QUALITY OF SERVICE MANAGEMENT IN ATM NETWORKS

A THESIS

Submitted in fulfilment of the requirements for the award of the degree
of
DOCTOR OF PHILOSOPHY
in
ELECTRONICS AND COMPUTER ENGINEERING

By

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Candidate's Declaration

I hereby certify that the work, which is being presented in the thesis, entitled "Quality of Service Management in ATM Networks" in fulfillment of the requirement for the award of the Degree of Doctor of Philosophy and submitted in the Department of Electronics and Computer Engineering of the Institute is an authentic record of my own work carried out during a period from Aug, 1997 to December, 2001 under the supervision of Dr. R.C. Joshi.

The matter presented in this thesis has not been submitted by me for the award of any other degree of this or any other Institute.

Mayank Dave 31/12/200

Signature of the Candidate

This is to certify that the above statement made by the candidate is correct to the best of my knowledge.

Date: 31/12/2001

Signature of the Supervisor

The Ph.D. Viva-Voce examination of Mayank Dave, Research Scholar, has been held on 5.7.02

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Abstract

Broadband applications over ATM networks generate traffic with varying traffic characteristics. These applications specify the required service guarantees in terms of quality of service (QoS) that the network has to provide. Several elements are needed to provide QoS guarantees. Examples are QoS specification, Admission control, QoS negotiation, Resource allocation and scheduling and Traffic policing. ATM networks should also support QoS adaptation and QoS renegotiation techniques. To ensure that all such requirements are satisfied, quality of service management is essential in ATM networks. The work presented in this thesis is an effort to study and propose different mechanisms for the management of quality of service in ATM networks.

In the first part, we propose a mechanism for QoS provisioning through dynamic bandwidth allocation and buffer control. Satisfaction of the different QoS requirements is a resource allocation problem. To make efficient use of resources and to satisfy QoS requirements like CLR or delay, efficient bandwidth and buffer allocation methods are needed. We assume that a large pool of buffer space exists that releases or receives back the buffer space as per requests received for buffer allocation. Under the proposed scheme the allocated buffer space of the connection is reallocated based on measurement of data from the system for certain time duration. Our scheme is based on the fact that there exists a relationship between queuing buffer

size, cell losses and delay. This scheme provides optimal bandwidth and buffer allocation to satisfy multiple QoS namely cell loss rate (CLR) and delay. We also investigate the use of cell loss priority (CLP) bit of cell header.

In the second part, we propose an algorithm for pre-computation of QoS routes. Path pre-computation schemes benefit from having multiple candidate routes to each destination, to balance the network load and have additional routing choices in case of a set-up failure. The QoS constraints for traffic are interrelated in a way that is determined by the network scheduling discipline. We have assumed that the scheduling policy in the network is rate based.

We compare the performance of our algorithm with that of pre-computation of k-constrained QoS routes using modified Bellman-Ford algorithm and with on-demand routing to compute route to the destination. We performed the simulation studies using minimum hop, widest-shortest path and shortest-widest path optimality criteria on two different topologies namely ISP and Switched cluster. We assume that a connection blocks after all k constrained paths are exhausted and result in connection setup failure. In our simulation after a failure in route extraction or connection establishment, the source triggers re-computation of the k constrained paths. We also carried out an experiment where the pre-computation rate was fixed and the link-state update period was varied.

In the third part we present a new adaptation protocol for QoS adaptation with renegotiation that allows an ATM network to recover from the QoS violations in order to satisfy end-to-end QoS requirements. Our protocol is applicable to PNNI based ATM network where the nodes are grouped together in peer groups hierarchically. We assume that every node has a QoS/Route Monitor unit that receives QoS/LSU updates from the network on a periodic or triggered basis. In addition to its usual functions, this monitor would also function as QoS agent, QoS manager or Connection QoS Manager depending on its location. The QoS/Route Monitor is responsible for sending or receiving the signalling messages required in our protocol.

There are two categories of QoS parameters that have been considered - concave and additive. The QoS violation may occur in both types of QoS parameters. The protocol works differently for each of these categories. To facilitate the functioning of the protocol, several signalling messages have been defined. These signalling messages characterize different control mechanisms required for the protocol and the network.

Finally, we conclude with a unified model for quality of service management in ATM networks that has QoS monitor as primary part of the ATM interface or switch. The model consists of components responsible for QoS provisioning, QoS monitoring and managing QoS routes. The model is also responsible for processing the proposed adaptation protocol between end points.

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Mayank Dave

List of Abbreviations

ABR	Available Bit Rate
AAL	ATM Adaptation Layer
API	Application Programming Interface
ATM	Asynchronous Transfer Mode
B-ISDN	Broadband Integrated Services Digital Network
CAC	Connection Admission Control
CBR	Constant Bit Rate
CLP	Cell Loss Priority
CDV	Cell Delay Variation
CDVT	Cell Delay Variation Tolerance
CLR	Cell Loss Rate
CQM	Connection QoS Manager
CTD	Cell Transfer Delay
GCRA	Generic Cell Rate Algorithm
IETF	Internet Engineering Task Force
IP	Internet Protocol
ISP	Internet Service Provider
ITU-T	ITU - Telecommunication Standardization Sector
LSU	Link State Update
MBF	Modified Bellman-Ford
MBS	Maximum Burst Size
MCR	Minimum Cell Rate

MMBP	Markov Modulated Bernoulli Process
MPEG	Motion Pictures Expert Group
NMS	Network Management System
OS	Operating System
PBS	Partial Buffer Sharing
PCR	Peak Cell Rate
PMT	Proposed Modified Topkis
PNNI	Private Network Node Interface
PO	Push Out
QoS	Quality of Service
RSVP	Resource Reservation Protocol
SCR	Sustainable Cell Rate
SVC	Switched Virtual Connection
UBR	Unspecified Bit Rate
UNI	User Network Interface
UPC	Usage Parameter Control
VBR	Variable Bit Rate
VC	Virtual Channel
VP	Virtual Path
VPI	Virtual Path Identifier

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Chapter 1

Introduction and Statement of the Problem

1.1 Introduction

Asynchronous Transfer Mode (ATM) is the widely used packet switching and connection oriented technology that meets diverse service and performance requirements of real-time applications using the broadband networks [86]. The traffic that these applications, for example multimedia database access or interactive continuous media applications generate are of varying characteristics. For example, traffic generated by interactive data and video is highly bursty; while some other traffic, such as large files, is continuous. In many data applications, real-time delivery is not of primary importance, but high throughput and strict error control are required. Some services, such as real-time video communications require error-free transmission as well as rapid transfer. Some services, for instance, real-time voice require rapid transfer through the network, but the loss of small amounts of voice information is tolerable. To provide a uniform framework for different applications to specify required performance guarantees and for systems to provide the required guarantees, the concept of quality of service (QoS) was introduced. QoS is the quantitative and qualitative specification of an application's requirement which a network should satisfy in order to achieve desired application quality. ATM supports the quality of service that Ethernet or traditional IP networks cannot provide. This is the primary reason because of which issues related to QoS in ATM have been important and active area of research for the past several years.

Each ATM connection contains a set of parameters that describes the traffic characteristics of the source. These parameters are called source traffic parameters. These parameters along with the required type of service and QoS parameters constitute the traffic descriptor for the connection [7][94]. This set of parameters is then negotiated between application and the network prior to connection establishment through signalling. If the negotiation is successful it is called traffic contract. The quality of service requirements for applications in ATM or other high speed networks are typically end-to-end requirements which impose corresponding performance requirements on both the network and the end systems [4]. Additionally, efficient use of network resources is desired for providing a cost-effective service. In the context of ATM networks, typical QoS measures include average cell loss rate (CLR), cell transfer delay (CTD) and cell delay variance (CDV). Network resources of interest include bandwidth requirements at each physical links as well as buffer space and processing capability at each switching node.

A major issue for any network is to use network resources as efficiently as possible. Thus it may be inferred that there are two aspects to QoS: applications specify QoS requirements and networks provide QoS guarantees [48]. The notion of QoS used in data communications e.g. to characterize the performance of data transmission in terms of throughput or delay do not cover all the requirements of multimedia communication and are used at the transport level only. The following elements in general are needed to provide QoS guarantees in ATM networks:

- A QoS specification mechanism for the applications to specify their requirements
- Admission control to determine whether the new application should be admitted without affecting the QoS of other ongoing applications
- QoS negotiation process to determine a set of QoS parameters acceptable for both the application and the network with the aim to support as many applications possible
- Resource allocation and scheduling to meet the QoS requirement of accepted applications
- Traffic policing to make sure that applications generate the correct amount of data within the agreed specification

ATM network contains both physical resources like buffers and logical resources like virtual paths. Quality of service and bandwidth are mainly related to physical resources. ATM cell occupies a buffer and cell transmission requires bandwidth. Thus buffer and bandwidth allocation to cells is required. In order to satisfy different QoS requirements, efficient bandwidth and buffer allocation methods are needed that utilize the available resources efficiently. Thus resource allocation becomes a management problem.

Routing is also an important issue in establishment of a connection in any network. But the routing protocols used in traditional networks are typically transparent to any particular QoS that different packets or flows may require. Under QoS routing the paths for the flows are determined based on some knowledge of resource availability in the network as well as the QoS requirements of the flows. Thus it enables a provision where a route can offer the requested QoS provided it has adequate available resources. QoS routing in the context of ATM based high speed networks has gained tremendous attention in the recent years. QoS routing supports alternate routing where if the best existing path can no longer provide the negotiated QoS or cannot admit a new flow, some adequate additional path could be used if it exists. Thus, the QoS routing could be very effective in increasing the network utilization.

In addition to the above basic elements, some other elements are also needed to meet diverse requirements of BISDN applications. These elements are also special because the notion that negotiated QoS for a connection remains the same for the lifetime of the connection is not always valid for multimedia applications.

- A QoS renegotiation mechanism is required so that applications can request changes in their initial QoS specifications.
- The actual QoS provided to the ongoing sessions should be monitored so that appropriate actions can be taken in case of any problem in providing specified QoS guarantees.
- Graceful quality degradation techniques are required together with the above mechanisms to provide satisfactory services to multimedia applications.

It is evident from the above discussions that the proper management of QoS is significantly important for efficient network utilization and for satisfying the specified QoS requirements. The management of quality of service involves all such techniques that contribute in providing the QoS guarantees to the applications. The QoS management techniques should be such that they should not get affected by the underlying protocols and mechanisms. We can therefore say that quality of service management is concerned with identifying appropriate characteristics and reserving the resources necessary to achieve the required functionality of a given service and to optimize the overall system performance. Examples of QoS management functions are QoS negotiation, admission control, resource reservation, QoS mapping, QoS renegotiation, QoS monitoring and QoS adaptation. QoS routing mechanism is also considered as one of the

QoS management functions. There have been have efforts in providing different QoS management techniques and QoS architectures have also been proposed but the performance of these management schemes can be improved.

1.2 Statement of the Problem

In the present work we have attempted to investigate and propose mechanisms for QoS management in ATM networks. The main objective of the present research work can be described by the statement of the problem which is expressed as follows "To investigate and propose schemes for providing the quality of service guarantees, maintaining the guarantees through QoS routing and managing these assurances under the conditions of violation and renegotiation of traffic contract in ATM networks". In order to explore the above problem, we divide it into a small number of objectives. These objectives are specified as follows.

1. To investigate mechanisms needed in ATM networks for satisfying the quality of service.

2. To examine the provisioning of quality of service guarantees through resource allocation and suggest improvements in existing techniques.

3. To investigate and propose algorithms for providing performance guarantees in ATM networks through QoS routing.

4. To improve and / or propose mechanisms for the management of quality of service subject to varying network conditions.

5. To propose a unified model for QoS management in ATM networks. This model should support mechanisms for maintaining quality of service through QoS routing and should be able to adapt to the renegotiation of traffic contract.

1.3 Organization of the Thesis

In Chapter 2 review and general considerations for the quality of service management in ATM networks are presented. We briefly discuss elements needed to provide QoS guarantees and review the related research reported by others. Chapter 3 proposes scheme for QoS provisioning through dynamic bandwidth allocation and buffer allocation. In Chapter 4, we address the issue of QoS routing in ATM networks and propose an algorithm for pre-computation of QoS routes. In Chapter 5, we propose a protocol for QoS adaptation with renegotiation. In the same chapter we propose a unified model for QoS management to integrate proposed mechanisms for QoS. In Chapter 6, conclusion of the whole work is explained and the scope for future research work in this area is outlined.

Chapter 2

Review and General Considerations

The introduction of ATM technology has made it feasible to support high speed multimedia applications like video conferencing or video on-demand. A key characteristics of these applications is that they have stringent service requirements which are specified in terms of quality of service guarantees. The quality of service provides a uniform framework for different applications to specify required performance guarantees and for systems to provide the required guarantees [39]. Multimedia applications specify the quality of service and the underlying network is required to support not only the real-time communication but also the quality of service [74]. In this chapter we briefly discuss elements needed to provide QoS guarantees and review the work done in the area of QoS management.

2.1 ATM Traffic Management

Traffic management in communication networks deals with the controlled use of network resources to prevent the network from becoming a bottleneck. In particular, when network resources are allocated to more connection or traffic than they can effectively support, network performance degrades. Therefore, it is necessary to manage resource utilization and control the traffic so that the network can operate at acceptable levels even at times when the offered load to the network exceeds its capacity. In ATM networks, the traffic management is difficult as there are several aspects that affect the traffic management functions:

1. Various B-ISDN VBR sources generate traffic at significantly different rates which also have time-varying levels.

2. A single source may generate multiple types of traffic with different characteristics.

3. Different services have different types of QoS requirements at considerably varying levels.

4. There is high delay-bandwidth product due to which there are very large number of cells in transit at any time in the network. Also, large propagation delays compared with small transmission times give rise to long periods between the onset of congestion and its detection by the network control elements.

Functions related to the implementation of QoS in ATM networks are usage parameter control (UPC) and connection admission control (CAC). The UPC function (implemented at the network edge) ensures that the traffic generated over a connection conforms to the declared traffic parameters: excess traffic may be dropped or carried on a best effort basis that is the QoS guarantees do not apply. The CAC function is implemented by each switch in an ATM network to determine whether the QoS requirements of a connection can be satisfied with the available resources. For non stationary traffic for example video, due to change in long term activity, it is generally not possible to find a single UPC that results in uniform quality for the entire duration of the session without significantly over-dimensioning the initially declared

UPC. Therefore bandwidth renegotiation is needed so as to simultaneously achieve high network utilization and maintain acceptable performance. The UPC is intended to control the traffic offered by an ATM connection to ensure conformance with the negotiated traffic contract. Considerable attention has been made to provide efficient connection admission control schemes so that the objective that a traffic source will never be able to exceed the traffic contract is met [24][25][26][46][61][81][91].

2.1.1 Traffic Conformance

A traffic contract specifies the negotiated characteristics of an ATM layer connection at a user network interface (UNI). A typical traffic contract consists of a connection traffic descriptor and a requested QoS class for each direction of the ATM layer connection and shall include the definition of a compliant connection. The connection traffic descriptor consists of all parameters and the conformance definition which is used to specify unambiguously the conforming cells of the ATM connection that is

- the Source Traffic Descriptor i.e. traffic and QoS parameters
- The CDV tolerance
- the Conformance Definition based on one or more applications of the Generic Cell Rate Algorithm (GCRA). This is used to specify unambiguously the conforming cells of an ATM connection at the UNI.

The UNI specification places no restriction on the possible combinations that a user may request for QoS class and parameters in the connection traffic descriptor. In order to achieve the desired QoS, ATM end-point may use a function called traffic shaping. This mechanism attains desired characteristics for the stream of cells emitted into a connection. Examples of traffic shaping are peak cell rate reduction, burst length limiting and reduction of cell clumping due to CDV by suitably spacing cells in time. Traffic shaping may be performed to remain in conformance with the connection traffic descriptor and associated parameter values that were negotiated with the network.

2.1.2 ATM Traffic and QoS Parameters

Each ATM connection contains a set of parameters that describes the traffic characteristics of the source. Source traffic parameters, coupled with CDVT (cell delay variation tolerance) and Conformance definition parameter. Not all these traffic parameters are valid for each service category. When an end-system requests an ATM connection (SVC) to be set up, it indicates to the ingress ATM switch the type of service required, the traffic parameters of data flow and QoS parameters. The traffic parameters consists of these parameters: Peak Cell Rate (PCR), Cell Delay Variation Tolerance (CDVT), Sustainable Cell Rate (SCR), Maximum Burst Size (MBS) and Minimum Cell Rate (MCR).

The existing QoS model in broadband ATM networks consists of five service categories [6]. These categories are differentiated according to whether they support constant or variable rate traffic and real-time or non real-time constraints. The service parameters include a characterization of the traffic and a reservation specification in the form of QoS parameters. Also, the traffic is policed to ensure that it conforms to the traffic. ATM provides the ability to tag non-conforming cells and specify whether tagged cells are policed and dropped or provided with best effort service. Under the UNI specification, the service categories are constant bit rate (CBR), real-time variable bit rate (rt-VBR), non real-time variable bit rate (nrt-VBR), unspecified bit rate (UBR) and available bit rate (ABR). Table 2.1 describes the characteristics of

Class	Characteristics
Constant Bit Rate (CBR)	Real-time applications requiring QoS guarantees (CTD, CDV and CLR)
Real-Time Variable Bit Rate (rt-VBR)	Real-time applications requiring QoS Allows statistical multiplexing guarantees (CTD, CDV and CLR)
Non-real-Time Variable Bit Rate (nrt-VBR)	Suitable for non real-time bursty traffic (CTD,CLR). Allows statistical multiplexing
Available Bit Rate (ABR)	Feedback control of transfer rate
Unspecified Bit Rate (UBR)	Connectionless data traffic, Delay tolerant. Best effort service, no bandwidth reservation or QoS guarantee required

Table 2.1: ATM service classes.

the ATM service classes.

A network may support any subset of the possible values for each of the QoS parameters. The QoS parameters characterize the network-level performance in terms of cell loss ratio (CLR), cell delay variation (CDV), maximum cell transfer delay (max CTD), and cell error rate (CER):

- Cell Loss Ratio (CLR): It is the value of the ratio of the number of cells lost to the total number of cells transmitted by the source end station that the network agrees to offer as an objective over the lifetime of the connection.
- Cell Delay Variation (CDV): It determines variance in the cell delay i.e. the amount of jitter.
- Maximum Cell Transfer Delay (max CTD): It is the sum of the cell delay on a switch-by-switch basis along the transit path of a particular connection.
- Cell Error Rate (CER): It is the ratio of the number or errored cells to the total number of cells transmitted by the source end station that the network agrees to offer as an objective

Attribute	ATM Service Category				
	CBR	rt-VBR	nrt-VBR	UBR	ABR
	Traffic Parameters				
PCR and CDVT	Yes	Yes	Yes	Yes	Yes
SCR and MBS	n/a	Yes	Yes	n/a	n/a
MCR	n/a	n/a	n/a	n/a	Yes
	QoS Parameters				
CDV	Yes	Yes	No	No	No
Max. CTD	Yes	Yes	No	No	No
CLR	Yes	Yes	Yes	No	No

Table 2.2: ATM traffic and QoS parameters.

over the lifetime of the connection.

Table 2.2 summarizes the traffic descriptor and QoS parameters relevant to each category within traffic management specification version 4.0 [7].

2.2 Service Disciplines

The term guaranteed service has been defined in [108]. To provide real-time guarantees in high speed networks packet-scheduling or queue service disciplines have been developed. The scheduling or queuing discipline for each class of cell allocates a bandwidth to the class dynamically where each class typically correspond to each ATM-layer service. The scheduler thus moves the bandwidth boundary between classes. The new packet scheduling disciplines are the rate based schemes that designed for fast packet networks like ATM. In [123], a survey is presented on various service disciplines for both work conserving (non-idling) and non-work conserving disciplines. It also compares different characteristics and trade-offs involved in the service disciplines. Rate based service disciplines control traffic burstiness and support deterministic delay and jitter bounds in fast packet networks. To guarantee delay bounds the traffic scheduler has to ensure that flows receive their guaranteed share of the link bandwidth. The fair queuing systems provide rate guarantees or distribute bandwidths depending on flow's state. A fair queuing need act only when all connections are backlogged and then provide rate guarantees. In [62], a scheme for rate based scheduling has been presented. Parekh and Gallager [88] provide a theoretical background to generalized processor sharing (GPS) scheme. GPS is an ideal work conserving scheme that has been emulated for providing fair queuing. Several schemes have been proposed for providing QoS guarantees using fair queuing service disciplines [124][111][4][35]. Some new schemes are also proposed for work conserving [59] and non-work conserving [119][71] for scheduling schemes. A scheme was presented by Ferrari and Verma in [40] for providing real-time services in packet networks. Their procedure is to conduct tests on the channel to determine feasibility for providing performance guarantees and then to run connection establishment process on per node basis. Guerin and Peris [52] identify different approaches and mechanisms used in packet networks to provide QoS guarantees. They consider different scheduling policies for packets - FCFS, Fair Queuing, Earliest Deadline First etc. and buffer management policies like Early Packet Discard (EPD), Random Early Discard and Fair Random Early Drop (FRED). They also suggest that the buffers may be partitioned among each of the virtual circuits on the connection but compared with the case when the buffers are shared the loss probability would be higher. They further suggest that the buffer management schemes should ensure that service guarantees are met, so cells are not dropped. They propose that systems that cannot or are incapable of maintaining individual flow state information, the QoS guarantee provisioning to individual flows becomes difficult. Accordingly per flow buffer accounting should provide better performance, when excess traffic is there.

The traditional method of providing QoS guarantees is the integrated services or the intserv model also called the per-flow model which provides QoS to individual traffic flows. The intserv model suffers from a drawback of maintaining vast amount of state information in the network for the individual flows. In the differentiated services or the diffserv model proposed for the Internet, service guarantees are given to aggregate flows, rather than on a per flow basis [90][15][23].

2.3 QoS Guarantees through Resource Allocation

ATM network contains both physical resources (e.g. buffers and bandwidths) and logical resources (e.g. virtual path identifiers and virtual channel identifiers). The QoS and bandwidth are mainly related to physical resources. The objects that use allocated physical resources in an ATM network include cells, bursts, virtual channel (VC), virtual path (VP) and signalling message. A cell occupies a buffer and cell transmission requires bandwidth. Thus it means that both the buffer and bandwidth allocations to cells are required. When a connection is setup, virtual channel and virtual paths are established between the end points. The stream of cells flowing in a VC/VP consume buffers and the transmission of cells require bandwidth. Buffers and bandwidth are allocated to each cell and each burst by allocating buffers and bandwidth to the VC/VP containing the cell or burst.

It is clear from the foregoing discussions that the satisfaction of different QoS requirements is a resource allocation problem. To provide QoS guarantees ATM networks allow applications to reserve resources for network connections based on traffic parameters [68] [75] [99] [118] and [125]. Saito[102] studies bandwidth allocation schemes and considers static and dynamic allocation strategies for both VC and VP connections. They also consider dynamic bandwidth allocation for dynamic admission control algorithm (dynamic CAC) and for controlling source rate in ABR. [97] present end-to-end QoS allocation policy using weighted round robin server as cell scheduler. The scheme uses deterministic bandwidth reservation (CAC) at VP level and statistical multiplexing of VCs within each VP. In [96] admission control schemes and routing algorithms are presented for heterogeneous traffic. The notion of effective bandwidth was proposed by Guerin et. al. [49] order to provide statistical guarantees. The effective bandwidth for a source is the amount of bandwidth that must be allocated to the source at each node and is based on its PCR, mean cell rate and average duration of a burst period.

Gun and Guerin [53] present methods for bandwidth management and congestion control using buffered leaky bucket and packet spacer schemes in high speed networks. Some algorithms for bandwidth allocation to the video sources multiplexed at the source node of an ATM network have been presented in [30]. For the purpose of allocating bandwidth, the time is partitioned into cycles and each cycle is partitioned into slots corresponding to cell transmission time. So the bandwidth is allocated to a connection depends on slot and cycle lengths. Reninger et. al. [100] propose a bandwidth renegotiation scheme for VBR traffic by monitoring VBR bit stream and by computing its parameters it determines new service rates to reduce buffer requirements at different renegotiation rates. New structures for ATM switch have also been proposed that conserve cell queues and schedule cell departures [63].

Schemes have also been proposed that monitor the buffer occupancy and decide call admission and cell routing. Marsan et. al. [80] present cell-level and call-level ATM network simulation tools to analyze different network behaviours. Yaprak et. al. [122] propose dynamic buffer allocation to different output ports of a switch with real-time and non real-time input traffic. They also propose a cell accommodation rule to an incoming cell to shared buffer with dynamic threshold updates. Similar kind of schemes have been proposed by [8],[32],[65],[72], [92] and [113] where the performance analysis of various priority queuing strategies and buffer allocation protocols is done and effort is made to optimize discarding threshold values and queue lengths.

Schemes have been used [17][69] to analyse delay-loss priority control mechanism for classes of arrival streams namely delay-sensitive real time traffic and loss-sensitive non real-time traffic. [120] employs a buffer allocation scheme based on complete sharing with virtual partition. In their method the buffer is shared by different types of cells such that underutilized buffer segments can be utilized by an over-subscribing traffic. The new cell arrivals belonging to the under-subscribed traffic push-out excess cells of other traffic types. The scheme proposed in [28] is to dynamically adjust queue threshold for rate based flow control in closed-loop system like ABR. The work of [95] uses CLP bit present in the header of ATM cell to maximize revenue or the network utilization. Arrivals from different links to an ATM multiplexer or output queue are modeled as 2-phase MMBP. They study the queuing buffers under PBS and PBS+PO selective discarding schemes. The method proposed by Rampal et. al. in [98] dynamically obtains the minimum bandwidth necessary to satisfy a specified QoS for a given arbitrary source. It adjusts the instantaneous service rate of a finite buffer queue dynamically in order to meet a specified QoS performance measure. Choi and Kim [27] propose an optimal bandwidth and buffer allocation method that satisfies both CLR and delay requirements of applications. Their queuing model is queue with separate buffers. To find out the optimal buffer size in order to satisfy QoS requirements the mathematical relationship between required bandwidth and allocated buffer size is known in advance.

2.4 Routing in ATM networks

To reduce the size of routing tables and the complexity of network functions the concept of virtual paths (VPs) is introduced in ATM networks. A VP is a logical direct link between two nodes in the network that are connected via two or more sequential physical links. The connections using the links between the two end nodes that a VP defines are bundled together and transported with a virtual path identifier (VPI) common to all virtual channels (VCs). The cells belonging to different VCs are switched using this common VPI along the nodes of the VP, reducing the routing table size. A VC or a connection or a call are considered synonymous. All the VCs on a VP share a buffer space that is dedicated to the VP. A VP network can actually be modeled as a directed graph of nodes and edges where nodes represent switches and edges represent directed VPs.

2.4.1 PNNI Protocol

The ATM Forum which is a consortium of telecommunication vendors has specified Private Network Node Interface (PNNI) that uses source routing among switching systems, in which a switching system is one or more ATM switches. It is the standard for use between private ATM switches and groups of private ATM switches and defines the interface between switching systems including the routing framework required for the interoperability among different switching systems. PNNI uses source routing i.e. the switching system that a connection request originates from its UNI is responsible for finding the end-to-end path to the destination end station. The routing scheme is based on the principles of link-state routing. Each switching system broadcasts link-state information about the outgoing links attached to it using link-state update (LSU) message to other switching systems in the network. Each switching system maintains a topology database that contains link-state information about the links in the network database whenever it receives an LSU. The topology database is the set of resources e.g. nodes, links and addresses which define the network. Resources are qualified by defined sets of metrics and attributes (delay, available bandwidth, jitter etc.) which are grouped by supported traffic class.

Nodes in PNNI protocol are collected into peer groups. All the nodes within a peer group exchange link information and obtain an identical topology database representing the peer group. Within a peer group, each node has the description of the topology of the peer group including descriptions of all nodes, links and destinations that can be reached from each node and the status of nodes between the nodes. The operation of PNNI routing in a parent group attempts to collapse a child peer group into a single node. However the route through an individual peer group is always completed locally by an entry border node of that peer group. Route computation algorithms are not specified in the PNNI protocol. The only requirement is that the selected route must be able to support the QoS requirements of the connection.

The PNNI specification is sub-divided into two protocols: a signalling and a routing protocol. The PNNI signalling protocol is used to establish point-to-point and point-to-multipoint connections and supports source routing, crankback and alternate routing. PNNI allows each implementation to use its own path computation algorithm. After finding the route, the source node uses PNNI signalling to request connection establishment from intermediate nodes along the route. Each node processes received connection request messages, makes connection admission decision and passes the signalling message to the next node along the route or denies the connection request and sends a reject message to the preceding node.

Along with QoS parameters another metric which is used by PNNI to compute routes is the administrative weight. This quantity indicates the relative desirability of using a link or node. The weights of all links at a particular layer are same. These weights are set by the network operator. The administrative weight is also associated with every layer. The administrative weight of a layer is computed as the aggregate of the weights of the actual routes at the lower layers relative to that layer. Therefore its value grows with the layer index. The aim of the route selection process in PNNI is to identify feasible route (satisfying QoS requirement) with minimum administrative weight. Thus it becomes NP complete problem.

2.4.2 QoS Routing

In traditional networks, the routing algorithms implemented are variants of shortest path algorithms that route packets from source to destination over a least cost path and they mainly differ in the cost criteria used. Some networks use fixed cost for each link in the network while some others use some measured metrics such as congestion, mean delay and link utilization [103][110][112][114]. The routing decision place also varies among different networks [79][13][14][29][34][121]. In some networks, each node has the responsibility of selecting an output link for routing packets as they arrive, referred to as distributed routing. On the other hand, in centralized routing, a central node is responsible for making all routing decisions. Under source routing strategy as followed in ATM, the originating node determines the complete path [58]. There is another strategy called flooding which is a distributed QoS routing technique.

Source routing could be static or dynamic depending on whether link-load information is used for path computation or not. In the static routing, static topology information is used in choosing a route. In case of dynamic routing, the information on the available resource on each link must be distributed throughout the network, so that any source can have access to the correct information on the resources available in the network. The information distributed is called link-state and thus source routing is also called link-state routing.

In conjunction with resource reservation and admission control, finding a route which can provide user-requested QoS is an issue. QoS based routing is a routing mechanism under which paths for flows are determined based on some knowledge of resource availability in the network as well as the QoS requirements of flows. It enables a provision where a route can offer the requested QoS provided it has adequate available resources. If during the connection time the assigned route can no longer support the QoS parameters then the connection is not terminated but rather it should be assigned another conforming route.

QoS based routing extends the current routing paradigm for both unicast and multicast ATM networks in three basic ways

- Additional routes are available to support B-ISDN traffic and routing metrics e.g. delay and available bandwidth. If routing metrics change then routing tables must be updated.
- Traffic will be shifted to better available path as soon as it is available. However frequently changing routes can increase delay variation and jitter.
- If best path is not available then adequate alternative path can be used, if it exists.

One common set for QoS specifications in ATM networks could be Cell Loss Ratio (CLR), Cell Transfer Delay (CTD) and Cell Delay Variation (CDV). QoS based routing cannot support QoS requirements that cannot be meaningfully mapped onto a reasonable combination of path metrics. There can be three approaches to QoS requirements of any connection or flow. The first approach is to allow flexible combination of parameters where different parameters are ordered by priorities. The second approach is to combine QoS parameters into single value. The third and the last approach is to consider a used parameter set of QoS parameters where parameters are treated equally. QoS of a route is then based on this set. These approaches however reduce the degree of freedom in the route selection, because information about the actual QoS requirements are not known before the connection request arrives.

One of the earliest work in QoS routing is by Wang and Crowcroft [117]. They show that finding QoS routes with multiple constraints simultaneously is NP. They present shortest-widest and bandwidth-delay constrained paths for distance vector and link-state and hop-by-hop methods. Crawley [33] define and considers various issues involved in QoS routing in packet networks including ATM networks. They also discuss issues involved in interdomain and intradomain routing. A different method for computing QoS routes is suggested by Vogel et. al. in [116]. Here all parameters are considered equal. Thus better- suited path selection is based on comparing all parameters simultaneously i.e. using AND operator. In case alternative routes exist, parameter sets have to be compared. A relative quantity called availability is introduced. Path computation is then based on comparing the minimum of availability parameters for all parameter sets. That path is refused which has the lower value of availability. Alternatives are also suggested in case minimum availability of two paths is equal. Scheduling schemes have also been suggested that obtain optimality criterion for path selections [82][87].

Orda [87] considers rate based schedulers for finding constrained paths. The work by Ma and Steenkiste in [76] is to identify means of determining QoS routes meeting multiple constraints. They describe four optimality criteria for path selections based on rate proportional service discipline and simulate them for two types of network topologies. They use iterative Bellman Ford algorithm to solve multiple constraints as it provides minimum hop objective inherently. Apostolopoulos et. al. [2] provide an excellent study on QoS routing and its overhead. They employ a disjoint multipath topology to pre-compute multiple disjoint paths. Chen and Nahrstedt[19] overview and present a good survey on different QoS routing algorithms. They later define multi-constrained path problem for QoS routing and present routing algorithm with polynomial time complexity [22]. In this work they consider delay and cost constraints which are both additive parameters. They also present distributed QoS routing algorithm algorithms [21] and [20]. In the latter work they use bandwidth metric. Constrained routing algorithms have also been presented in [36] [64] [84] [83] and [70]. They consider several path selection algorithms for multiple constraints and determine QoS guaranteed and best-effort routes for various load conditions [77][78].

Sivabalan and Mouftah [109] investigate the impact of alternate path routing on network performance and suggest schemes to control the performance degradation at higher loads. Efficient algorithms for finding k-shortest paths from one node to all other nodes are presented in [115] and [73]. Guo and Matta [54] find out k delay-cost constrained routes by computing the path weight as a function of both path cost and delay. In [3] issues in pre-computation of QoS routes are considered. QoS routing with on-demand and pre-computed path selections has also been considered in [51]. They consider rate based scheduling to handle delay constraint.

Route pre-computation for QoS routing has also been considered in [106]. They also study the effect of link state update (LSU) policies on the performance of their algorithm. Some new LSU generation techniques are presented in [93]. They use both triggered and periodic update strategies. The effect of frequency of LSUs on QoS routing has also been considered in [1], [105] and [107]. [50] study the complexity of rate based and delay based model for inaccurate or stale link states. [66] suggests to distribute link state information by flooding where intermediate nodes do not keep link state information but keep minimum-hop path through each outgoing link to every other node. In [37] algorithms for QoS routing with multiple constraints are presented. To remove inaccuracies in the link state information, they use triggered updates. [5] is a similar study where they consider different traffic sources and path selection algorithm.

2.5 QoS Management

As discussed in the previous chapter, the quality of service is the collective effect of service performances which determine the degree of satisfaction of a user of the specified service. The role of QoS negotiation is to find an agreement on the required values of QoS parameters between the network and the users. The role of QoS adaptation is to keep providing the negotiated quality of service, eventually lowering it in case when resources are not available. The renegotiation may be user-initiated that is the user requests a better quality or it may be network initiated that is the network can no longer support the negotiated QoS and quality drops below the acceptable limit. In such a case, the user is asked to accept a lower quality. Some fundamental schemes for graceful adaptation of QoS of a real-time connection to modify the route established for a channel can be found in [89]. They propose an adaptation policy where the needed resources are equally divided among the clients those consent to adaptation.

Campbell et. al. [16] present a generalized framework to QoS and evaluate various QoS architectures in distributed multimedia systems. In general QoS architectures couple path establishment and resource reservation however some architectures like IETF Intserv model decouple them and then provide QoS. IETF intserv model provides soft QoS via RSVP and IPv6 flows. According to [16], QoS management can be divided into two categories static and dynamic. They involve activities that allow the support by the service provider of the desired QoS. The static QoS management techniques include QoS mapping, admission control and resource reservation. Dynamic QoS management include QoS monitoring, QoS maintenance (adaptation / renegotiation). Some fine grain schemes for QoS maintenance are loss via buffer manager, queuing delays via flow schedulers and throughput via flow regulators. The work presented by Hafid et. al. in [56] is an excellent reference on quality of service management in distributed multimedia applications for dynamic QoS management. They extensively study the issues in QoS architectures and QoS management techniques. [47] and [101] propose models to establish and manage data session flows and maintain QoS. Hafid et. al. present in [57] a new negotiation approach to QoS based on duration of the requested service. Their approach uses

QoS manager and agents to calculate amount of resources required to be reserved for providing the desired QoS. It uses the available QoS projections of involved components to provide the end-to-end QoS requirements of the user. In a related work by [55] and [12] identify characteristics and properties of multimedia applications and propose techniques for graceful adaptation of QoS. The work of [11] is similar in spirit to these works of providing dynamic schemes for QoS adaptation with renegotiations. An architecture for QoS management is also proposed in [41]. Their scheme is based on distributed and coordinated QoS management by using agents located on all the nodes of the system.

In [42] schemes for QoS management for multimedia applications in the MBONE environment are implemented. A graphical user interface is available to give users the ability to specify the desired quality of service. A language system called QUAL is described by Florissi and Yemini et. al. in [43] where QoS attributes are specified via language constructs and system is monitored using dedicated monitoring processes called CMP. Their system is based on SNMP protocol for network management and it augments the management information base (MIB) that contain application statistics to include information about application QoS for global use.

The QoS manager as a separate entity is is crucial in the architecture proposed by Barzilai et. al. [10] for RSVP based system. Their QoS manager performs various control functions like maintaining reservation status, managing buffers and policing traffic. Their protocol however do not manage the QoS parameters. According to [52] interoperability between environments employing different QoS mechanisms must be considered. Chatzaki and Sartzetakis [18] propose a routing management mechanism for a multidomain environment supporting multi-class services. Network environment is structured hierarchically and the QoS routes are determined using on-demand source routing algorithm. They employ QoS manager for managing local resources that interacts with QoS managers of neighbouring network domains for information exchange and maintenance. According to Francis-Cobley et. al. [45] mapping of QoS parameters is important because of interworking of different network technologies (IP / ATM) as the user requirements must be translated across network boundaries when heterogeneous networks are there. In the work by Lakshman and Yavatkar [67] ATM signalling framework is extended and APIs are proposed to allow an application to modify a QoS reservation for an ongoing connection. The changes are propagated downstream to all switches in the connection path. They do not consider QoS renegotiation initiated by the network or the system. Bansal et. al. [9] describe a QoS management system by proposing an ATM API called ATM Service Manager (ASM). This interface provides applications or users choice to specify traffic and QoS parameters and transport protocol preferences. The ASM then decides best transport protocol options based on user preferences.

Chapter 3

QoS Provisioning Through Dynamic Bandwidth Allocation and Buffer Control

3.1 Introduction

Asynchronous transfer mode (ATM) networks support a wide variety of applications with different traffic characteristics and quality of service (QoS). Since network resources are finite, efficient bandwidth and buffer allocation methods utilizing the limited resources efficiently and satisfying different cell loss ratio (CLR) and delay requirements are needed. Satisfaction of the different QoS requirements is a resource allocation problem. To provide QoS guarantees ATM networks allow applications to reserve resources for network connections based on traffic parameters. As discussed in chapter 2, the quality of service and bandwidth are mainly related to physical resources. Objects using allocated physical resources in an ATM network include cells, bursts, virtual circuits, virtual paths, and signalling message. A cell occupies a buffer and cell transmission requires bandwidth. Therefore both buffer and bandwidth allocation to cells is required. A whole stream of cells in an ATM connection uses buffers and this transmission also requires bandwidth. Buffers and bandwidth are allocated to each cell and each burst by allocating buffers and bandwidth to the virtual channel containing the cell or burst.

In bandwidth allocation methods, the bandwidth is calculated under the assumptions that the buffer size is fixed. This means that if an arriving cell finds the buffer busy or full, the cell is dropped. Buffer allocation to provide an efficient and fair use of the available buffer spaces is critically important for ATM networks. Under the proposed scheme the allocated buffer space of the connection is reallocated based on measurement of data from the system for certain time duration. Rampal et. al. have shown that there exists a relationship between queuing buffer size and cell losses [98]. In the same work the authors have proposed an alternative algorithm to allocate resources and providing QoS guarantees. They proposed an algorithm called REQS that performs better than the equivalent bandwidth approach [49].

We have also investigated the use of cell loss priority (CLP) bit of cell header. The CLP capability is used in ATM networks for the purpose of congestion control. If the CLP bit in the header of a cell is set to 1 then the network discards this cell in case of congestion. The CLP bit has another use also. It allows applications to declare two types of traffic with different QoS constraints namely the precious, high priority traffic with CLP=0 and the less precious, low priority traffic with CLP=1.

3.2 Provisioning of Quality of Service

3.2.1 Metric Selection

The QoS metrics reflect the basic characteristics of the network. The resource requirements specified by the applications are often diverse and application dependent. The computational complexity of any QoS configuration is determined by the composition rules of the metrics. Three basic composition rules are additive, multiplicative and concave. Different parameters generally fall into either of these categories. The delay, delay jitter and cost follow the additive composition rule, and bandwidth follows the concave composition rule. The cell loss rate can be represented as a multiplicative metric. The satisfaction of these parameters at the same time has shown to be in NP [117]. One strategy is to utilize sequential filtering of parameters according to some kind of priority.

3.2.2 Resource Allocation Method

The algorithm REQS dynamically obtains the minimum bandwidth necessary to satisfy a specified QoS for a given arbitrary source. It dynamically adjusts the instantaneous service rate μ_t of a finite buffer queue in order to meet a specified QoS performance measure Q_l. The basic step is given as

$$\mu_{n+1} = \mu_n + K_n.E(P_n, Q_l)$$

In this equation term μ_n is the service rate in the nth update interval, $E(\,)$ is an error function which is a measure of how far the current cell loss rate is from the targeted QoS and K_n is a scalar term that depends on the interval.

If cell loss rate is taken as the QoS parameter then the error term $E(P_n, Q_l)$ is taken to be $\log(P_n, Q_l)$ where P_n means cell loss rate in the nth update interval and Q_l is target QoS performance measure that is the cell loss rate. Figure 3.1 shows model of the queuing system under consideration.

The algorithm has two distinct modes of operation. In the first mode the length of the update intervals are kept fixed and K_n is varied. In the second mode, the algorithm tries to gradually converge exactly to the desired cell loss rate. In this mode K_n is kept fixed, but the size of the

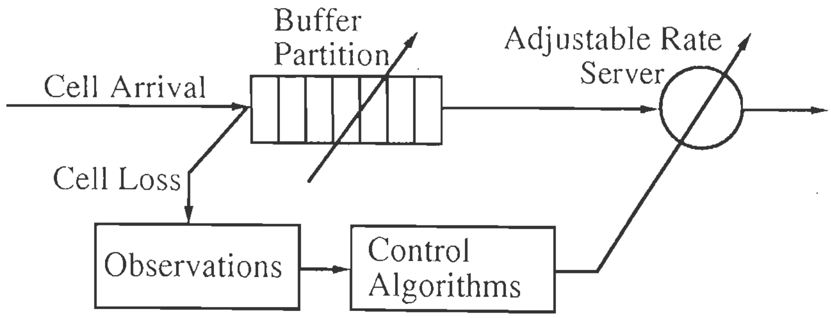

Figure 3.1: Model of the queuing system.

update intervals is varied. Decisions about adjustment to the resource allocations are made at discrete instants t_i; the interval between decision points t_n and t_{n+1} is referred to as the nth update interval U_n. The service rate over the nth update interval, μ_n is assumed to be constant. The algorithm converges when the resource value remains within 5% of the final value. The time required for the resource value to arrive at and stay within this limit in the steady state is called as the convergence time.

In this scheme bandwidth (service rate) reallocation is not continually required during the duration of a session. The algorithm determines a steady effective bandwidth so that once the algorithm has converged, no bandwidth reallocation is necessary.

3.3 Proposed Scheme

The buffer space is generally allocated based on the parameters negotiated at the connection setup time or estimated based on traffic monitoring. Buffering may be done by shared central buffers. Ideally it would be best as all the available storage can be dynamically allocated to arriving packets that is the buffer space being allocated on the basis of each packet received. We assume that there is a shared pool of buffer space that releases or receives back the buffer space as per requests received for buffer allocation. The required bandwidth of a buffer decreases with respect to increase in buffer size. If the required bandwidth can be represented as a function of the allocated buffer size then optimal buffer space can be allocated using a method like in [27]. But this can be done only when the exact mathematical relationship between bandwidth and buffer space is known. Also there exists a relationship between queuing buffer size and cell loss. In the proposed scheme initially some buffer space is allocated and then this buffer space in the queuing system is controlled through the measurement of cell loss rate periodically. This measurement is done only after a certain initial setup time. Depending on the monitored CLR, buffer space is reallocated. Under this scheme if the cell loss rate is found to be very poor when the measurement of data is done then the service rate is increased so as to guarantee the target cell loss rate (CLR). Service rate will increase if buffer size is decreased. Alternatively service rate decreases if the buffer size is increased. This is done as expressed below.

$$\begin{aligned} if(clr > 2.0 \times Q_l)\ BUFPSIZE &\ -= \ abs(\ln(clr/Q_l)) \\ if(clr < 2.0 \times Q_l)\ BUFPSIZE &\ += \ abs(\ln(clr/Q_l)) \end{aligned} \tag{3.1}$$

where $BUFPSIZE$ is the size of buffer partition and clr is the monitored cell loss rate. Under our scheme this change in the buffer requirement is done only after initial setup time and once the cell loss rate is satisfied, the buffer size does not change for the remaining duration of the session. The extra buffer space required during the monitoring phase may be obtained from the shared buffer space. Similarly any freed up buffer space may be returned to the shared buffer space. The scheme thus helps in optimal use of buffer space.

While investigating the use of CLP capability of ATM, effectiveness of the proposed scheme was also investigated in push-out (PO) and partial buffer sharing (PBS). These schemes are used to distinguish traffic with different cell loss requirements and to increase the utilization of the network [92][86][44]. In the push-out scheme cells are admitted into the buffer regardless of their priorities until it becomes full. Once full, newly arriving low-priority cells are prohibited in the buffer and discarded, but newly arriving high-priority cells replace low-priority cells within the buffer.

The PBS scheme accepts cells into the buffer regardless of their priorities up to a pre-assigned threshold. Beyond this threshold, only high-priority cells are allowed to enter. In the case of this scheme the threshold was varied if the buffer space was reallocated. This is required so that there is little likelihood of cell loss rate for CLP=1 traffic to increase for the newly allocated buffer space.

3.4 Simulation Process

The queuing behaviour and hence the cell loss rate depends on the nature of the arrival processes. A two state Markov modulated Bernoulli process (MMBP) was used as the traffic source [85][104]. The 2-state MMBP is in either of two states. In each state cells are generated according to a Bernoulli process. The duration of two states is geometrically distributed. While investigating CLP capability we assume that both CLP=0 and CLP=1 cells can arrive in either phase of the modulating Markov process. We also assume that different classes of cells have same service requirements, cells get lost only when the buffer is full or may be exceed some threshold and the server is not left idle when the buffer is not empty. The various simulation parameters assumed were as follows.

Average cell generation rate in state 0 of MMBP, $\lambda_0 = 200\, cells/s$,

Average cell generation rate in state 1 of MMBP, $\lambda_1 = 1000\, cells/s$,

Average duration of state 0 of MMBP, $\mu_0 = 0.02\, s$,

Buffer Size (cells)	Cell Service Rate (cells/ms)		Convergence Time (s)	
	Static Buffer	Dynamic Buffer	Static Buffer	Dynamic Buffer
10	0.3790	0.3827	48	42
20	0.4156	0.4234	325	294
40	0.4512	0.4637	437	406
80	0.5210	0.5023	512	483
120	0.4972	0.5294	725	710

Table 3.1: Steady state cell service rate and convergence time with variation in the initial buffer size for static and dynamic buffer allocation.

Average duration of state 1 of MMBP, $\mu_1 = 0.01\ s$,

$Q_l = 10^{-3}$,

$U_0 = 0.5\ s$,

$BUFSIZE = 80\, cells$.

The initial and final value of the scalar K_n were set to 1% of λ_1. The simulation run was performed for a duration of 5 hours of simulation time.

3.5 Discussion of Results and Conclusions

The service rate in our simulations was modified using the REQS algorithm. For all the values for the cell service rate and cell loss rate, the 90% confidence intervals were computed.

Table 3.1 shows the cell service rate for the steady state and convergence time for satisfaction of target cell loss rate for different initial buffer sizes. Figure 3.2 shows cell service rate for the two cases where the cells arrive to a static buffer that is the buffer size is kept fixed and to a dynamically allocated buffer where buffer space is modified according to Equation 3.1. It may be seen in this figure that the cell service rate increases sharply during the initial phase due to increase in the second term of equation for cell service rate. As the resource allocation method enters into the second mode the error term approaches zero and the cell service rate becomes

U_0 (sec)	Convergence Time (sec)	
	Static Buffer	Dynamic Buffer
0.5	138	132
5	195	190
50	512	483
200	1137	1125
500	2678	2506

Table 3.2: Convergence time with variation in the initial update interval, U_0 for static and dynamic buffer allocation.

constant. For the results shown in the figure the buffer size was reallocated to 77 cells as against the initially allocated 80 cells. The service rate is higher and the convergence is smaller in the reallocated buffer case. An effect of this is that average delay of a cell occupying the buffer gets decreased. As shown in Figure 3.3, the resource allocation method is such that the final cell loss rate is same as the target cell loss rate. Similar result is obtained for cumulative cell loss rate. We also measured the convergence time for Table 3.2 shows the convergence time for satisfaction of target cell loss rate for different initial update intervals. It is observed that convergence time is smaller in the case of dynamically allocated buffer space.

In the case of push-out (PO) and PBS cell discarding schemes the system quickly converges to a final steady state service rate. Figure 3.5 shows the cell service rate for the PO scheme. While investigating the use of CLP capability, the CLR for CLP=0 cells was specified as the target CLR (Figure 3.6) and no target cell loss rate was specified for CLP=1 cells (Figure 3.7). This is because the CLR for CLP=1 traffic is more than CLP=0 traffic and hence when CLR for CLP=0 traffic is met (severe QoS requirement), the service rate is such that CLR for CLP=1 traffic gets satisfied itself. In the case of PBS scheme the threshold was kept to 80% of the buffer size. The cell service rate for the PBS scheme is shown in Figure 3.8.

The proposed scheme thus allocates minimum bandwidth dynamically that satisfies QoS requirement namely the CLR and utilizes buffer space efficiently. The scheme was applied on cell discarding schemes namely push-out and PBS schemes using CLP capability of ATM networks. Our scheme could be used in conjunction with other methods to guarantee or improve QoS. We have taken stationary traffic source in the simulation. If however the source is non-stationary then feedback from the monitored parameters could be used by the traffic sources to change its operation mode to satisfy QoS. For example if the traffic source is MPEG video then the source could change its GOP pattern and transmit frames with cells of different priority similar to controlled CLP=1 traffic [95] that guarantees cell loss rate of CLP=0 traffic while accepting substantial CLP=1 traffic. This would improve the QoS. The technique like partial packet discard could further improve QoS as it prevents useless cells (tail) to get transmitted and congest the network.

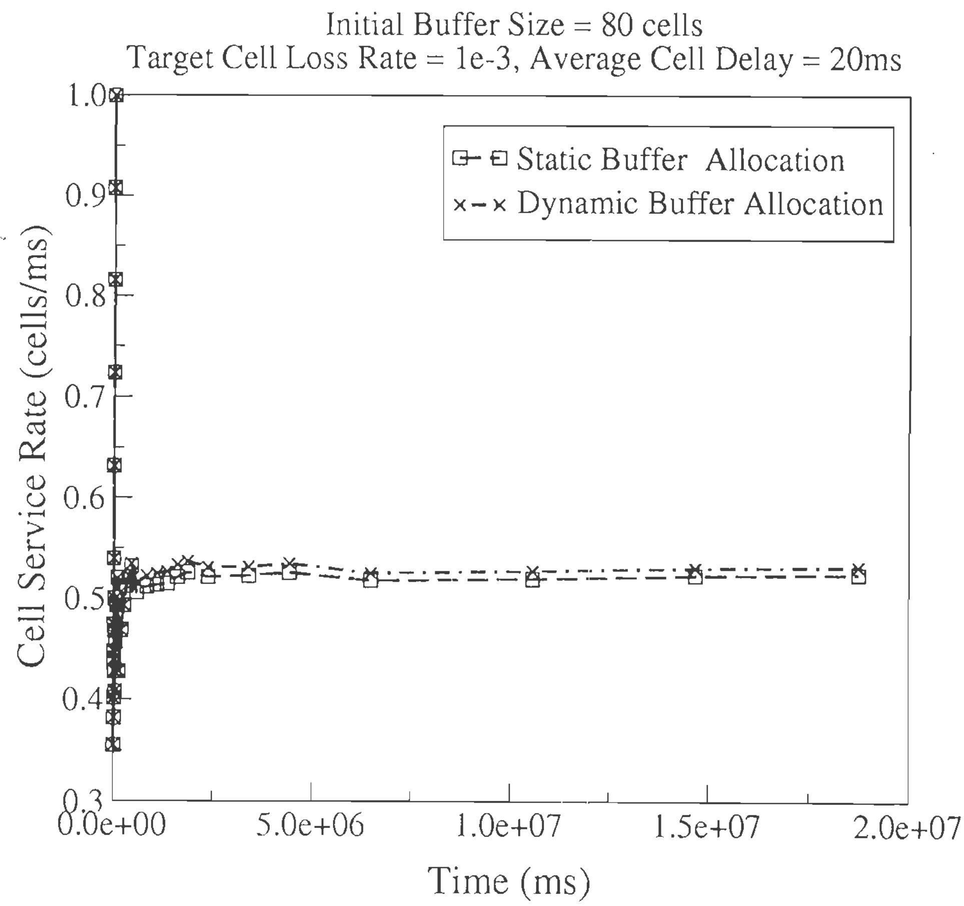

Figure 3.2: Cell service rate for static and dynamic buffer allocations.

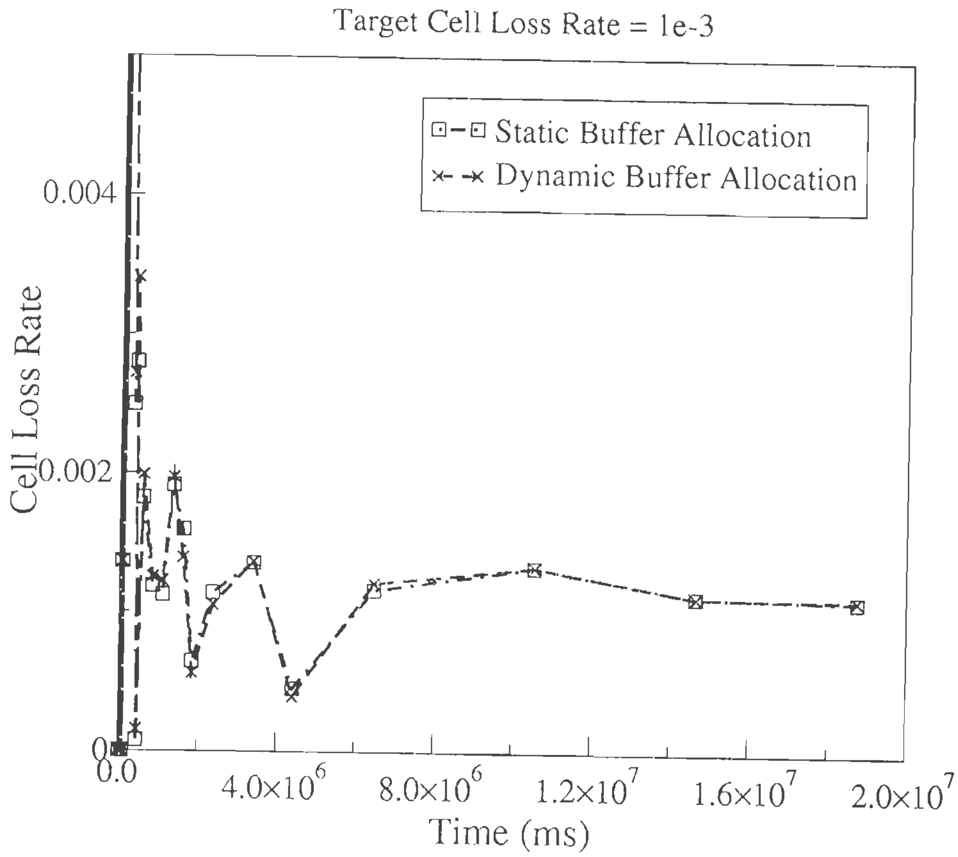

Figure 3.3: Cell loss rate for static and dynamic buffer allocations.

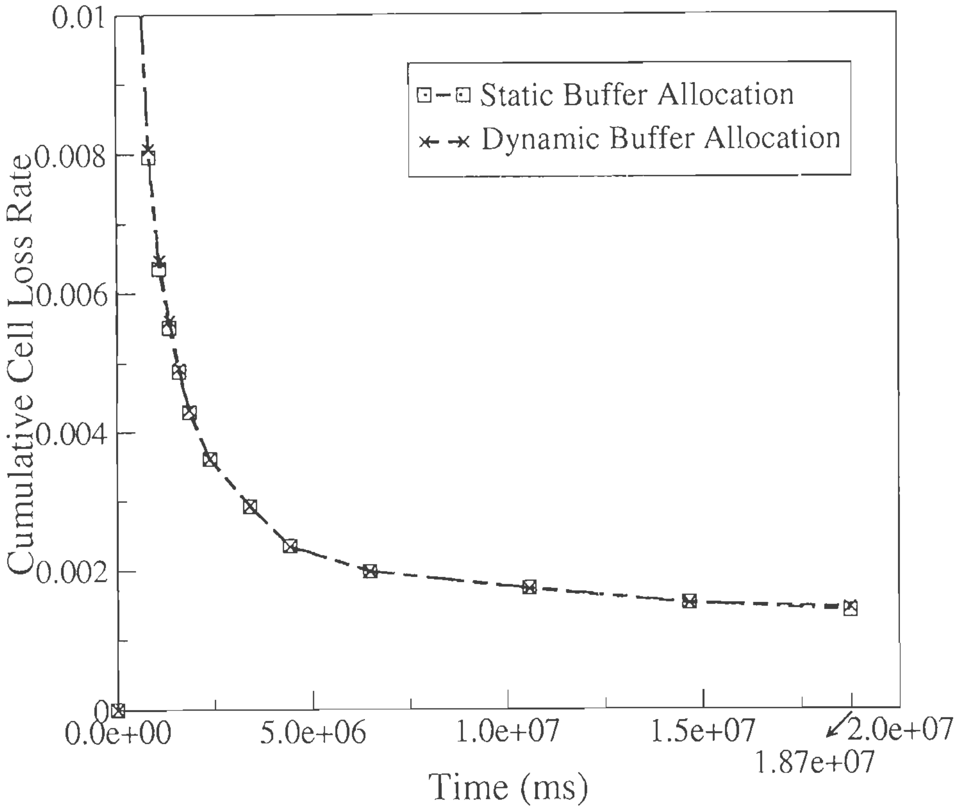

Figure 3.4: Cumulative cell loss rate for static and dynamic buffer allocations.

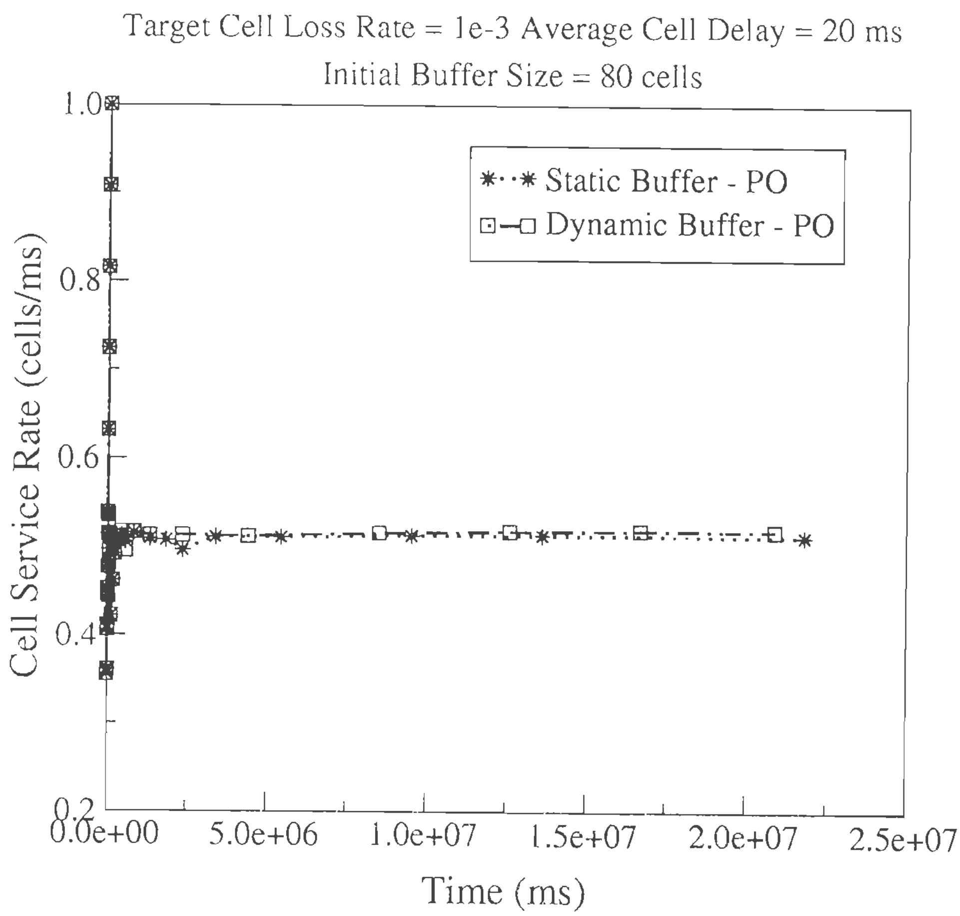

Figure 3.5: Cell service rate for static and dynamic buffer allocations with push out cell discarding scheme.

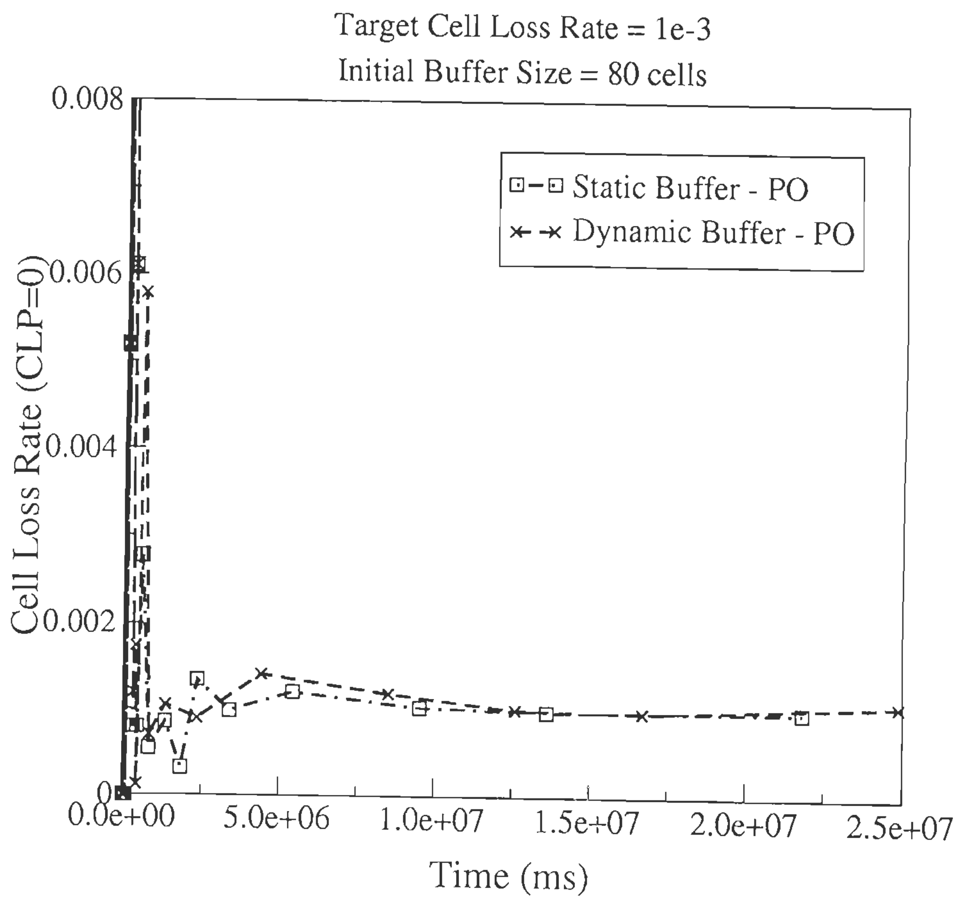

Figure 3.6: Cell loss rate for CLP=0 cells for static and dynamic buffer allocations with push out cell discarding scheme.

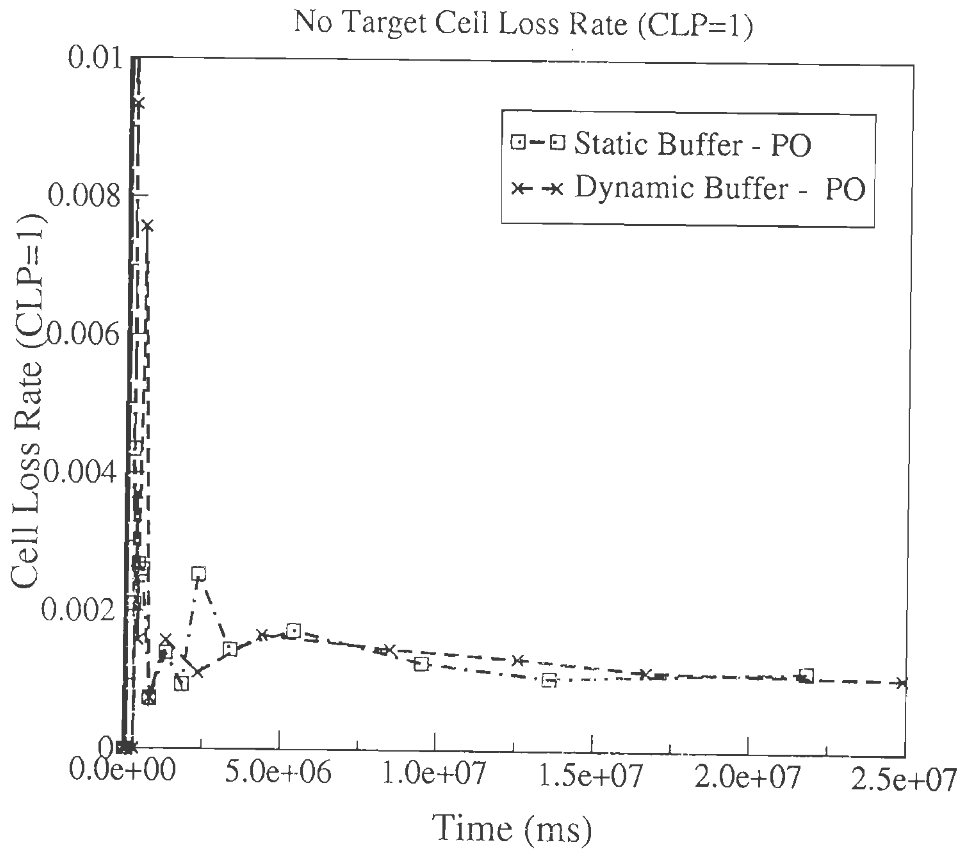

Figure 3.7: Cell loss rate for CLP=1 cells for static and dynamic buffer allocations with push out cell discarding scheme.

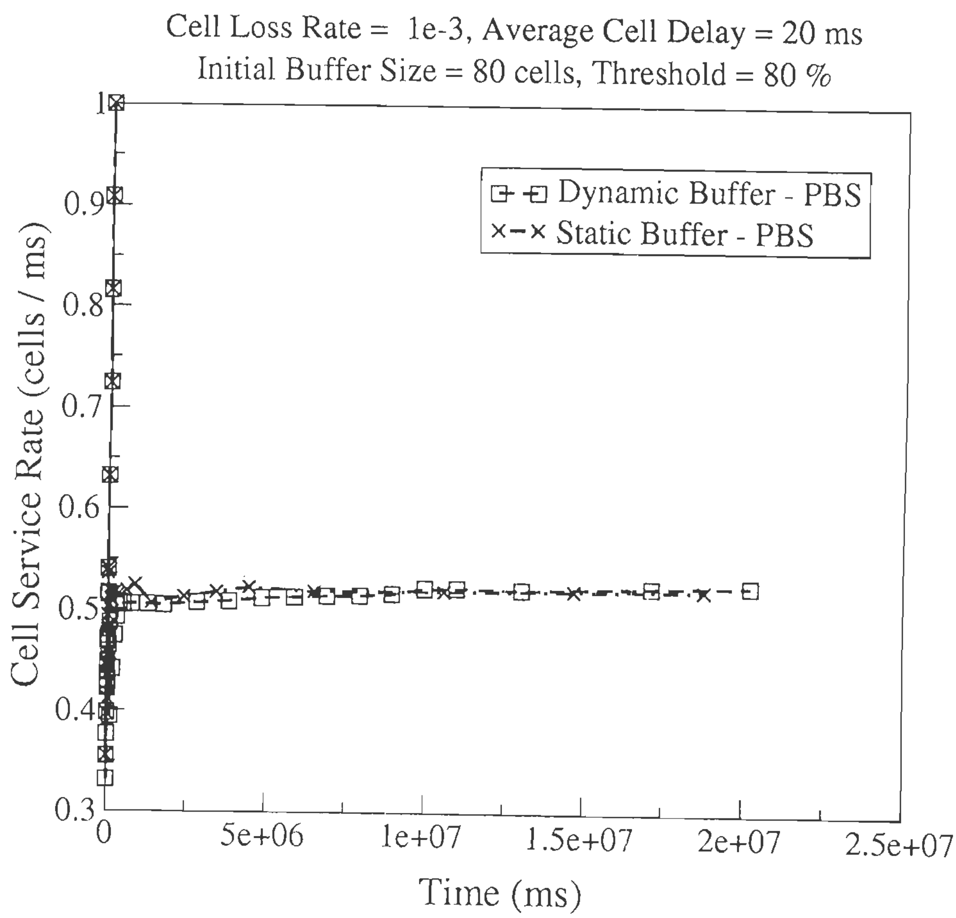

Figure 3.8: Cell service rate for static and dynamic buffer allocations with PBS cell discarding scheme.

Chapter 4

Performance Guarantees through Pre-Computed QoS Routes in ATM Networks

4.1 Introduction

To guarantee a certain QoS in ATM, resource management based on traffic characteristics is required. During flow setup, at first a route has to be determined. Then resource managers at the selected nodes accept requests for the desired QoS, test whether this QoS can be provided and report back the result. This process is repeated hop-by-hop until all destinations are reached. The QoS parameters are exchanged in the form of flow specification and have to be mapped onto link metrics. To avoid violations in the quality of a distributed multimedia application, the set of parameters should provide all QoS parameters of the application. However, the communication, computation and storage complexity of the routing method is increased by every additional, equally treated, not correlated parameter. QoS routing is the process of selecting the path to be used by the packets of a flow based on its QoS requirements and the resource availability of the network [76][19][84][117]. QoS routing is therefore used in conjunction with some form of resource reservation or resource allocation scheme as discussed in the previous chapter. In this chapter we propose a new algorithm for pre-computation of k-constrained QoS routes and evaluate its performance.

4.2 Characteristics of QoS Routing

For QoS based routing certain objectives must be achieved before it is successfully implemented. These are dynamic determination of feasible paths, optimization of resource usage and graceful performance degradation. The QoS requirements of an application are specified either as a set of path-constraints or a set of link constraints. A path with sufficient resources to satisfy the QoS requirements of an application is called a feasible path. The connection admission control (CAC) algorithm determines if a path is feasible, which depends on the dynamic network load at that particular moment.The QoS routing algorithm should be capable of identifying feasible paths so as to satisfy the maximum possible number of flows with QoS requirements.

QoS routing algorithms achieve resource efficiency by limiting resource consumption while balancing the network load. Under light traffic load when network resources are readily available, QoS routing is more useful in balancing the traffic than searching feasible paths. Balanced traffic distribution helps to increase the call admission ratio of future connections and improve the response time of best effort traffic. However when the network load is heavy and dynamic, efficient algorithm for finding feasible paths for the current request is critical as it also relates to increasing the chance of accepting future requests.

Important aspect in minimizing the impact of QoS routing is to develop a solution that has the smallest possible computational overhead. QoS routing incurs additional cost. This cost has two components, namely computational cost and protocol overhead. The former is due to more more frequent path selection computations and the latter is caused by the need to distribute updates on the state of the network resources that are of relevance to path selection e.g. available link bandwidth. Such updates translate into additional network traffic and processing. Additional computations are unavoidable, but it is desirable to keep the total cost of QoS routing at a level comparable to that of traditional routing algorithms [51]. Several studies in the recent past have shown that quality of service routing can provide increased utilization compared to routing that is not sensitive to QoS requirements of the traffic [2][76][117]. QoS routing attempts to improve network utilization by diverting traffic to paths that would not been discovered by traditional non QoS sensitive routing. QoS routing is more useful and more effective in environments where traffic and network capacity are mismatched and alternative paths with lower load exist. The following parameters strongly influence the computational cost of a QoS routing solution

- Path selection criteria : Optimize multiple criteria as well as satisfy additional constraints.
- Trigger for path selection: Periodically/ In response to significant change.
- Flexibility in supporting alternate path selection choices. The unavoidable inaccuracy in the network state information has implications for the path selection process. It may thus be desirable to maintain and alternate between several choices in order to avoid single choice poor path.

The following could be some path selection metrics:

Link Available Bandwidth - Current amount of available (unallocated) bandwidth to satisfy a new flow's requirements.

Hop-Count - Measure of the path cost to the network. A path with a smaller number of hops is preferable, since it consumes fewer network resources.

Policy - Use of policies to handle specific requirements allows considerable simplification in the optimization task to be performed by the path selection algorithm. Policy could be for example to prune links from the network that are incompatible with the requirements of a flow.

QoS routing relies on state information specifying resource availability at network nodes and links in the form of a database, and uses it to find paths with enough free resources to accommodate new flows. In turn, the successful routing of new flows together with the termination of existing ones, induce constant changes in the amount of resources available. These must be communicated back to QoS routing to ensure that it makes its decision based on updated information. These changes are communicated using link-state information which can be propagated periodically or in response to a significant change (called trigger) in the link-state metric. The significant change ensures that link becomes congested or at least has some traffic. The significant change may also be due to the establishment or teardown of a connection.

The link-state updates are accomplished by having every node broadcast the load values of its links i.e. link-state metric to all other nodes either using flooding or along a minimum spanning tree. This broadcast is done upon the establishment, modification or teardown of a channel. If there are no channel establishments, modifications or teardown within a specified time interval, periodic link updates are sent by each node to assure other nodes that link is still active. A routing failure or setup failure is said to occur if the source cannot find a route based on link-state information. A signalling failure occurs if one of intermediate nodes on the route determined by the source cannot reserve the resources required to satisfy QoS of the requested connection. Periodic propagation of link-state information, that is, some minimum time between update messages ensures that overloading effects due to network bandwidth and processing resources during rapid fluctuations in link bandwidth are avoided. However coarse triggers and large periods result in stale link-state information or inaccurate information which can cause a switch to select a suboptimal route or a route that cannot accommodate the new connection. This may also incur setup failures which may require additional resources for computing and signalling an alternate route for the connection. According to Guerin and Orda [50] there are two main components to the cost of timely distribution of changes in network state, the number of nodes/links generating such updates and the frequency at which each entity generates updates.

QoS routes can be either be selected on-demand or they can be pre-computed. With on-demand routing, the path selection algorithm is executed for every request. With pre-computed paths, the path selection algorithm is executed periodically as new routing information is received. The path pre-computation has been found as a means of reducing the computational cost of QoS route computation [3][106].

4.3 Multi-constrained Routing Problem

There is a class of routing problems called multi-constrained routing problem which is a multi-objective optimization problem and is typically in NP. QoS constraints for traffic requiring delay guarantees include end-to-end delay, delay-jitter and buffer space bounds. These constraints are considered independent. The multi-constrained problem is finding a path such that it would satisfy all the QoS requirements. That is,

$R(\mathbf{p}) = \min[r_i] \geq R$, $D(\mathbf{p}) = \sum d_i \leq D$, $J(\mathbf{p}) = \sum j_i \leq J$, and $b_i \geq$

B, for all $i \in \mathbf{p}$

where $R(\mathbf{p})$, $D(\mathbf{p})$ and $J(\mathbf{p})$ represent the bottleneck rate, delay and delay-jitter on the path $\mathbf{p}$ respectively. The respective QoS constraints are R, D and J. The buffer constraint is specified as B. The multi-constrained routing problem is therefore difficult because different constraints can conflict with one another. An example of multi-constrained problem is delay-bandwidth constrained routing problem which is NP-complete [117].

It has been observed [123] that QoS constraints are interrelated in a way that is determined by the network scheduling discipline. Rate-proportional service disciplines ensure that flows sharing an output link get their proportional shares of the link capacity and therefore, the end-to-end queuing delay, the delay-jitter and the required buffer space in each network node is determined by the bandwidth reserved for the flow and the characteristics of the traffic source. We assume that the scheduling policy in the network is rate based and therefore finding a path that satisfies delay, delay-jitter and buffer-space constraints is solvable in polynomial time [22][78].

4.3.1 QoS Constraints under Rate based Scheduling

Given a path $\mathbf{p}$ of n hops with link capacity C_i for link i and a traffic source constrained by a token bucket $\langle \sigma, b \rangle$, where σ is the average token rate and b is token bucket size, the provable end-to-end delay bound (i.e. including all links) is given by

$$D(\mathbf{p}, r, b) = \frac{b}{r} + \frac{n.L_{max}}{r} + \sum \frac{L_{max}}{C_i} + \sum prop_i \tag{4.1}$$

where r ($r \geq \sigma$) is the bandwidth to be reserved, L_{max} is the maximal packet size in the network and $prop_i$ is the propagation delay. The end-to-end delay-jitter bound and buffer space requirements at the h-th hop are given by

$$J(\mathrm{p}, r, b) = \frac{b}{r} + \frac{n.L_{max}}{r} \tag{4.2}$$

$$B(\mathrm{p}, b, h) = b + h.L_{max} \tag{4.3}$$

A path is feasible for traffic with delay guarantees, if it meets the delay, delay-jitter and buffer space requirements.

4.4 Proposed Scheme

Path pre-computation schemes offer multiple candidate routes to each destination. These choices are useful in case of set-up failure. In the proposed scheme multiple routes are pre-computed. These are selected with the criteria described in [76]. With pre-computed paths, the path selection algorithm is executed periodically as new routing information is received. The pre-computation period should be same or more than the link state update period. If the pre-computation period is less than the link state update period then it means that several pre-computations are performed on same link state information and thus would not generate any newer route information.

Under pre-computation of QoS routes, the source node pre-computes constrained paths to each destination using some routing algorithm. As we have considered rate based scheduling, it is clear from Equation 4.1 that there is a dependency between the bandwidth, the delay and the path being selected. Due to this reason, a traditional shortest path algorithm is not useful. The two common shortest paths algorithms used to determine the route to destination in networks are Dijkstra and Bellman-Ford algorithm [31]. Both of these algorithms can be modified to find out constrained paths [22][76]. However, Bellman-Ford algorithm is preferred as it directly gives the hop count to the destination. Pre-computation of paths is costly as paths to all destination are determined in advance and this requires processing and additional storage space. Instead of pre-computing all possible candidate paths to destinations, we consider k-constrained paths ($k > 1$) as this reduces search space, and reduces processing and storage costs. When a connection request arrives best path is selected among these k-constrained paths. To compare the performance, we also used Bellman-Ford algorithm to find out k-constrained paths with constraints based on rate-based scheduling. This version of the algorithm is called Modified Bellman-Ford (MBF) algorithm. This algorithm can be implemented in time $O\,(cen)$, where c is some integer, n the number of links and e the number of links in the network.

The Equation 4.1 also suggests that a path with low end-to-end delay should have few hops and a high reservable bandwidth (r). Thus the criterion for selecting a path is to select path with the minimum hop count, minimum end-to-end delay and minimum load. Minimum load path is one that has the maximum reservable bandwidth (mrb). This also emphasizes distributing the load in the network. The $mrb_{\mathbf{p}}$ of a path is $min\,\{R_j \, \forall \, j \in \mathbf{p}\}$, where $\mathbf{p}$ is the path and R_j the residual bandwidth i.e. the amount of reservable bandwidth of link j. There could be several heuristics by combining the optimality criteria discussed above [76][78][87][117].

In [2] a disjoint multipath topology has been used for pre-computation of alternate QoS routes. The advantage of this topology is that it results in multiple disjoint routes. Due to these disjoint routes this particular topology resulted in improved performance over ISP topology. Instead of relying on the topology we propose a routing algorithm for computing disjoint QoS routes.

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4.4.1 Algorithm

For pre-computation of k-constrained paths, our algorithm is based on the algorithms presented by Topkis [115] for k-shortest paths problem. The k shortest paths are the paths with distinct initial links that is the links from the source node. We improved upon this algorithm to pre-compute k QoS routes. We implemented this algorithm for pre-computing k minimum hop, widest-shortest and shortest-widest optimal routes.

The algorithm considers a network with a set V of nodes and a set E of links. Some other useful notations referred in the algorithm are

n - $|V|$, e - $|E|$, $[i,\ j]$ - link from node i to node j,

$A\,(i)$ - nodes j such that $[i,\ j] \in E$,

$F\,(h)$ - network resulting from deleting all $[s,\ j]$ with $j \in A\,(s) \setminus \{h\}$

$L\,(P)$ - second node on path P

$z\,(i,\ j)$ - length of link $[i,\ j]$

$C\,(h,\ X,\ i)$ - minimum path length over all loopless paths from s to i with $L\,(P)$=h and using only nodes $X \bigcup \{s,\ i\}$

$T\,(i)$ -For node i it a routing table or a list of distinct second nodes from s

$V\,(h)$ - nodes i with $h \in T\,(i)$

$D\,(h, i)$ - Equal to $C\,(h, V\,(h)\,, i)$

$d\,(i)$ - minimum of $D\,(h, i)$ over h available for $T\,(i)$

$H\,(i)$ - minimizer of $D\,(h, i)$ over candidates h available for $T\,(i)$

$b\,(h,\ i)$ - pointer used to construct paths

The source node is specified as $s \in V$ and positive integer k with $k \leq |A\,(s)|$. For $i \in V \setminus \{s\}$, the k shortest path problem is to find an optimal set of paths for each $i \in V \setminus \{s\}$ with distinct initial links from the source node. For $i \in V \setminus \{s\}$, a routing table $T\,(i)$ is a list of distinct elements of $A(s)$. The following steps describe the algorithm:

Step 1:Initialization. Set $D(h,i) \leftarrow \infty$ for all $h \epsilon A(s)$ and $i \epsilon N \setminus \{s, h\}$; $d(i) \leftarrow \infty$ for each $i \epsilon N \setminus (A(s) \cup \{s\})$; $D(i,i) \leftarrow z(s,i)$, $d(i) \leftarrow D(i,i)$, $H(i) \leftarrow i$, and $b(i,i) \leftarrow s$ for each $i \epsilon A(s)$; and $T(i) \leftarrow \emptyset$ for each $i \epsilon N \setminus \{s\}$.

Step 2: While some candidate exists,

a) pick I to minimize $d(i)$ over all i with $|T(i)| < k$, and add $H(I)$ to $T(I)$;

b) for each $j \epsilon A(I)$ with $H(I)$ available for $T(j)$ and $D(H(I), I) + z(I,j) < D(H(I), j)$,

b.1) set $D(H(I), j) \leftarrow D(H(I), I) + z(I,j)$ and $b(H(I), j) \leftarrow I$;

b.2) if $D(H(I), j) < d(j)$, then set $d(j) \leftarrow D(H(I), j)$ and $H(j) \leftarrow H(I)$;

c) if $|T(I)| < k$, then set $d(I)$ to be the minimum value of $D(h, I)$ over all h available for $T(I)$; if also $d(i) < \infty$, then set $H(I)$ to minimize $D(h, I)$ over all candidates h for $T(I)$.

The run time of this algorithm is $O(ke(\log_2 n))$, when binary heaps are used for implementation.

4.4.2 Simulation Model

To evaluate the performance of our algorithm we developed an event-driven simulator that models QoS routing at connection level. Our scheme is based on dynamically adaptive routing. The dynamically adaptive routing uses an adaptive routing algorithm that automatically routes traffic around congested, damaged or destroyed nodes and links and allows the system to continue to function over the remaining portions of the network.

We considered two network topologies in our simulation. The first topology is the Internet Service Provider - ISP topology with 18 nodes and 70 links (Figure 4.1). This topology has been considered in several QoS routing studies [1][5][37][76]. We also considered a Switched cluster topology with 16 nodes and 56 links (Figure 4.2) [76].

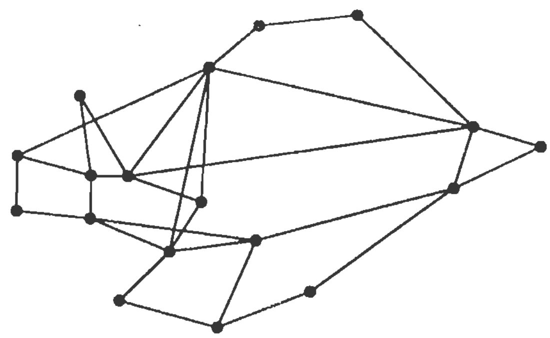

Figure 4.1: ISP network topology.

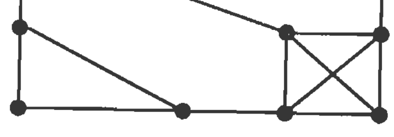

Figure 4.2: Switched cluster network topology.

Several simulation parameters are assumed during the simulation run. Most of the parameters assumed here are obtained from the studies made by [37] and [76]. However different studies assume different parameters but in general the parameters that we have considered are on the lines of most of the QoS routing studies. We have assumed periodic link state updates.

The link residual bandwidths that is the bandwidths offered to the traffic are assumed to be in the range [1.5, 6 $Mbps$]. These rates are the typical requirements of packet video and multimedia database access [86]. Other simulation parameters assumed during the simulation are given below:

Average Token rate, $\sigma = 1\, Mbps$,

Token Bucket Size, $b = 4\, kB$ (This is approximately equal to 80 cells),

Maximal Packet Size, $L_{max} = 53\, B$ (ATM cell size),

Jitter Bound, $J = 5\, ms$, Delay Bound, $D = 80\, ms$

Link Capacity, $C_i = 80\, Mbps$, for all links i

Link Residual Bandwidth between [1.5, 6 $Mbps$]

Connection Arrival Rate (Network Load) = [1.2, 1.5, 1.8, 2.0, 2.2] connections/sec. (The connection arrival rate is assumed Poisson distributed i.e. the interarrival times are exponentially distributed.).

Connection Setup Time = 15 ms

Connection Holding Time = 3 min (Exponentially Distributed)

Connection Tear Down Cost = 1 ms/hop

Connection Re-routing Overhead = 1 ms/hop.

The simulations were run till a total of 50000 connection requests arrived in the network.

We consider two different approaches to determine reservable bandwidths. In the first approach, the link reservable bandwidth is uniformly distributed in the interval [1.5, 6 $Mbps$].

The incoming connection request is allocated a bandwidth from this interval. Here we run the routing algorithm as per the parameters mentioned above for obtaining the results. We compare the performance of our algorithm with that of pre-computation of k-constrained QoS routes using the MBF algorithm and with that of on-demand routing to compute minimum hop route to the destination for the two topologies.

In the second approach, as an optimization we do not consider all possible residual bandwidths but consider a small set of residual bandwidth like mentioned in [76]. This approximation also puts a limit on the amount of bandwidth that a flow can reserve. The set of bandwidth values is $S = \{v_1, v_2, \cdots, v_n\}$ such that $v_1 > v_2 > \cdots > v_n$. The value v_1 is the maximum bandwidth that can be reserved. The algorithms are iterated over these different values in decreasing order of v_i. We selected 12 different values for the set S of bandwidth values which are $\{6, 5, 4, 3.5, 3, 2.5, 2, 1.9, 1.8, 1.7, 1.6, 1.5\}$. All these bandwidths are in $Mbps$. The pre-computation interval was varied from $5s$ to $50s$. For all the values for the connection blocking rate, the 90% confidence intervals were computed and the mean are shown in the results.

4.5 Implementation

4.5.1 Computation of Paths

The path pre-computation reduces the processing cost by performing a costly single source multiple-destination path computation periodically and using the resulting paths to route requests. According to Apostolopoulos [3] for the same topology, the cost of path pre-computation depends solely on the pre-computation period and it can become arbitrary small by increasing this period. Path pre-computation schemes benefit from having multiple candidate routes to each destination, to balance the network load and have additional routing choices in case of a setup failure. However if the multiple routes have dissimilar costs then the selected paths may make inefficient use of network resources. When a new request at that source comes for some destination then best feasible path to that destination is allocated from the set of pre-computed paths and the resources are reserved accordingly for the traffic flow to that destination. In case of setup failure the next best available route is selected. In case of renegotiated QoS parameters, the current path is checked first for resource availability. If the newly required resources are not available on the current path then the next alternate path is selected and verified for resource availability and resources are reserved.

We also determined on-demand QoS routes. In on-demand routing the QoS routes between source and destination pair are determined only when the request arrives. Since the on-demand routing makes use of the latest available link-state information for route computation and connection setup done is at the same time hence the connection blocking rate should be low. However its overhead should be high as the routes are computed on every request. We have determined number of failed or blocked connections as total of routing failures and setup failures.

We have considered minimum hop, widest-shortest and shortest-widest path heuristics as the path selection algorithms. Minimum hop path is a feasible path with minimum hop count. The bandwidth is randomly allocated. Widest-shortest path is a feasible path with minimum hop count. If there is more than one path with the minimum hop count, the one with the maximum reservable bandwidth is selected. Shortest-widest path is a feasible path with the maximum reservable bandwidth. If there are several such paths, the one with the minimum hop count is selected.

The scheduling policy is rate based and therefore we consider the hop count bounds as determined by the jitter and buffer constraints [19][123]. A path is feasible for traffic with delay guarantees, if it meets the delay, delay-jitter and buffer space requirements as given by the set of equations from Equation 4.1 to 4.3. Therefore to satisfy the delay-jitter constraint, the maximum number of hops is given as $h_{max} = \min\left(|V|-1,\ \lfloor (J.r-b)/L_{max} \rfloor\right)$ where J is the end-to-end delay-jitter bound. To satisfy the buffer space constraints, a link i has to ensure its hop count is less than or equal to $(B_h - b)/L_{max}$ where B_h is the buffer space requirement at the h-th hop.

Following steps specify our implementation.

1. The link lengths $z(i,\ j)$ are set to one for all $(i,\ j) \in E$, for i and j.

2. The list $T[s][i]$ contains second nodes $h \in A(s)$ for all destinations $i \in V \setminus \{s\}$.

3. The array $b[s][h][i]$ contains preceding node of i.

4. The heap $DP[s][i]$ contains hop counts of paths to destination i.

5. We pre-computed k constrained paths to every node $i \in V \setminus \{s\}$ for all nodes s as per above constraints and above algorithm.

6. The limit on maximum number of pre-computed constrainted paths i.e. $k \leq |A(s)|$.

7. The k-constrained paths for every source node are stored in a corresponding heap. The optimal paths for a particular source-destination pair are then extracted when a connection request arrives.

The algorithm so modified is called Proposed Modified Topkis (PMT) algorithm.

4.5.2 Delayed Extraction and recomputation of pre-computed Routes

The input traffic pattern is not generally available in advance so allocated bandwidth is from a set. Under delayed extraction of routes, we compute and store the k-constrained paths for each $h \epsilon A(s)$. During path extraction we apply the most recent link-state information in determining the resource availability on the selected route. If the resources are not available then connection setup fails and alternate path is tried. Thus the accurate information is used to select the route.

The frequency of recomputation of routes is a design consideration. When a request arrives the source extracts a route and initiates signalling for path setup. The source blocks the request if the extraction process does not produce a route. In our scheme the source triggers recomputation of the k-constrained paths after a failure in route extraction or connection establishment.

4.6 Discussion of Results and Conclusions

Case I: Minimum hop objective

In the case when the link bandwidth is uniformly distributed between [1.5, $6Mbps$]. The paths are determined by Minimum hop objective and constraints as per rate based scheduling. The bandwidth allocated to a connection request is selected at random from this interval. Table 4.1 and Table 4.2 show values for connection blocking rate for minimum hop objective for the two algorithms for the two topologies.

Connection blocking rates for the two topologies were plotted and are shown in Figure 4.3 to Figure 4.5. It can be seen from these figures that the connection blocking rate increases for MBF algorithm as pre-computation interval is varied. This is because the route information that is pre-computed and stored becomes stale. Therefore when connection requests arrive this

Arrival Rate (con./s)	1.5		1.8		2.0	
Pre-comp. Inter. (sec)	MBF	PMT	MBF	PMT	MBF	PMT
5	0.145	0.132	0.185	0.167	0.235	0.217
10	0.155	0.141	0.195	0.179	0.248	0.228
15	0.165	0.153	0.205	0.190	0.260	0.239
20	0.175	0.164	0.220	0.203	0.270	0.248
25	0.183	0.171	0.230	0.214	0.279	0.257
30	0.189	0.179	0.237	0.221	0.284	0.265
35	0.195	0.186	0.245	0.227	0.293	0.272
40	0.200	0.190	0.250	0.234	0.297	0.277
45	0.205	0.195	0.254	0.237	0.303	0.285
50	0.210	0.200	0.259	0.243	0.309	0.290

Table 4.1: Values for connection blocking rate for minimum hop objective for MBF and Proposed Modified Topkis (PMT) algorithms for different arrival rates for ISP topology.

Arrival Rate (conn./sec)	1.5		1.8		2.0	
Pre-comp. Inter. (sec)	MBF	PMT	MBF	PMT	MBF	PMT
5	0.195	0.185	0.257	0.239	0.360	0.336
10	0.198	0.188	0.262	0.243	0.365	0.344
15	0.201	0.191	0.267	0.249	0.370	0.350
20	0.205	0.194	0.272	0.253	0.376	0.355
25	0.207	0.198	0.275	0.259	0.381	0.361
30	0.211	0.201	0.279	0.263	0.386	0.366
35	0.214	0.205	0.282	0.266	0.390	0.370
40	0.217	0.208	0.285	0.270	0.394	0.373
45	0.220	0.211	0.288	0.273	0.397	0.377
50	0.223	0.215	0.295	0.277	0.400	0.381

Table 4.2: Values for connection blocking rate for minimum hop objective for MBF and Proposed Modified Topkis (PMT) algorithms for different arrival rates for Switched cluster topology.

results into setup failures. Same effect is observed for Modified Topkis algorithm but here the connection blocking rate is lower. The routing is faster in case of Modified Topkis.

However if we look into the performance of on-demand routing, the connection blocking is much less and is independent of pre-computation. Total simulation time required for on-demand routing is less for lower pre-computation intervals then that of Modified Topkis and MBF algorithms as seen in Figure 4.9 to Figure 4.11. But as pre-computation interval increases the simulation time becomes smaller for Modified Topkis and MBF algorithms. This is due to less overhead because of pre-computations. The on-demand results that are shown in all the graphs are for MBF algorithm.

Similar results were observed Switched cluster topology (Figure 4.6 to Figure 4.8). However here the connection blocking rate was higher than the corresponding blocking rates for ISP topology. Simulation times are less than that of ISP topology (Figure 4.12 to Figure 4.14). These effects are due the network topology.

The results from the simulation study shows that the proposed Modified Topkis algorithm performs better than MBF as it routes more connections in the simulated run time and also the connection blocking probability of the proposed algorithm is smaller as against MBF algorithm when pre-computation period is varied.

Case II: Widest-shortest and shortest-path optimal paths with link residual bandwidths limited to a fixed set of values

As per the optimization discussed above the link residual bandwidths were taken from a small set of bandwidths. The simulations for this case were performed for the same set of assumptions and confidence but the link reservable bandwidth is now limited to a set of discrete values. For this case different optimality condition was chosen, as the minimum hop

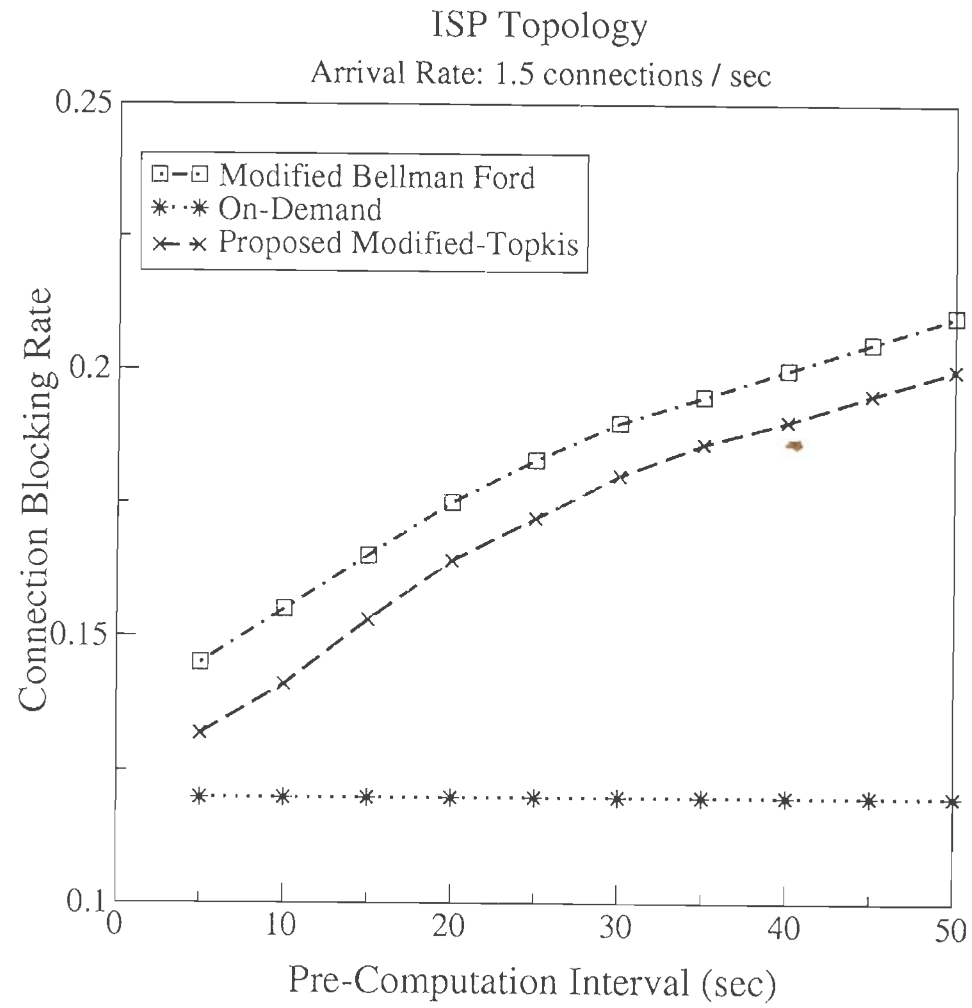

Figure 4.3: Connection blocking rate for different pre-computation intervals for 1.5 connection arrivals per sec and Minimum hop objective for ISP topology.

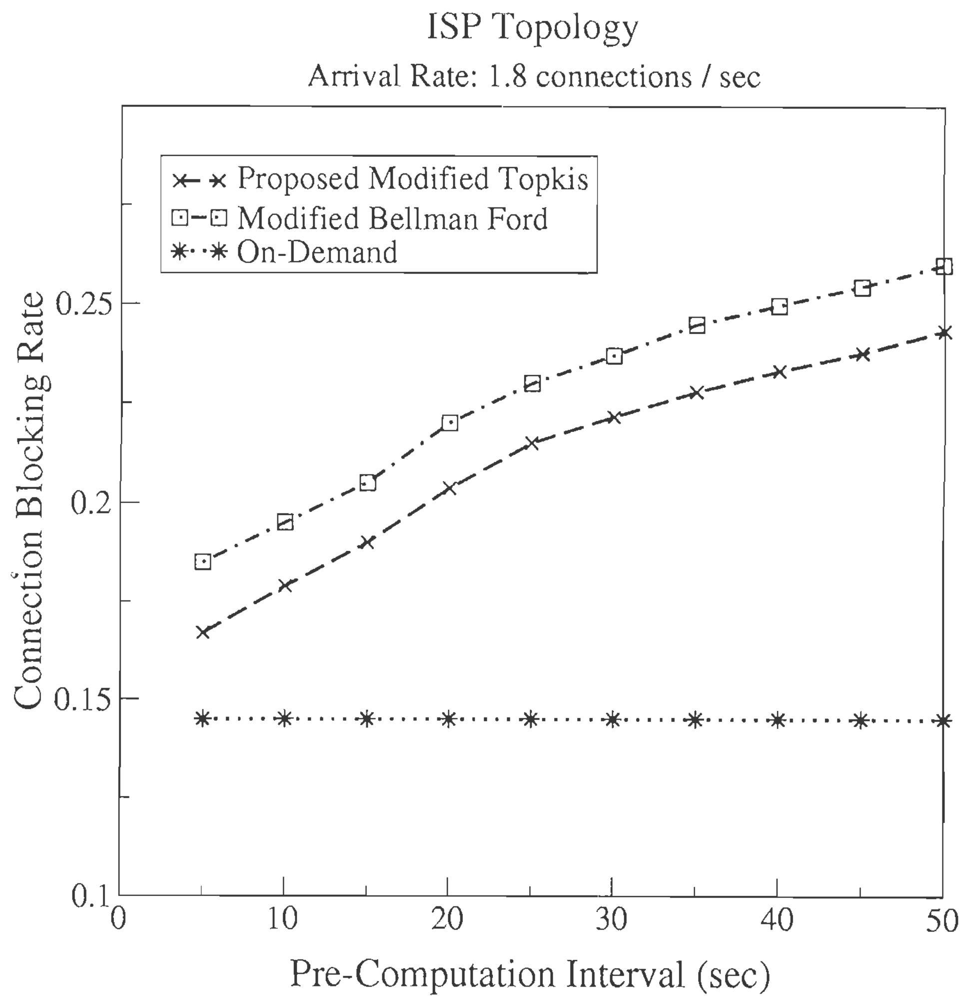

Figure 4.4: Connection blocking rate for different pre-computation intervals for 1.8 connection arrivals per sec and Minimum hop objective for ISP topology.

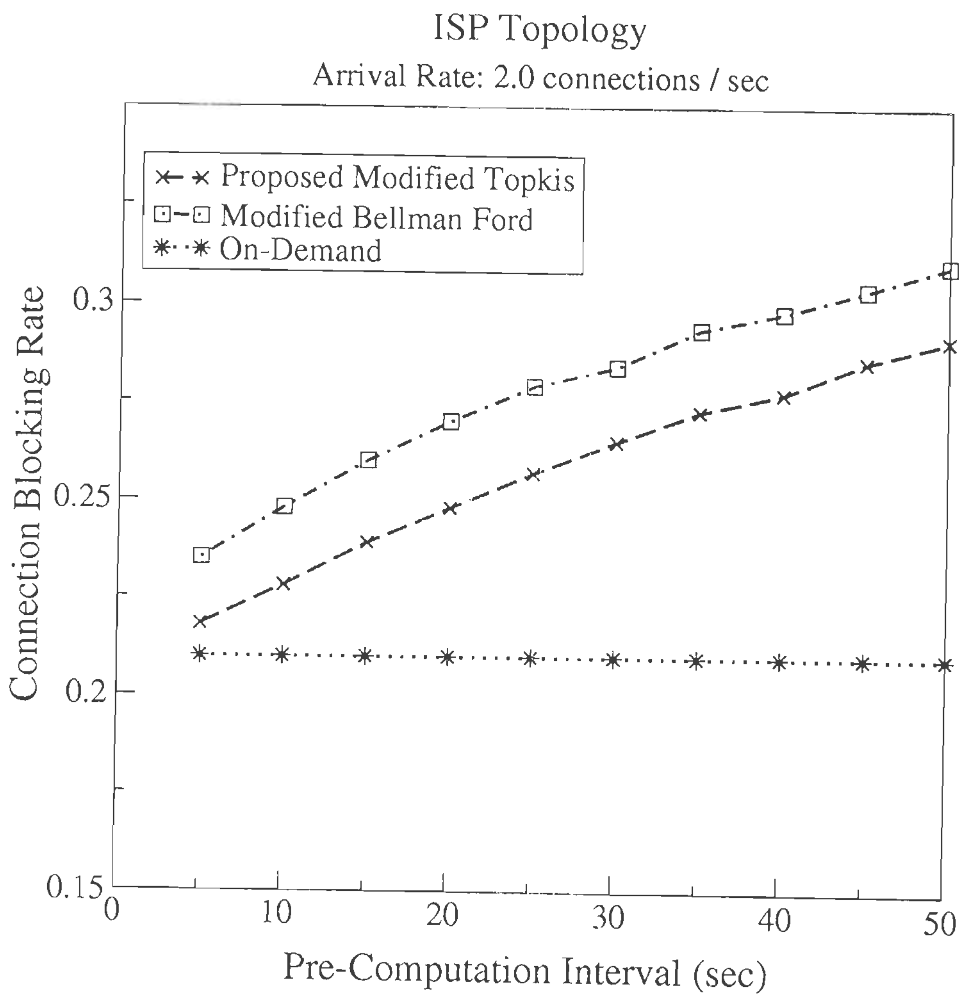

Figure 4.5: Connection blocking rate for different pre-computation intervals for 2.0 connection arrivals per sec and Minimum hop objective for ISP topology.

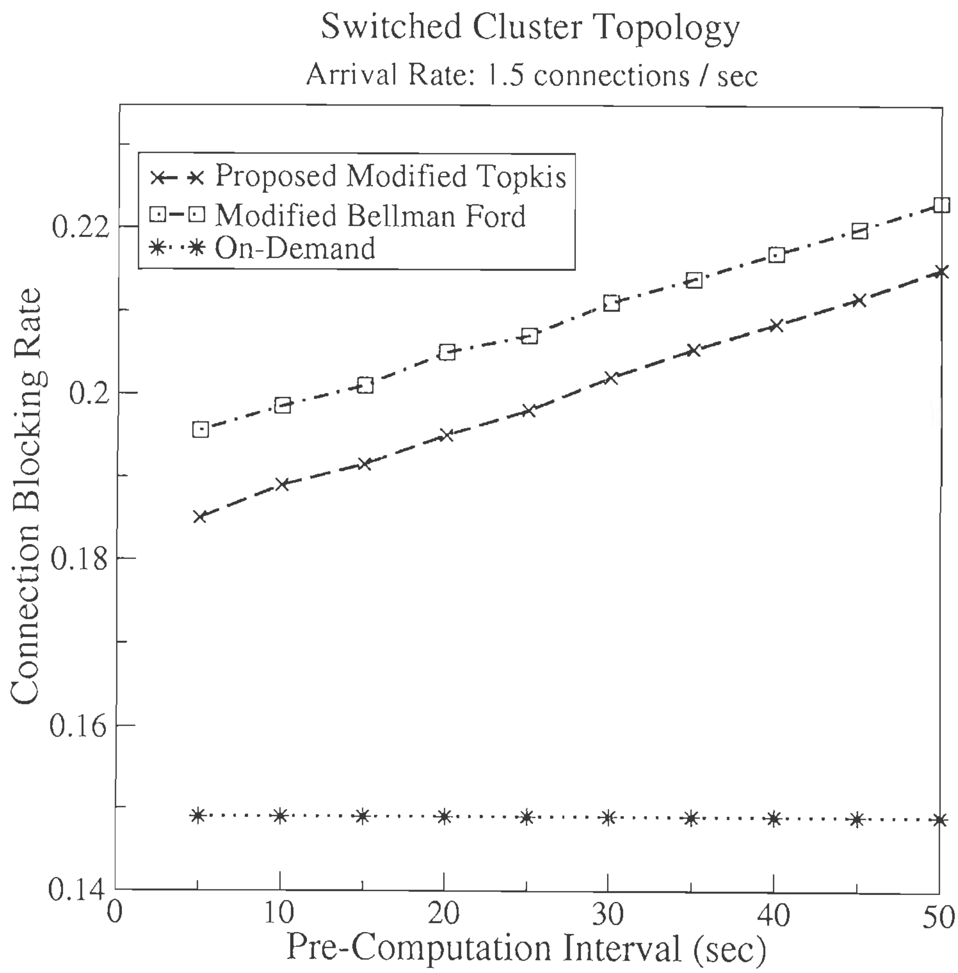

Figure 4.6: Connection blocking rate for different pre-computation intervals for 1.5 connection arrivals per sec and Minimum hop objective for Switched cluster topology.

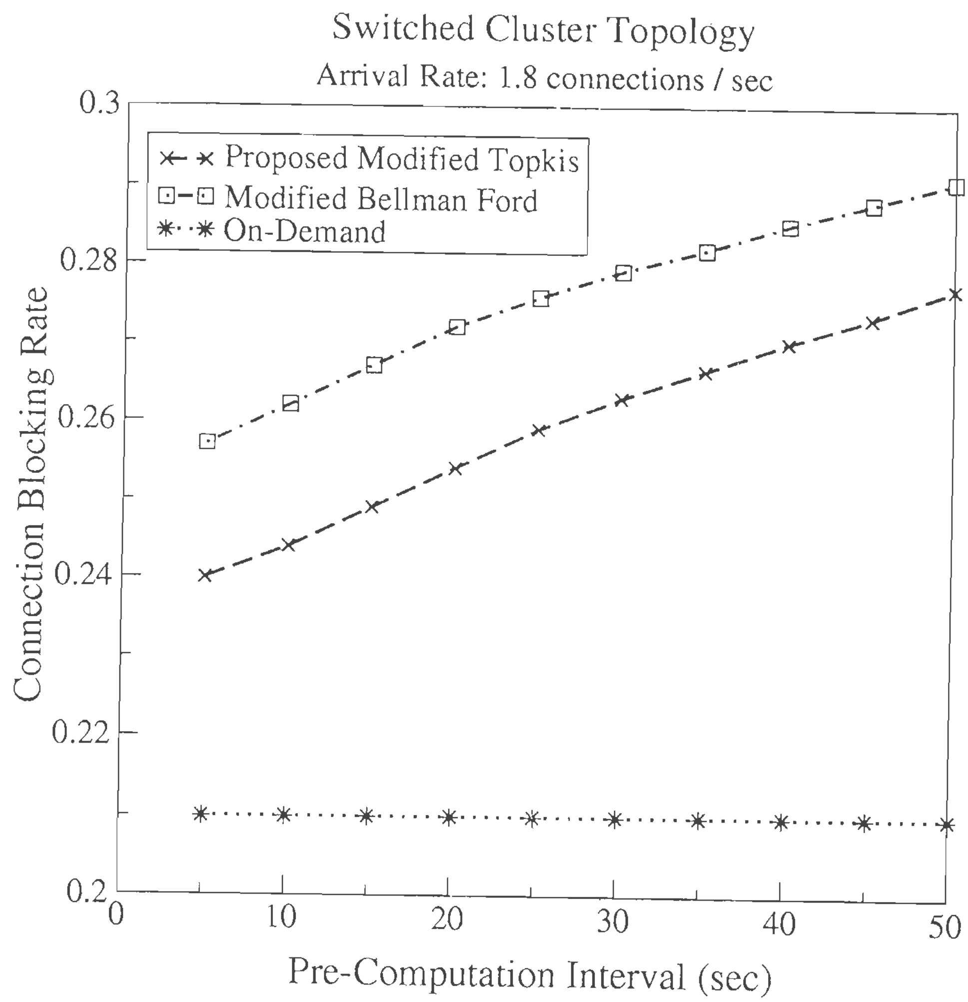

Figure 4.7: Connection blocking rate for different pre-computation intervals for 1.8 connection arrivals per sec and Minimum hop objective for Switched cluster topology.

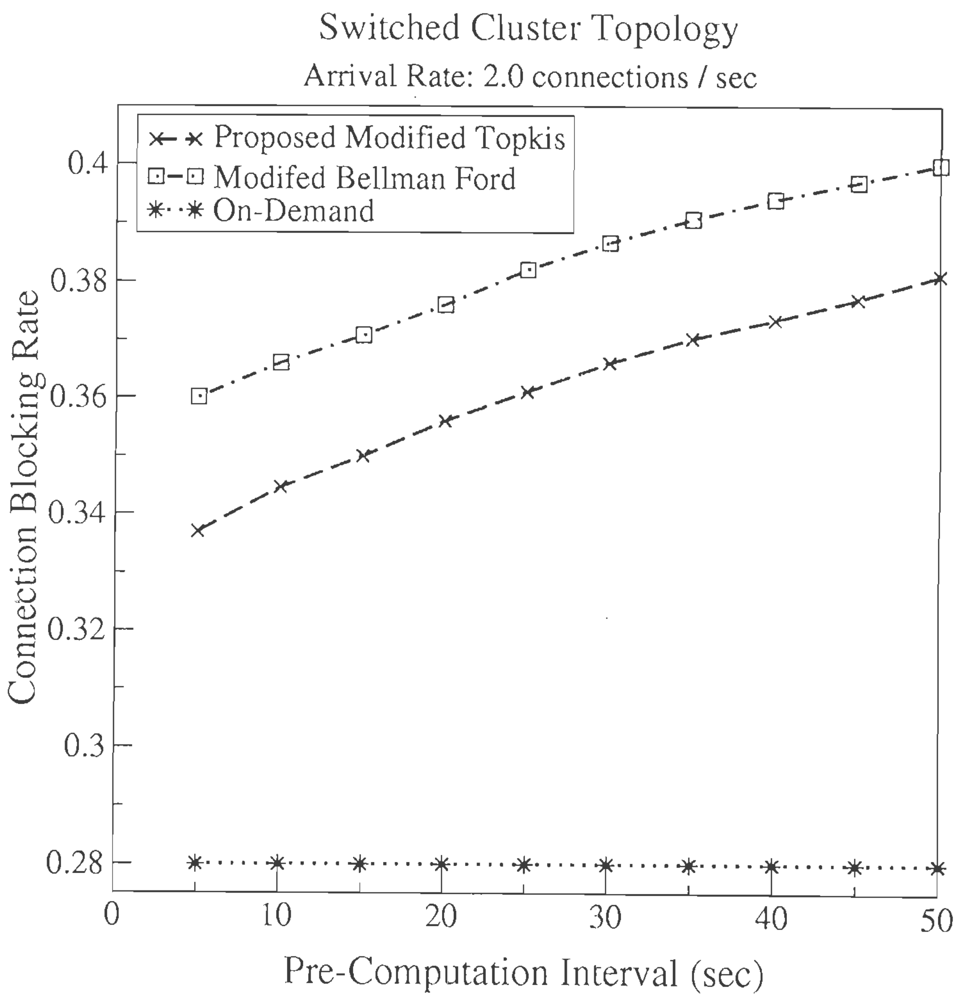

Figure 4.8: Connection blocking rate for different pre-computation intervals for 2.0 connection arrivals per sec and Minimum hop objective for Switched cluster topology.

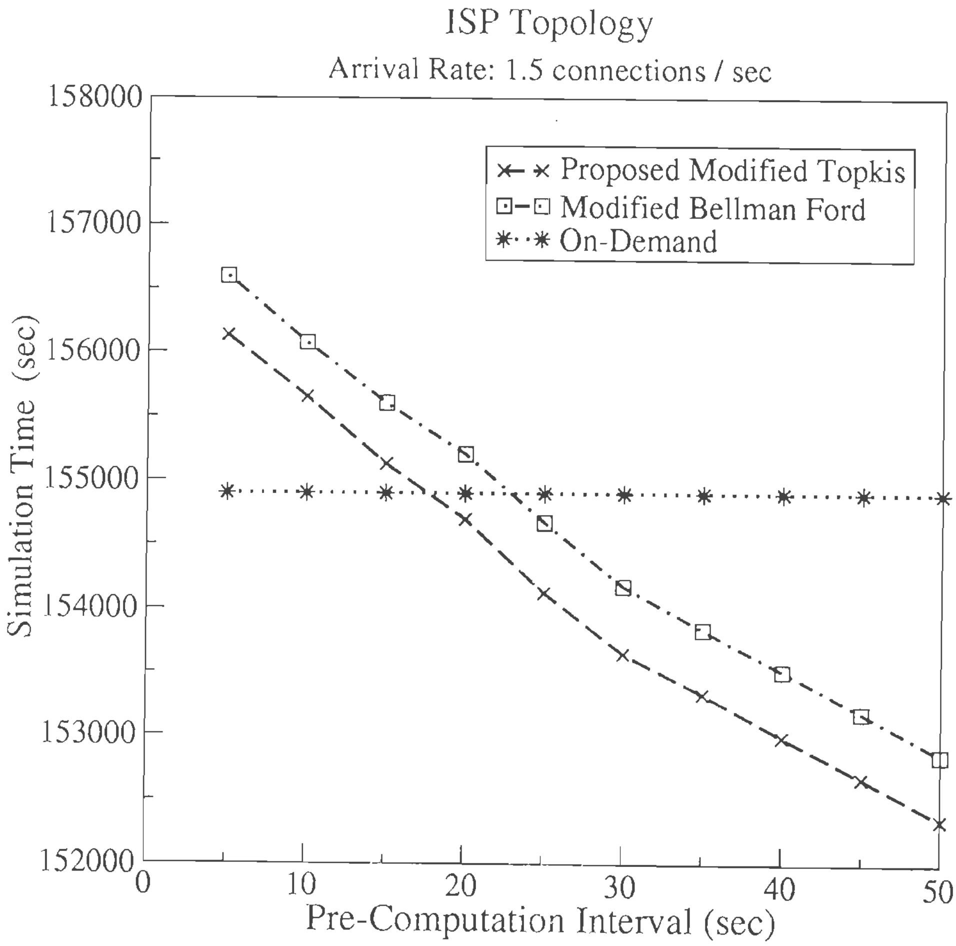

Figure 4.9: Total simulation time till 50000 connection requests arrive for routing for an arrival rate of 1.5 connections per sec and Minimum hop objective for ISP topology.

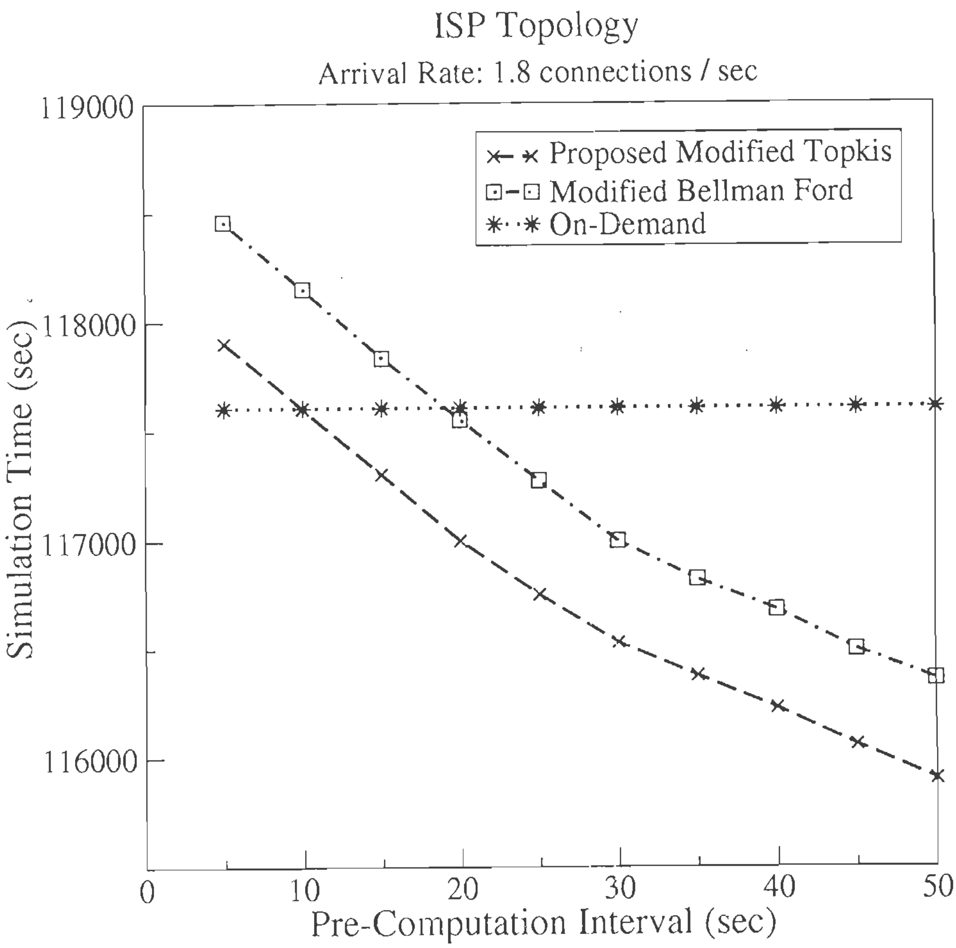

Figure 4.10: Total simulation time till 50000 connection requests arrive for routing for an arrival rate of 1.8 connections per sec and Minimum hop objective for ISP topology.

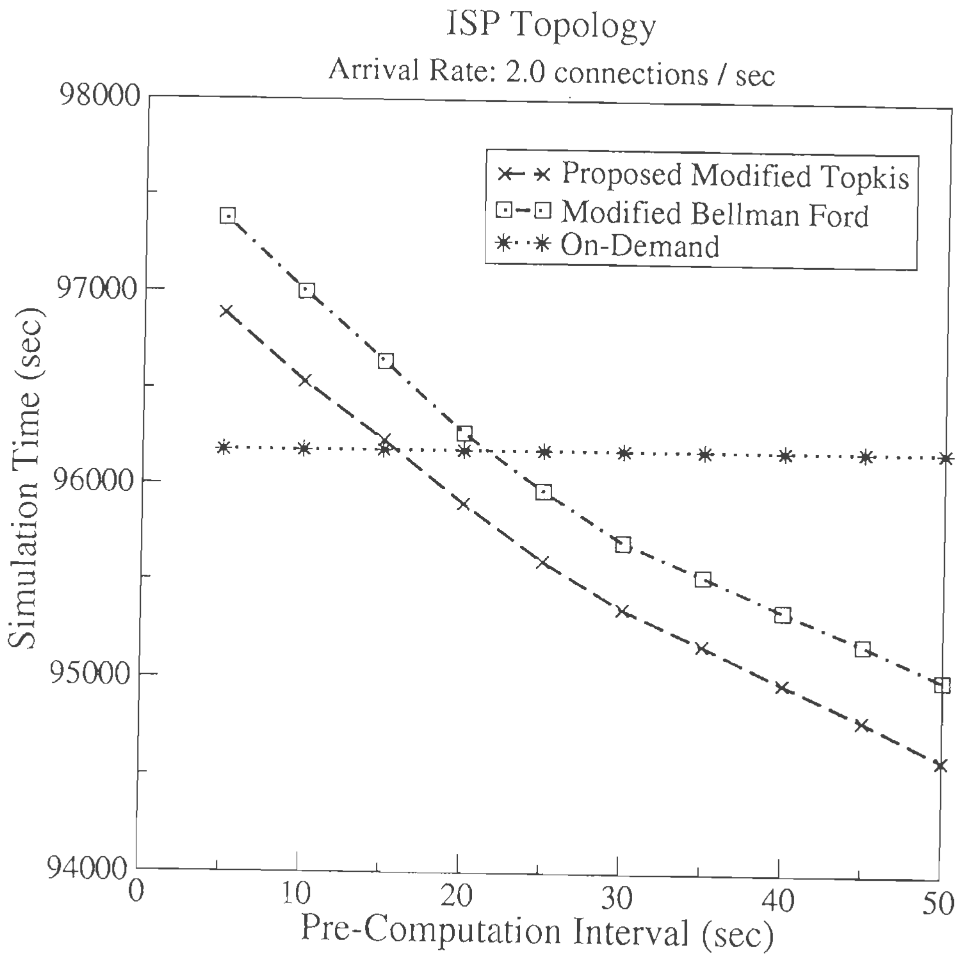

Figure 4.11: Total simulation time till 50000 connection requests arrive for routing for an arrival rate of 2.0 connections per sec and Minimum hop objective for ISP topology.

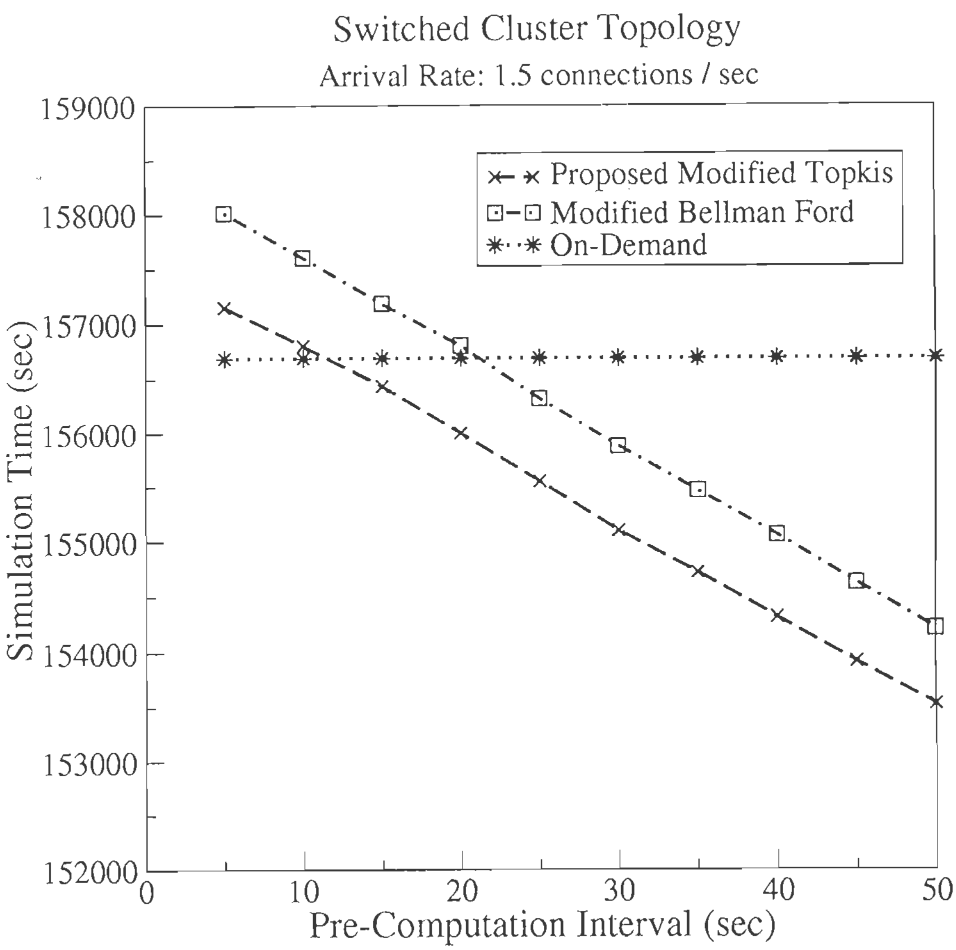

Figure 4.12: Total simulation time till 50000 connection requests arrive for routing for an arrival rate of 1.5 connections per sec and Minimum hop objective for Switched cluster topology.

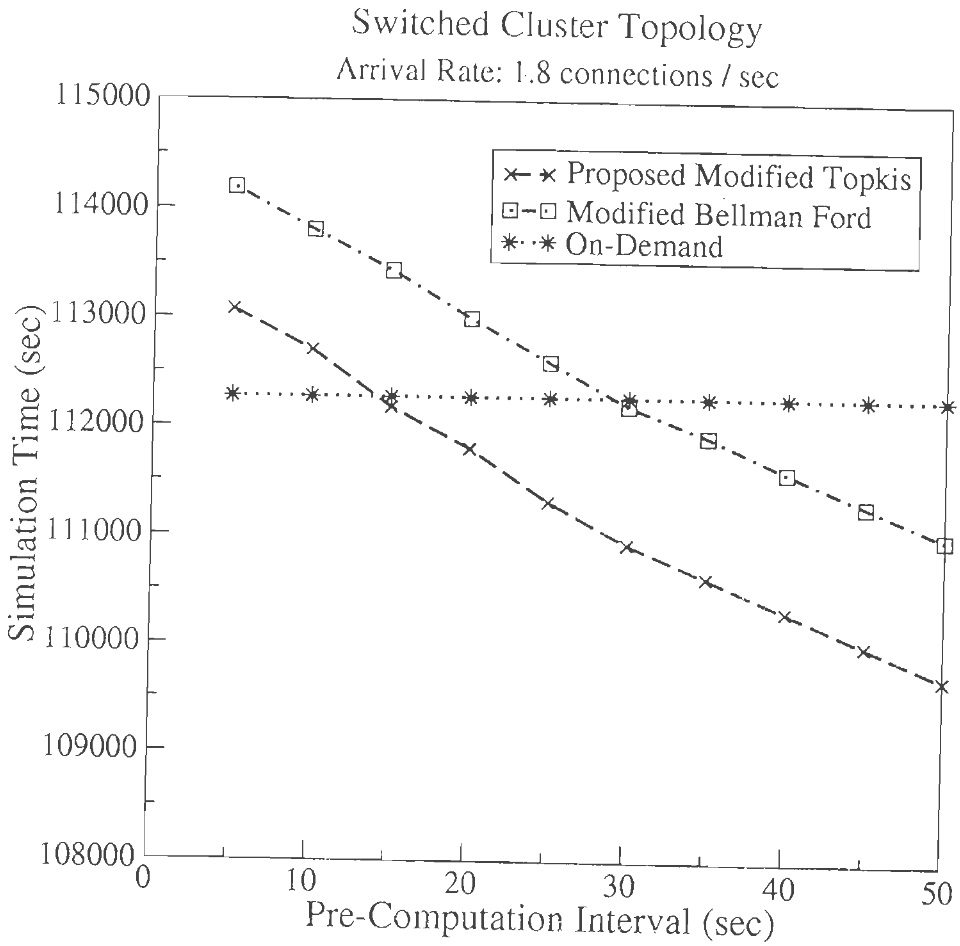

Figure 4.13: Total simulation time till 50000 connection requests arrive for routing for an arrival rate of 1.8 connections per sec and Minimum hop objective for Switched cluster topology.

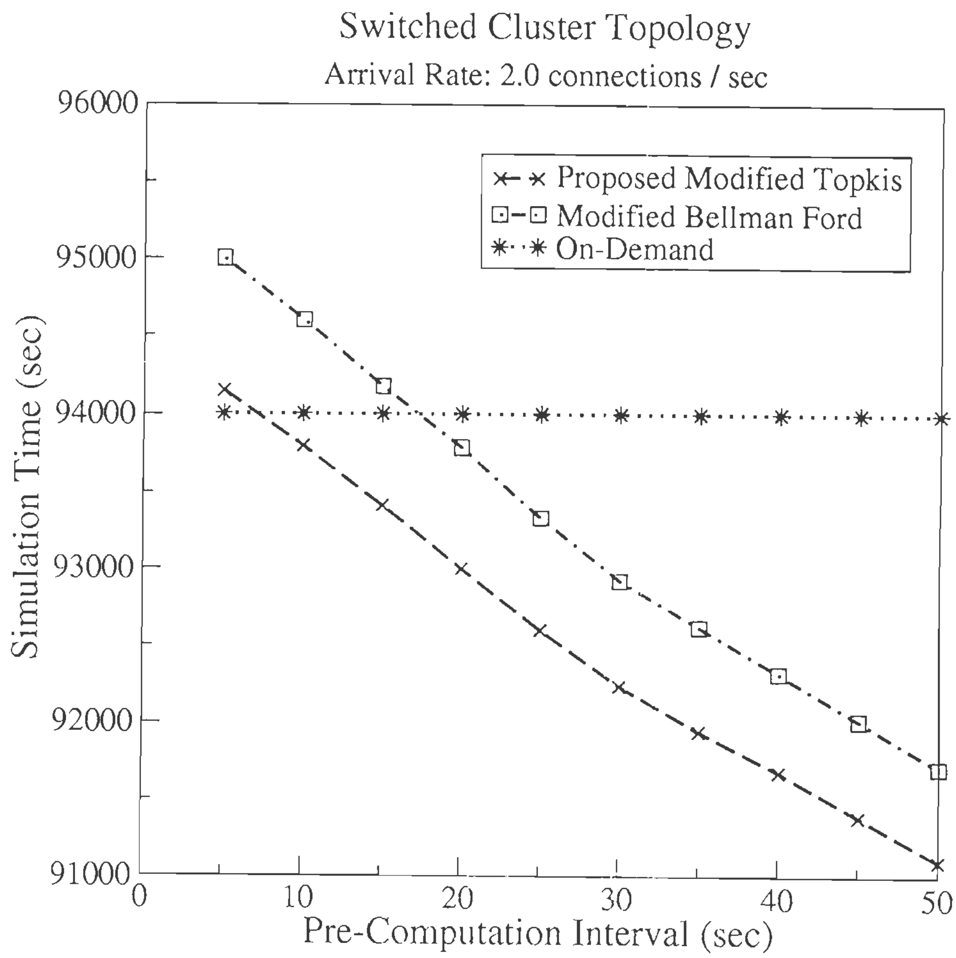

Figure 4.14: Total simulation time till 50000 connection requests arrive for routing for an arrival rate of 2.0 connections per sec and Minimum hop objective for Switched cluster topology.

Arrival Rate (connections/sec)	ISP Topology			
	MBF		PMT	
	Widest-shortest	Shortest-widest	Widest-shortest	Shortest-widest
1.2	0.0100	0.0134	0.0045	0.0049
1.5	0.0224	0.0250	0.0124	0.0135
1.8	0.0381	0.0372	0.0230	0.0250
2.0	0.0534	0.0575	0.0314	0.0350
2.2	0.0692	0.0760	0.0422	0.0480

Table 4.3: Values for connection blocking rate for pre-computation interval = 20 sec for widest-shortest and shortest-widest paths for MBF and Proposed Modified Topkis (PMT) algorithms for ISP topology.

objective can be improved to give satisfactory performance. Therefore the widest-shortest and shortest-widest optimality criteria were used here as described in subsection 4.5.1. Table 4.3 and Table 4.4 gives the connection blocking rate values for widest-shortest and shortest-widest path objectives for the ISP and Switched cluster topologies respectively.

The two optimality criteria so considered, combine two different constraints - link reservable bandwidth and the minumum hop. Hence the performance is much improved but the nature of the results are same i.e. the Modified Topkis algorithm performs much better than the MBF algorithm. The connection arrival rates were varied from 1.2 connections/s to 2.2 connections/s. As evident from Figure 4.15 the shortest-widest paths do not perform well for both the topologies specially at higher arrival rates. The performance is better for the widest-shortest path compared with shortest-widest paths. All the remaining figures in this chapter are drawn for the widest-shortest optimal routes.

Table 4.5 and Table 4.6 show the values for connection blocking rate for widest-shortest path objective for MBF and Proposed Modified Topkis (PMT) algorithms for ISP and Switched cluster topology respectively. The connection blocking rate for the widest-shortest optimal

Arrival Rate (connections/sec)	Switched Cluster Topology			
	MBF		PMT	
	Widest-shortest	Shortest-widest	Widest-shortest	Shortest-widest
1.2	0.0401	0.0410	0.0325	0.0325
1.5	0.0584	0.0615	0.0524	0.0535
1.8	0.0873	0.0925	0.0821	0.0845
2.0	0.1224	0.1232	0.1180	0.119
2.2	0.1534	0.1602	0.1440	0.154

Table 4.4: Values for connection blocking rate for pre-computation interval = 20 sec for widest-shortest and shortest-widest paths for MBF and Proposed Modified Topkis (PMT) algorithms for Switched cluster topology.

Arrival Rate (connections/sec)	1.5		1.8		2.0	
Pre-computation Interval (sec)	MBF	PMT	MBF	PMT	MBF	PMT
5	0.0200	0.0110	0.0350	0.0199	0.0511	0.0299
10	0.0209	0.0113	0.0360	0.0210	0.0520	0.0304
15	0.0215	0.0119	0.0369	0.0219	0.0529	0.0309
20	0.0224	0.0124	0.0381	0.0230	0.0534	0.0314
25	0.0230	0.0129	0.0389	0.0234	0.0541	0.0320
30	0.0234	0.0135	0.0395	0.0239	0.0546	0.0324
35	0.0240	0.0138	0.0399	0.0243	0.0550	0.0328
40	0.0242	0.0142	0.0403	0.0248	0.0553	0.0329
45	0.0246	0.0147	0.0410	0.0252	0.0555	0.0334
50	0.0251	0.0152	0.0414	0.0258	0.0560	0.0340

Table 4.5: Values for connection blocking rate for widest-shortest path objective for MBF and Proposed Modified Topkis (PMT) algorithms for different arrival rates for ISP topology.

Arrival Rate (connections/sec)	1.5		1.8		2.0	
Pre-computation Interval (sec)	MBF	PMT	MBF	PMT	MBF	PMT
5	0.0560	0.0499	0.0848	0.0802	0.1200	0.1159
10	0.0566	0.0507	0.0855	0.0809	0.1208	0.1166
15	0.0576	0.0514	0.0866	0.0817	0.1214	0.1173
20	0.0584	0.0524	0.0873	0.0821	0.1224	0.1180
25	0.0589	0.0533	0.0883	0.0829	0.1237	0.1189
30	0.0594	0.0537	0.0890	0.0837	0.1242	0.1198
35	0.0599	0.0542	0.0898	0.0846	0.1247	0.1203
40	0.0603	0.0545	0.0905	0.0852	0.1253	0.1209
45	0.0607	0.0549	0.0912	0.0859	0.1257	0.1214
50	0.0613	0.0554	0.0919	0.0867	0.1263	0.1219

Table 4.6: Values for connection blocking rate for widest-shortest path objective for MBF and Proposed Modified Topkis (PMT) algorithms for different arrival rates for Switched cluster topology.

path for the ISP and Switched cluster topologies for different arrival rates were plotted. The corresponding graphs are shown in Figure 4.16 to Figure 4.18 and Figure 4.19 to Figure 4.21 respectively. In this where there is a limit on the number of link residual bandwidths, the connection blocking rates for both the topologies are much lower than the minimum hop paths. The connection blocking rate for the Proposed Modified Topkis algorithm is again lower than the MBF algorithm.

The simulation time spent by the algorithms till 50000 connection requests arrived at the network are shown in Figure 4.22 to Figure 4.24 for the ISP topology and in Figure 4.25 to Figure 4.27 for the Switched cluster topology.

It can be deduced from these results that the difference in routing performance of on-demand and pre-computation is less for smaller arrival rates. The periodic pre-computation of routes take less simulation run times compared with the on-demand route computations.

Case III: Effect of link state updates on widest-shortest pre-computed paths

The connection blocking rate increases as a function of the link state update period. For this case the pre-computation period was fixed and link state update period was varied. It can be seen from Figure 4.28 and Figure 4.29 that as the link state update periods are increased it leads to larger connection blocking rates. When the link state information in the system is outdated, it is also inaccurate. This causes the source nodes to mistakenly consider infeasible links as feasible. Hence the source node selects infeasible path. There may be other feasible path available at that time but due to optimality conditions they were not selected. These are mostly the setup failures and result into poor use of network resources.

The increase for the large update periods compared with smaller initial update periods is less. When an link update message indicates a low utilization or availability of resources, the rest of the network reacts by routing more traffic to the link. Blocking remains low during this interval of available resources. But once the link is saturated and this information is distributed then the no more connections are admitted hence blocking rate does not increase by much.

It can be concluded from the above results that pre-computation of routes provide faster route computation but their blocking rate is higher. The Proposed Modified Topkis algorithm gives a much improved performance over MBF algorithm which is used traditionally. The performance gap for these algorithms and optimality criteria is large and varies with the network load and topology. To achieve low blocking rates, optimization criteria are necessary. Studies can be made to determine effects of link update periods on message overheads and processing cost.

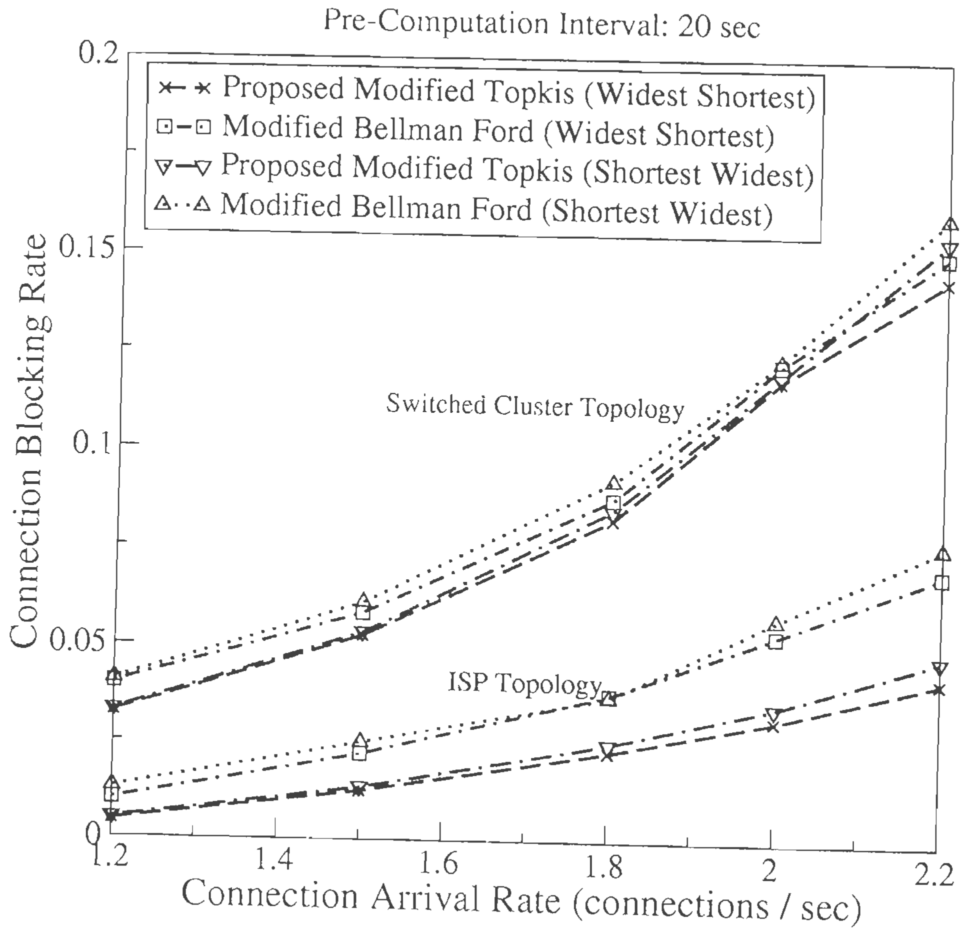

Figure 4.15: Connection blocking rates for different pre-computation intervals for ISP and Switched cluster topology.

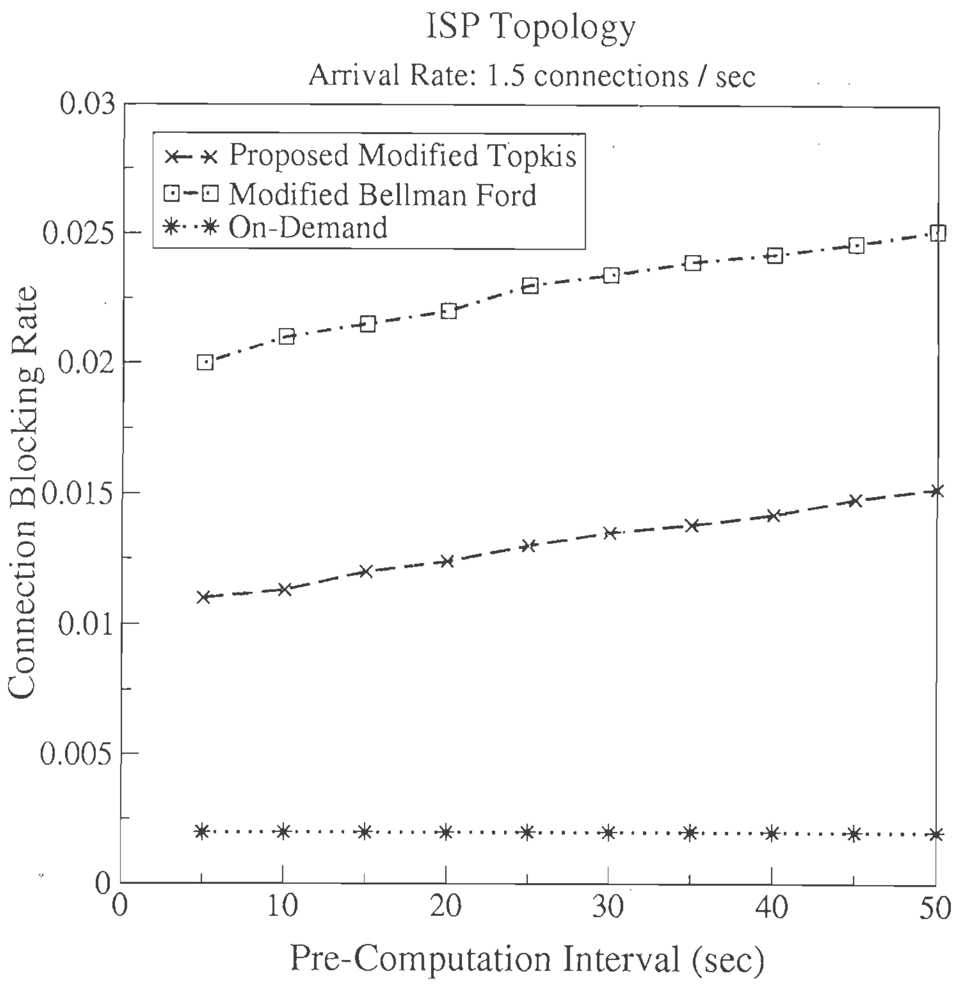

Figure 4.16: Connection blocking rate for different pre-computation intervals for 1.5 connection arrivals per sec and Widest-shortest path objective for ISP topology.

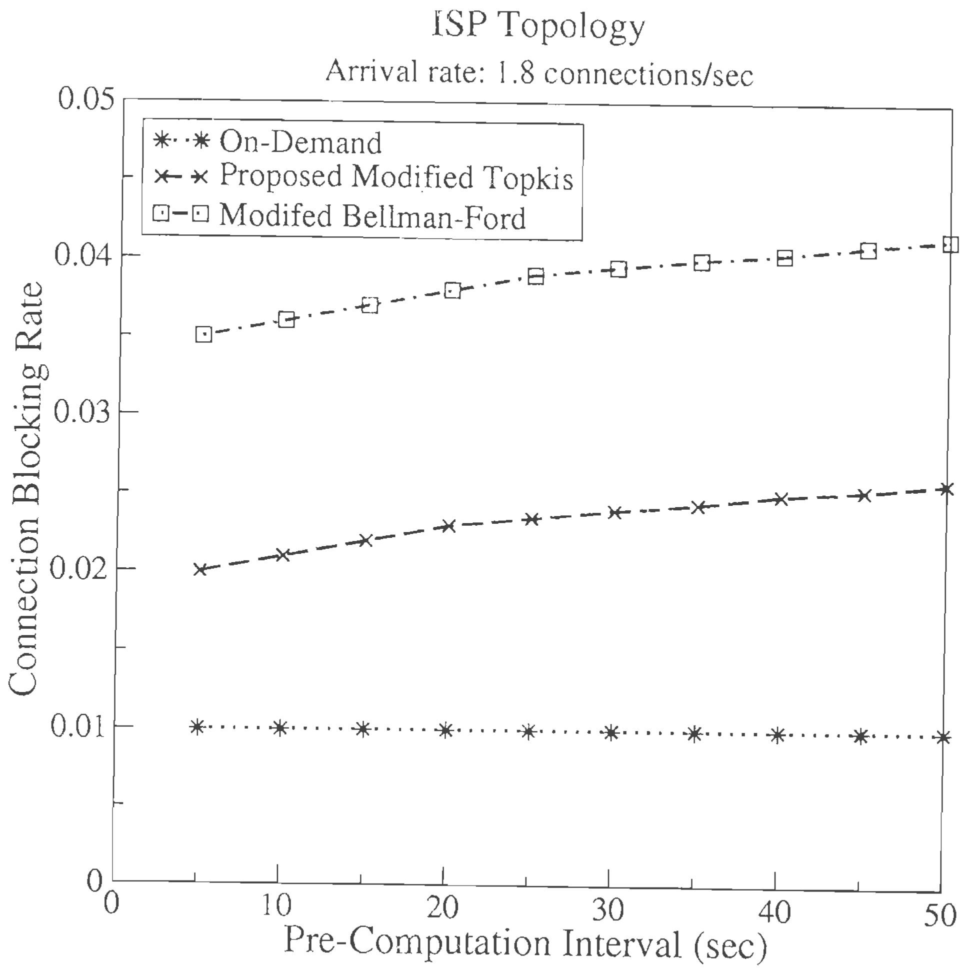

Figure 4.17: Connection blocking rate for different pre-computation intervals for 1.8 connection arrivals per sec and Widest-shortest path objective for ISP topology.

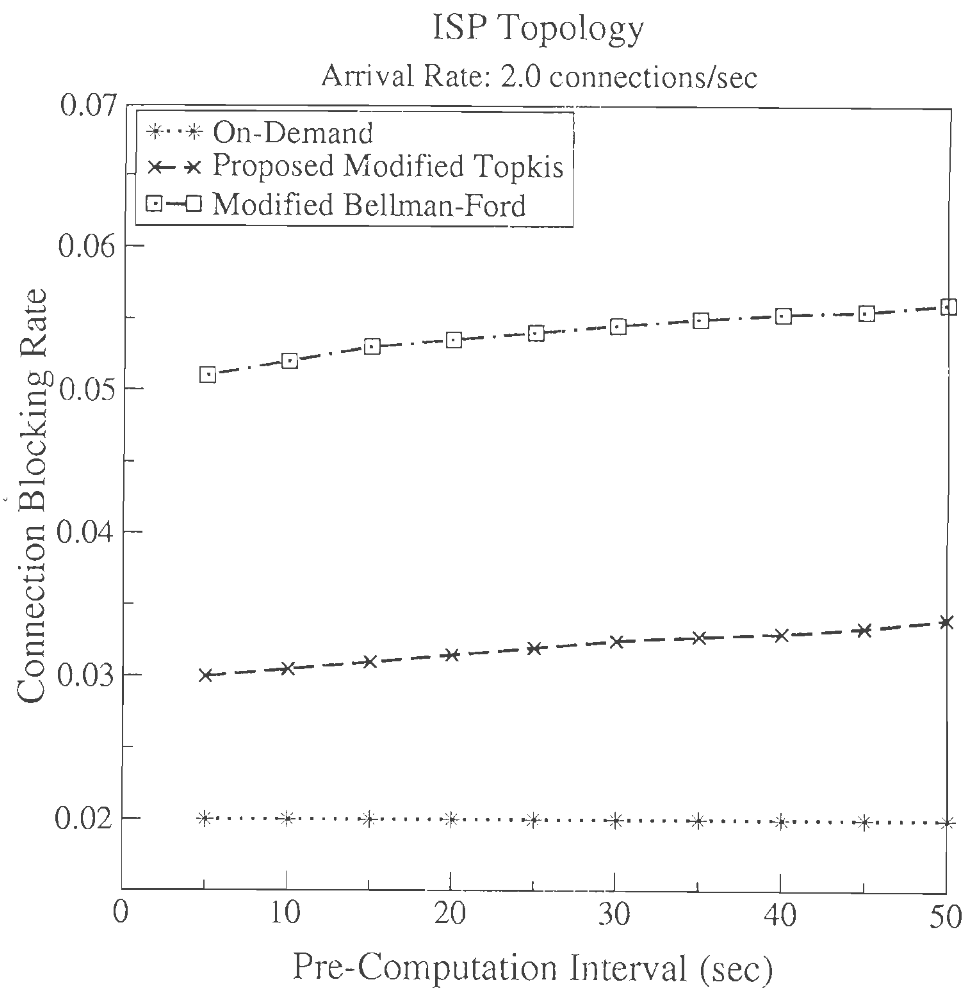

Figure 4.18: Connection blocking rate for different pre-computation intervals for 2.0 connection arrivals per sec and Widest-shortest path objective for ISP topology.

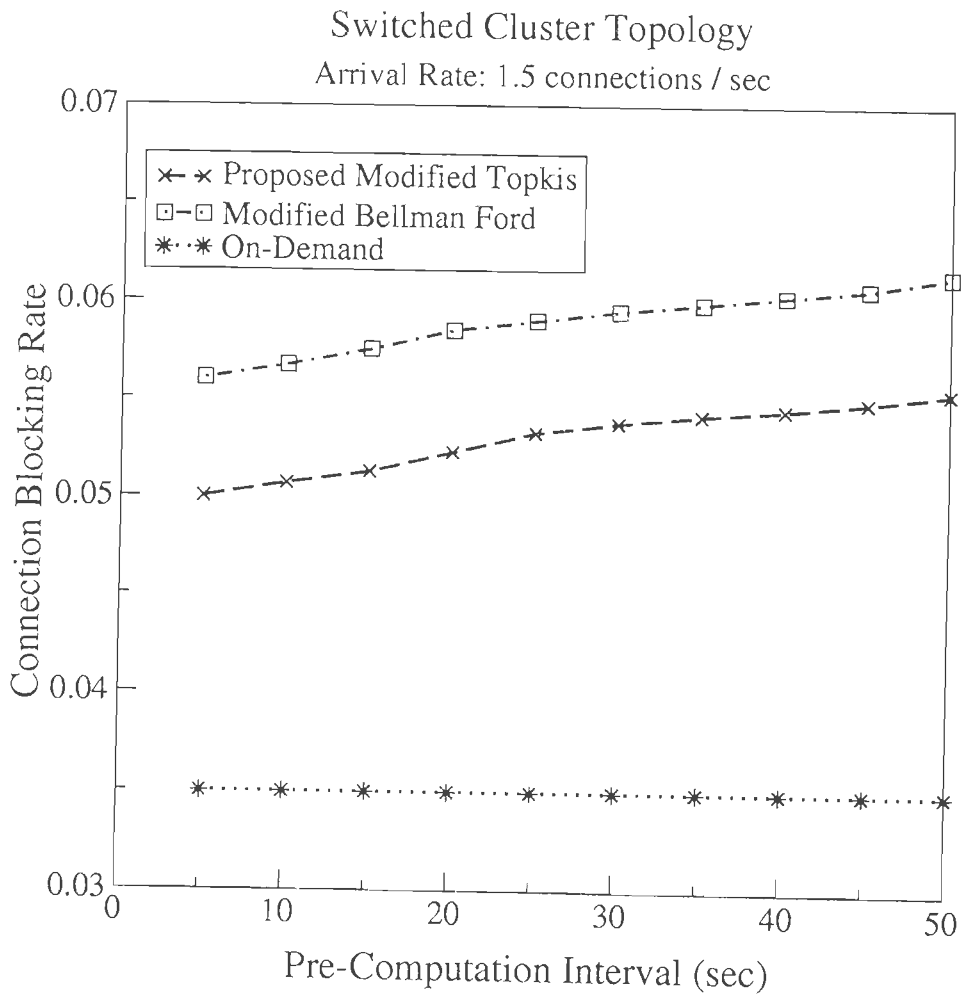

Figure 4.19: Connection blocking rate for different pre-computation intervals for 1.5 connection arrivals per sec and Widest-shortest path objective for Switched cluster topology.

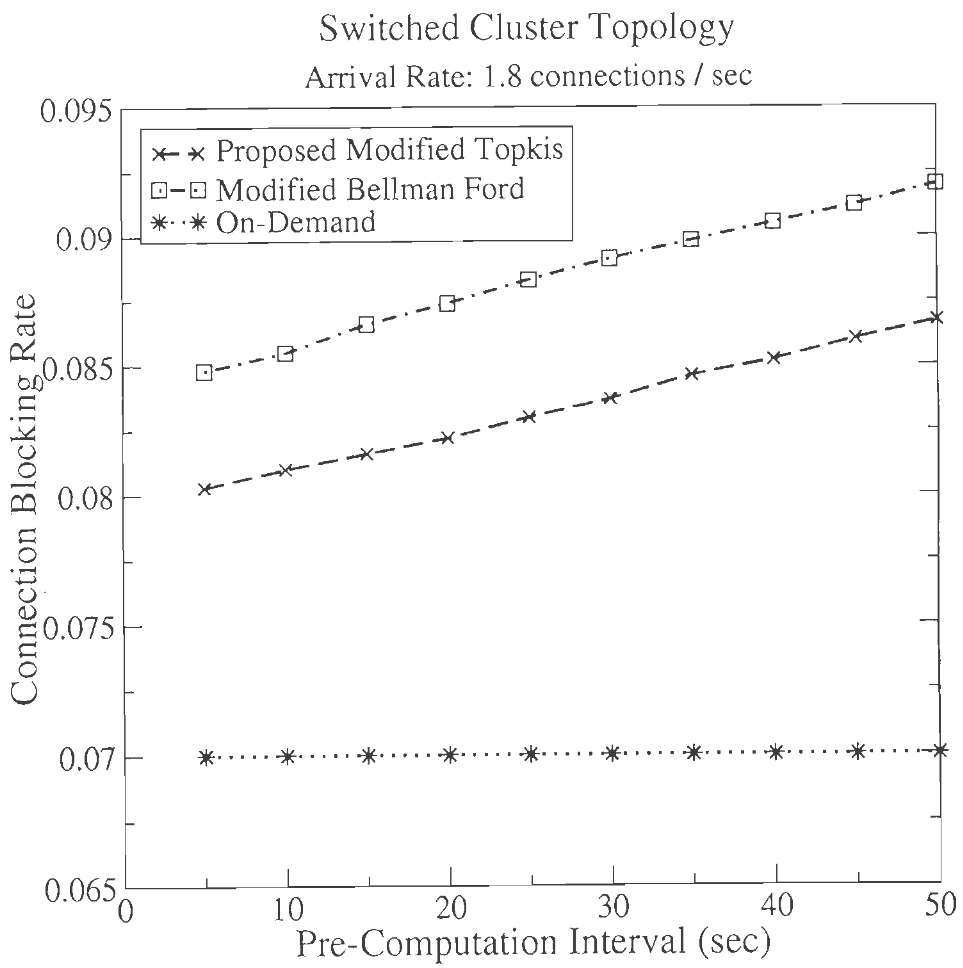

Figure 4.20: Connection blocking rate for different pre-computation intervals for 1.8 connection arrivals per sec and Widest-shortest path objective for Switched cluster topology.

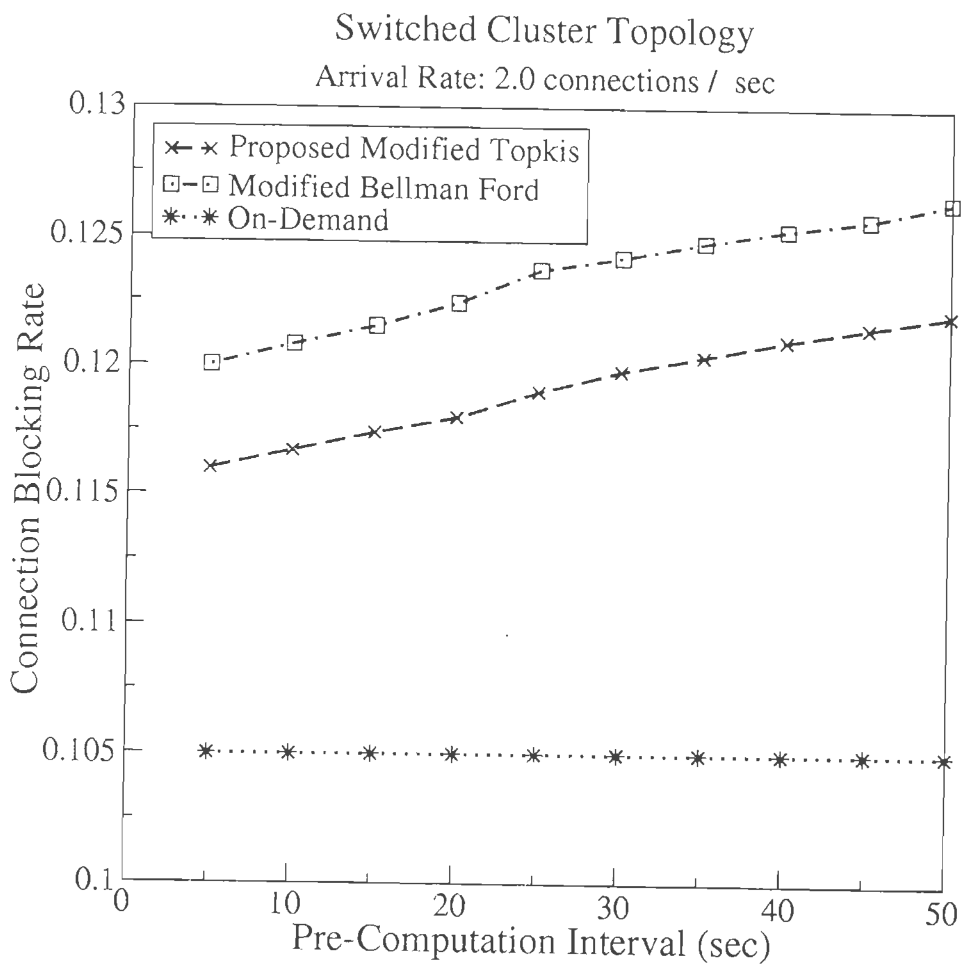

Figure 4.21: Connection blocking rate for different pre-computation intervals for 2.0 connection arrivals per sec and Widest-shortest path objective for Switched cluster topology.

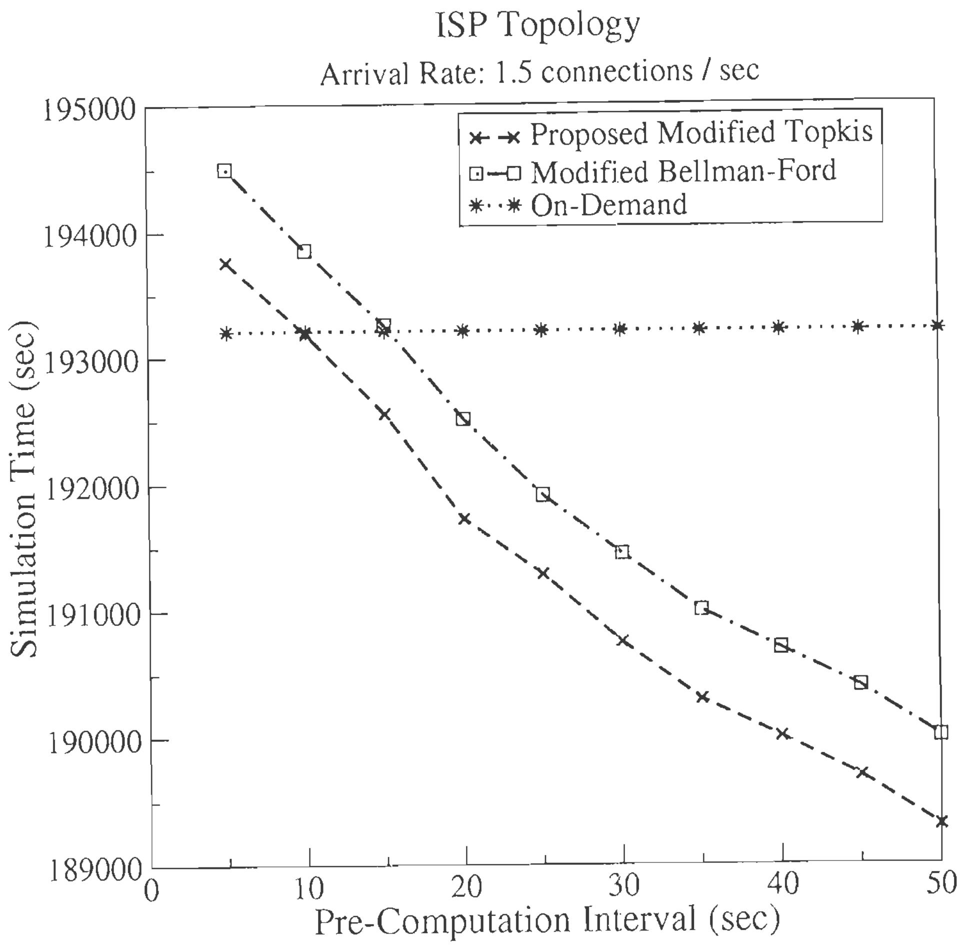

Figure 4.22: Total simulation time till 50000 connection requests arrive for routing for an arrival rate of 1.5 connections per sec and Widest-shortest path objective for ISP topology.

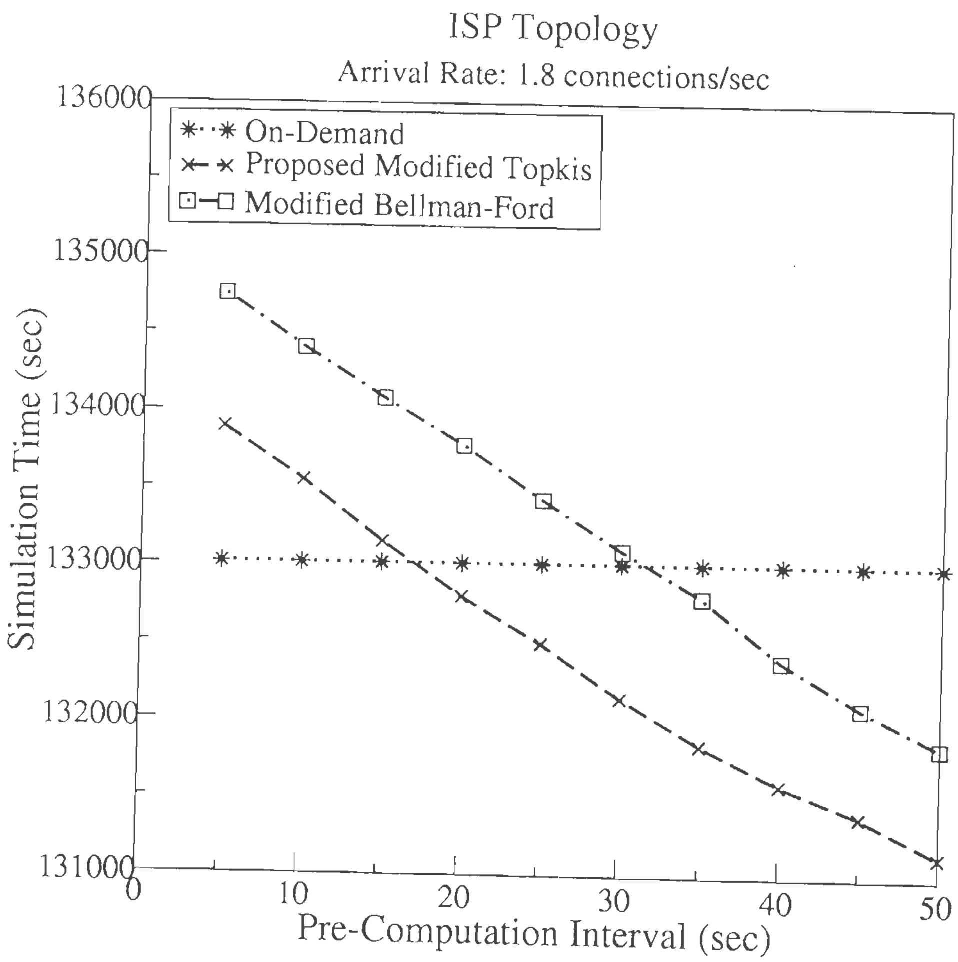

Figure 4.23: Total simulation time till 50000 connection requests arrive for routing for an arrival rate of 1.8 connections per sec and Widest-shortest path objective for ISP topology.

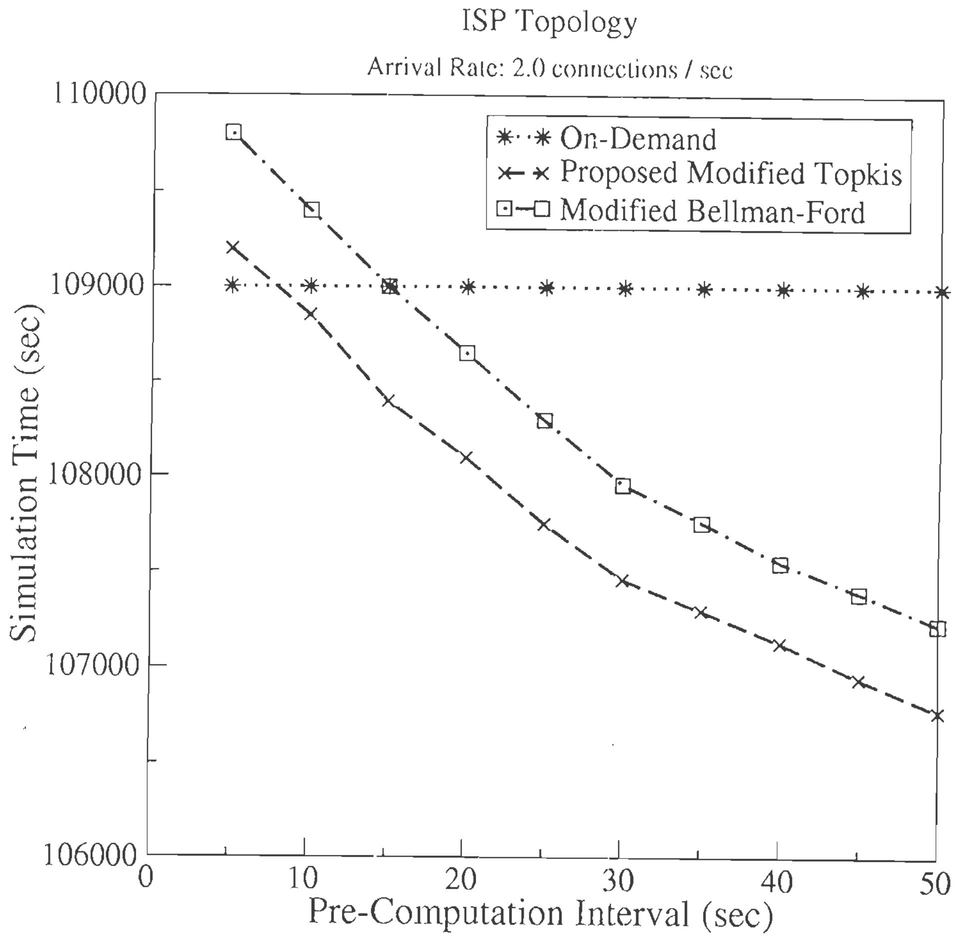

Figure 4.24: Total simulation time till 50000 connection requests arrive for routing for an arrival rate of 2.0 connections per sec and Widest-shortest path objective for ISP topology.

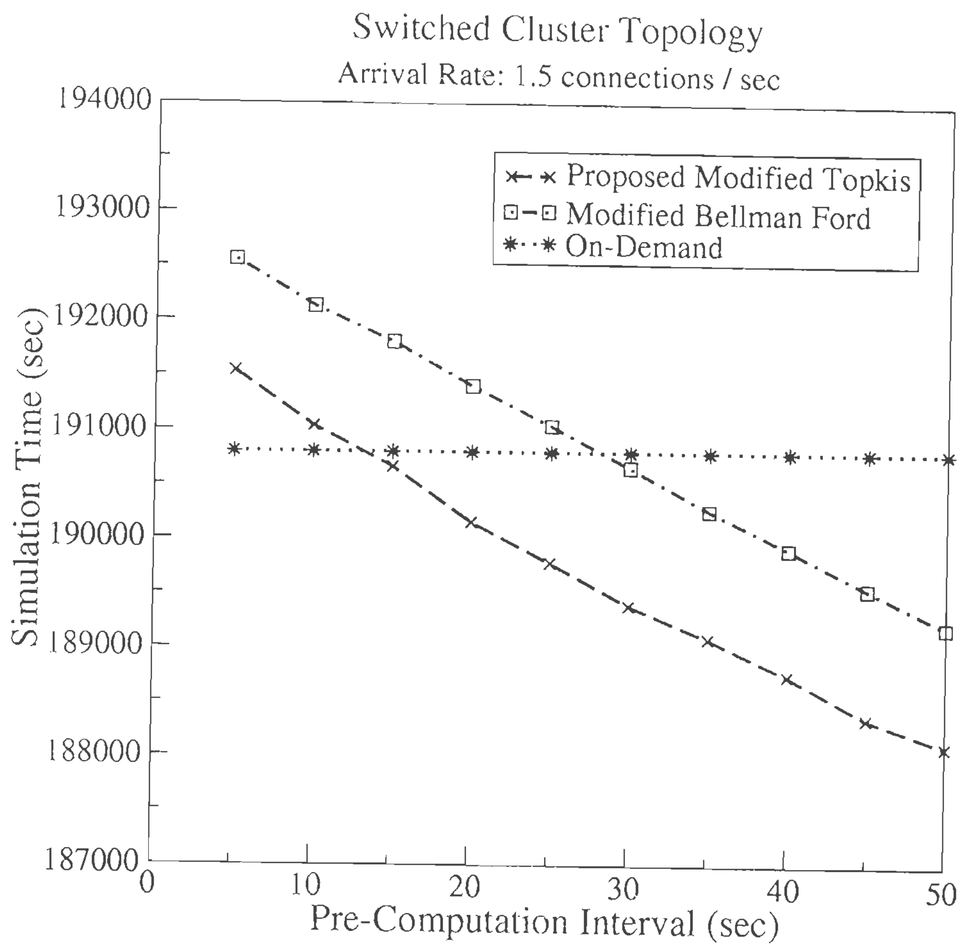

Figure 4.25: Total simulation time till 50000 connection requests arrive for routing for an arrival rate of 1.5 connections per sec and Widest-shortest path objective for Switched cluster topology.

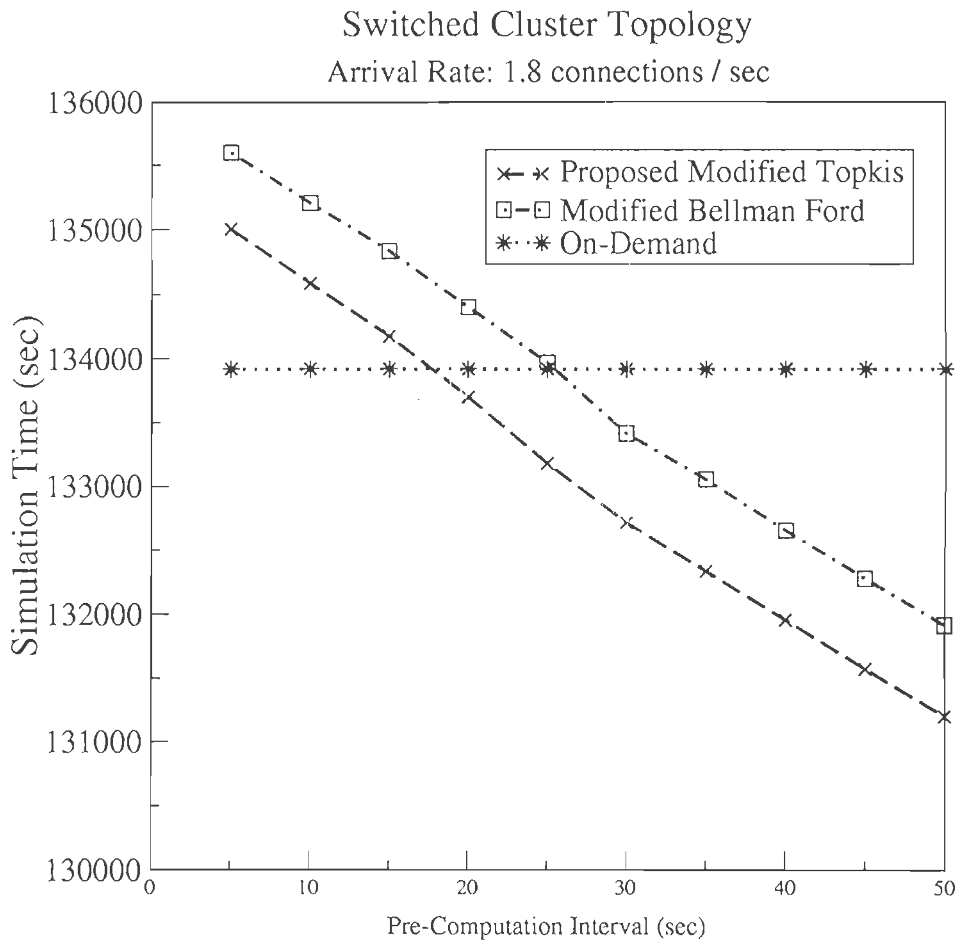

Figure 4.26: Total simulation time till 50000 connection requests arrive for routing for an arrival rate of 1.8 connections per sec and Widest-shortest path objective for Switched cluster topology.

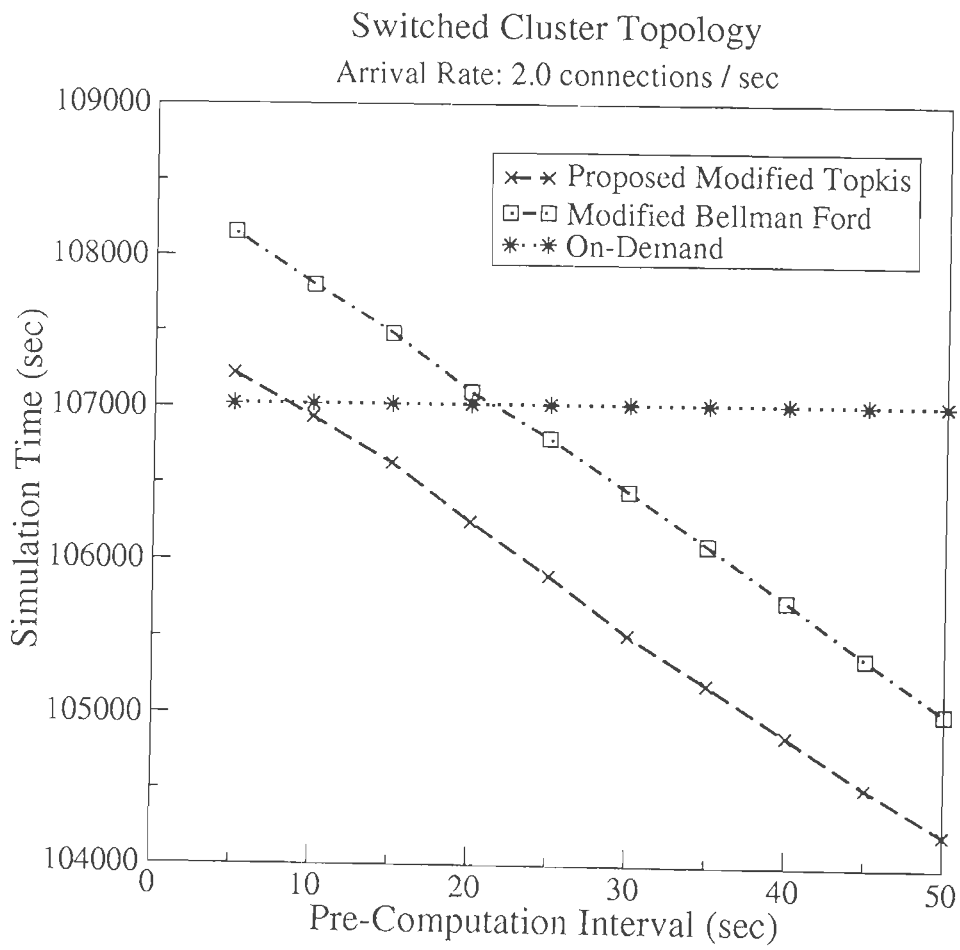

Figure 4.27: Total simulation time till 50000 connection requests arrive for routing for an arrival rate of 2.0 connections per sec and Widest-shortest path objective for Switched cluster topology.

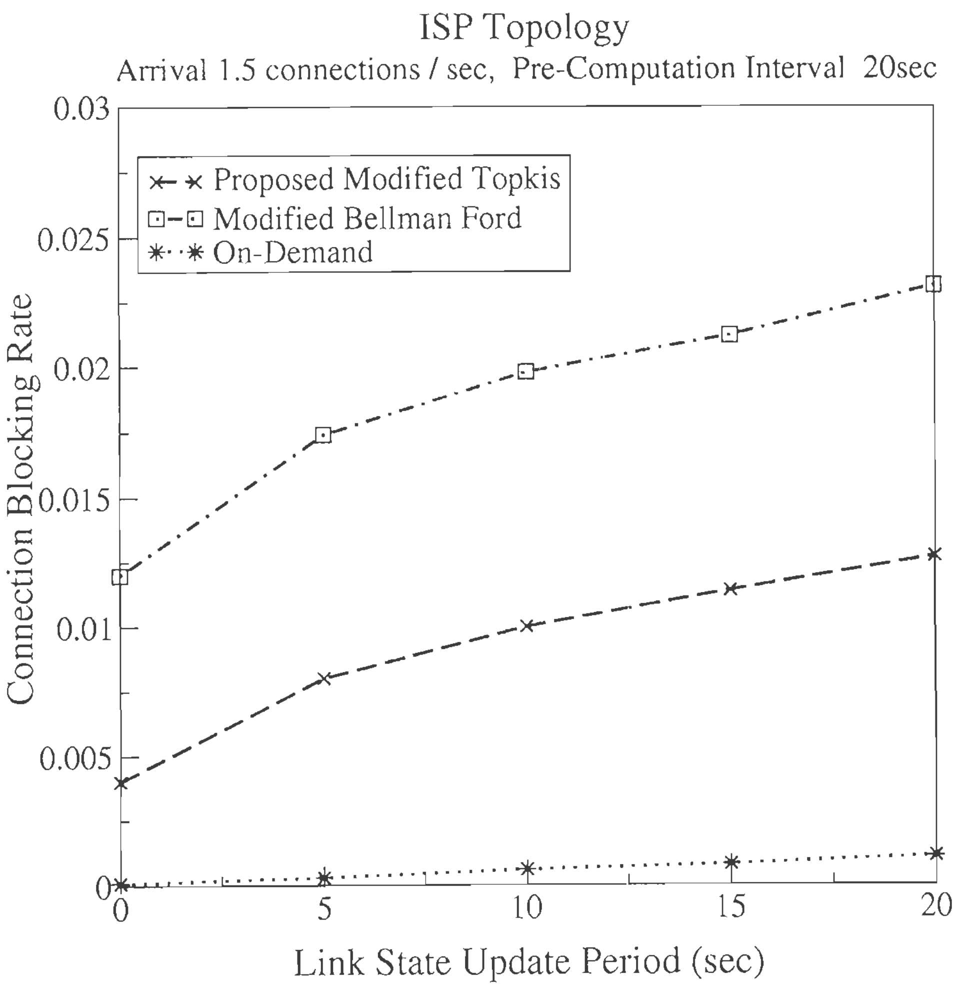

Figure 4.28: Connection blocking rates for Widest-shortest paths for different link state update periods for ISP topology.

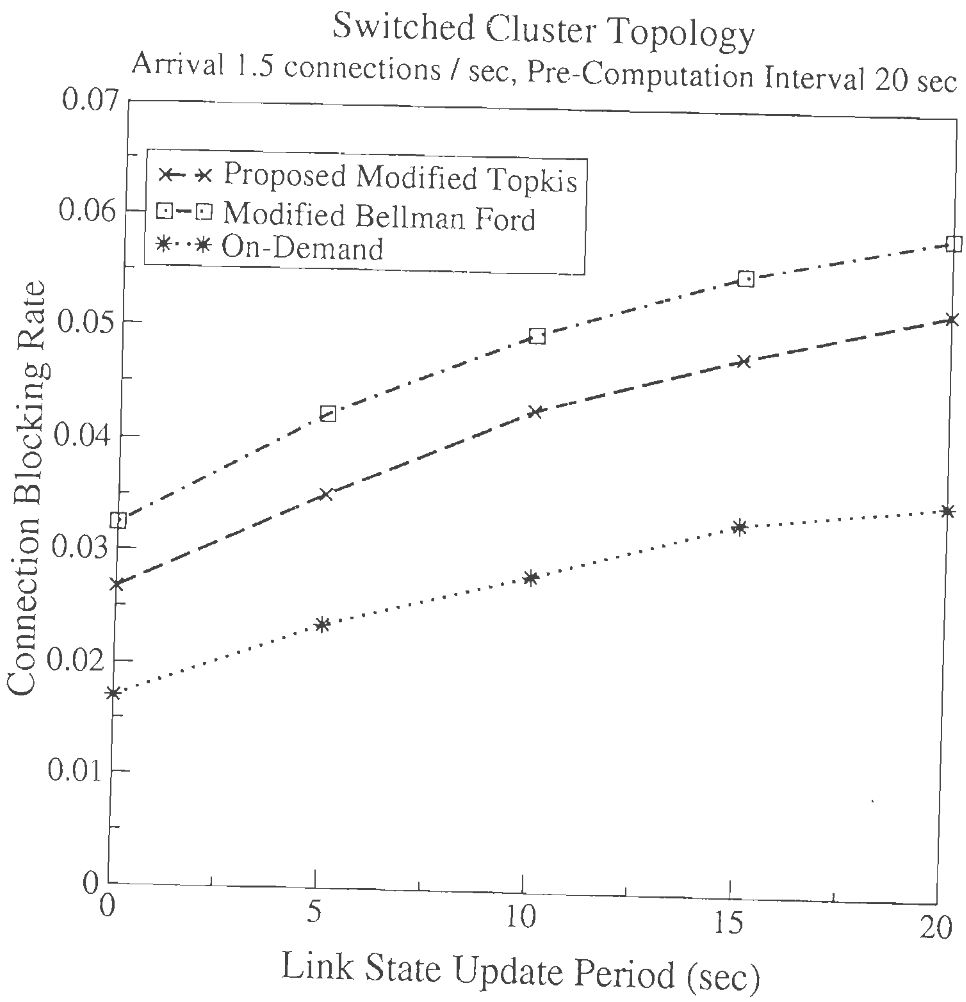

Figure 4.29: Connection blocking rates for Widest-shortest paths for different link state update periods for Switched cluster topology.

Chapter 5

Protocol for Quality of Service Adaptation with Renegotiation

5.1 Introduction

The notion that negotiated QoS for a connection remains the same for the lifetime of the connection is not always valid for multimedia applications. Applications should be able to renegotiate QoS during the lifetime of the connection. Renegotiation is crucial for those applications also that cannot specify the desired QoS for the whole duration of the service. Examples are medical applications or teleconferencing applications. The role of QoS adaptation is to keep providing the negotiated quality of service, eventually lowering it in the case when resources are unavailable. In this chapter we present a new adaptation protocol for QoS adaptation with renegotiation that allows an ATM network to recover from the QoS violations in order to satisfy end-to-end QoS requirements. We also propose a unified model for managing QoS routes and QoS parameters.

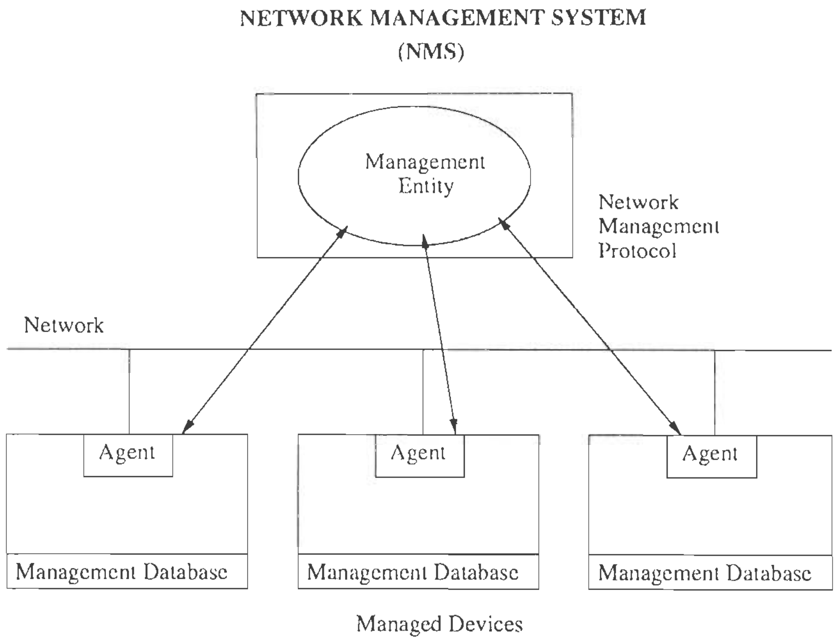

Figure 5.1: Manager/Agent model of the network management system.

5.2 Network Management System

The network management interface that has been defined for use in ATM networks is called Interim Local Management Interface or ILMI. The ILMI is intended for use on the ATM UNI and does not address any network management issues on the NNI interface. The ILMI [6] is based on a protocol for network management, called Simple Network Management Protocol (SNMP), which defines information to be collected by access agents [38]. These agents respond to requests for information and to action directives received from a manager as shown in Figure 5.1. It can be seen in the figure that there are agents in the system that manage different network devices. These agents require software for management purposes. This software is a crucial part of network management software as it has to perform effectively in the presence of large volume of interactions and negotiations so that desired QoS configuration may be provided.

For a comprehensive network management system that performs QoS management, QoS monitoring mechanism is also necessary. It monitors the actual QoS provided to the ongoing sessions so that appropriate actions can be taken in case of any problem in providing specified QoS guarantees. A QoS mapping functionality is also needed to convert actual application QoS to cell level QoS defined in ATM. QoS mapping is also required to allow the service provider to handle and manage meaningful parameter representations. For example the service provider may not know how to handle and manage the frame rate required for video. But it may know how to handle and manage the bandwidth or cell service rate. Thus a mapping of frame rate into throughput is necessary to allow the service provider to support the services requested by the application. In [42], a graphical user interface is available to give users the ability to specify the desired quality of service. Thus mapping becomes necessary. In the this thesis we have assumed that a mapping function is available and the protocol needs to handle the QoS parameters only.

5.2.1 Routing Mechanism

As seen in Chapter 2, ATM uses PNNI protocol for routing purposes. We use a model for QoS management similar to one shown in Figure 5.1. The motivation for using agents in support of QoS management is to perform dedicated processing on behalf of applications inside the network at particular links or in particular regions of node clusters, which may not implement the required algorithms and services.

The PNNI protocol views nodes in an ATM network as a collection of peer groups. All the nodes within a peer group exchange link information and obtain an identical topology database representing the peer group. Peer groups are organized into a hierarchy in which one or more peer groups are associated with a parent peer group. Parent peer groups are grouped into higher layer peer groups with some identifier. Within a peer group, each node has the description of the topology of the peer group including descriptions of all nodes, links and destinations that can be reached from each node and the status of nodes between the nodes. At each level of hierarchy, the topology is represented by logical nodes and logical links. At the lowest level of hierarchy, each node represents a real switching system. At higher layers, each node may represent either a real system or a group of switching systems.

Each node in PNNI protocol advertises a set of parameters (link state updates or LSU) including information about the links attached to it, QoS parameters it can guarantee, administrative weights of the link and its capability and desirability to carry out particular types of connections. Source node uses this information during the route selection process in determining end-to-end routes. All the nodes within a peer group exchange link information and obtain an identical topology database representing the peer group. Routing outside a peer group follows the same link state operation but at the higher layers of the hierarchy [86][87]. We have not considered administrative weights in our protocol.

5.3 QoS Adaptation Protocol

A connection from a source s to a destination t passes through different clusters before terminating at the cluster where destination node is located. Let $m(i,j)$ to be a QoS metric or parameter for the link (i,j). For a path $P = (s,i,j,...,l,t)$, metric m is concave if $m(P) = min\{m(s,i), m(i,j), ..., m(l,t)\}$. Metric m is additive if $m(P) = m(s,i)+m(i,j)+ ... + m(l,t)$. While the bandwidth is concave; delay, delay jitter and cost are additive [117].

When we consider additive QoS parameter like delay then the end-to-end delay of a connection or QoS configuration would be the sum total of the individual delays due to all clusters in that connection. The service requirement is to keep the end-to-end delay within the agreed delay bound, D. Now if some cluster or node violates its component of this delay bound due to some resource allocation problem then there are two methods to resolve the problem:

1. Use alternate QoS configuration. The alternate QoS configuration could be in the form of pre-computed QoS routes as discussed in the previous chapter. This method requires that the new QoS route should be setup, followed by the teardown of the existing connection. It involves cost of both of these operations.

2. Use QoS adaptation mechanism. Here the other clusters or nodes involved in the connection need to compensate for this QoS violation. For example one cluster may offer a smaller delay service. Thus it may decrease its part of delay by an amount that may completely or partially compensate for the violation. If the network QoS management mechanisms are unable to compensate this violation completely then system may either initiate reconfiguration mechanism as specified in the first method or renegotiate with the application.

5.3.1 Features of Management System

It is possible for the multimedia applications to modify the agreed traffic contract. The ITU-T Q.2963.2 recommendation [60] provides the capability for the connection owner (i.e the originator of the call who initiates the setup message) to use the modify message and adjust the traffic descriptor dynamically on an active connection. A new information element, dynamic bandwidth management option, in the existing setup message is used by all parties to indicate their support of this feature. This information element is then included in the connect message to signal the connection owner of the end-to-end support for dynamic bandwidth management. The recommendation also defines messages with appropriate information elements for the user-network interface (UNI) signalling. Once the renegotiation protocol completes, the QoS controller modifies the stream's traffic shaper and effectively flow controls the bit rate to the available network bandwidth.

The protocol that we propose extends the functionality of the above protocol by ITU-T. We assume that there is a QoS agent at every node. It has a QoS/Route Monitor unit that receives QoS/LSU updates from the network on a periodic or triggered basis. The QoS information available at a node is the QoS information of the link from it to the next node. There can be several connections passing through a node at any given time but there will be a single QoS agent. In addition to its usual functions, this entity would also function as QoS manager or Connection QoS Manager.

The QoS/Route Monitor would also send or receive the signalling messages required in our protocol. These signals are exchanged on a control VC established at the time of connection setup. This control VC setup between the source and the destination is maintained throughout the life of the connection. Upon receiving a signalling message on the control VC, the monitor passes the signalling information to its protocol processing entity for processing.

Following are the key characteristics of our management system.

1. A single QoS agent exists at each node that manages all the connections.

2. There is a Connection QoS Manager (CQM) maintained at the source node of the connection and a QoS manager maintained at every cluster for managing QoS of that peer group or cluster.

3. QoS agent of the source node acts as the CQM and QoS agent of the ingress node at every cluster acts as the peer group leader or the QoS manager of that cluster.

Figure 5.2 shows the behaviour of QoS agent and CQM for a connection. The behaviour of

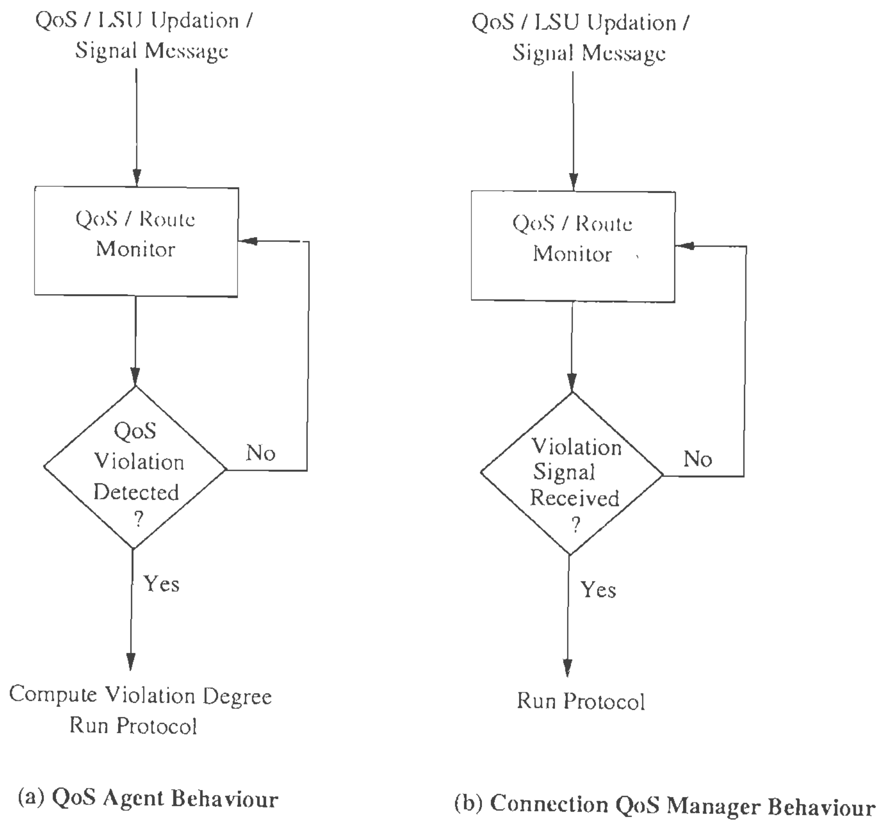

Figure 5.2: Behaviour of QoS agent and Connection QoS Manager (CQM) of a connection.

QoS manager is similar but it is not shown in the figure. As shown in the figure that there is a QoS/Route Monitor unit inside the QoS agent. This entity detects the violation in the QoS and initiates the adaptation protocol as described below in the next sub-section. This entity also receives routing related information which it passes to the QoS Route Manager unit that manages the QoS routes.

5.3.2 Description of the Protocol

As discussed above there are two categories of QoS parameters - concave and additive. The QoS violation may occur in both types of QoS parameters.

(a) When a violation in the concave QoS parameter (i.e. bandwidth) is detected by the QoS agent then the end-to-end QoS cannot be satisfied completely. Therefore a QoS renegotiation mechanism is required. Hence, the QoS agent sends a change QoS request signal to the CQM. The CQM on receiving this signal tries a different QoS configuration for example, an alternate QoS configuration or route to the destination that satisfies all QoS constraints or the end-to-end QoS requirements. If the CQM cannot find a QoS configuration or a QoS route to the destination then it lowers the QoS metric i.e. the bandwidth below the acceptable limit and establishes a connection based on this metric, transfers traffic on this connection and informs the user or the application which may decide to keep or reject this new connection and abort.

(b) When the QoS violation is detected in a QoS parameter that has the additive concatenation property (e.g. delay or cost), the protocol for QoS adaptation works as described below.

1. When a QoS agent detects a QoS violation, that is, it can no longer contribute towards the end-to-end QoS, it computes a violation degree. The violation degree is difference between the targeted QoS and the actual measured QoS. It then sends a violation signal to the next node with in its cluster. The QoS agent of this node now attempts to reserves resources to completely compensate the violation. This agent would be able to either completely or partially compensate for the violation. It may also fail to compensate the violation if it is at its maximum utilization. It would next compute a violation degree which would be the remaining QoS to be compensated and send further a violation signal to next node under that connection which would then attempt to compensate the violation and so on till the last node of the cluster. If the violation is completely compensated i.e. the violation degree becomes zero or the violation signal reaches the end node and still violation degree is not equal to zero, a signal containing the remaining violation degree is sent to the QoS agent from where the violation signal had been originated. This scheme

succeeds if the violation degree becomes zero.

2. When the above QoS adaptation scheme fails i.e. the violation degree is not equal to zero, the originating QoS agent sends the violation signal containing the remaining violation degree to the CQM. The CQM checks then the resource availability at all clusters by sending proper signals to the respective QoS managers. These QoS managers collect the QoS information from their agents and send it to the CQM. The CQM then computes the necessary QoS changes for every cluster and sends the QoS change signal to the QoS manager of that cluster. On receiving the QoS change signal from the CQM, the QoS manager at each cluster requests its QoS agents to commit the changes.

3. If the CQM fails to compute suitable QoS changes for peer groups then QoS renegotiation is required. CQM tries an alternate QoS configuration to satisfy end-to-end requirements. If that is not possible, the CQM lowers the QoS metric by a suitable value, establishes a new connection, transfers the traffic on it and informs the application i.e. higher layer which may decide to keep or reject this new connection.

5.3.3 Signalling Messages

To facilitate the functioning of the protocol described above, several signalling messages are needed. The signalling messages characterize different control mechanisms required for the protocol and the network. Each signalling message is a request for a specific function or it is a response to a specific request. A signalling message is composed of a number of informational elements (IE) with each IE identifying a specific aspect of the function requested by the message. The following IEs are used in the signalling messages.

a) ConnId - Identifier for the connection

b) AgentId - Identifier for the QoS agent (including that of QoS manager)

c) ViolationDegree - Difference between the desired QoS and the currently provided QoS

d) MaximumQoS - Maximum amount of QoS that could be supported by some agent

e) QoSToSupport - Amount of QoS needed to be reserved by a cluster as computed by CQM

f) LowerQoS - QoS for alternate degraded QoS configuration tried by CQM.

The signalling messages are described as follows.

1. VIOLATION(ConnId, AgentId, ViolationDegree)- When a QoS violation is detected by a QoS agent, The violation could be in either concave or additive parameter. If the violation is in the additive parameter then the QoS agent needs to computes a violation degree that is equal to the difference between the targeted QoS and the actual measured QoS. This signal is then sent by the QoS agent to the next node in its cluster. This signal is also generated by the last node of the cluster.

2. REQUEST_RESOURCE(ConnId, AgentId) - This signal is sent to QoS managers of all clusters except to the one from where the violation signal was originated. Through this signal CQM solicits from all other QoS managers information about the maximum QoS that they can support.

This similar to triggering the QoS managers for generating link state updates (LSU s) but with maximum QoS contribution information. AgentId is for identifying QoS managers.

3. QOS_AVAILABLE(ConnId, AgentId, MaximumQoS) - When the REQUEST_RESOURCE signal is received, the QoS manager computes the maximum QoS that it can contribute and informs CQM using this signal.

The MaximumQoS element sent by a QoS manager could be zero.

4. CNF_QOS(ConnId, AgentId, QoSToSupport) - CQM on receiving QOS_AVAILABLE from all the requested QoS managers, decides the best QoS value for each QoS manager so

that end-to-end QoS requirement may get satisfied. One approach is to distribute the required QoS equally among the clusters. The QoS managers would then distribute the QoS among the nodes in their respective clusters. This is QoS adaptation.

5. DEGRADE_QOS(ConnId, AgentId, LowerQoS) - When the effective contribution of each QoS manager is negative, this means that QoS violation cannot be compensated. The CQM then tries alternate QoS configuration for the application between end points. This is QoS reconfiguration.

If the alternate configurations are impossible then this signal is sent by CQM to QoS managers to lower QoS by a minimum value and the application is informed. This is called graceful degradation.

6. ABORT_CONNECTION(ConnId, AgentId) - If the application decides to reject the new degraded QoS configuration then CQM sends this signal to all peer leaders to abort the connection (ConnId).

Table 5.1 summarizes the types of signals and their definitions for QoS agents and QoS managers. Some of the signals will be sent to all agents except to the one that detected a violation. Similarly, certain signals will be sent by all agents except the one that had detected a violation.

5.4 Unified Model for Managing QoS

We have assumed that there is a QoS agent at every node that monitors and collects the QoS information available at that node. We propose a new model for quality of service management that has QoS agent as the crucial entity in the model. This model integrates the resource allocation/buffer management schemes for providing QoS, the QoS route manager that han-

Signal Message Type	Informational Elements	Definition	Comment
VIOLATION	ConnId, AgentId, ViolationDegree	On detection of QoS violation, this signal sent to next node / CQM	Sent by agent that detected violation or last node of the cluster
REQUEST_RESOURCE	ConnId, AgentId	On receiving violation signal CQM solicits from other QoS managers information about the maximum QoS that they can support	Sent to the other QoS managers except to the one from where violation signal received
QOS_AVAILABLE	ConnId, AgentId, MaximumQoS	On receiving request resource signal, different QoS managers inform CQM about their maximum contribution	Sent by all QoS managers that received request to contribute more QoS
CNF_QOS	ConnId, AgentId, QoSToSupport	CQM decides best QoS value for each QoS manager to be allocated therein to satisfy end-to-end requirements	Sent by CQM to QoS managers for contributing QoSToSupport
DEGRADE_QOS	ConnId, AgentId, LowerQoS	When effective contribution of each QoS manager is negative and alternative configuration is impossible then CQM sends this signal to degrade QoS	Sent by CQM to QoS managers (peer group leaders) to lower QoS
ABORT_CONNECTION	ConnId, AgentId	CQM sends this signal to various peer leaders to abort connection	Application decides to abort the new degraded QoS configuration

Table 5.1: Signal message types and definitions.

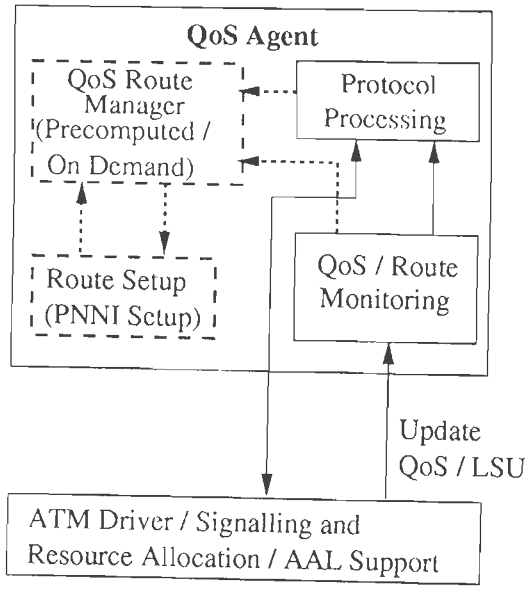

Figure 5.3: Model of the QoS Agent maintained at every node

dles pre-computed/on-demand QoS routes and Protocol Processing entity that adapts to QoS violations or system initiated renegotiation. Some of these entities could be part of switch software or switch OS. The resource allocation method may be performed in conjunction with rate control methods is necessary for QoS conformance. This has to be done at cell level or the transport level.

The entities QoS Route Manager and Route Setup shown in dashes in Figure 5.3 are active only when the node is the source node of a connection. When a node becomes a source node or needs to compute QoS routes, QoS Agent acts like the Connection QoS Manager (CQM). Figure 5.4 shows the proposed unified model of the Connection QoS Manager.

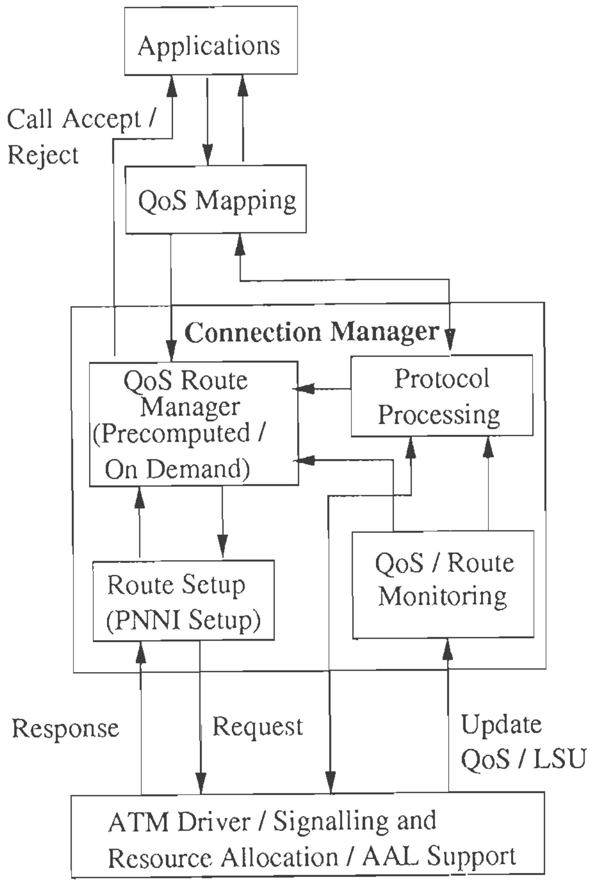

Figure 5.4: Model for Connection QoS Manager

5.5 Discussion and Conclusions

The protocol discussed in section 5.3 was implemented on a simulated ATM network. The main advantage of the PNNI protocol is that it provides distinct administrative domains for each cluster. This helps in separating the management costs per cluster basis. The proposed protocol for the QoS adaptation may be used for revenue maximization of the network. The concept of QoS improves the network resource utilization. Though the topology of the network can change but it is relatively infrequent compared to those QoS constraints like bandwidth or delay. Therefore QoS adaptation and QoS renegotiation scheme become important. These schemes are also important, as they can be more frequent compared to the connection establishment as connection establishment involves extra overhead of allocating VCs and state creation in the switches. We suggest that the unified model for the management of QoS techniques which includes schemes proposed by us should be implemented.

Chapter 6

Conclusions and Scope for Future Work

6.1 Conclusions

In this thesis we have investigated schemes for the quality of service management in ATM networks. An efficient scheme for the provisioning of QoS parameters through resource allocation and buffer management has been proposed. This scheme can be integrated with cell scheduling schemes like rate based scheduling. We have proposed a pre-computation algorithm for determining quality of service routes. The results of the comparative study made with the traditional Modified Bellman-Ford algorithm show that the performance of the proposed algorithm is better.

The pre-computation of QoS routes provides alternate QoS configurations. When a QoS reconfiguration is required the choice could be to use an alternate route if available or to lower QoS on the existing connection. A protocol for adaptation of quality of service has been proposed that accommodates to the QoS violations and performs graceful degradation of the quality of service.

The major findings of our work can be summarized as follows:

1. For the provisioning of quality of service parameters dynamic bandwidth allocation and buffer management is required.

2. The buffer management with proper allocation policies should be incorporated inside ATM nodes so as to satisfy QoS.

3. The Modified Topkis algorithm proposed by us for finding k-constrained QoS routes has better performance than the generally used Modified Bellman Ford algorithm. The connection blocking rate of the proposed algorithm is lower compared with Modified Bellman Ford algorithm.

4. On-demand QoS routing has much lower connection blocking rate however this algorithm requires route computation on every request. We found that the pre-computation schemes can have smaller computational times than on-demand routing.

5. There is a considerable performance gap for the QoS routing algorithms. The performance of these algorithms varies according to the network load and topology.

6. We have proposed a protocol for QoS adaptation with renegotiation in ATM networks. This protocol can be used for QoS adaptation and QoS reconfigurations.

7. For quality of service management in ATM networks, we propose a unified model for QoS management that integrates resource allocation/buffer management schemes for providing QoS, QoS route manager that handles pre-computed/on-demand QoS routes and Protocol Processing entity that adapts to QoS violations or system initiated renegotiation.

6.2 Scope for Future Work

As the high speed networks are deployed, newer applications are emerging like telesurgery and continuous multimedia database access. These applications have stringent service requirements and need efficient provisioning and management of quality of service. Hence QoS management becomes critical for the overall performance in the network. We outline following specific areas in QoS management where further research is required.

1. Studies should be made to improve upon the resource allocation scheme proposed by us to simultaneously satisfy different QoS parameters for different traffic conditions.

2. The pre-computation algorithms should be studied to find out the cost and other overheads, and determining the optimal level of pre-computations.

3. Efforts should be made to implement the proposed unified model for QoS management in ATM switch software (OS). Efforts should be made to design APIs at ATM switches that implement the proposed adaptation protocol and the unified model for QoS management.

4. The cost-benefit analysis of alternate QoS configurations, QoS adaptation and reconfigurations need to be investigated for the optimum use.

5. The differentiated services model recently proposed for the Internet does not rely on any flow based state representation inside the core network and there are no explicit resource reservations. Studies should be made to determine the performance of resource allocation and other mechanisms for QoS in networks supporting differentiated services as against the integrated services schemes. Studies should also be made for QoS management in such networks.

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