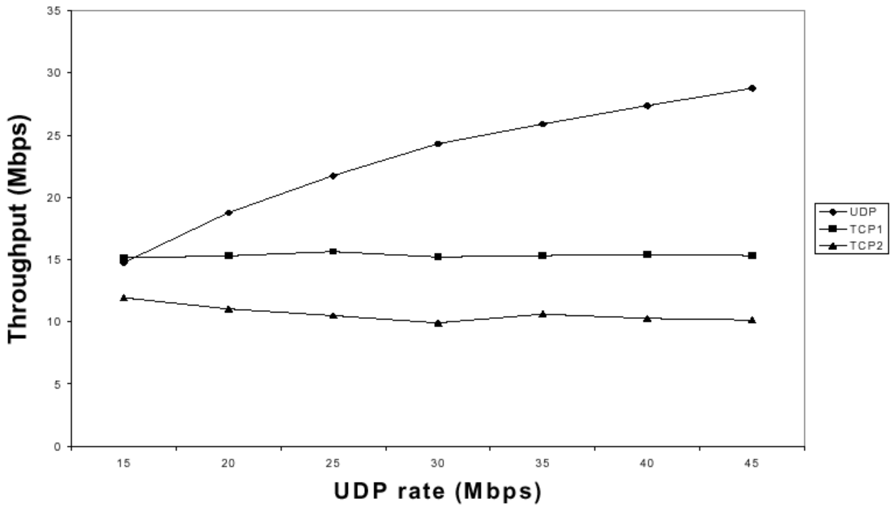


Figure 4.25: Results using a separate trunk for different flows

Here it can be seen that the increase in the UDP source rate does not affect either of the TCP sources and both are able to get a fairly constant throughput. Thus, by isolating UDP and TCP flows, a given Quality of Service can be guaranteed to sources which are responsive to congestion control also. Although this is achieved at the overhead of maintaining separate queues for each of the trunks at each of the routers. According to Li and Rekhter [185], the number of trunks within a given topology is within the limit of (N*(N-1)*C), where N is the number of routers in the topology and C is the number of traffic classes. This means the overhead is within a reasonable limit.

It is worth to note that although one flow is used in each trunk, this is not the general case. There could be multiple flows in a trunk. The simulations suggest that UDP flows from TCP flows should be separated using different trunks.

4.6.3.4 Case 4: Non end-to-end Trunks

In this case a scenario where the trunks are not end-to-end is analyzed. Specifically, the trunks are being initialized only from R2. In this case, the flows interfere with each other for part of the path since R1 does not distinguish between the various flows. The results are shown in Figure 4.26.

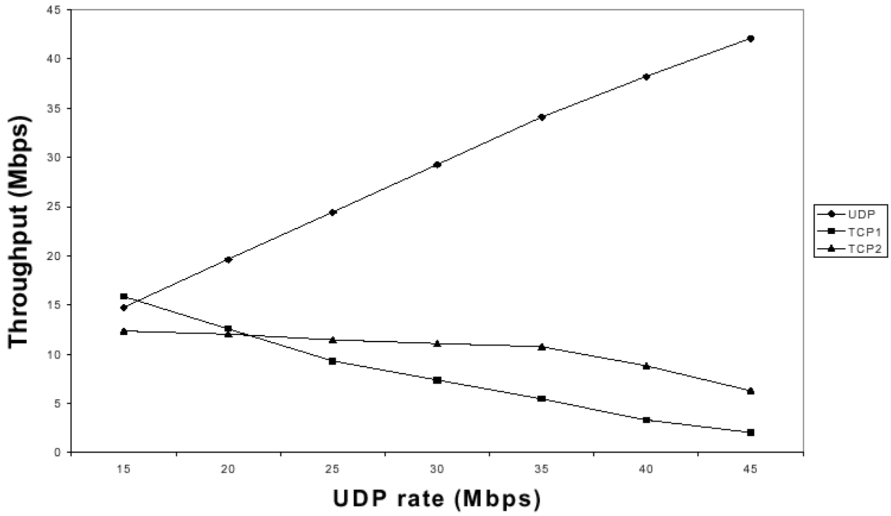


Figure 4.26: Results using Non end-to-end trunks

The above results show that the network really cannot guarantee anything in such a case. This happens because at R1 the flows are treated identically and during periods of congestion the TCP sources reduce their source rates and this leads to very poor throughput for the TCP sources even though they are treated distinctly at R2. Thus, for such a scheme to be successful, the flows should be separated as soon as they share a common link in the network.

4.7 Summary

Satellite IP QoS classes and performance objectives are described. Quality of Service architectures for satellite IP, simulation study and full factorial analysis for TCP and UDP traffic with Differentiated Services over different satellite network configurations and throughput performance analysis for TCP and UDP for MPLS over satellite network was studied in this chapter.

In Section 4.4 three QoS architectures for satellite IP are proposed. These architectures are (a) IntServ based, (b) DiffServ based, and (c) IntServ/DiffServ with aggregate RSVP. It is concluded that the architectures 1 and 2, based on IntServ and DiffServ respectively, are the simplest to implement at the terminals and the gateways. They provide end-to-end QoS. Architecture 3 is a scalable architecture. It provides an end-to-end QoS with dynamic aggregate RSVP reservations. However, this aggregate RSVP is not widely deployed in terrestrial environments. Considering the advantages of the aggregate RSVP as opposed to individual flow RSVP, this architecture 3 is a compromise between 1 and 2.

In Section 4.5 a QoS simulation model with Differentiated Services for satellite network is developed. The objective of this simulation study is to investigate the influence of several parameters on TCP and UDP performance over GEO, MEO and LEO networks. A wide range of simulations varying several factors to identify the significant ones influencing fair allocation of excess satellite network resources among congestion sensitive TCP and congestion insensitive UDP flows were developed. Appendix 4A.1 provides LEO simulation results.

Multiple drop precedence levels in an Assured Forwarding traffic class on a satellite network in GEO network are to ensure that all customers achieve their reserved rate and a fair share of excess bandwidth. The study was extended for network access satellite architecture, without on-board processing and switching and the impact analysis of BER and other factors on the performance of assured forwarding (Appendix 4A.2). The key conclusions in this study are:

- The key performance parameter is the level of green (reserved) traffic. The combined reserved rate for all customers should be less than the network capacity. Network should be configured in such a manner so that in-profile traffic (colored green) does not suffer any packet loss and is successfully delivered to the destination.
- If the reserved traffic is overbooked, so that there is little excess capacity, two drop precedence give the same performance as three.
- The fair allocation of excess network bandwidth can be achieved only by giving different treatment to out-of-profile traffic of congestion sensitive and insensitive flows. The reason is that congestion sensitive flows reduce their data rate on detecting congestion however congestion insensitive flows keep on sending data as before. Thus, in order to prevent congestion insensitive flows from taking advantage of reduced data rate of congestion sensitive flows in case of congestion, excess congestion insensitive traffic should get much harsher treatment from the network than excess congestion sensitive traffic. Hence, it is important that excess congestion sensitive and insensitive traffic is colored differently so that network can distinguish between them. Clearly, three colors or levels of drop precedence are required for this purpose and is independent of the orbital selection. Classifiers have to distinguish between TCP and UDP packets in order to meaningfully utilize the three-drop precedence.
- RED parameters and implementations have significant impact on the performance. Further work is required for recommendations on proper setting of RED parameters.
- BER levels do not influence the nature of TCP and UDP results
- The behavior is the same for GEO, MEO, and LEO networks.

MPLS traffic engineering was applied to study TCP and UDP throughput improvements over a satellite network. Simulation results show that the total network throughput improves significantly with proper traffic engineering. Congestion-insensitive (UDP) flows affect the throughput of congestion-sensitive (TCP) flows. Therefore, different types of flows should be isolated in different trunks in order to guarantee quality of service. For the scheme discussed in Section 4.6 to be effective the trunks need to be end-to-end, otherwise the advantages of isolation in other parts of the network are eliminated or reduced significantly.

5 Quality of Service for Satellite Internet Access Networks

High speed Internet access was previously limited to enterprise, using technologies such as leased T-1, frame relay or ATM. But with exponential growth of the Internet access for residential users, service providers have recognized the great opportunity in the residential broadband and more broadband enterprise markets. The telephone, cable, xDSL and satellite companies have been developing access technologies. Table 2.6 of Chapter 2 provides a partial list of access satellite systems for regional coverage.

Certain technical challenges are yet to be solved for a complete realization of the broadband satellite access systems. The issues include:

- What is the right choice of protocol for the return channel?
- What are the similarities and differences between the existing protocols?
- How is interoperability achieved between the different return channel protocols?
- What is the end-to-end QoS architecture for such a 2-way interactive satellite IP network?

In this chapter specifically the following are addressed:

Problem 1: MF-TDMA Return Channel Protocol Analysis.

The 2-way interactive satellite Internet access network presents a network asymmetry environment. Two Multi Frequency - Time Division Multiple Access (MF-TDMA) based protocols for return channel standard protocols are proposed by ETSI/DVB-RCS and Cable Labs. However, a proper selection of the return channel protocol for Internet access is extremely critical for system design. In this contribution, the MF-TDMA based return channel protocols are analyzed.

Problem 2: New CDMA based Spread ALOHA Protocol for Return Channel and Analysis.

A new CDMA based Spread ALOHA protocol for satellite Internet return channel is proposed and analyzed. The spread ALOHA with single code multiple access repeats the code or spread sequence every symbol. On the other hand code reuse multiple access or Spread ALOHA One Long Code uses one long code as long as the packet. For the first time the performance analysis of these spread ALOHA protocols for return channel are analyzed using Monte Carlo simulations.

5.1 MF-TDMA Return Channel Protocol Analysis

Figure 5.1 shows an example of broadband satellite network using the Digital Video Broadcasting via Satellite (DVB-S) protocol standard for the forward channel and the Digital Video Broadcast - Return Channel via Satellite (DVB-RCS) standard for the return channel [62, 63]. The *forward* channel refers to the link from the gateway that is received by the user terminal and the *return* channel is the link from the user terminal to the gateway.

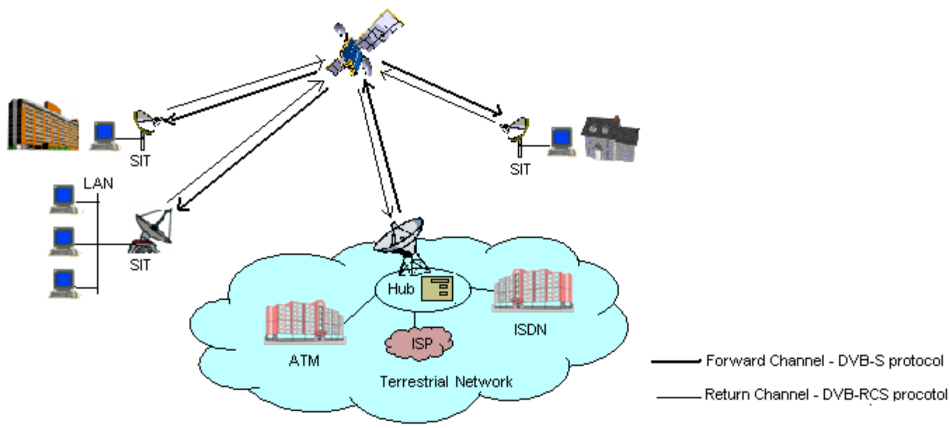


Figure 5.1: Broadband Satellite Access – DBV-RCS

The 2-way interactive satellite Internet access network also presents a network asymmetry environment. The DVB-S forward link is about 38-45 Mbps and return link using terrestrial dialup or cellular or satellite is with around 10:1 asymmetry. To accommodate such network path asymmetry, several mitigation techniques have been proposed [134, 113].

The DVB network elements consist of an Enterprise Model, Service Level Agreements (SLA), and TCP Protocol Enhancement Proxy (PEP), the Hub station and the Satellite Interactive Terminal (SIT). The target applications of the DVB network could be Small and Medium Enterprises and residential users. One of the major advantages of DVB-RCS is that multicasting is possible at a low cost using the existing Internet standards. The multicast data is tunneled over the Internet via a multicast streaming feeder link from a streaming source to a centralized multicast streaming server and is then broadcasted over the satellite medium to the intended target destination group. The DVB-RCS system supports relatively large streaming bandwidths compared to existing terrestrial solutions (from 64 kbps to 1 Mbps).

In the DVB network, *a satellite* forward and return links typically use frequency bands in Ku (12-18 GHz) and/or Ka (18-30 GHz). The return links use spot beams and the forward link global beams are used for broadcasting and Ku-band. Depending on the frequency bands (Tx/Rx), three popular versions are available: (a)Ku/Ku (14/12 GHz) (b)Ka/Ku (30/12 GHz) and (c) Ka/Ka (30/20 GHz). In business-to-business applications, the SIT is connected to several user PCs via a LAN and a Point-of Presence (POP)

Router. The Hub station implements the forward link via a conventional DVB-S chain (similar to Digital TV broadcasting) whereby the IP packet is encapsulated into DVB-streams, IP over DVB. The return link is implemented using the DVB-RCS standard "MF-TDMA Burst Demodulator bank", IP over ATM like. The HUB station is connected to the routers of several ISP's via a Broadband Access Server. The HUB maps the traffic of all SITs belonging to each ISP in an efficient way over the satellite. The selection of a suitable residential access technology depends on the type of application, site location, required speed, and affordable cost.

5.1.1 DVB-RCS

The DVB Return Channel System via Satellite (DVB-RCS) was specified by an ad-hoc ETSI technical group founded in 1999. The DVB-RCS system specification in ETSI EN 301 790, v1.2.2 (2000-12) specifies a satellite terminal (sometimes known as a Satellite Interactive Terminal (SIT) or Return Channel Satellite Terminal (RCST) supporting a two-way DVB satellite system [62, 63]. Another CDMA based spread ALOHA has been proposed for return channel access [266]. This section describes DVB-RCS protocol. The use of standard system components provides a simple approach and should reduce time to market.

Customer Premises Equipment (CPE) receives a standard DVB-S transmission generated by a satellite gateway. Packet data may be sent over this forward link in the usual way (e.g. MPE, data streaming, etc.) DVB-RCS provides transmit capability from the user site via the same antenna. The transmit capability uses a Multi-Frequency Time Division Multiple Access (MF-TDMA) access scheme to share the capacity available for transmission by the user terminal. Return channel is coded using rate ½ convolution FEC and Reed Solomon coding. The standard is designed to be frequency independent and it does not specify the frequency band(s) to be used - thereby allowing a wide variety of systems to be constructed. Data to be transported may be encapsulated in ATM cells, using ATM Adaptation Layer 5 (AAL-5), or use a native IP encapsulation over MPEG-2 transport. It also includes a number of security mechanisms.

5.1.2 Data over Cable Service Interface Specification (DOCSIS)

The DOCSIS was developed by the North American Cable Industry under the auspices of Cable Labs. to create a competitive market for cable modem equipment. It was developed as a cheap web-serving platform. The main specification work for DOCSIS 1.0 was completed in March 1997 (27).

A cable data system consists of multiple cable modems (CM), in subscriber locations, and a cable modem termination system (CMTS), all connected by a Community Antenna Television (CATV) plant. The CMTS can reside in a headend or a distribution hub. The DOCSIS products have been available since 1999. The DOCSIS 1.1 version has enhanced the specification in terms of Quality of Service (QoS), IP multicast and security [65]. The DOCSIS 2.0 version has been released in 2002 downstream. The DOCSIS supports an upstream of 320 Kbps – 10.24 Mbps and downstream rates of 36 Mbps.

5.2 Performance Analysis

At the Physical layer DVB-RCS and DOCSIS have a very similar structure, burst mode TDMA upstream and TDM downstream. In particular, the choice of FEC for the downstream is similar in both DVB-RCS and DOCSIS but for upstream it is different. Heavily concatenated coding or turbo coding is used in DVB-RCS while DOCSIS has narrow filtering with Reed-Salmon coding. DVB-RCS supports ATM, IP and MPEG formats thus supporting consumer and business markets. DOCSIS supports only IP and so only the consumer market.

At the MAC layer, the basic features such as, capacity allocation based on demand and free assignments, capacity request by piggy-back, request slot (except random-access request allowed in DOCSIS), synchronization/ranging functions, and registration, are similar in both DVB-RCS and DOCSIS.

The specific MAC formats are different in both of them. In DOCSIS, the head-end station handles the capacity management, ranging, and registration of the terminals. DVB-RCS, on the other hand, has three elements, NCC, terminal and head-end station, with each element handling each function. The head-end station takes care of registration. Capacity management involves both head-end station and NCC since in satellite, many service providers share resources. The service provider pay for a grand total capacity. The actual dynamic capacity allocation is controlled by NCC according to a contract. Synchronization and ranging is under the control of NCC.

Table 5.1 provides a comparison of the two return channel protocols.

Table 5.1: Protocol Comparison Summary

Feature	DVB-RCS	DOCSIS
Downstream Rates	QPSK with block turbo (plans) 8 MHz Channelization, OOB R-S & Convolutional Turbo	64-QAM: 27 Mbps 256-QAM: 42 Mbps 6 MHz Channelization Reed Solomon
Upstream Rates	1.544 Mbps; 3.088 Mbps Differential QPSK 5-65 MHz	.320, .640, 1.280, 2.560 & 5.120 Mbps QPSK & .640, 1.280, 2.560, 5.120, 10.24 Mbps 16-QAM; 200 KHz – 3.2 MHz 5-42 MHz, Reed Solomon
Multiple Access	MF-TDMA	MF-TDMA
Services	Internet access, interactive set-top box	Internet access, interactive set-top box, voice over IP
Management	SNMP	SNMP
Standards Status	Standard approved, Dec 2000 Enhancement Plans	Well established for cable modem None for satellite implementation today

In summary, DOCSIS as is cannot be used for broadband satellite communication network. Considerable modifications at the Physical and MAC layers have to be developed, for DOCSIS to be used for satellite access return channel. In the next section a CDMA based Spread ALOHA protocol has been developed and analysis and simulation results are presented.

5.3 Spread ALOHA Multiple Access for Satellite Network

In this section a CDMA based Spread ALOHA Multiple Access is proposed and developed. This channel is actually the first resource all users must share to access the multimedia information to be delivered by the forward channel on a per-user basis. The efficient use of this channel impacts the overall efficiency and latency along with the forward channel dynamic resource assignment on "bent pipe" and partial or full processor satellites. Figure 5.2 shows the configuration of forward and return channels for multimedia satellite networks.

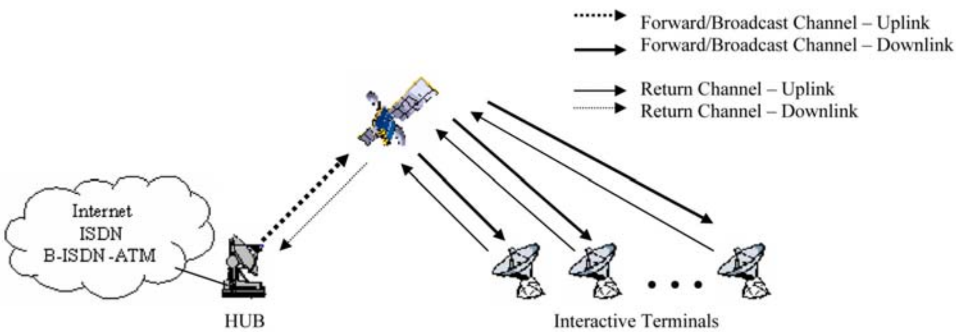


Figure 5.2: Configuration of interactive forward and return channel for multimedia satellite networks

5.3.1 Multiple Access Architectures for Return Channel

The interactive communication channel in the forward direction (HUB-Sat-terminals) is one-to-many, i.e. a broadcast channel and its dimensioning is a relatively simple problem. This channel is generally time-division multiplexed (TDM) like the Direct Video Broadcasting-Satellite (DVB-S) standard of the European Telecommunications Standards Institute (ETSI) [62, 63]. The communication channel in the return direction (terminals-Sat-HUB) is many-to-one, i.e. a multiple-access channel to be shared by all the users within the satellite beam. Its optimum architecture depends on several interdependent variables making it a challenging design problem especially in GEO networks scenario.

The key design parameters to be taken into account are the latency and overhead introduced by the multiple access candidates. This point is especially critical in a GEO constellation that inherently introduces a round-trip-delay (RTT) of about half a second. A LEO (Low-Earth-Orbit) constellation introduces smaller delays albeit time variant. Terrestrial networks have potential faster setup connectivity but in turn are subjected to be degraded by congestion that eventually the satellite network is able to bypass. Another characteristic of broadband satellite is the asymmetry between the forward and the return channel (can be 10:1 or greater). Even with this asymmetry the return multiple access channel should serve as many terminals as the forward channel does. An optimum satellite multiple access channel should also be flexible enough to offer the wide range of transmission rates needed for multimedia data transmission (dependent on the particular global system architecture) and scalable for future changes in traffic demand and service requirements. Finally, interactive terminals should be low-power and low-cost to be competitive with or to complement terrestrial access terminals.

Multiple access architectures can be categorized as connection-oriented architectures or contention-oriented architectures.

5.3.2 Connection-Oriented Architecture

In this architecture the system assigns part of the return channel to each user after a setup process. This assignment avoids contention but introduces overhead and setup latency. The figure of merit of a connection-oriented multiple access architecture is the efficiency, which is the ratio between the multiple access channel transmission rate and a continuous transmission rate. Examples are the well-known Frequency-Division Multiple Access (FDMA) also called Single Channel per Carrier (SCPC) and Time Division Multiple Access (TDMA). The assignment can be fixed or on demand (Demand-Assigned Multiple Access, DAMA).

The main disadvantage of FDMA is its lack of flexibility for variable transmission rates. Instead, TDMA gives sufficient flexibility to allow transmission rate variability down to a frame-by-frame basis. The main disadvantage of TDMA however is the required high peak transmit power in a packet transmission due to the high burst rates. In addition, TDMA setup connection time may be high since it requires significant overhead for burst synchronization along with a setup connection time to assign the channel. In order to limit the peak transmission power of the terminals a combination of FDMA and TDMA called Multi Frequency (MF-TDMA) is used. The MF-TDMA consists of a TDMA scheme over more than one carrier. Figure 5.3 shows an example of a MF-TDMA frame.

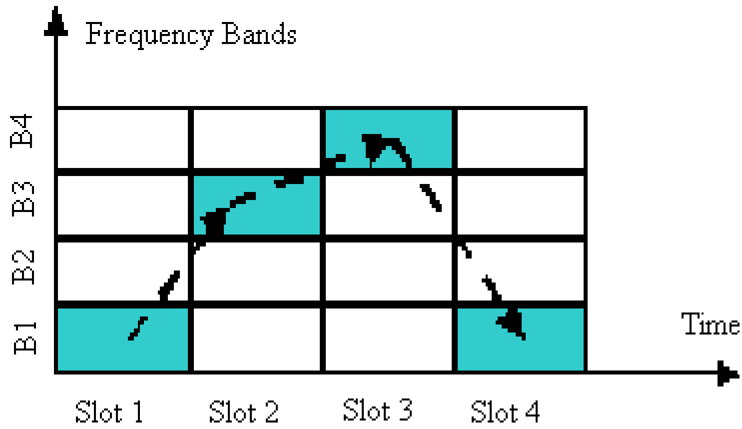


Figure 5.3: Example of multi-frequency TDMA frame

Also Code Division Multiple Access (CDMA) can be considered as a connection multiple access scheme when the assignment of the available set of codes is made on a per-user basis after an initial setup process. For voice applications or continuous flow of information this strategy can be efficient. However, it may lack flexibility when dealing with bursty traffic. In this case contention policies also based on CDMA become more suitable.

5.3.3 Contention-Oriented Architecture

In contention architecture the terminals access the channel without any previous assignment and therefore there will be packet collisions. Normalized offered load, *G*, is defined [266] as the total traffic expressed as a fraction of the maximum data rate, i.e. $G=\lambda T_p$ where λ is the packet arrival rate and T_p is the packet duration. It is normally assumed that throughput is the figure of merit of a contention architecture. The normalized multiple access throughput, *S*, can be defined as the same fraction but referring to successful traffic, $S=G.P_s$, where P_s is the probability of success. Packet arrival is considered to be Poisson-distributed.

The basic contention or random access scheme is *pure ALOHA* proposed by Abramson in 1970 [267]. In pure ALOHA each user transmits without any synchronization with other users. After sending the packet, the user waits the round-trip delay for an acknowledgement (ACK) from the receiver, if no ACK is received, the packet is assumed to be lost in a collision and it is re-transmitted after a random delay. The relationship between *S* and *G* in this case is [267]

$$S_{ALOHA} = Ge^{-2G} \tag{5.1}$$

It can be observed that the maximum normalized throughput is *1/2e* occurring at $G^*=0.5$. Note that error correction techniques (channel coding) are not used in ALOHA since it does not have multiple-access capability in a common channel.

If terminals are only allowed to transmit within a certain time pattern, the scheme is called *slotted ALOHA* and the throughput increases to

$$S_{slotted\ ALOHA} = G e^{-G} \tag{5.2}$$

Now the maximum normalized throughput is *1/e* occurring at $G^*=1$. If instead of one single narrow band channel of pure ALOHA there are N independent channels chosen at random, the system is called *Multiple ALOHA Multiple Access (MAMA)* [266]. In this case the throughput is given by

$$S_{MAMA} = G e^{-2\frac{G}{N}}; \qquad S_{sloted\ MAMA} = G e^{-\frac{G}{N}} \tag{5.3}$$

This means that N ALOHA channels have N times the throughput of one single channel when serving N times the offered load. Note that for an arbitrary number of channels N, the maximum normalized throughput is $N/2e$ occurring at $G^*=N/2$. Figure 5.4 shows a comparison of pure ALOHA, slotted ALOHA and MAMA throughputs.

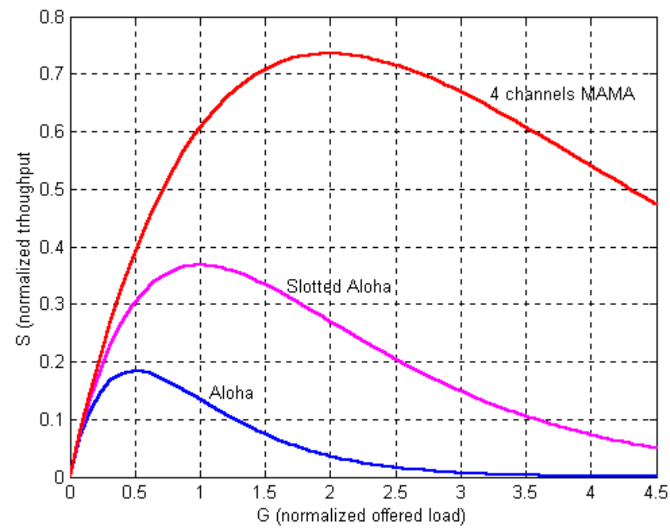


Figure 5.4: Throughput comparison of pure ALOHA, slotted ALOHA and MAMA (4 channels)

If the bandwidth of a narrow band ALOHA channel is increased by a factor F, there is a High Bandwidth ALOHA multiple access scheme**,** where the packets will be shorter thus decreasing the number of collisions. There is however a practical problem that excludes ALOHA for high bit user data rates. Bit energy, E_b, must keep unchanged since digital quality does not depend on power but on bit energy. This means that power during the packet burst must be increased by F since the packet has shorter length now due to the larger data rate. For values of F in the order of 100 and above the power to be transmitted can be extremely high. This fact limits pure ALOHA (either unslotted or slotted) to narrowband operation (up to few tens of kilobits per second). For broadband systems, different solutions can be applied. Spread spectrum technology on top of ALOHA can be used to provide a high bandwidth interactive return channel. The following section describes CDMA-based multiple access schemes.

5.3.4 CDMA based Contention Multiple Access Schemes

Fundamentally, there are two spread ALOHA multiple access schemes. These are distinguished by the code assignment as follows

- A different code is assigned to each user
- All users use the same code

In the first case, the receiver distinguishes a user for its code. ALOHA is applied by selecting randomly a code sequence. Note that this approach is different from the connection oriented CDMA where the code is assigned on a per-user basis during a setup procedure. To make this solution efficient, in [268] the inhibit sense multiple access (ISMA-CDMA) protocol is introduced, where the HUB broadcasts the status (free or busy) of the various available code sequences. In [268] a performance comparison between Slotted ALOHA CDMA and MAMA is presented. Authors conclude that by using error control coding the CDMA packet radio exhibits higher peak throughput, less delay (for low offered load) and much better stability than MAMA.

In the second case, the receiver distinguishes between the received packets if there are sufficient time offsets between them. Figure 5.5 shows how symbols from two different packets can be distinguished even when different users employ the same code.

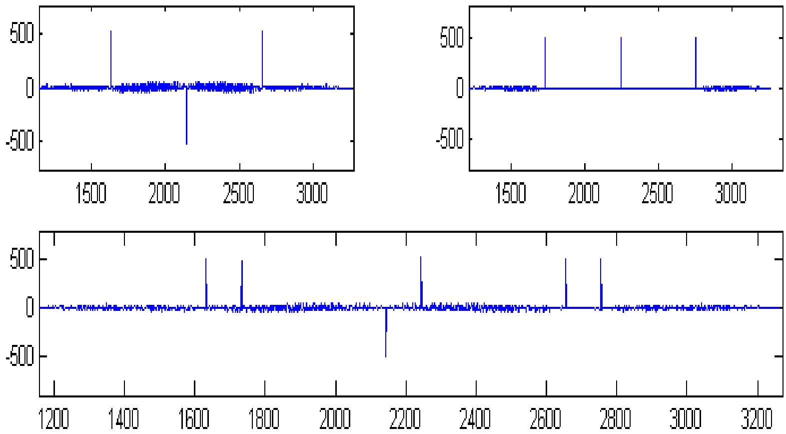


Figure 5.5: Outputs of two matched filters of two users using different codes (top) and outputs of one matched filter of two users using the same code

The use of an unique code implies that the channel is still a contention one, similar in nature to ALOHA. They differ however in two essential factors:

– The probability of having contention collisions is drastically reduced
– there can be several users simultaneously on the channel as long as the multiuser interference keeps below a given level. Channel coding can be used to mitigate error due to the interference of the multiple packets on-the-air

Spread spectrum ALOHA with one single code can be implemented in two ways: Spread Aloha Multiple Access (SAMA^{TM}) and Code Reuse Multiple Access (CRMA^{TM}). SAMA^{TM} repeats the code or spread sequence every symbol. Conversely, CRMA^{TM} uses one single code as long as the packet. For convenience, henceforth the following notation will be used:

– SAOC (Spread ALOHA One Code) will be SAMA^{TM}
– SAOLC (Spread ALOHA One Long Code) will be CRMA^{TM}

5.3.5 Performance of SAOC and SAOLC

The throughput performance analysis addresses the following issues:

– Is throughput limited by collision or by multiple access interference?
– What is the maximum number of simultaneous packets that the channel can admit? and
– How are transmitted power and probability of correct packet related to the maximum number of simultaneous packets that the channel can admit?

The performance is analyzed through Monte Carlo simulations. A single-hop spread spectrum return channel is considered with the following assumptions:

- An infinite number of independent ISTs (Interactive Satellite Terminals) transmitting asynchronously (unslotted ALOHA), a hub and one GEO satellite
- A common spreading sequence which either repeats every symbol (SAOC) or is as long as the packet (SAOLC)
- Data rate is R=128 kb/s and spreading factor, F can be 12, 30, 36, 60, 100, 200 or 300
- Packets are Poisson distributed with an arrival rate of λ. Packets are of fixed length of L bits
- Every transmitted packet is received with equal power

The bit error of asynchronous spread spectrum multiple access is due to both interference and additive gaussian noise (AWGN). The bit error probability is given by [269]

$$P_b(k) = \frac{2}{3}Q\left[\left(\frac{k-1}{3F} + \frac{N_0}{2E_b}\right)^{-0.5}\right] + \frac{1}{6}Q\left[\left(\frac{k-1F/3 + \sqrt{3}\sigma_k}{F^2} + \frac{N_0}{2E_b}\right)^{-0.5}\right]q - 0.5$$
$$+ \frac{1}{6}\frac{k-1F/3 + \sqrt{3}\sigma_k}{F^2} + \frac{N_0}{2E_b}^{-0.5} \tag{5.4}$$

$$\sigma_k^2 = k\left[F^2\frac{23}{360} + F\left(\frac{1}{20} + \frac{k-1}{36}\right) - \frac{1}{20} - \frac{k-1}{36}\right]$$

where $Q(\cdot)$ is the error function, E_b is the bit Energy, N_o the spectral noise density, F is the spreading gain and k is the number of simultaneous users. If Gaussian noise is neglected equation 5.4 reduces to

$$P_b(k) = \frac{2}{3}Q\left[\sqrt{\frac{3F}{k-1}}\right] \tag{5.5}$$

The probability of correct packet given that there are k simultaneous users on-the-air is:

$$P(\text{correct packet}/k \text{ simultaneous users}) = [1 - P_b(k)]^L \tag{5.6}$$

If the packet includes block error control capability that can correct t or fewer errors, the probability of packet success is

$$P(\text{correct packet}/k \text{ simultaneous users}) = \sum_{i=0}^{t}\binom{L}{i}(P_b(k))^{L-i}P_b(k)^i \tag{5.7}$$

Monte Carlo simulations are carried out by stepping through a vector of arrival times (Poisson-distributed). For each new arrival, one index tracks the packets still on-the-air while other index checks whether the new arrival incurs in either a multiple access collision or multiuser interference.

Let us note that the unslotted ALOHA case is an interesting scheme for a satellite scenario since synchronization would introduce a significant additional delay in a GEO scenario. However, since the power fluctuates during packet transmission the throughput and maximum number of simultaneous users is specially difficult to calculate as shown in Figure 5.7. Actually, the time evolution of the number of packets on-the-air is a stochastic process. In [270] and [271] this process is modeled as a birth-death process and a statistical treatment of the problem is given. The authors however disregard collision probability which may not be totally neglected in SAOC while it is in SAOLC (see Figure 5.6).

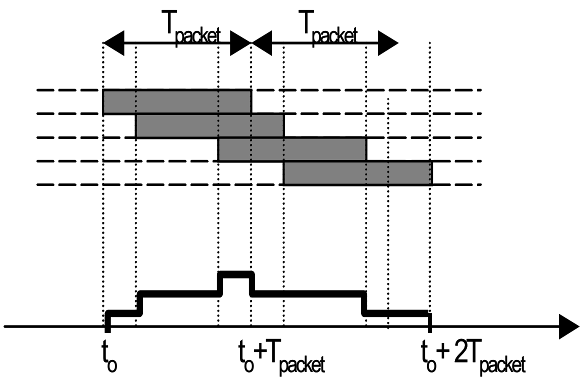


Figure 5.6: Power fluctuations in asynchronous spread ALOHA

To demonstrate this, the collision probability is calculated as follows. The probability of two or more packet arrivals during one bit duration (*P_bit*) and during one chip duration (*P_chip*) are:

$$P\_bit = \sum_{k=2}^{\infty} \frac{(\lambda Tb)^k}{k!} e^{-\lambda Tb} = \sum_{k=2}^{\infty} \frac{\left(\frac{G}{L}\right)^k}{k!} e^{-\frac{G}{L}} \tag{5.8}$$

$$P\_chip = \sum_{k=2}^{\infty} \frac{(\lambda Tc)^k}{k!} e^{-\lambda Tc} = \sum_{k=2}^{\infty} \frac{\left(\frac{G}{LF}\right)^k}{k!} e^{-\frac{G}{LF}} \tag{5.9}$$

Figure 5.7 represents equations 5.8 and 5.9 with $F = 30$ and packet lengths of 512 bits up to 512×5 bits. Note that these probabilities do not depend on the data rate.

There is a need to find out what the collision windows are. In SAOC a collision occurs when one terminal starts transmitting during the first chip of any symbol of the packets on-the-air. In SAOLC a collision occurs when one terminal starts transmitting during the first chip of the first symbol of the packets on-the-air. Collision windows are plotted in Figure 5.8.

Therefore, applying equations 5.8 and 5.9:

$$P_{colision\_SAOC} = L \cdot P\_chip \tag{5.10}$$

$$P_{colision\_SALOC} = P\_chip \tag{5.11}$$

With $F = 30$ and packet length of 512 bits $P_{colision\_SAOC} = 512 \times 2 \cdot 10^{-6} \approx 10^{-3}$ while $P_{colision\_SAOLC} = 10^{-6}$ for $G = 30$.

It can therefore be concluded that throughput can be limited by collisions in SAOC for small packet lengths and high offered load. Throughput is only limited by interference in SAOLC.

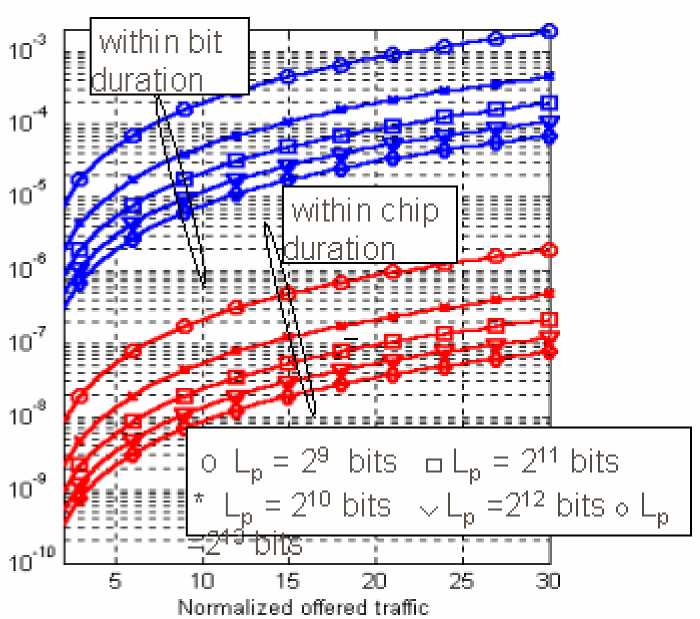


Figure 5.7: Probability of two or more packet arrivals during one bit duration, *P_bit* or during one chip duration, *P_chip*

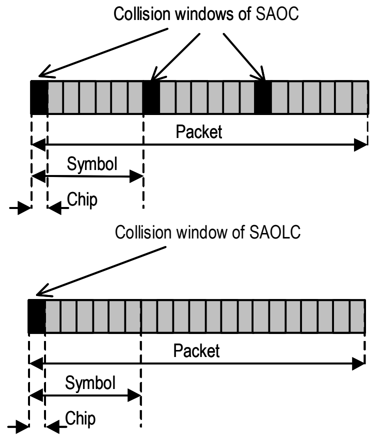


Figure 5.8: Collision windows for SAOC and SAOLC

5.3.6 Simulation Results

In this section SAOC and SAOLC multiple access performance are accessed. The following algorithm has been considered:

- When a collision is detected all the packets involved are considered lost (note that collisions only happen in SAOC)
- When the new arriving packet exceeds the maximum permitted number of packets on-the-air, all the packets are considered lost, including the new arriving one (channel coding was not taken into account)

Figures 5.9 and 5.10 show the performance for different maximum number of simultaneous users without considering interference effects. Note that throughput and offered traffic are normalized to that number of users for comparison. The following conclusions can be drawn: [272]

- SAOLC throughput increases above SAOC throughput as the maximum number of simultaneous users increases. This demonstrates that SAOLC is limited by interference while SAOC is limited by collisions
- SAOLC follows the curve $Y = X$ from low loads up to about 0.5 for 12 users and almost 0.85 for 60 users up to 200 users. These regions present virtually no packet lost
- SAOLC degrades rapidly as the load approaches N_{max} since the interference increases significantly. Thus SAOC behaves better for higher normalized offered traffic.

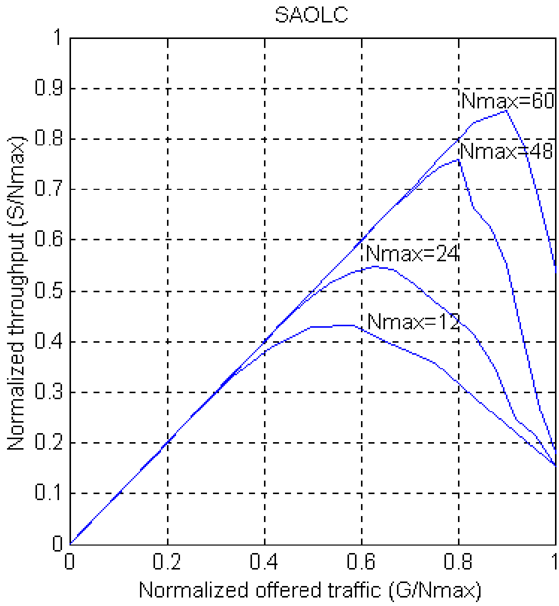


Figure 5.9: SAOLC normalized throughput for different number of users and *F* = 60

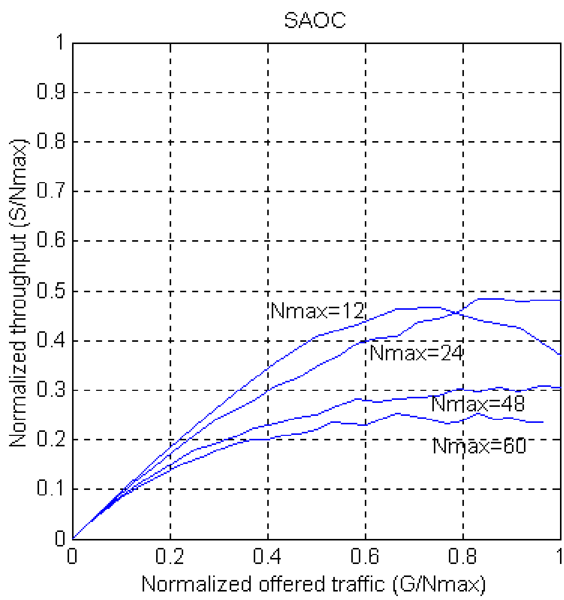


Figure 5.10: SAOC normalized throughput for different number of users and *F* = 60

However, throughput in Figures 5.9 and 5.10 does not take into account bit error probability due to interference, only shows the effect of number of users and collisions.

In order to introduce the effect of the bit error caused by interference, the number of maximum permitted simultaneous users from equations (5.4)–(5.5) is given in Table 5.2. Figure 5.11 shows the normalized throughput variation with normalized offered traffic for both SAOC and SAOLC using $F = 60$ and different E_b/N_o.

Table 5.2: Maximum number of simultaneous users for a given bit error probability

Pb	E_b/N_o (dB)	SPREADING FACTOR, F					
		12	30	60	100	200	300
10^{-6}	20	2	5	7	13	25	38
	60	2	7	9	15	28	42
10^{-3}	20	5	15	18	33	66	98
	60	5	16	22	35	69	100

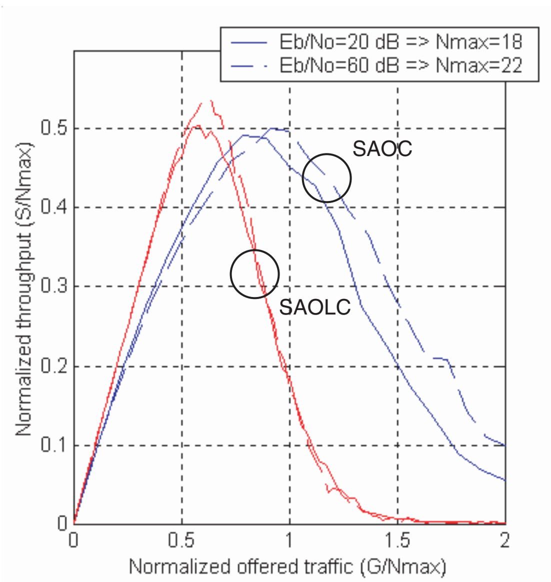


Figure 5.11: SAOC and SAOLC for $F = 60$ and two different E_b/N_o

By comparing Figure 5.11 with figures 5.9 and 5.10, it can be observed that for SAOLC, up to an 85% throughput could be achieved for a channel with about 60 simultaneous users. However, bit error due to multiuser interference reduces the throughput because the maximum permitted number of simultaneous users is actually restricted by the E_b/N_o. In addition, as it was concluded before, SAOLC throughput increases above SAOC initially but then decreases more rapidly than SAOLC.

5.4 Summary

Currently, for 2-way satellite IP Internet access networks DVB-RCS protocol for return channel is the preferred choice. However, other systems like WildBlue proposed to use DOCSIS with modifications to suit satellite access. There is a need for performance analysis of the two protocols from a reliability point of view. In this section, a comparative analysis of the DVB-RCS and DOCSIS is given. [273]

A novel CDMA based spread ALOHA single long code multiple access protocol proposed in this chapter, provides better throughput than spread ALOHA single code multiple access but is subjected to multiple user interference. This result is demonstrated through Monte Carlo simulations. The throughput performance results show that:

– Spread ALOHA One Long Code (SAOLC) throughput increases above Spread ALOHA One Code (SAOC) throughput as the maximum number of simultaneous users increases. This demonstrates that SAOLC is limited by interference while SAOC is limited by collisions.
– SAOLC follows the curve $Y = X$ from low loads up to about 0.5, and up to 12 users. The throughput increases almost to 0.85 for 60 users up to 200 users. These regions present virtually no packet loss.
– SAOLC degrades rapidly as the load approaches N_{max} while the interference increases significantly. Thus SAOC behaves better for higher normalized offered traffic.

Future satellite broadband systems could consider these return channel protocols for high bandwidth return interactive channel since they present good normalized throughput characteristics. Issues such as channel coding, delay distribution, power efficiency or economical aspects compared to other return channel protocols need to be assessed.

6 Conclusions and Future Work

This chapter concludes the thesis contributions and results. To realize future broadband satellite networks for new bandwidth intensive applications, future research issues are discussed.

6.1 Thesis Conclusions

Many of the future satellite communication networks are intended to provide global connectivity and regional access. One of the main future applications is high speed Internet access for applications which require high bandwidth. Some of the future systems are being designed at Ka-band, employ onboard processing and either ATM or fast packet switching. The flexibility and efficiency of operation is enhanced by these new system architectures but the complexity of implementation increases. The new system architectures must support the application Quality of Service levels.

This thesis analyzed the technical challenges for broadband satellite networks supporting ATM and IP technologies. Technical issue analysis, buffer models for ATM networks, analytical Demand Assignment Multiple Access (DAMA) model for bandwidth allocation were studied. Extensive simulation studies for TCP/UDP traffic for satellite IP with differentiated services were performed. IntServ, DiffServ, and MPLS based new QoS architectures for satellite based IP networks were proposed. A Spread ALOHA based protocol was proposed and analyzed for return channel of the broadband satellite Internet access network.

Chapter 2 studied the technical challenges for the design and implementation of the next-generation multimedia satellite networks. An architecture selection involves questions such as GSO system vs. NGSO (LEO or MEO) systems vs. hybrid systems (employing both GSO and NGSO), bent pipe vs. on-board processing; global vs. regional coverage, and operating frequency bands (Ka, Ku, C, X, and/or Q/V). Tradeoffs must be made between a wide area coverage and narrow spot beams. Wide coverage satellite antennas have lower antenna gain, and therefore require larger user terminals with perhaps higher terminal RF power. Narrow spot beam satellite antennas have higher gain, and thus require smaller user terminals, but are more complex. The user antenna size is another important criteria. Large antennas are expensive, undesirable for consumer applications and in some cases are prohibited by local building codes or other regulations. Antennas for high frequencies such as Ka-band are more expensive due to the need for greater precision in manufacturing. The narrow beam widths of Ka-band antennas require accurate pointing to the satellite and maintaining pointing alignment during the service life of the unit in the presence of wind, snow, animals and settling of the building where it is mounted. The larger the antenna the smaller the beamwidth and thus the more severe the pointing problem. Satellite communication systems operating at Ka-band are susceptible to RF link degradation due to attenuation by rain. The magnitude of the loss increases with frequency and challenges Ka-band system designers in several ways. Adaptive coding techniques and power control requirements were reported.

For the satellite Internet, Transmission Control Protocol (TCP) performance is degraded due to long propagation delays, link errors, and high degree of bandwidth asymmetry. Several mechanisms have been presented in the literature to improve TCP performance. However, optimal buffer design and management techniques for TCP over satellite have not been analyzed for different satellite configurations. The buffer design and TCP and UDP performance analyses were presented in the next chapters.

In Chapter 3 of this thesis, buffer requirements, TCP enhancements analysis by full factorial simulations, and bandwidth allocation for a satellite ATM network supporting UBR service were studied. The simulation study for buffer requirements for TCP over ATM UBR for LEO and GEO configurations was performed. Three latencies (5 ms, 50 ms, and 275 ms) with various number of sources and number of buffer sizes are considered. The simulation study assumed per VC buffer allocation policies of Selective Drop at the switch and end system policy of TCP SACK. TCP performance over UBR for sufficiently large buffer sizes is scalable with respect to the number of TCP sources. The simulation results indicate that a buffer size of about 0.5 x RTT to 1 x RTT is sufficient to provide 98% throughput to infinite TCP traffic for long latency networks and a large number of sources. The buffer requirement is independent of the number of sources. The fairness is high for a large number of sources because of per-VC buffer management performed at the switches.

As link delay increases, SACK clearly comes out superior to New Reno in terms of efficiency. For LEO, SACK and New Reno have similar efficiency values. For MEO, SACK performs a little better than New Reno and for GEO, SACK already outperforms New Reno. The reason for this is that New Reno needs N RTTs to receiver from N packet losses in a window whereas SACK can recover faster and start increasing CWND again. This effort becomes more and more pronounced as RTT increases.

Selective Discard (SD) does not always lead to increase in fairness as compared to Early Packet Discard (EPD). This result can be attributed to nature of WWW traffic. SD accepts packets of only under-represented VCs after crossing the threshold R. For sufficient buffer size, many of these VCs are under represented in switch buffer because they do not have a lot of data to send. Thus, SD fails to cause significant increase in fairness. It can be concluded that for large delay links, end system policies are more important than switch drop policies in terms of efficiency and fairness for WWW traffic.

To increase system utilization and flexibility, DAMA scheme has been proposed. A DAMA model for supporting CBR and VBR services is developed. Two possible resource (slot) allocations, contiguous and non-contiguous, for a TDMA based satellite ATM network are presented. Analytical and simulation results for blocking probability vs. offered load for non-contiguous and contiguous cases are presented. An example considered with 30% request for 16 Kbps CBR, 50% requests for 64 Kbps CBR and 20% for 128 Kbps CBR, non-contiguous i.e., totally flexible slot assignment results in an 8% increase in uplink channel utilization at an offered load of 80%. Another example with 95% requests for an average of 16 Kbps and 5% requests for an average of 192 Kbps show, that non-contiguous case affords roughly a 10% gain in uplink utilization.

In Chapter 4, Quality of Service architectures for satellite IP, simulation study and full factorial analysis for TCP and UDP traffic with Differentiated Services over GEO, MEO and LEO satellite network and throughput performance analysis for TCP and UDP for MPLS over satellite network were studied.

The IP based broadband satellite network for both global connectivity and access must support user QoS. IETF proposed QoS architectures to provide guaranteed service level to different applications - Integrated Services (IntServ), Differentiated Services (DiffServ) and Multiprotocol Label Switching (MPLS). But these architectures mainly address terrestrial networks. There is an urgent need for developing Quality of Service architectures for broadband satellite network and identifying the challenges for realizing Ka-band satellite systems.

In this thesis, three new QoS architectures for satellite IP were proposed in Chapter 4. These architectures are (a) IntServ based, (b) DiffServ based, and (c) IntServ/DiffServ with aggregate RSVP. It is concluded that the architectures 1 and 2, based on IntServ and DiffServ respectively, are the simplest to implement at the terminals and the gateways. They provide end-to-end QoS. Architecture 3 is a scalable architecture. It provides an end-to-end QoS with dynamic aggregate RSVP reservations. However, this aggregate RSVP is not widely deployed in terrestrial environments. Considering the advantages of the aggregate RSVP as opposed to individual flow RSVP, this architecture 3 is a compromise between 1 and 2.

A simulation model with a wide range of simulations varying several factors to identify the significant ones influencing fair allocation of excess satellite network resources among congestion sensitive TCP and congestion insensitive UDP flows were developed for GEO, MEO and LEO networks. Also the influence of bit error rate (BER) on the performance was studied.

One of the goals of deploying multiple drop precedence levels in an Assured Forwarding traffic class on a satellite network in GEO network, is to ensure that all customers achieve their reserved rate and a fair share of excess bandwidth. The study was extended for network access satellite architecture, without on-board processing and switching and the impact analysis of BER and other factors on the performance of assured forwarding. The key conclusions in this study are:

- The key performance parameter is the level of green (reserved) traffic. The combined reserved rate for all customers should be less than the network capacity. Network should be configured in such a manner so that in-profile traffic (colored green) does not suffer any packet loss and is successfully delivered to the destination.

- If the reserved traffic is overbooked, so that there is little excess capacity, two drop precedence give the same performance as three.
- The fair allocation of excess network bandwidth can be achieved only by giving different treatment to out-of-profile traffic of congestion sensitive and insensitive flows. The reason is that congestion sensitive flows reduce their data rate on detecting congestion however congestion insensitive flows keep on sending data as before. Thus, in order to prevent congestion insensitive flows from taking advantage of reduced data rate of congestion sensitive flows in case of congestion, excess congestion insensitive traffic should get much harsher treatment from the network than excess congestion sensitive traffic. Hence, it is important that excess congestion sensitive and insensitive traffic is colored differently so that network can distinguish between them. Clearly, three colors or levels of drop precedence are required for this purpose and is independent of the orbital selection. Classifiers have to distinguish between TCP and UDP packets in order to meaningfully utilize the three-drop precedence.
- RED parameters and implementations have significant impact on the performance. Further work is required for recommendations on proper setting of RED parameters.
- BER levels do not influence the nature of TCP and UDP results
- The behavior is the same for GEO, MEO, and LEO networks.

MPLS traffic engineering was applied to study TCP and UDP throughput improvements over a satellite network. Simulation results show that the total network throughput improves significantly with proper traffic engineering. Congestion-unresponsive (UDP) flows affect the throughput of congestion-responsive (TCP) flows. Therefore, different types of flows should be isolated in different trunks in order to guarantee quality of service. For the above scheme to be effective the trunks need to be end-to-end, otherwise the advantages of isolation in other parts of the network are eliminated or reduced significantly.

In Chapter 5, a new Spread ALOHA based protocol is defined and analyzed for return channel of the broadband satellite Internet access network. Currently, for 2-way satellite IP Internet access networks DVB-RCS protocol for return channel is the preferred choice. However, other systems like WildBlue proposed to use DOCSIS with modifications to suit satellite access. There is a need for performance analysis of the two protocols from a reliability point of view. In this section, a comparative analysis of the DVB-RCS and DOCSIS is given.

A CDMA based spread ALOHA single long code multiple access protocol, as proposed in Chapter 5, provides better throughput than spread ALOHA single code multiple access but is subjected to multiple user interference. This result is demonstrated through Monte Carlo simulations. The throughput performance results show that:

- Spread ALOHA One Long Code (SAOLC) throughput increases above Spread ALOHA One Code (SAOC) throughput as the maximum number of simultaneous users increases. This demonstrates that SAOLC is limited by interference while SAOC is limited by collisions.
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Future satellite broadband systems can consider these return channel protocols for high bandwidth return interactive channel since they present good normalized throughput performance. Issues such as channel coding, delay distribution, power efficiency or economical aspects compared to other return channel protocols need to be assessed.

6.2 Future Research

The current Internet infrastructure must be architected to handle future media-rich, content rich applications. The success of applications such as video-on-demand, multicast and content distribution depends on Quality of Service and bandwidth guarantees. Satellites will continue to play a significant role in the development of such an Internet infrastructure. Though the satellite systems provide more power and bandwidth through the use of spot beam technology, new "killer" applications drive the architectures. Traffic Management functions; especially, congestion control algorithms and performance models for real-time and non-real-time applications (TCP/UDP based) must be developed. Several TCP enhancements proposed by the IETF need to be evaluated for satellite IP network infrastructure. New QoS architectures for satellite IP networks, DVB networks, and access networks should be studied. Application QoS has to be defined in order to meet the service level agreements. Standards for the satellite IP QoS and access control protocols for the return channel should be developed and coordinated with the organizations such as IETF, ITU, TIA, and ETSI.

The following research issues have to be further studied to realize broadband satellite IP networks in future:

1. Quality of Service for satellite IP Internet
   - New QoS architectures for satellite IP based in DiffServ and MPLS traffic engineering over satellite
   - QoS models for non-real-time and multimedia applications over satellite configuration
   - ATM over MPLS architectures
2. Bandwidth allocation
   - New bandwidth allocation algorithms based on differentiated services for various high bandwidth applications
3. TCP Enhancements for satellite IP networks
   - TCP spoofing: router processing load increase and end-to-end IP security issue
   - Link layer protocol performance
   - Bandwidth asymmetry: throughput performance and traffic burstiness
   - TCP enhancements for real-time applications

4. Return channel protocols for access networks
   - CDMA based return channel access protocol for mobile IP applications
5. Interoperability
   - Interoperability between hybrid satellite and terrestrial networks
   - Satellite interface definitions and interworking functions
6. Standardization
   - QoS for IP over satellite
   - MAC protocols
   - Performance of DVB-RCS
   - Encoding techniques e.g., turbo coding
   - Coordination among ITU-T, ITU-R, IETF, and ETSI/DVB-RCS
   - Extension of the standards research to onboard processing and switching systems

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Appendix 1 A List of Acronyms

AAA	Authentication, Authorization and Accounting
AAL	ATM Adaptation Layer
ABR	Available Bit Rate
ACK	Acknowledgment
ACTS	Advanced Communications Technology Satellite
ADSL	Asymmetric Digital Subscriber Line
AF	Assured Forwarding
AFSATCOM	Air Force Satellite Communications
AH	Authentication Header
AINI	ATM Inter-Network Interface
ANOVA	Analysis of Variation
AR	ACK Reconstruction
ARPA	Advanced Research Projects Agency
ATM	Asynchronous Transfer Mode
BB	Bandwidth Broker
BER	Bit Error Rate
BDP	Bandwidth Delay Product
B-ICI	B-ISDN Inter Carrier Interface
B-ISDN	Broadband Integrated Services Digital Network
BT	Burst Tolerance
CAC	Connection Admission Control
CATV	Community Antenna Television
CBR	Constant Bit Rate
CBQ	Class Based Queuing
CCR	Current Cell Rate
CCSDS	Consultative Committee of Space Data Systems
CDD	Content Delivery Distribution

CDMA	Code Division Multiple Access
CDV	Cell Delay Variation
CDVT	Cell Delay Variation Tolerance
CER	Cell Error Ratio
CFDAMA	Combined Free/Demand Assignment Multiple Access
CLP	Cell Loss Priority
CLR	Cell Loss Ratio
CM	Cable Modem
CMR	Cell Misinsertion Rate
CMTS	Cable Modem Termination System
COPS	Common Open Policy Service
CPE	Customer Premises Equipment
CRMA	Code Reuse Multiple Access
CRRMA	Combined Reservation Multiple Access
CTD	Cell Transfer Delay
DACK	Duplicate Acknowledgement
DAMA	Demand Assignment Multiple Access
DOCSIS	Data over Cable Service Interface Specification
DoD	Department of Defense
DSCP	DiffServ Code Point
DSCS	Defense Satellite Communication Systems
DSL	Digital Subscriber Line
DTH	Direct-to-Home
DVB	Digital Video Broadcast
DVB-RCS	Digital Video Broadcast - Return Channel via Satellite
DVB-S	Digital Video Broadcasting via Satellite
ECN	Explicit Congestion Notification
EF	Expedited Forwarding
EHF	Extremely High Frequencies
EPD	Early Packet Discard
ER	Explicit Rate
ERICA	Explicit Rate Indicator Algorithm
ESP	Encapsulated Security Payload
ETSI	European Telecommunications Standards Institute
FCC	Federal Communications Commission
FDD	Frequency Division Duplex
FDMA	Frequency Division Multiple Access
FEC	Forward Error Correction
FECN	Forward Explicit Congestion Notification

FIFO	First-In First-Out
FODA	FIFO Ordered Demand Assignment
FRR	Fast Retransmit and Recovery
FQ	Fair Queuing
FSS	Fixed Satellite Service
GCRA	Generic Cell Rate Algorithm
GEO	Geostationary Earth Orbit
GFR	Guaranteed Frame Rate
GPRS	Generic Packet Radio Services
GS	Guaranteed Services
GSM	Global Satellite for Mobile Communications
GSO	Geostationary Orbit
GTS	General Traffic Shaping
GW	Gateway
HDTV	High Definition Television
HRP	Hypothetical Reference Path
HTTP	Hyper Text Transfer Protocol
IEEE	Institute of Electronics and Electrical Engineers
IETF	Internet Engineering Task Force
IP	Internet Protocol
IPDV	IP Packet Delay Variation
IPER	IP Packet Error Ratio
IPLR	IP Packet Loss Ratio
IPSec	IP Security
IPTD	IP Packet Transfer Delay
ISAKMP	Internet Security Association Establishment and Key Management Protocol
ISL	Inter-Satellite Link
ISMA	Inhibit Sense Multiple Access
ISO	International Standard Organization
ISP	Internet Service Provider
IST	Interactive Satellite Terminals
ITU-R	International Telecommunication Union – Radio Communication
ITU-T	International Telecommunication Union – Telecommunications
IW	Initial Window
LAN	Local Area Network
LDP	Label Distribution Protocol
LEO	Low Earth Orbit
LLC	Logical Link Control

LSP	Label Switched Path
LSR	Label Switched Routers
MAC	Media Access Control
MAMA	Multiple ALOHA Multiple Access
MAMT	Multiple Accounting, Multiple Threshold
MAN	Metropolitan Area Network
MAST	Multiple Accounting, Single Threshold
MBS	Maximum Burst Size
MCR	Minimum Cell Rate
MF-TDMA	MultiFrequency-Time Division Multiple Access
MEO	Medium Earth Orbit
MILSTAR	Military Strategic and Tactical Relay
MP	Measurement Point
MPEG	Moving Picture Expert Group
MPLS	MultiProtocol Label Switching
MSS	Mobile Satellite Service (chapter 2)
MSS	Maximum Segment Size (chapter 3)
NASA	National Aeronautical and Space Administration
NCC	Network Control Centre
NCR	Network Clock Reference
NCS	Network Control Station
NGSO	Non-Geostationary Orbit
N-ISDN	Narrow-band Integrated Digital Network
NMS	Network Management System
NPC	Network Parameter Control
nrt-VBR	non-real-time Variable Bit Rate
OSI	Open System Interconnect
OSPF	Open Shortest Path First
PCR	Peak Cell Rate
PDP	Policy Distribution Point
PEP	Performance Enhancing Proxies
PHB	Per-Hop Behavior
PILC	Performance Implications of Link Characteristics
PKIX	Public Key Infrastructure
PMTU	Path Maximum Transmission Unit
PNNI	Private Network-Network Interface
PODA	Priority-Oriented Demand Assignment
POP	Point-of Presence
PSK	Phase Shift Keying

PUNI	Public User Network Interface
QAM	Quadrature Amplitude Modulation
QoS	Quality of Service
QPSK	Quadrature Phase Shift Keying
RCST	Return Channel Satellite Terminal
RFC	Request For Comments (IETF Document)
RM	Resource Management
RRR	Round-Robin Reservation
RSVP	Resource Reservation Protocol
RTT	Round Trip Time
rt-VBR	real-time Variable Bit Rate
SACK	Selective Acknowledgment
SAMA	Spread Aloha Multiple Access
SAMT	Single Accounting, Multiple Threshold
SAOC	Spread ALOHA One Code
SAOLC	Spread ALOHA One Long Code
SAST	Single Accounting, Single Threshold
SCC	Satellite Control Centre
SCPC	Single Channel per Carrier
SCPS-TP	Space Communications Protocol Specifications - Transport Protocol
SCR	Sustained Cell Rate
SCS	Switching Control System
SCTP	Stream Controlled Transmission Protocol
SD	Selective Drop
SECBR	Severely Errored Cell Block Ratio
SIMP	Satellite IMP
SIT	Satellite Interactive Terminal
SLA	Service Level Agreement
SNACK	Selective Negative Acknowledgement
SNMP	Simple Network Management Protocol
SOHO	Small Office/ Home Office
SONET	Synchronous Optical Network
SRS	Switching Router System
SS7	Signaling System 7
SSL	Secure Socket Layer
TBTP	Terminal Burst Time Plan
TCP	Transmission Control Protocol
TDD	Time Division Duplex
TDM	Time Division Multiplexing

TDMA	Time Division Multiple Access
TIU	Terminals Interface Unit
TLS	Transport Layer Security
TOS	Type of Service
TWTA	Traveling-Wave Tubes Amplifier
UBR	Unspecified Bit Rate
UDP	User Datagram Protocol
UNI	User Network Interface
UPC	Usage Parameter Control
UT	User Terminal
VC	Virtual Connection
VOD	Video on Demand
VPN	Virtual Private Network
VSAT	Very Small Aperture Terminal
VS/VD	Virtual Source/Virtual Destination
WAN	Wide Area Network
WAP	Wireless Application Protocol
W-LAN	Wireless LAN
WPM	Window Prediction Mechanism
W-WAN	Wireless WAN
WWW	World Wide Web

Appendix 2.A Satellite TCP/IP

This section provides the baseline TCP protocol, link characteristics affecting TCP, TCP enhancements including PEP and their evaluation in brief.

2.A.1 Baseline TCP Protocol

Several flavors of TCP have been proposed and/or implemented over the years. TCP specifies four congestion control algorithms: (a) Slow start (b) Congestion avoidance (c) Fast retransmit (d) Fast recovery [97]

2.A.1.1 Slow Start and Congestion Avoidance

These algorithms are operated by two sender-side variables and one receiver-side variable. The sender maintains the congestion window (cwnd) and the slow start threshold (ssthresh) while the receiver advertises its maximum receive window (rwnd). The sender ensures that at no time does the amount of in-flight data exceed the minimum of cwnd and rwnd. After a TCP connection has been established, slow start is activated and the value of cwnd is set to 1. The sender transmits a single TCP segment and increments cwnd by 1 when the segment has been acknowledged. The sender is then allowed to transmit two segments. The variable cwnd is incremented for each received ACK, providing exponential growth during the slow start phase (in reality, most receivers implement a delayed ACK approach, acknowledging every other segment which slows the growth function for cwnd).[274] The slow start threshold ssthresh is typically initialized to either 64 KB or the receiver's advertised window and defines the point at which the sender transitions from slow start to congestion avoidance. When cwnd is smaller than ssthresh, the slow start phase is in effect. Otherwise, the sender is in the congestion avoidance phase. During congestion avoidance, the growth of cwnd is slowed to a linear pace. Upon each received ACK, the sender increases cwnd by segsize*segsize/

cwnd, which increments cwnd by approximately 1 every RTT (or 1 every 2 RTTs with delayed ACKs). This behavior continues until the connection is terminated or a loss is detected. The sender determines that a loss has occurred through either a timeout or reception of duplicate ACKs. When a retransmission timer expires, the sender sets the value of ssthresh to cwnd/2 (but no less than 2) and then reduces the value of cwnd to 1. The lost segment is retransmitted and the slow start algorithm is then used to increase cwnd up to the new value of ssthresh, whereby congestion avoidance is used thereafter. When the sender has received 3 duplicated acknowledgements, the fast retransmit algorithm, described below, is invoked.

2.A.1.2 Fast Retransmit and Fast Recovery

The receiver only acknowledges the last received in-order segment. When an out of order segment is received, the receiver generates a duplicate ACK informing the sender that segments up to the last in-order segment were correctly received. The sender may receive duplicate ACKs either because a segment was lost or because the network did not deliver them in order. Since it has no way of distinguishing between the two cases, the sender waits for three duplicate ACKs before assuming that a loss has occurred (this is 4 identical ACKs with no other further segment ACKs arriving in between). The fast retransmit algorithm resends what it believes to be the missing segment at this time, without waiting for the retransmission timer to expire. The value of ssthresh is set to cwnd/2 (but no less than 2). In the older Tahoe version, TCP would then set cwnd to 1 and begin slow start. However, since the receiver has generated duplicate ACKs, it must have been receiving segments from the sender so the congestion within the network cannot be that severe. Therefore, it does not make sense to enter the slow start phase at this point, as the sender does under a retransmission timeout. Instead, new segments are clocked out with incoming duplicate ACKs, which defines the fast recovery algorithm. The value of cwnd is inflated to ssthresh + 3, accounting for the number of out of order segments that have already been received. Thereafter, each time a duplicate ACK is received, cwnd is inflated by 1 and a new segment is transmitted if allowed by the values of cwnd and rwnd. When the retransmitted segment is finally acknowledged, cwnd is deflated to ssthresh (which is half the old value of cwnd) and the fast recovery phase is terminated. The congestion avoidance phase is entered and the growth of cwnd is linear.

2.A.2 TCP Enhancements for Satellite Networks

In this section, the TCP enhancements that appear most promising for satellite networks are considered.

2.A.2.1 Enhancements to Slow Start and Congestion Avoidance

Ideally, the sender would like to increase the value of cwnd as quickly as possible to maximize throughput. The duration of slow start can be a limitation to performance, as seen above. The enhancements in this section have been proposed to enable quicker growth of the congestion window.

2.A.2.2 Large Initial Window

The initial phase of a TCP connection begins with a value of cwnd equal to 1 at the sender's side (although the standards track does allow an initial window of 2 segments).[97] As the slow start phase progresses, cwnd is increased based upon the incoming ACKs. The Large Initial Window experimental proposes to begin with a larger value of cwnd, allowing more segments to be transmitted during the first RTT of the connection.[97, 98] This in turns speeds up the number of ACKs received at the sender, allowing the congestion window to increase at a faster rate. The initial value of cwnd is defined as follows:

$$\text{cwnd} = \min\{4*\text{SMSS}, \max\{2*\text{SMSS}, 4380 \text{ bytes}\}\},$$

where SMSS defines the sender's maximum segment size. This formula allows the sender to transmit up to 4 segments initially, which can be particularly beneficial for short-lived connections such as those used for http transactions.

2.A.2.3 Delayed ACKs after Slow Start

Receivers do not typically ACK every incoming segment, but rather ACK every other segment. While this can be advantageous on asymmetric links with low bandwidth in the return direction, this strategy slows the growth of the congestion window and can limit throughput, since new transmissions are clocked out based on received acknowledgements. Delayed ACKs after slow start is a strategy designed to counter the effects of delayed ACKs on long delay paths.[99] The receiver acknowledges every incoming segment during the slow start phase, and returns to the delayed ACK algorithm during congestion avoidance. Thus, the slow start phase is shortened since the sender's congestion window reaches its maximum value more quickly. While throughput generally improves, a moderate increase in loss may occur since the sender is more aggressively injecting segments into the network. Furthermore, an implementation may be difficult to achieve since it is not clear how the receiver will know when the sender has terminated the slow start phase. Explicit notification by the sender has been suggested, although the details are still undetermined.

2.A.2.4 Byte Counting

This is another enhancement designed to counter the effects on throughput that delayed ACKs can cause. Byte counting proposes to increase the congestion window based on the number of transmitted bytes acknowledged by incoming ACKs, rather than on the number of ACKs received.[97, 98] In this way, the congestion window is opened according to the amount of data transmitted, rather than on the receiver's ACK interval. For long delay paths in particular, this scheme has been shown to reduce the amount of time it takes to reach the optimal congestion window size.

2.A.2.5 Fast Retransmit and Fast Recovery

The TCP Reno algorithm does not generally recover well in the presence of multiple segment losses within one window's worth of data. Only the first lost segment is recovered via the fast retransmit algorithm, while the rest normally must wait for a retransmission timeout before recovery (the fast recovery phase is terminated after the first retransmission is acknowledged). Most of the enhancements in this section improve upon the fast retransmit/fast recovery mechanism when multiple losses are incurred. These ideas are particularly important in large bandwidth*delay product networks where a large number of segments are in-flight at any given time, increasing the probability of multiple segment loss within a window.

2.A.2.6 TCP NewReno

In the absence of a selective acknowledgement mechanism, the sender has limited information regarding lost segments. This proposal modifies the fast retransmit/fast recovery algorithms of TCP Reno to allow recovery from multiple segment losses. An ACK for a retransmitted segment will normally acknowledge all segments transmitted prior to entering fast retransmit (loss detection from duplicate ACKs). However, if there were multiple segments lost, then the ACK will only acknowledge some of the segments that were in-flight. This type of acknowledgement for a retransmitted segment is known as a partial acknowledgement. TCP NewReno [101] is a modification to the fast retransmit/fast recovery algorithms in response to partial acknowledgements. When a partial ACK is received, the first unacknowledged segment is retransmitted and the fast recovery algorithm continues (duplicate ACKs cause an increase in cwnd by 1 and a transmission of a new segment if allowed by cwnd and rwnd). This is an aggressive policy since the first unacknowledged segment is retransmitted without waiting for three duplicate acknowledgements or a retransmission timeout. It represents an important consideration since such duplicates may never arrive at the sender because new segments were not sent to the receiver. Without this modification, the sender would likely have to wait for a retransmission timeout, which impacts throughput severely. TCP NewReno is limited to the retransmission of at most one dropped segment per RTT, since the second loss is not discovered until the first retransmission is acknowledged.

2.A.2.7 TCP SACK

TCP SACK [102, 103] generally refers to a selective acknowledgement option at the receiver and a selective retransmit mechanism at the sender. The receiver has the ability to acknowledge all segments that have been received, allowing the sender to retransmit only those segments that have been lost. A flag in the SYN segment is set to indicate that the SACK option should be used during the connection. The receiver uses the TCP options portion of the ACK header to indicate the (possibly out of order) received segments. The segments are acknowledged in non-contiguous groups of blocks. TCP SACK behaves like TCP Reno except when multiple segments are lost during a single window of data. The difference is, during fast recovery, the sender has complete information about which segments to retransmit. TCP SACK decouples the identity of the segments requiring retransmission from the ACK clock that it maintains. The sender will maintain a flag for each segment in its retransmission buffer. When an acknowledgement with a SACK option is received, the sender marks each segment that has been (S)ACK'd. If a retransmission is required later on (due to duplicate ACKs or a retransmission timeout), the sender can skip over those segments that have been marked, avoiding unnecessary retransmissions. TCP SACK is an extremely effective enhancement to TCP Reno and is better able to recover from multiple segment losses than TCP NewReno. The latter algorithm, however, does not require any changes to the receiver, whereas the former does. An interesting combination of both TCP SACK and TCP NewReno has shown promising results over the use of either one independently.

2.A.2.8 TCP Vegas

TCP Vegas reduces the time to retransmit a segment when waiting for three duplicate ACKs.[100] It also provides a mechanism for retransmission that avoids a long timeout when duplicate ACKs are not forthcoming (due to a lack of transmissions in-flight after the loss). For each segment, Vegas computes the RTT based on the recorded system time at transmission and the ACK time of arrival. When a duplicate ACK is received because a segment was lost, the computed RTT of the lost segment is checked to see if a timeout should have occurred. If so, the segment is retransmitted without waiting for more duplicate ACKs. Recall that the purpose of waiting for three duplicate ACKs in Reno is to avoid a retransmission simply because segments were reordered in the network. Because of the RTT calculation, Vegas does not violate this policy. After a retransmission, a second check based on calculated RTTs is used to retransmit any additional lost segments that may have occurred. This check is used only on the first or second non-duplicate acknowledgement received after a retransmission.

2.A.2.9 Other TCP Enhancements

This section describes enhancements to the TCP protocol that do not necessarily fit into the congestion control algorithms of slow start, congestion avoidance, fast retransmit, and fast recovery.

Window Scaling

The standard maximum TCP window size is 2^{16} = 65 KB, which presents a severe limitation on the achievable throughput for a long delay connection. The throughput is limited to the ratio of the window size to the RTT, which defines the maximum amount of in-flight data allowed. For a connection traversing a satellite hop, the throughput will be limited to approximately 900 kbps using this window size. Notice that a larger window size is also needed to realize the potential of high bandwidth links traversing low delay paths (e.g. fiber optic). With RFC 1323 [106], the window size can achieve an effective size of 2^{30} = 1.1 GB through the use of scaling the existing 16 bit field. A satellite connection can now theoretically achieve a throughput upwards of 15 Gbps.

TCP for Transactions

TCP for Transactions attempts to reduce the connection handshaking latency for most connections from two to one RTT.[107] This reduction in latency can be significant for short web and satellite links.

2.A.2.10 Forward Error Correction (FEC)

TCP interprets all losses are due to congestion. In case of losses due to transmission errors, TCP does not need to reduce its congestion window. For TCP to operate efficiently, nearly all transmission losses should be eliminated. Several forward error correction (FEC) techniques have been proposed and implemented to improve the BER performance on satellite links.[24 , 80]

Explicit Congestion Notification (ECN)

Explicit Congestion Notification allows routers to inform TCP senders about imminent congestion without dropping segments.[110] In case of congestion, routers mark IP packets with a special tag and the TCP receiver echoes the congestion information back to the sender in an ACK packet. ECN mechanism can be used in satellite networks to differentiate between congestion losses and error losses. ECN also improves the performance of TCP connections.

TCP/IP Header Compression

With TCP/IPv4 headers of at least 40 bytes and TCP/IPv6 headers of at least 60 bytes, there is a significant amount of overhead found in each TCP segment. Some of this information is constant or slowly changing during a connection and need not be repeated with each transmitted segment. This includes the source and destination addresses, port numbers, and other static information. Header compression [275, 276] typically reduces TCP/IPv4 headers from 40 to 3 – 5 bytes. In its most general form, header compression involves the process of intermittently transmitting a full header, with subsequent compressed headers referencing the contents of the previous full header. A sender may send a full header when one or more of the header fields must be updated. Since compression happens at the link layer, routing is not affected since the header is expanded before being passed to the IP layer. In addition to improving throughput of user data, compression can reduce segment loss due to bit errors since the probability of a header error is reduced.

2.A.3 Space Communications Protocol Specifications-Transport Protocol

Space Communications Protocol Specifications-Transport Protocol (SCPS-TP) was developed by the MITRE corporation and later was standardized by Consultative Committee of Space Data Systems (CCSDS) to account for the “stressed environment” i.e., space environment with delay, and multiple sources of data loss.[136] SCPS-TP uses a congestion control algorithm that does not depend on packet loss as a way to signal congestion in the network. SCPS-TP can react to explicit signals of the two sources of packet loss (congestion, and the satellite disappearing over the horizon). The ability for SCPS-TP to tailor its response to the nature of the loss allows for better network utilization and better end-to-end performance without harming the overall network stability.

Among several experiments conducted to measure and verify the performance of SCPS-TP, were two “bent-pipe” tests and one onboard test in which the spacecraft hosted the SCPS software. Results proved that SCPS-TP is well suited to the long-delay, potentially high bit-error rate environment of satellites. Using options such as Header Compression, Selective Negative Acknowledgement (SNACK), and TCP Timestamps produces varying effects under different conditions.

Appendix 3.A Delay Analysis Model for GEO and LEO Satellite Networks

Interactive voice requires very low delay (ITU-T specifies a delay of less than 400 ms to prevent echo effects) and delay variation (up to 3 ms specified by ITU-T). GEO systems have a high propagation delay of at least 250 ms from ground terminal to ground terminal. If two GEO hops are involved, then the inter-satellite link delay could be about 240 ms. Other delay components are additionally incurred, and the total end-to-end delay can be higher than 400 ms. Although the propagation and inter-satellite link delays of LEOs are lower, LEO systems exhibit high delay variation due to connection handovers, satellite and orbital dynamics, and adaptive routing. Non-interactive voice/video applications are real-time applications whose delay requirements are not as stringent as their interactive counterparts. However, these applications also have stringent jitter requirements. As a result, the jitter characteristics of GEO and LEO systems must be carefully studied before they can service real time voice-video applications.

The advantage of continuous visibility and fixed geometry has made GEO satellites one of the principle means of distribution of television, telephone and data communications throughout the world. Most of the commercial telecommunications satellites currently in operation are in GEO orbit. A GEO satellite in GEO orbit can provide service to a very large area. If the look angle of the ground station is restricted to a minimum of 20^{o} above the horizon, the surface area capable of being served by one satellite corresponds to about 135 million square kilometers, or 26% of the total surface of the Earth. Only 3 GEO satellites are required to provide service to all tropical and temperate zones of the Earth. Due to their position above the equator, GEO satellites cannot provide service beyond 81 degrees North or South latitudes (76 degrees latitude if the minimum ground station is restricted to an elevation of 5^{o} above the horizon).

For non-geosynchronous satellites (MEO and LEO), to avoid interruptions at the end of each pass, data buffering must be employed at the ground station or the next visible satellite must be acquired and tracked before communications are handed over from the setting satellite. The first solution requires large data buffers which introduce data transmission delays. The latter solution requires two independently steered earth station antennas and duplicate receiving equipment which increases the cost of the earth station.

Compared to a GEO satellite, a LEO satellite, because of its lower altitude and closer horizon, can provide service only to a small area at any given time. For example, for Teledesic, the surface area capable of being served by one satellite at a given time corresponds to about 3% of the total surface of the Earth for the same 20^o minimum ground station elevation look angle. For Teledesic's advocated 40^o minimum ground station look angle, the surface area capable of being served drops to about 1% of the Earth's surface. Inter-satellite links to relay the transmission from the source to the destination are required to provide anything beyond local service. An advantage of LEO satellites operating in high inclination orbits is that they can provide service to the near-polar regions.

As the satellites go in and out of view of the ground stations at both ends of the transmission path, the choice of intermediate relay satellites also must be changed to maintain a continuous connectivity between ground stations. Inter-satellite link communications must be periodically handed over from one satellite to another to maintain this connectivity. To avoid data loss during these handovers, several independent inter-satellite link units must be utilized and data buffering on board the satellites must be employed as well. This buffering introduces additional data latency.

Because of the constantly changing network geometry, the total path length, and corresponding transmission delay, is constantly varying. Handovers from one satellite to another cause discrete step changes in the path length and in transmission delay. Great care must also be employed in the design of routing and handover algorithms to avoid intermittent data loss and throughput degradation due to the retransmission of lost packets. Routing tables at the earth stations and at all of the relay satellites must be frequently updated to reflect the connection changes and maintain the continuity of communications. To avoid data loss during handovers, generous data buffering must be employed not only at the ground station but also at each of the relay satellites. This buffering causes additional data latency.

Thus, while LEO satellite networks have less propagation delay than GEO networks, they suffer from additional delays introduced by data buffering at the ground stations and at relay satellites which is required to compensate for unavoidable changes in network geometry and frequent satellite handovers.

The performance of TCP/IP file transfer applications is throughput dependent and has very loose delay requirements. As a result, both GEOs and LEOs with sufficient throughput can meet the delay requirements of file transfer applications. It is often misconstrued that TCP is throughput limited over GEOs due to the default TCP window size of 64K bytes. The TCP extend windows option allows the TCP window to increase beyond 64K bytes and results in the usage of the available capacity even in high bandwidth GEO systems. The efficiency of TCP over GEO systems can be low because the TCP window based flow control mechanism takes several round trips to fully utilize the available capacity. The large round trip time in GEOs results in capacity being wasted during the ramp-up phase. To counter this, the TCP spoof protocol is being designed that splits the TCP control loop into several segments. However this protocol is currently incompatible with end-to-end IP security protocols. Several other mechanisms are being developed to mitigate latency effects over GEOs as discussed in Chapter 2.

The TCP congestion control algorithm inherently relies on round trip time (RTT) estimates to recover from congestion losses. The TCP RTT estimation algorithm is sensitive to sudden changes in delays as may be experienced in LEO constellations. This may result in false timeouts and retransmits at the TCP layer. More sophisticated RTT measurement techniques are being developed for TCP to counter the effects of delay jitter in LEO systems.

3.A.1 End-to-End Delay Model

The end-to-end delay experienced by a data packet traversing a satellite network is the sum of the following elements:

- Buffering and format conversion delay at the originating earth station
- Transmission delay from the earth station to the first satellite in the path (the currently visible satellites at the origin)
- On board switching, processing and buffering delays at each of the satellites in the transmission path
- Transmission delays between each pair of satellites in the transmission path (i.e. inter-satellite link delays)
- Transmission delay from the last satellite in the path (the currently visible satellite at the destination to the destination earth station)
- Buffering and format conversion delays at the destination earth station

The total end-to-end delay experienced by a data packet is the cumulative sum of all of the above plus the delay of the terrestrial network connecting the end users to the earth stations of the satellite network. In the following sections is discussed the delay contributions only due to the satellite network. Delay variations in both GEO and LEO satellite networks are caused by buffering and on-board processing. LEO satellite networks inherently suffer from unavoidable additional delay variations, both of a continuous and discrete nature, which are introduced by the dynamically changing network geometry.

Delay model of a satellite network can be used to estimate the end-to-end delay of GEO, MEO and LEO satellite networks.

The end-to-end delay (D) experienced by a data packet traversing the satellite network is the sum of the transmission delay (t_t), the uplink (t_{up}) and downlink (t_{down}) ground segment to satellite propagation delays, the inter-satellite link delay (t_i), the on-board switching and processing delay (t_s) and the buffering delay (t_q). The inter-satellite, on-board switching, processing and buffering delays are cumulative over the path traversed by a connection. In this model, only the satellite component of the delay is considered. The total delay experienced by a packet is the sum of the delays of the satellite and the terrestrial networks. This model does not incorporate the delay variation experienced by the cells of a connection. The delay variation is caused by orbital dynamics, buffering, adaptive routing (in LEOs) and on-board processing. Quantitative analysis of delay jitter in satellite systems is beyond the scope of this study. The end-to-end delay (D) is given by:

$$D = t_t + t_{up} + t_i + t_{down} + t_s + t_q \tag{3.A.1}$$

3.A.1.1 Transmission delay

The transmission delay (t_t) is the time taken to transmit a single data packet at the network data rate.

$$t_t = \frac{Packet\_length(l)}{Data\_rate(r)} \tag{3.A.2}$$

For broadband networks with high data rates, the transmission delays are negligible in comparison to the satellite propagation delays. For example, a 9180 byte TCP packet is transmitted in about 472 microseconds. This delay is much less than the propagation delays in satellites.

3.A.1.2 Propagation delay

The propagation delay for the cells of a connection is the sum of the following three quantities:

- The source ground terminal to source satellite propagation delay (t_{up})
- The Inter-satellite link propagation delays (t_i)
- The destination satellite to destination ground terminal propagation delay (t_{down})

The uplink and downlink satellite-ground terminal propagation delays (t_{up} and t_{down} respectively) represent the time taken for the signal to travel from the source ground terminal to the first satellite in the network (t_{up}), and the time for the signal to reach the destination ground terminal from the last satellite in the network (t_{down}).

$$t_{up} = \frac{Source\_Satellite\_Distance(d1)}{Speed\_of\_light(c)} \tag{3.A.3}$$

$$t_{down} = \frac{Destination\_Satellite\_Distance(d2)}{Speed\_of\_light(c)} \tag{3.A.4}$$

The inter-satellite link delay (t_i) is the sum of the propagation delays of the inter-satellite links (ISLs) traversed by the connection. Inter-satellite links (crosslinks) may be in-plane or cross-plane links. In-plane links connect satellites within the same orbit plane, while cross-plane links connect satellites in different orbit planes. In GEO systems, ISL delays can be assumed to be constant over a connection's lifetime because GEO satellites are almost stationary over a given point on the earth, and with respect to one another. In LEO constellations, the ISL delays depend on the orbital radius, the number of satellites-per-orbit, and the inter-orbital distance (or the number of orbits). Also, the ISL delays change over the life of a connection due to satellite movement and adaptive routing techniques in LEOs. As a result, LEO systems can exhibit a high variation in ISL delay.

$$t_i = \frac{\sum l}{c} \qquad \text{where } 1 = \text{Inter Satellite Link distance} \tag{3.A.5}$$

3.A.1.3 Switching and processing delays

The data packets may incur additional delays (t_s) at each satellite hop depending on the amount of on-board switching and processing. For high data rate networks with packet/cell switching, switching and processing delays are negligible compared to the propagation delays.

3.A.2 GEO Propagation Delay Model

For the GEO network 4 satellites, equally spaced around the equator, located at 0^o, 90^o, 180^o East and 90^o West longitude are assumed.

3.A.2.1 GEO Propagation Delay Elements

The propagation delay between the earth station and the satellite is a function of the line-of-sight path length, which is a function of the satellite orbit altitude and the "look angle" (i.e. pointing elevation) of the earth station antenna. For reasons of atmospheric attenuation and providing unobstructed view of the satellite, ground stations of typical satellite networks are preferentially located where the look angle is 20^o or better above horizontal.

The antenna look angles and propagation delays from selected cities to each of the satellites in the assumed network are shown in Tables 3.A.1 and 3.A.2.

Table 3.A.1: Ground station antenna elevation angle for each satellite in the sample GEO constellation (dashed lines indicate satellite is not in view of the ground station)

	0^o E satellite	90^o E satellite	180^o E satellite	90^o W satellite
New York	3.6^o	-	-	40.21^o
Tokyo	-	23.67^o	30.85^o	-
Paris	33.87^o	-	-	-
London	31.02^o	-	-	-
Seoul	-	31.94^o	20.37^o	-
Los Angeles	-	-	14.73^o	40.24^o
Toronto	-	-	-	38.54^o
Mexico City	-	-	0.51^o	64.92^o
Sydney	-	15.19^o	39.98^o	-
Chicago	-	-	-	41.60^o

Table 3.A.2: Propagation delay (ms) to each satellite in the sample GEO constellation (numbers in parenthesis correspond to path below the 20^o earth station pointing angle elevation limit)

	0^o E satellite	90^o E satellite	180^o E satellite	90^o W satellite
New York	(138)	-	-	126
Tokyo	-	131	129	-
Paris	128	-	-	-
London	128	-	-	-
Seoul	-	128	132	-
Los Angeles	-	-	(134)	126
Toronto	-	-	-	126
Mexico City	-	-	(139)	121
Sydney	-	(134)	126	-
Chicago	-	-	-	126

The propagation delay between two GEO satellites is a function of the line-of-sight path between the satellites, which is a function of the difference in longitude between the satellites. Table 3.A.3 lists the total one-way propagation delay as a function of the number of satellites in the transmission path for the sample GEO constellation with 4 satellites evenly spaced around the equatorial plane.

Table 3.A.3: One way inter-satellite propagation delay (ms) as a function of number of GEO satellites in the transmission path

Number of Satellites in the path	Cumulative Inter-Satellite Link Distance (km)	Cumulative Inter-Satellite Link Delay (ms)
1	0	0
2	59,629	199
3	119,258	398

3.A.2.2 GEO Route Calculation

It is assumed that each GEO satellite is equipped with two inter-satellite links, one connecting to the next adjacent satellite to the East and the other to the next adjacent satellite to the West. To calculate the route between city pairs, one city as the source and the other as the destination are assigned. Then all the satellites that provide a minimum ground station look angle elevation of 20^o when viewed from the source city are found. This process is repeated at the destination city. If any satellites appear on both lists, the connection can be established using only a single satellite. If more than one satellite appears on both lists, the ones having the lowest elevation to either city are discarded in turn until only one satellite remains.

If no satellite appears on both lists, the next adjacent satellite to the source which is closest to the destination city is selected. If this satellite appears in the list corresponding to the destination city, routing is determined. If not, the process is repeated with the adjacent satellite, continuing in the direction of the destination city until the selected satellite appears in the destination city's list.

3.A.2.3 GEO Propagation Delay Summary

Table 3.A.4 indicates the resulting total propagation delay between city-pairs for the assumed 4 satellite GEO constellation. In a GEO network, the longest propagation delay encountered is 652 milliseconds in the London-Sydney and Paris-Sydney routes.

Table 3.A.4: Total propagation delay (ms) between city-pairs for the sample 4 satellite GEO constellation

	New York	Tokyo	Paris	London	Seoul	Los Angeles	Toronto	Mexico City	Sydney	Chicago
New York	252									
Tokyo	454	258								
Paris	453	458	256							
London	453	458	256	256						
Seoul	457	259	455	455	256					
Los Angeles	252	454	453	453	457	252				
Toronto	252	454	453	453	457	252	252			
Mexico City	247	449	448	448	452	247	247	242		
Sydney	451	255	652	652	258	451	451	446	252	
Chicago	256	454	453	453	457	252	252	247	451	252

Table 3.A.5 indicates the number of satellites in the transmission path for the same city-pairs and GEO constellation. In this example, only two city pairs (Paris-Sydney and London-Sydney) required relaying by more than 2 satellites.

Taulukko 1. Table 3.A.5: Number of satellites in the transmission path between city-pairs for the sample 4 satellite GEO constellation

	New York	Tokyo	Paris	London	Seoul	Los Angeles	Toronto	Mexico City	Sydney	Chicago
New York	1									
Tokyo	2	1								
Paris	2	2	1							
London	2	2	1	1						
Seoul	2	1	2	2	1					
Los Angeles	1	2	2	2	2	1				
Toronto	1	2	2	2	2	1	1	1		
Mexico City	1	2	2	2	2	1	1	1		
Sydney	2	1	3	3	1	2	2	2	1	
Chicago	1	2	2	2	2	1	1	1	2	1

GEO systems are at an altitude of about 36,000 km above the equator. For GEOs, t_{up} and t_{down} can be approximated to about 125 ms each for ground terminals near the equator. Inter satellite propagation delays are stable and depend on the number of satellites in the constellation. As few as three GEOs are sufficient to cover the earth. Table 3.A.6 lists the inter-satellite link distances and propagation delays for GEO systems with N satellites evenly spaced around the equatorial plane. For ground terminals farther away from the equator, the propagation delay from ground station to ground station through a single satellite is about 275 ms.

Table 3.A.6: GEO Inter Satellite Delays

Number of Satellites (N)	Inter-Satellite Link Distance (km)	Inter-Satellite Link Delay (ms)
3	73,030	243
4	59,629	199
5	49,567	165
6	42,164	141
7	36,589	122
8	32,271	108
9	28,842	96
10	26,059	87
11	23,758	79
12	21,826	73

3.A.3 LEO Propagation Delay Model

In this section, a simple model for the propagation delays of LEO systems is given. This model calculates the total propagation delay from source ground terminal to the destination ground terminal, through the LEO network. A LEO geometry and network topology model is used to determine the total number of satellites and the total propagation delay for communication between two ground points. The model uses the following information:

– Number of orbit planes
– Number of satellites per orbit plane
– Satellite altitude
– Orbit plane inclination angle[3]
– Ground terminal coordinates

3.A.3.1 LEO Orbital Model

Low Earth Orbit (LEO) communication satellites are arranged in a constellation in the following manner. The satellites are organized into a set of p orbit planes, each containing n satellites. The orbits are assumed to be circular and to have a common, high inclination angle (α). The inclination angle, combined with the electronic-horizon reach of the satellites directly determines the range of latitudes for which the system can provide service. The satellites within a given orbit plane are evenly spaced by using a delta anomaly between in-plane satellite orbits:

3. Inclination angle is the angle made by the satellite orbital plane with the equatorial plane.

$$\text{delta anomaly } (\delta) = \frac{360^\circ}{n} \qquad \text{where } n = \text{number of satellites per plane} \quad (3.A.6)$$

The orbit planes are approximately evenly spaced about the earth polar axis by using a delta right ascension and a correction term between orbit planes:

$$\theta = \frac{180^\circ}{p} + \phi \qquad \text{where } \phi \text{ is RAcorrection and } p \text{ is number of orbit planes} \quad (3.A.7)$$

Spreading the right ascension of the orbit planes over 180 degrees means that the satellites in adjacent planes are roughly traveling in parallel, with the exception that between the last and first planes, the satellites are traveling in opposite directions. The interval between the last and first orbit plane is called the "seam" of the constellation. The fact that the satellites in the last and first orbit planes of the constellation are traveling in opposite directions means that any cross-plane links spanning the seam will have to change connectivity to a different satellite every few minutes. This will result in frequent handovers causing additional delays and delay variations.

The *RA correction* term in the previous formula is necessary for the following reason. LEO communication constellations use inclination angles of less than 90 degrees. Because of this, the last orbit plane tends to tilt in the opposite direction as the first orbit plane by roughly twice the complement of the inclination angle (i.e., $2 \times \alpha$). Without the correction term for θ, a "hole" results in the ground coverage of the constellation in the two areas of the earth where the seam orbit-planes are tilting away from each other. In the opposite hemisphere of the two holes, the seam orbit-planes are tilting towards each other, resulting in overlapping, or redundant, ground coverage. Trade-offs can be made between how much of the serviced latitude range will be provided continuous, uninterrupted service, and the amount of redundant coverage suffered. The model described here currently uses the following simple correction term.

$$\text{RA correction } \phi = \frac{1.5(90 - \alpha)}{p} \qquad (3.A.8)$$

The inter-plane satellites are phased by about one-half of the in-plane satellite spacing. This staggered phasing provides a more optimal and uniform coverage pattern, and maximizes inter-plane satellite distances near the extreme latitudes where the orbit planes cross. The current model uses the following delta inter-plane phasing.

$$\text{delta inter plane phasing} = 0.5x\delta + \phi x Sin(90 - \alpha) \qquad (3.A.9)$$

The model uses an Earth Centered Inertial (ECI) right -handed coordinate system. The first and second axes lie on the equatorial plane with the first axis aligned with the prime meridian. The third axis is aligned with the earth's polar axis pointing north. The first satellite of the first orbit plane (satellite(1,1)) is arbitrarily placed at latitude, longitude coordinates of (0,0), giving this satellite a position vector in ECI coordinates of (r, 0, 0), where r is the sum of the equatorial earth radius and the satellite altitude. The position vectors of the remaining satellites are obtained by an appropriate series of rotations of the satellite(1,1) position vector involving the angles described above.

3.A.3.2 LEO Route Calculation

This model is used to calculate routes and propagation delays across the satellite constellation. First the above procedure is used to create a constellation pattern providing continuous ground coverage for most of the latitude range covered by the constellation. Circular orbits with staggered inter-plane phasing are assumed as described above. Each satellite is assumed to have four crosslinks (inter-satellite links) providing connectivity to the in-plane satellites immediately leading and following, and to the nearest satellites in the two adjacent orbit planes—in navigational terms, these would be the fore, aft, port, and starboard satellites. Cross-plane connectivity constraints in the area of extreme latitudes or across the constellation seam (i.e., where satellites in the last orbit plane and the first orbit plane are traveling in opposite relative directions) are not considered. Anticipatory routing to reduce hand-offs and routing around congested paths is not considered.

The following simple algorithm is used to determine the route between two ground points through the satellite constellation. One of the ground points is designated as the source node and the other ground point is assigned as the destination node. The satellite nearest to the source node, and the satellite nearest to the destination node are first determined. This assumes minimal redundant satellite ground coverage, which is usually the case for LEO communication systems. Starting at the satellite nearest the source node, of the satellites with which it has connectivity, the satellite nearest to the destination node's satellite is selected. The process is repeatedly applied at each selected satellite, with backtracking precluded, until the destination node's satellite is reached. The algorithm then counts the number of satellites in the end-to-end, source-to-destination path. The distances between successive path-nodes, beginning at the source terminal and ending at the destination terminal, are computed, converted to link propagation delays by dividing by the speed of light, and accumulated to provide the path end-to-end propagation delay. The number of satellites in the route path and the total propagation delay are then reported. While the routing algorithm just described is strictly geometry based and only locally optimal, of the limited cases examined thus far, the results appear to be generally coincident with a globally optimal solution.

The model can also generate three-dimensional orthographic-projection displays showing satellite orbits, satellite positions, cross-links, ground terminal positions, path-links followed, and earth model from any desired viewing direction. Figure 3.A.1 shows an example path of a 6-plane, 11-satellites per plane LEO system path from Los Angeles to London. The associated configuration parameters are given in Table 3.A.7.

Table 3.A.8 shows the resultant number of path-satellites, the individual link delays, and the total end-to-end delay for the example. Table 3.A.9 shows the end-to-end propagation delays for a 6-plane, 11-satellites per plane constellation, between 10 cities

of the world ranked by Gross Domestic Product. Table 3.A.10 shows the number of satellites in the path between the same set of cities. Table 3.A.11 and 3.A.12 show the same information for a 12-plane, 24-satellites per plane constellation at an altitude of 1400 km and an inclination angle of 82 degrees.

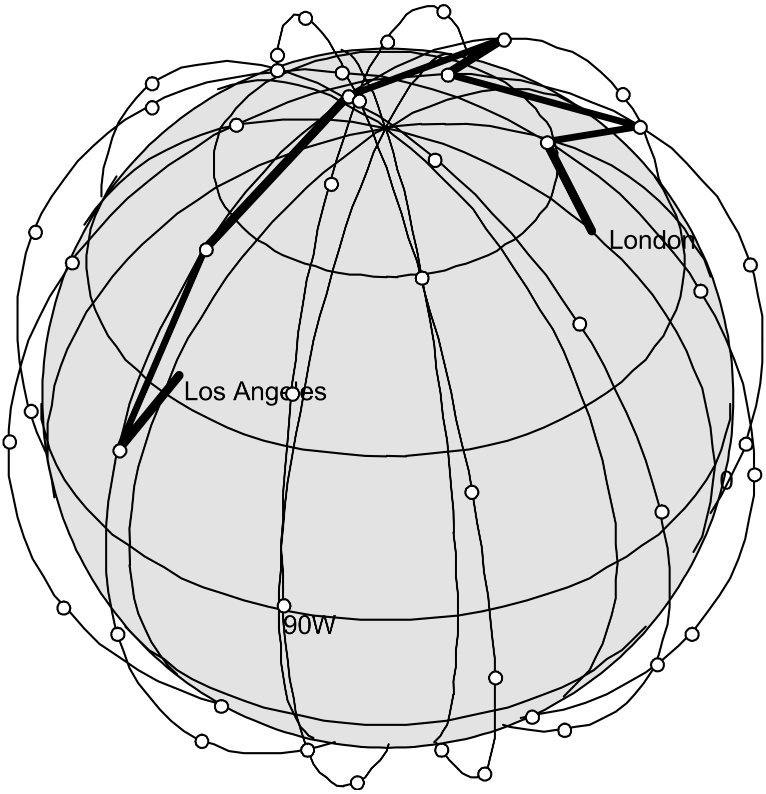

Figure 3.A.1: Example path through LEO constellation

Table 3.A.7: LEO Configuration Parameters

Orbit planes	6
Satellites per plane	11
Total satellites	66
Altitude (km)	780
Inclination (deg)	86

Table 3.A.8: LEO Propagation Delays

Source	Los Angeles
Destination	London
Satellites in path	7
Propagation delays (ms):	
Uplink	5.87
Downlink	6.37
ISL 1	13.44
ISL 2	13.44
ISL 3	13.44
ISL 4	7.80
ISL 5	13.44
ISL 6	9.63
Total propagation delay	83.45

Table 3.A.9: Total propagation delays in milliseconds in city-to-city path for 6x11 constellation

	New York	Tokyo	Paris	London	Seoul	Los Angeles	Toronto	Mexico City	Sydney	Chicago
New York	11									
Tokyo	58	13								
Paris	60	60	13							
London	39	47	26	13						
Seoul	62	37	83	68	11					
Los Angeles	24	71	69	83	71	12				
Toronto	10	57	54	38	63	23	9			
Mexico City	26	73	51	61	79	39	25	14		
Sydney	62	37	91	71	36	49	61	77	7	
Chicago	8	56	53	36	61	22	7	23	59	6

Table 3.A.10: Number of satellites in city-to-city path for 6x11 constellation

	New York	Tokyo	Paris	London	Seoul	Los Angeles	Toronto	Mexico City	Sydney	Chicago
New York	1									
Tokyo	5	1								
Paris	5	5	1							
London	4	4	2	1						
Seoul	5	3	7	6	1					
Los Angeles	2	6	6	7	6	1				
Toronto	1	5	5	4	5	2	1			
Mexico City	2	6	4	5	6	3	2	1		
Sydney	5	3	7	6	3	4	5	6	1	
Chicago	1	5	5	4	5	2	1	2	5	1

Table 3.A.11: Total propagation delays in milliseconds in city-to-city path for 12x24 constellation

	New York	Tokyo	Paris	London	Seoul	Los Angeles	Toronto	Mexico City	Sydney	Chicago
New York	10									
Tokyo	57	11								
Paris	77	109	10							
London	78	110	11	12						
Seoul	58	24	75	76	11					
Los Angeles	41	57	99	100	59	10				
Toronto	10	57	73	74	58	44	11			
Mexico City	24	67	87	88	68	30	24	10		
Sydney	92	52	89	89	53	115	92	113	10	
Chicago	22	54	83	84	57	30	23	23	101	10

Table 3.A.12: Number of satellites in city-to-city path for 12x24 constellation

	New York	Tokyo	Paris	London	Seoul	Los Angeles	Toronto	Mexico City	Sydney	Chicago
New York	1									
Tokyo	8	1								
Paris	10	15	1							
London	10	15	1	1						
Seoul	8	3	12	12	1					
Los Angeles	6	9	15	15	9	1				
Toronto	1	8	10	10	8	6	1			
Mexico City	3	10	12	12	10	4	3	1		
Sydney	14	7	12	12	7	19	14	18	1	
Chicago	3	8	12	12	8	4	3	3	13	1

3.A.4 Delay Variation

Although LEO networks have smaller propagation delays than GEO networks, there is a clear tradeoff between delay and jitter characteristics of GEO and LEO systems, especially for interactive real-time applications.

3.A.4.1 GEO Delay Variation

GEO systems exhibit relatively stable delay characteristics because they are nearly stationary with respect to the ground terminals. Connection handovers are rare in GEO systems and are mainly due to traffic balancing or fault recovery reasons.

3.A.4.2 LEO Delay Variation

The delay variation in LEO systems can arise from several factors:

Handovers: The revolution of the satellites within their orbits causes them to change position with respect to the ground terminals. As a result, the ground terminal must handover the connections from the satellite descending below the horizon to the satellite ascending from the opposing horizon. Based on the velocity, altitude and the coverage of the satellites, it is estimated that call handovers can occur on an average of every 8 to 11 minutes. The handover procedure requires a state transfer from one satellite to the next, and will result in a change in the delay characteristic of the connection at least for a short time interval. If the satellites across the seam of the constellation are communicating via crosslinks, the handover rate is much more frequent because the satellites are traveling in opposite directions.

Satellite Motion: Not only do the satellites move with respect to the ground terminal, they also move relative to each other. When satellites in adjacent orbits cross each other at the poles, they are now traveling in opposite sides of each other. As a result, calls may have to be rerouted accordingly resulting in further changes in delays.

Buffering and Processing: A typical connection over a LEO system might pass through several satellites, suffering buffering and processing delays at each hop. For CBR traffic, the buffering delays are small, but for bursty traffic over real time VBR (used by video applications), the cumulative effects of the delays and delay variations could be large depending on the burstiness and the amount of overbooking in the network.

Adaptive Routing: Due to the satellite orbital dynamics and the changing delays, most LEO systems are expected to use some form of adaptive routing to provide end-to-end connectivity. Adaptive routing inherently introduces complexity and delay variation. In addition, adaptive routing may result in packet reordering. These out of order packets will have to be buffered at the edge of the network resulting in further delay and jitter.

GEO systems exhibit relatively stable delay characteristics because they are almost stationary with respect to the ground terminals. Connection handovers are rare in GEO systems and are mainly due to fault recovery reasons. As a result, there is a clear trade-off between delay and jitter characteristics of GEO and LEO systems, especially for interactive real-time applications.

3.A.5 Buffering delay

Section 3.4 provided the buffering delay for TCP over Satellite ATM – UBR service.

3.A.6 End-to-End Delay Analysis

In this section end-to-end delay is calculated from equation (3.A.1), using propagation delays for GEO and LEO configurations from Section 3.A.2 and 3.A.3. The buffering delays are included from Section 3.4. This calculation has been performed for the same selected city pairs for the sample GEO and LEO networks.

3.A.6.1 GEO Total Delay

The estimated total transmission delay ranges between selected city pairs using the sample 4-satellite GEO constellation are shown in Table 3.A.13.

Table 3.A.13: Total transmission delay (ms) for sample 4-satellite GEO network

	New York	Tokyo	Paris	London	Seoul	Los Angeles	Toronto	Mexico City	Sydney	Chicago
New York	252-756									
Tokyo	454-1816	258-774								
Paris	453-1812	458-1832	256-768							
London	453-1812	458-1832	256-768	256-768						
Seoul	457-1828	259-777	455-1820	455-1820	256-768					
Los Angeles	252-756	454-1816	453-1812	453-1812	457-1828	252-756				
Toronto	252-756	454-1816	453-1812	453-1812	457-1828	252-756	252-756			
Mexico City	247-741	449-1796	448-1792	448-1792	452-1808	247-741	247-741	242-756		
Sydney	451-1804	255-765	652-3260	652-3260	258-774	451-1804	451-1804	446-1784	252-756	
Chicago	256-768	454-1816	453-1812	453-1812	457-1828	252-756	252-756	247-741	451-1804	252-756

3.A.6.2 LEO Total Delay

The estimated total transmission delay ranges between selected city pairs using the sample 288-satelltie LEO constellation are shown in Table 3.A.14.

Table 3.A.14: Total transmission delay (ms) for sample 288-satellite LEO network

	New York	Tokyo	Paris	London	Seoul	Los Angeles	Toronto	Mexico City	Sydney	Chicago
New York	10-30									
Tokyo	57-570	11-33								
Paris	77-924	109-1853	10-30							
London	78-936	110-1870	11-33	12-36						
Seoul	58-580	24-120	75-1050	76-1064	11-33					
Los Angeles	41-328	57-627	99-1683	100-1700	59-649	10-30				
Toronto	10-30	57-570	73-876	74-888	58-580	44-352	11-33			
Mexico City	34-120	67-804	87-1218	88-1232	68-816	30-180	24-120	10-30		
Sydney	92-1472	52-468	89-1246	89-1246	53-477	115-2415	92-1472	113-2260	10-30	
Chicago	22-110	54-540	83-1162	84-1176	57-570	30-180	23-115	23-115	101-1515	10-30

Appendix 4.A DiffServ QoS for Satellite IP

4.A.1 LEO Satellite IP Network Simulation Results (On-Board Switching)

In this section, simulation configuration, simulation parameters and simulation results for TCP/UDP in a LEO satellite networks are presented. See Section 4.5. [277]

Figure 4.A.1 shows the simulation configuration for LEO satellite IP networks. Table 4.11 provides the configuration parameters similar to those described in Sections 4.5.3 and 4.5.4.

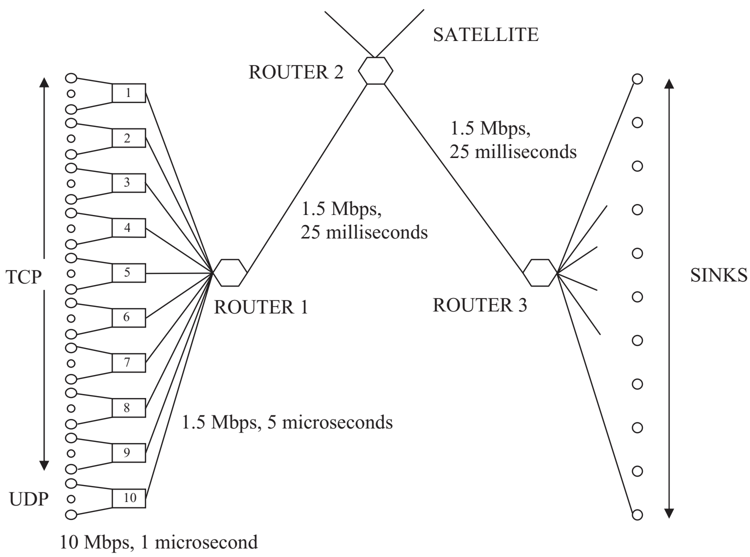

Figure 4.A.1: Simulation Configuration of LEO

Table 4.11: LEO: Simulation Configuration Parameters

Simulation Time	100 seconds
TCP Window	64 packets
IP Packet Size	576 bytes
UDP Rate	1.28 Mbps
Maximum queue size (for all queues)	60 packets
Link between Router 1 and Router 2:	
Link bandwidth	1.5 Mbps
One way delay	25 milliseconds
Drop policy	From Router 1: RED_n To Router 1: DropTail
Link between Router 2 to Router 3:	
Link bandwidth	1.5 Mbps
One way delay	25 milliseconds
Drop policy	DropTail
Link between Router 3 and Sinks:	
Link bandwidth	1.5 Mbps
One way delay	5 microseconds
Drop policy	DropTail
Link between UDP/TCPs and Customers:	
Link bandwidth	10 Mbps
One way delay	1 microsecond
Drop policy	DropTail
Link between Customers & Router 1:	
Link bandwidth	1.5 Mbps
One way delay	5 microseconds
Drop policy	DropTail

4.A.1.1 Fairness: Two-Color Vs Three-Color

Table 4.13: LEO: Main Factors Influencing Fairness Results in Three Color Simulations

Factor/Interaction	Allocation of Variation (in %age)
Yellow Rate	50.33
Yellow Bucket Size	24.17
Interaction between Yellow Rate and Yellow Bucket Size	21.50

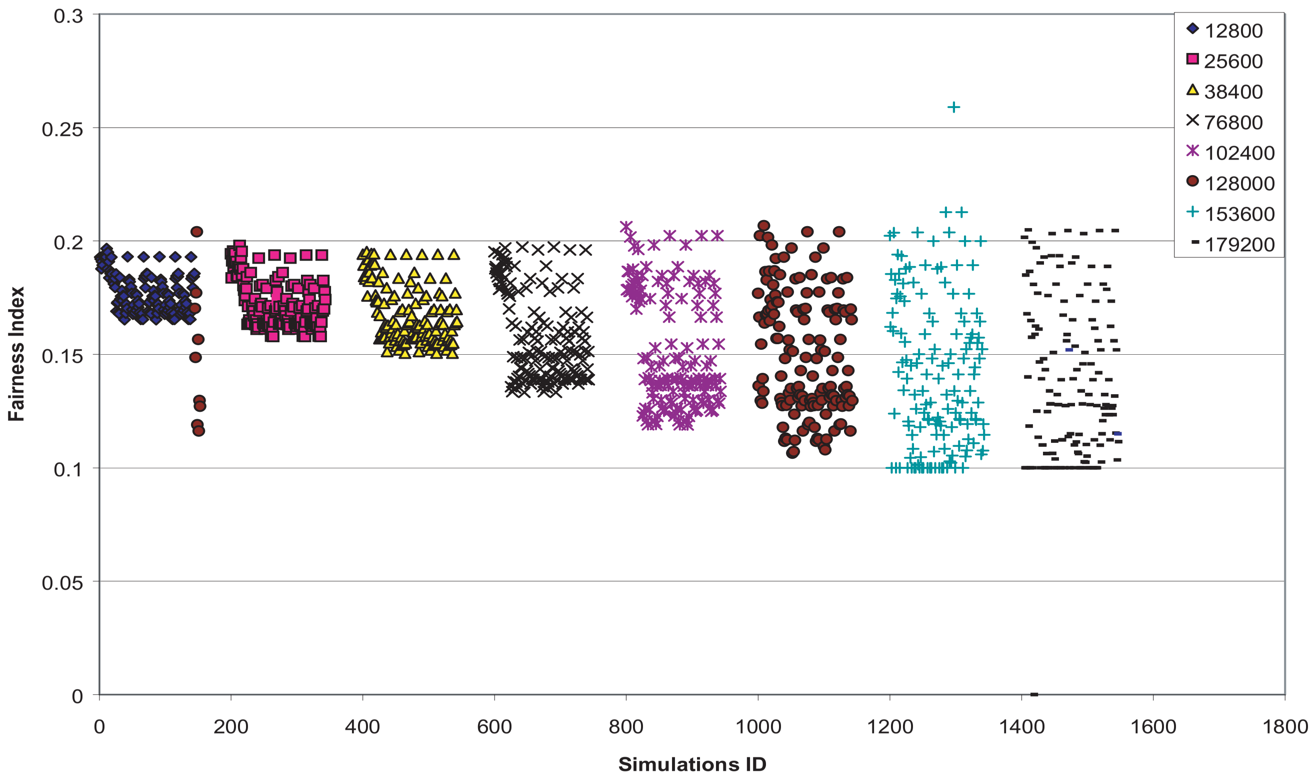

Figure 4.A.2: LEO Simulation Results: Fairness achieved in Two Color Simulations with Different Reserved Rates

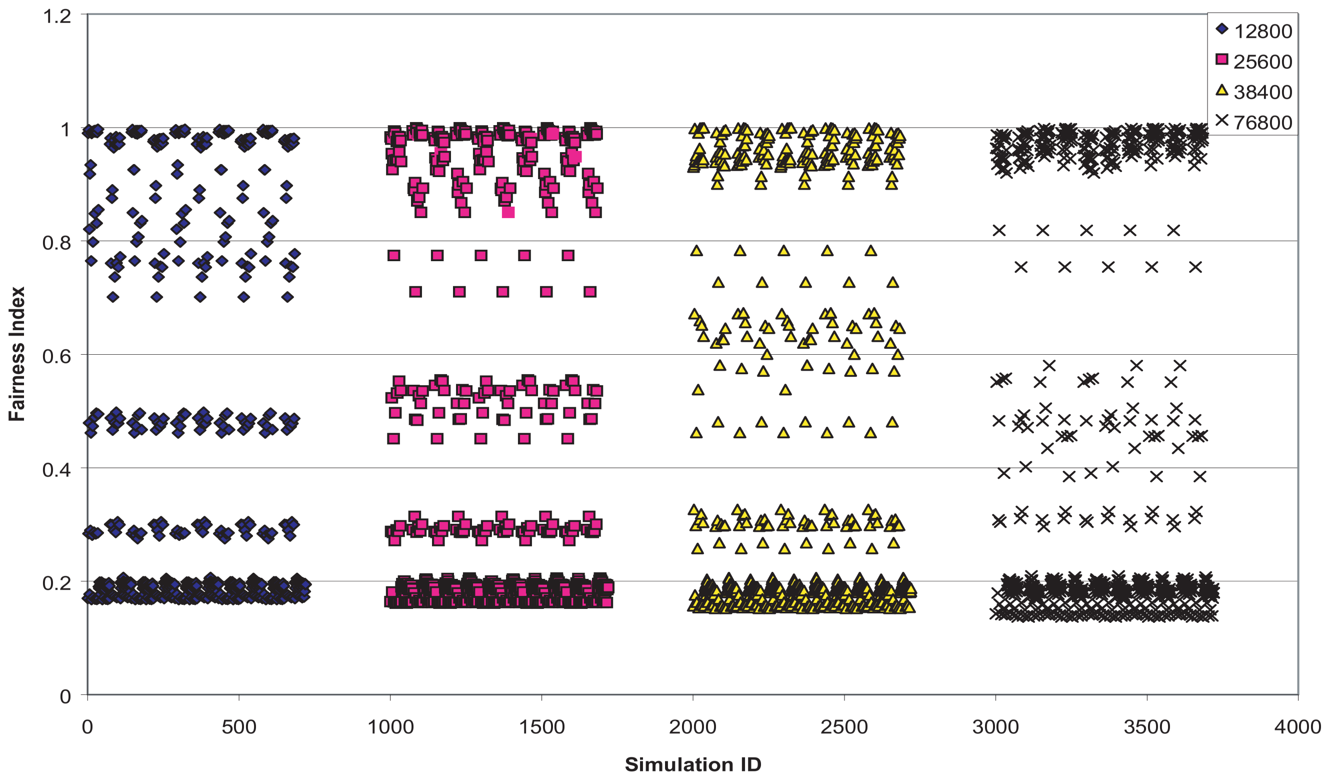

Figure 4.A.3: LEO Simulation Results: Fairness achieved in Three Color Simulations with Different Reserved Rates

4.A.1.2 Reserved Rate Utilization – Two-Color Vs Three-Color

Table 4.12: LEO: Main Factors Influencing Reserved Rate Utilization Results

Factor/Interaction	Allocation of Variation (in %age)			
	2 Colors		3 Colors	
	TCP	UDP	TCP	UDP
Green Rate	3.59	38.05	1.67	19.51
Green Bucket Size	94.49	34.58	94.63	63.48
Green Rate - Green Bucket Size	1.47	26.89	1.28	16.98

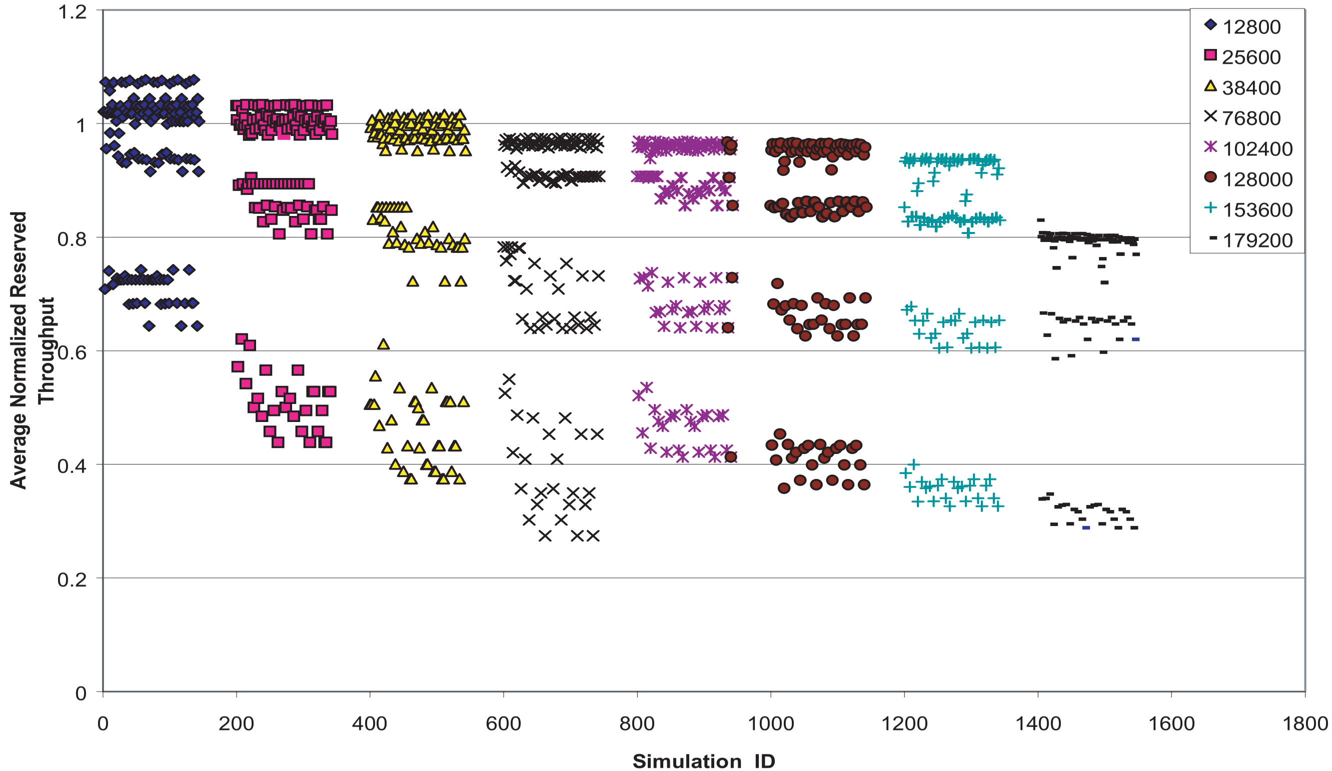

Figure 4.A.4: LEO Simulation Results: Reserved Rate Utilization by TCP Customers in Two Color Simulations

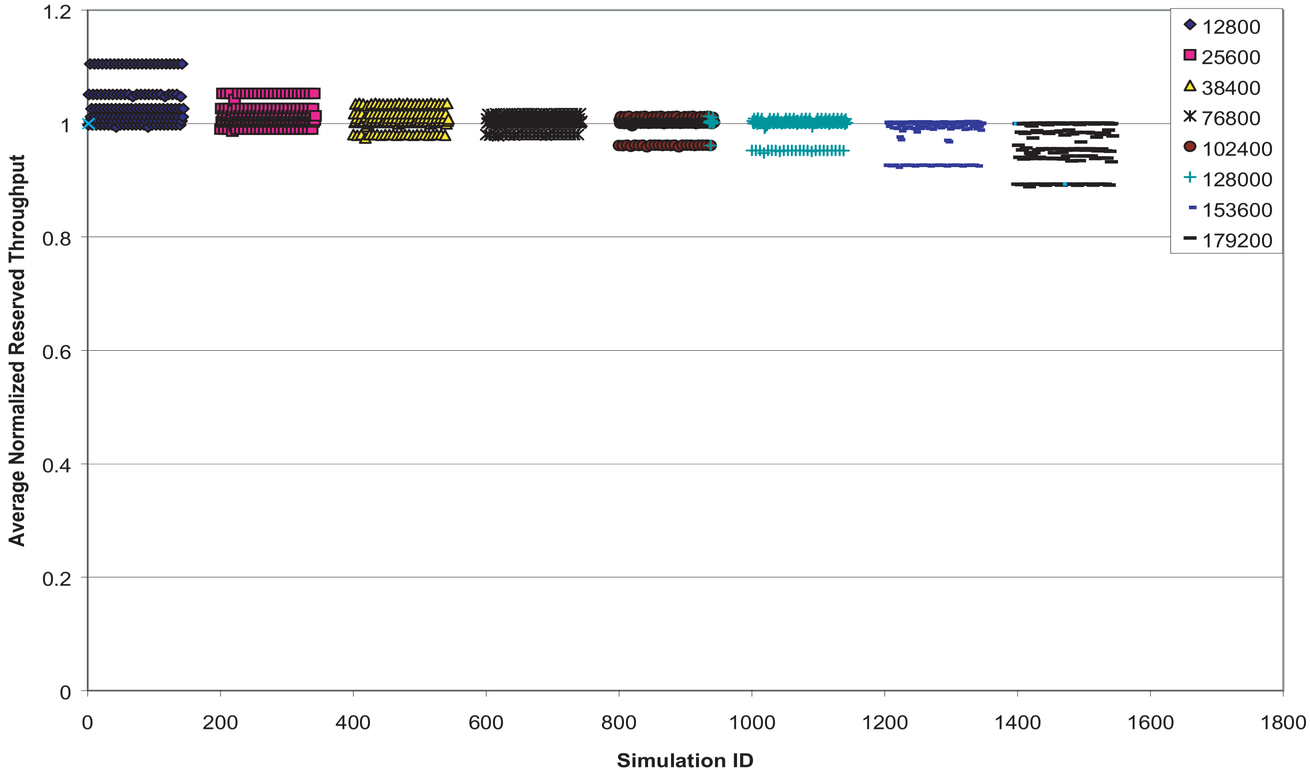

Figure 4.A.5: LEO Simulation Results: Reserved Rate Utilization by UDP Customers in Two Color Simulations

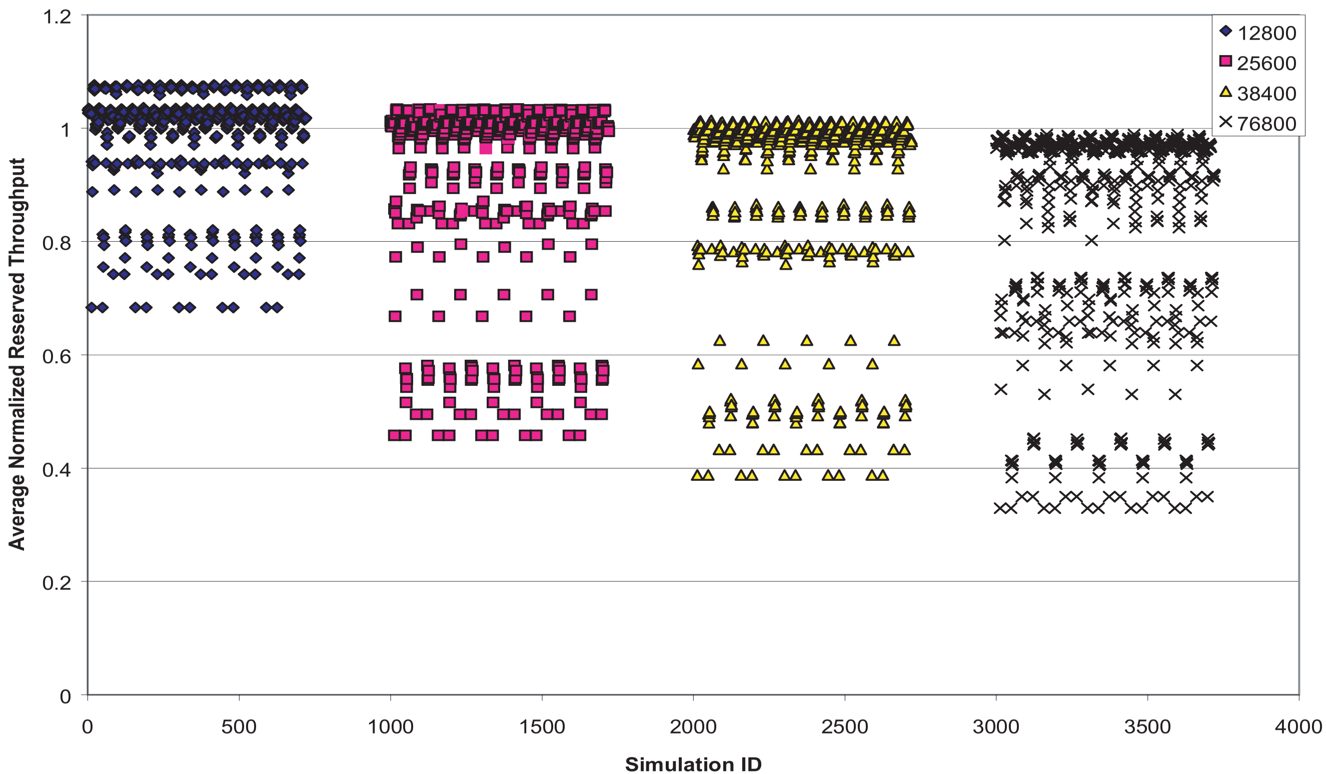

Figure 4.A.6: LEO Simulation Results: Reserved Rate Utilization by TCP Customers in Three Color Simulations

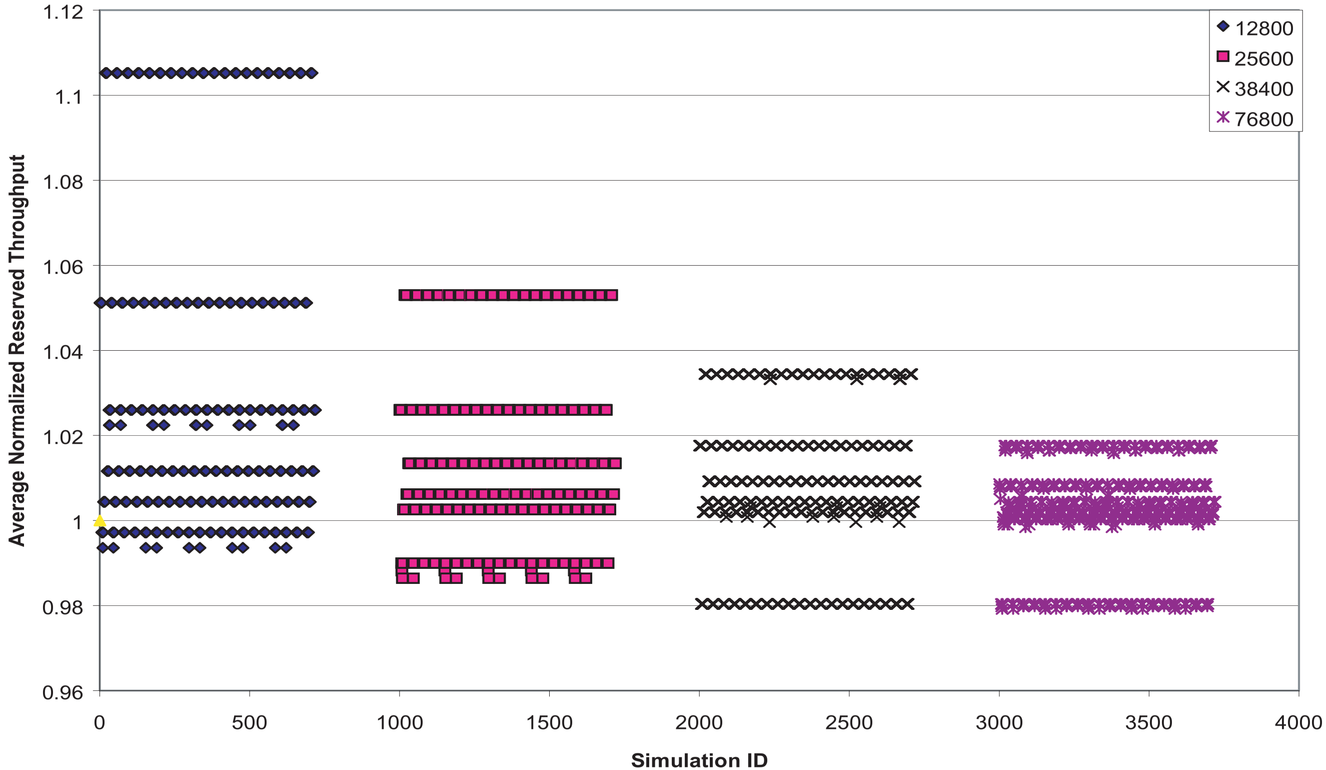

Figure 4.A.7: LEO Simulation Results: Reserved Rate Utilization by UDP Customers in Three Color Simulations

4.A.1.3 LEO Simulation Results Discussion

It is clear from figures 4.A.2 and 4.A.3 that fairness is not good in two-color simulations. With three colors there is a wide variation in fairness results with the best being close to 1. It can be noted that fairness is 0 in some of the two-color simulations as the total reserved traffic uses all the bandwidth and there is no excess bandwidth available to share.

Figures 4.A.4 and 4.A.6 show the reserved rate utilization for TCP traffic and it is obvious that three-color simulations is better than two colors. Figures 4.A.5 and 4.A.7 show the reserved rate utilization for two-color UDP and three-color UDP. As discussed in Section 4.5.3, in three-color simulation, the UDP customer always has a yellow rate of 0. However, TCP packets coming from customers 1-9 are colored green, yellow or red (three precedence levels) and UDP packets are colored green or red (two precedence levels). It can be noted that in some cases reserved rate utilization is slightly more than 1. This is because token buckets are initially full which results in all packets getting green color in the beginning.

Clearly, three levels of drop precedence (colors) are required for high reserved rate utilizations. The fair allocation of excess network bandwidth can be achieved only by giving different treatment to congestion-sensitive TCP and congestion-insensitive UDP packets.

4.A.2 Performance of TCP/UDP with Link Error Rate (Bent-Pipe Satellite)

This section provides simulation results for studying the influence of Bit Error Rate (BER) over TCP/UPD transport over GEO network with DiffServ as discussed in Section 4.5. The nature of the results remain the same. [278]

GEO Simulation results with a bit error rate of 10^{-8} are shown in Figures 4.A.8-4.A.13.

Figures 4.A.8 and 4.A.9 show reserved rate utilization by TCP and UDP customers in two color for GEO networks. Figures 4.A.10 and 4.A.11 show reserved rate utilization for TCP and UDP customers in three colors. For TCP customers, the average reserved rate utilization in each simulation has been plotted. Note that in some cases, reserved rate utilization is slightly more than one. This is because token buckets are initially full which results in all packets getting green color in the beginning. Figures 4.A.9 and 4.A.11 show that UDP customers have good reserved rate utilization. In contrast, TCP customers show a wide variation in reserved rate utilization. Figures 4.A.12 and 4.A.13 show the fairness results achieved for GEO networks, in 2 colors and three colors.

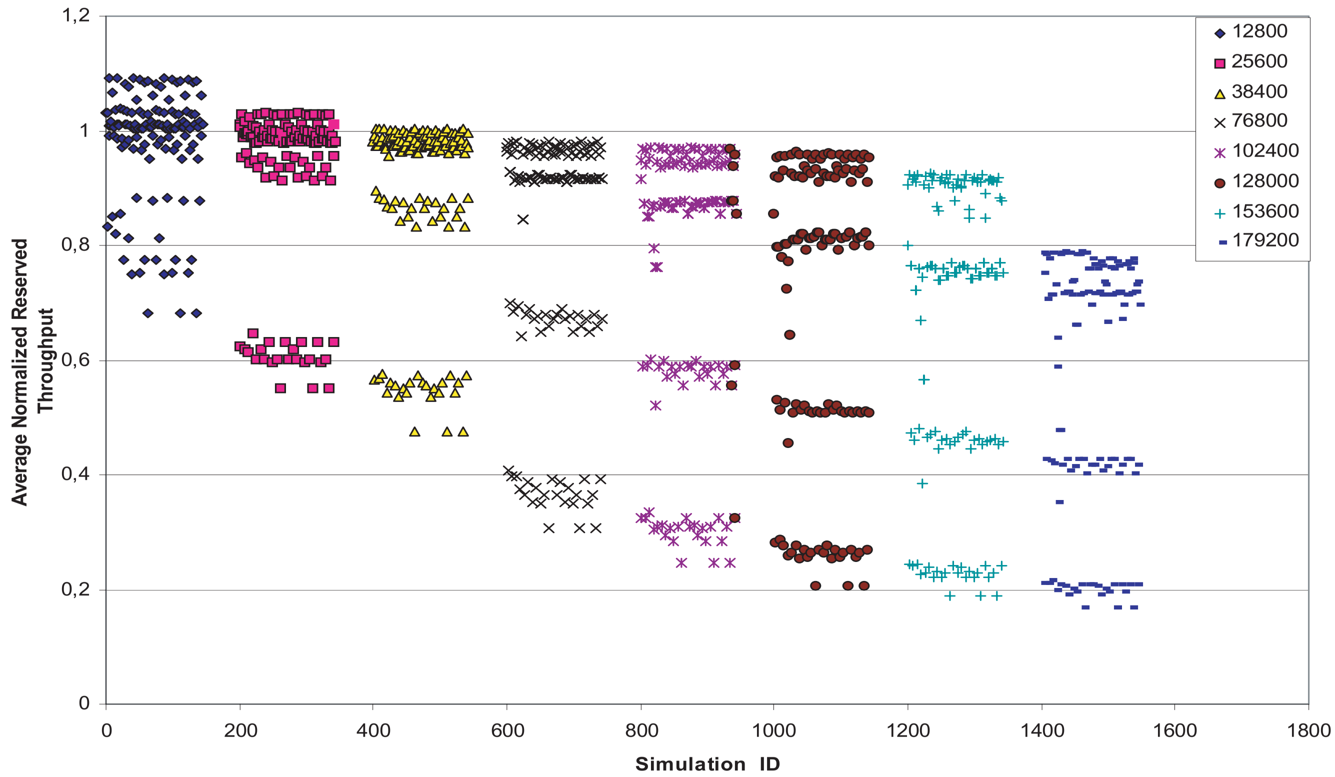

Figure 4.A.8: GEO Simulation Results: Reserved Rate Utilization by TCP Customers in Two Colors with BER = 10^{-8}

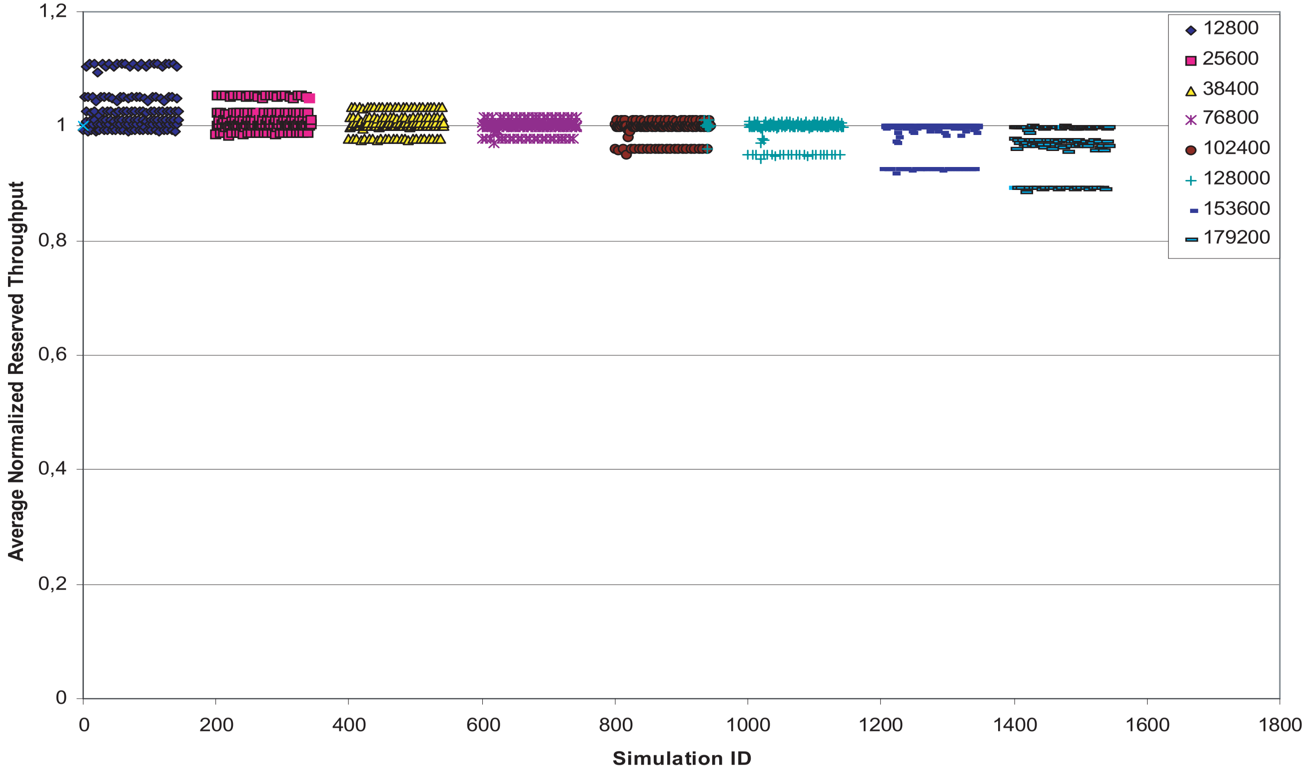

Figure 4.A.9: GEO Simulation Results: Reserved Rate Utilization by UDP Customers in Two Colors with BER = $10^{-\delta}$

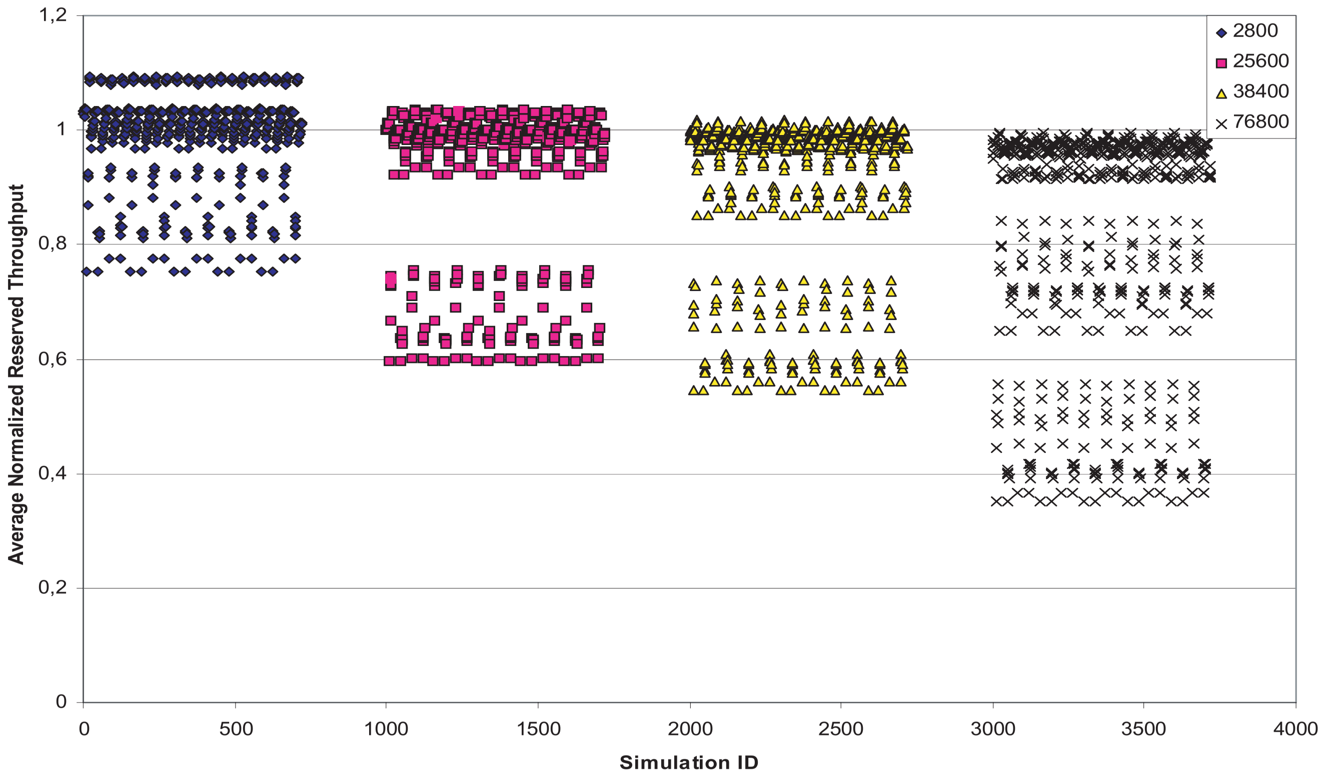

Figure 4.A.10: GEO Simulation Results: Reserved Rate Utilization by TCP Customers in Three Colors with BER = 10^{-8}

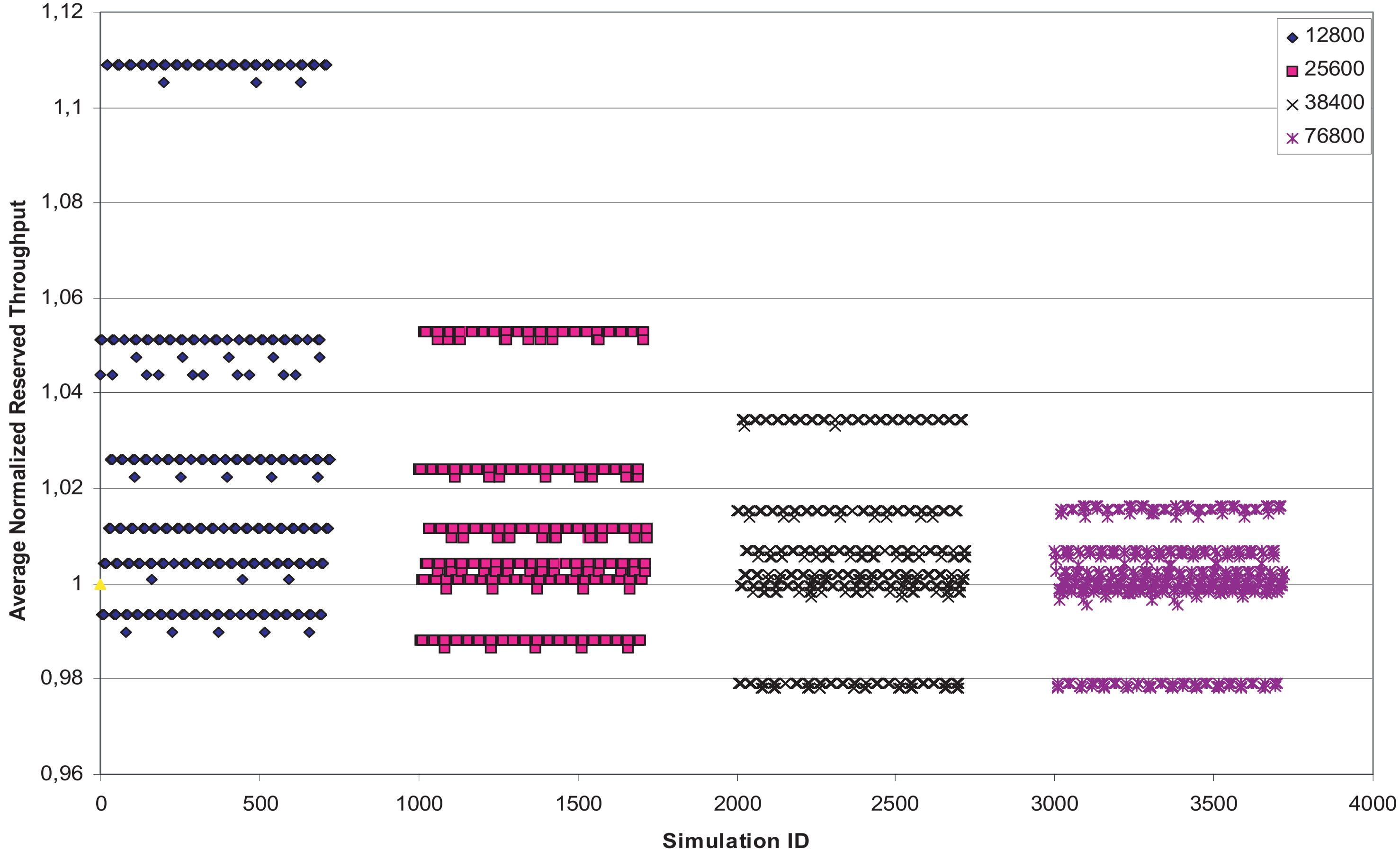

Figure 4.A.11: GEO Simulation Results: Reserved Rate Utilization by UDP Customers in Three Colors Simulations with BER = 10^{-8}

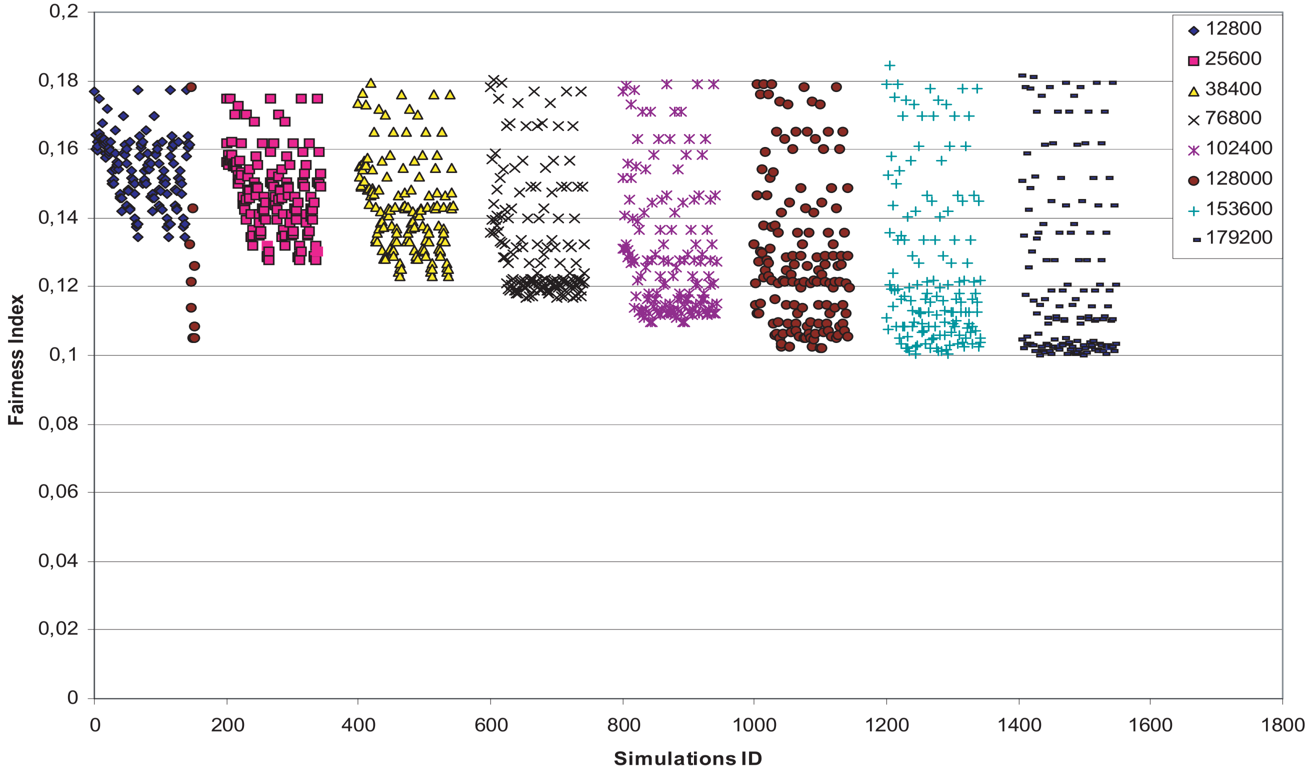

Figure 4.A.12: GEO Simulation Results: Fairness in Two Colors with BER=10^{-8}

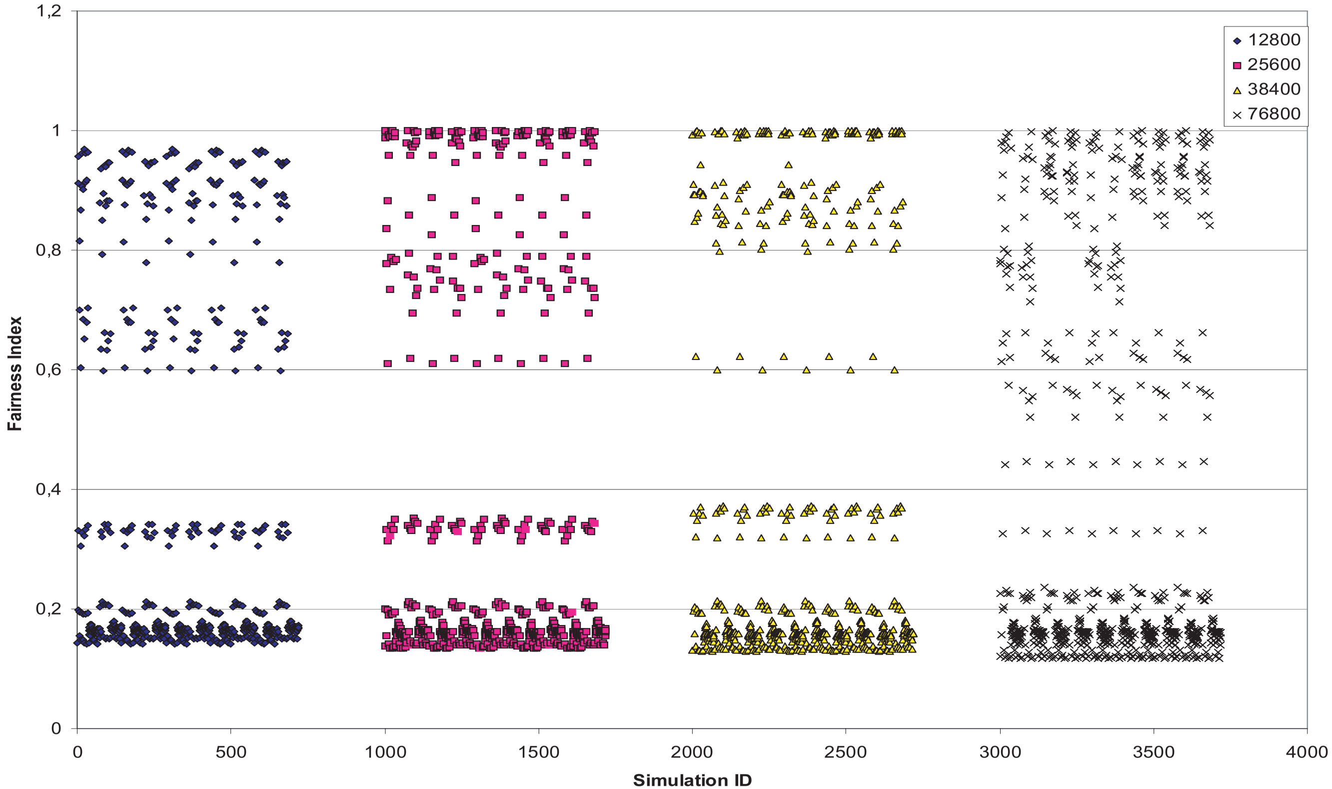

Figure 4.A.13: GEO Simulation Results: Fairness in Three Colors with BER=10^{-8}

4.A.2.1 BER Results Discussion

The BER does not influence that much the behaviour of TCP and UDP as discussed in [278]. This performance level is as expected, because DiffServ provides the relative QoS of how each flow behaves with respect to each other. The error rate affects both the TCP and UDP in a similar fashion.

As shown from the simulation results, UDP behaves better than TCP in both two-color and three-color cases. This is due to the fact that UDP takes over the network resources available. So to protect the reserved UDP, two colors are enough. However, three colors are required to protect the reserved TCP against UDP.