Scalable quality of service (QoS) over broadband internet and next generation mobile network

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**Scalable Quality of Service (QoS)
over Broadband Internet and
Next Generation Mobile Network**

Zaw Min Naing

School of Electrical & Electronic Engineering

A thesis submitted to the Nanyang Technological University
in fulfillment of the requirement for the degree of
Doctor of Philosophy

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Abstract

The increasing need of providing QoS over broadband Internet and mobile network has driven the intensification of the research efforts in this area. The thesis mainly focuses on modeling and performance analysis of scalable QoS over broadband Internet and MPLS based mobile network.

The first focus of this thesis is modeling and performance analysis of aggregate reservation to provide required QoS and scalability over broadband Internet. Performance results obtained by numerical analysis and simulation experiments show that aggregation is able to provide performance of QoS as that of IntServ model. The second focus of this thesis is to present the framework for efficient mobility with continuous QoS support and dynamic handover using MPLS protocol. Major performance measures obtained from the simulation experiments show that the proposed LSP scheme is able to provide QoS support with scalable mobility in MPLS based mobile access networks. A reliable handoff scheme is elaborated. The third focus of the thesis is modeling and performance analysis of IntServ (IS) and DiffServ (DS) traffic with multiple QoS classes over MPLS based mobile network. The performance analysis is done using fluid flow technique. The numerical results demonstrate that the proposed queuing model and the differentiated service based on the priority schemes defined in the MPLS packet header are effectively able to provide the multi-class QoS requirement for supporting multimedia traffic over MPLS based mobile network.

STATEMENT OF ORIGINALITY

I hereby certify that the following research works embodied in this thesis have been carried out originally by the candidate and the thesis has not been submitted for a higher degree award to any other University or Institution. This thesis includes the following major research works that have been done originally by the candidate:

1. A novel resource reservation mechanism called aggregate resource reservation scheme combining with IntServ model has been developed. IntServ architecture provides guaranteed QoS for individual traffic flow but it has significant scalability problem due to per flow reservation request. The proposed scheme is able to provide QoS support on aggregation basis plus flexible scalability over broadband Internet. The performance results obtained from numerical analysis and simulation experiments show the significant improvement of QoS and scalability over the conventional IntServ model.
2. Mobility with QoS support based on MPLS technique in mobile access network is investigated. First of all, a framework of efficient mobility with continuous QoS support in mobile access networks using MPLS is presented. The proposed strategy is implemented using scalable LSP configuration which is able to support IP DiffServ model. Secondly, a reliable handoff scheme is elaborated. Major performance measures obtained from the simulation experiments show that the proposed LSP scheme based on MPLS is able to provide QoS support with scalable mobility in mobile access networks.
3. Modeling and performance analysis for MPLS based mobile network has been carried out. The modeling and performance analysis are focused on mixed IP traffic with multiple QoS classes. The analysis is done by fluid flow queuing technique. The numerical results show that differentiated services based on the MPLS technique are effectively able to provide the multi-class QoS requirement for supporting multimedia traffic over MPLS based mobile network.

17-03-2005
Date

Zaw Min Naing

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SUMMARY

The increasing need of providing QoS over broadband Internet and mobile network has driven the intensification of the research efforts in this area. Therefore, how to provide scalable QoS in broadband Internet and mobile network introduces a new challenge. The thesis mainly focuses on the modeling and performance analysis of scalable QoS over broadband Internet and MPLS based mobile network.

The first major focus of this thesis is the modeling and performance analysis of aggregate reservation combining with IntServ to provide QoS support and flexible scalability. The performance analysis of aggregate reservation is evaluated on three performance metric: overload probability, reserved resource utilization and the normalized available bandwidth. The performance results obtained from numerical analysis and simulation experiments show that if the control time scale and demand time scale are separated by two orders of magnitude and variance of traffic is moderate, aggregation can provide performance close to conventional IntServ model.

The second major focus of this thesis aims to provide an efficient mobility with continuous QoS support and dynamic handover in mobile access networks using MPLS protocol. The proposed strategy is implemented using scalable LSP configuration which is able to support IP DiffServ model. It provides simple and fast handoff by updating the gateway's forwarding table while supporting the same QoS after handoff. Moreover, a reliable handoff scheme is also elaborated. The evaluation has been focusing network mobility in terms of new arrival call blocking probability and handover call blocking probability. Major performance measures obtained from the simulation experiments show that the proposed LSP scheme offers lower handoff call blocking probability (i.e., significantly improves probability of successful handover) under different traffic load scenarios. Therefore, the proposed LSP scheme is able to provide QoS support with scalable mobility in MPLS based mobile access networks.

The third major focus of the thesis is the modeling and performance analysis of IntServ (IS) and DiffServ (DS) traffic with multiple priority classes over MPLS based mobile network. Multi-class QoS is implemented on differentiated service basis using loss priority scheme and delay priority scheme incorporated in the MPLS packet header. Note that mobility is included in queuing model. The analysis is done by fluid flow queuing technique. The obtained numerical results from the performance analysis demonstrate that the proposed queuing model and the differentiated service based on the priority schemes defined in the MPLS packet header are able to provide the multi-class QoS requirement for supporting multimedia traffic over MPLS based mobile network.

The fundamental theme of this research uses modeling and analysis techniques for scalable QoS in support of multimedia traffic over broadband Internet and MPLS based mobile network. Throughout this thesis, the proposed mechanisms and algorithms such as aggregate resource reservation coupling with IntServ (IS) and queuing models with multi-class QoS over MPLS based mobile network are evaluated using numerical experiments and simulations based on real-life video trace.

CHAPTER 1 INTRODUCTION

1.1 Motivation

Provisioning of scalable quality of service (QoS) in emerging high speed networks has gained much popularity in recent years. Unlike traditional data networks, broadband wide area networks will be required to carry a broad range of traffic classes ranging from bursty, variable-rate sources, such as voice and variable rate coded video, to smooth and constant bit rate sources. On the other hand, these networks will have to do so while providing a guaranteed performance or quality of service (QoS) to many of these traffic classes and the ability to guarantee performance is particularly important in supporting real-time applications over networks. The need to provide QoS guarantees remains an important issue in broadband networking. Thus, work is underway on a wide variety of high bandwidth applications such as medical diagnosis, multimedia conferencing, etc. One of the burning issues in this evolution deals with the provision of adequate service for these new applications. In addition, from the network performance point of view the scalability also plays an important role in high speed networks.

Current IP networks are only able to provide the best effort datagram delivery. Several recent studies have focused on QoS as the Internet evolves into a global communication infrastructure and thus demands for IP QoS have rapidly developed. In order to support QoS in the Internet, the Internet Engineering Task Force (IETF) defined the quality of service support mechanisms for the Internet IP networks. Two types of architectures have been developed for providing QoS, including IntServ Architecture (ISA) and DiffServ (DS). In ISA, flows must request service from the network and are accepted or rejected based on the resources available. In this technique, RSVP signaling is used to carry the reservation request to the routers along the path. Such architectures provide excellent QoS but significant scalability problem. Routers must process per flow reservation requests and keep per flow state. In the limit of many flows, an unbearable burden will be placed on

routers. Hence, the scalability of these ISA remains an open question. Differentiated Services (DS) is another model for providing QoS. This method requires no signaling, nor maintenance per flow state at core routers and resources are not explicitly reserved. Instead, traffic is differentiated into a set of classes, and network routers provide priority based service according to these classes. The disadvantage of this approach is that upon overload in a given service class, all flows in that class suffer a degradation of service. Therefore, comparative study has been made in this thesis to achieve engineering benefits.

Currently, there is a significant interest in the development of solutions for performance guarantees in broadband networks. New applications such as video conferencing and medical imaging have strict performance requirements in terms of throughput, delay, delay jitter, and packet loss rate. Current packet switched network (such as the Internet) offers only a best-effort service where the performance of each session can degrade significantly when the network is overloaded. Hence, it is important to provide network services with performance guarantees. One of the most important issues in providing guaranteed performance services is the choice of the packet service discipline at the router. Service discipline must be simple in order that they can be implemented at very high speeds. Therefore, the thesis presents three effective solutions for providing performance guarantees over broadband Internet. They are namely:

1. Dynamic Packet State (DPS)
2. Rate Controlled Service Discipline (RCSD)
3. Aggregate Base Resource Reservation Scheme

DPS technique provides the scalable and robust network because edge routers perform per flow management and core routers do not perform per flow management. RCSD relates to the provision of per connection end-to-end delay guarantees in a high speed network. In this discipline, traffic from each connection is reshaped at every hop. The main disadvantage of this service discipline is that the end to end delay guarantees are obtained as the sum of the worst case delays at each node. But the problem can be eliminated by using good traffic shapers. With a suitable choice of shaper envelopes the RCS discipline

can provide the same end-to-end delay guarantees that the best delay bound for GPS can provide. In this research work, the study shall involve why RCS discipline has become a powerful solution for providing end to end delay guarantee and throughput. Aggregate base resource reservation scheme is useful for providing the QoS of the network in a scalable way. By using aggregation of resource reservation, the network can simultaneously be achieved scalability and high utilization.

To provide guaranteed QoS for a particular data flow, it is necessary to create and maintain resource reservation on each link. Integrated Service Architecture (ISA) has been designated as a flow based resource reservation scheme and provides QoS requirements for each traffic flow. However, such kind of scheme needs to request reservation at core routers for each flow. This feature is not efficient in high speed networks, hence aggregation has been proposed for achieving scalability [1]. Aggregation of flows has no negative effect on the mean delay of the flows and actually results a reduction of delay in the “tail” of the delay distribution for the flows. By implementing the concept of aggregation, the high speed networks can acquire the per flow quality of service (QoS) without necessary to process per flow signaling at core routers. A key concentration and distribution of the first part of this thesis is mainly focused on aggregate resource reservation scheme combining with ISA to provide scalable QoS in broadband Internet and performance analysis of aggregate reservation.

IETF has proposed many service models and mechanisms to meet the demand for QoS. Significantly among them are the Integrated Services/RSVP model [2], the Differentiated Services (DS) model [3-4], MPLS [5], Traffic Engineering [6-7] and Constrained Based Routing [8]. The IS model and DS have already been mentioned above. MPLS uses a label switching technique to forward data through the network. A small fixed format label is encapsulated within each data packet on its entry into the MPLS network and the subsequent classification, forwarding, and services for the packets are based on labels. At the egress node of MPLS network, label is removed and forwards it as a regular IP packet.

Most current network routing protocols such as Internet Gateway Protocol (IGP) are based on shortest path algorithm. The shortest paths from different sources overlap at some links, causing congestion on those links. The purpose of traffic engineering [7-9] is to enhance network utilization and to improve the architecture (topology and link capacity) of a network in a systematic way, so that the network is robust, adaptive, and easy to operate. For example, non-shortest paths can be chosen to forward traffic. An important result of traffic engineering is the avoidance of congestion on any path. Constrained based routing (CBR)[8] computes routes that are subject to constraints such as link characteristics (bandwidth, delay), hop count and QoS requirement. CBR can be used in conjunction with MPLS to set up LSPs. The LSPs that are established could be CBR-LSPs, where the constraints could be explicit hops or QoS requirements. When using CBR, it may select a longer (in terms of cost) but lightly loaded path better than a heavily loaded shortest path. In order to improve the performance of IP routers, the thesis presents label swapping algorithm and traffic aggregation mechanism as a highly efficient scheme for forwarding data packets since packets of different flows can be labeled differently and received different QoS and hence achieving network scalability by implementing aggregation of traffic flows of the same class inside LSP.

With the proliferation of mobile networks technologies, mobile users are expected to demand the same QoS available to fixed users. The continuous support of minimum QoS guarantees of multimedia application poses a major challenge in heterogeneous, mobile environment. Such a support requires the development of a flexible and adaptable network resources management framework. The framework must provide efficient QoS mechanisms as mobile units move from one coverage to another and accommodate applications with different QoS requirements in response to network performance degradation. In order to efficiently support QoS guarantees in a mobile network, bandwidth must be reserved in advance in those cells that are most likely to be visited by the mobile unit. Therefore, resource reservation mechanism is essential for QoS provisioning in mobile environment. To overcome mobility impact on service guarantee,

mobile hosts need to make resource reservation in advance at the location where it may visit during the lifetime of the connection.

A number of micro-mobility schemes address the fast handover issues with a good degree of success but lacking the capabilities for QoS. Therefore, the second part of the thesis concentrates on the development of Mobility-Aware MPLS strategy for mobility with capabilities of providing continuous QoS and fast handover performance within the IP based mobile access MPLS domain. Scalable LSP configuration approach is used within the administrative domain to obtain more efficient use of wireless bandwidth. The thesis also provides the reliable handoff scheme for implementing the proposed mobility aware MPLS strategy in the mobile access network.

Provision of end-to-end QoS is one of the major challenges to Mobile Internet environment. The implementation of Multi-Protocol Label Switching (MPLS) enables faster switching, QoS guarantees and traffic engineering capabilities has led to more efficient technologies for QoS service mapping. Furthermore, Mobile MPLS scheme [10] improves the scalability of the mobile IP data forwarding because in the backbone MPLS network, the whole data forwarding process is done at the MPLS layer. With the modifications and extensions to MPLS including QoS mapping, prioritized packet scheduling, pre-established all the CR-LSP between MPLS end-points will enable to the provision of end-to-end QoS in mobile environment.

The final part of the thesis first highlights the framework combining the DiffServ (DS) ideas and modified mobile enabled MPLS in order to provide end-to-end QoS delivery between correspondent node of mobile user and mobile node. Afterwards, the thesis provides proposed QoS mapping, queuing model developing and the performance analysis for heterogeneous traffic with multi-class QoS over mobile enabled MPLS.

1.2 Objectives

The first focus of this research is to study two important QoS service models for current IP networks from comparative point of view including the investigation of the effective solutions for performance guarantee of scalable QoS over broadband Internet. A novel solution namely- aggregate base resource reservation scheme coupling with strong service model IntServ Architecture (ISA) with the major aim of providing the scalable quality of services for real time traffic flows in high speed networks is investigated. The problem of aggregate resource reservation which plays an important issue for performance analysis over broadband Internet is addressed and analyzed the scenarios under which aggregation can provide high performance approximation to the baseline Integrated Service Architecture (ISA).

The second focus of this research is to investigate the effective routing mechanisms and algorithm choices for granularity of QoS routing in the MPLS network describing the desirable feature of MPLS for supporting traffic engineering and QoS provisioning in IP-based mobile access networks using Mobility-Aware MPLS scheme. Based on the study of MPLS as a versatile solution to scalability, QoS management and traffic engineering discussed above, the thesis proposes a novel technique to implement the label swapping algorithm and traffic aggregation mechanism of MPLS for scalable end-to-end QoS behaviors.

In addition, the second major focus of the thesis is to design a mobility management framework that provides efficient mobility management integrating with mobile access MPLS networks providing continuous QoS support and handover performance. The aim is to develop a new strategy called "Mobility-Aware MPLS" by combining the advantage of MPLS such as IP QoS support with the advantage of host-based mobility such as low-latency handoffs in order that the proposed strategy can be applied within the mobile

access MPLS domain for a mobility solution. Moreover, the reliable handoff scheme which is an extremely important system features for mobile environment to implement with Mobility-Aware MPLS strategy is provided.

As the third focus of this thesis, the architectural model is explored for providing end-to-end QoS in MPLS based mobile environment including the desirable QoS mapping of IntServ (IS), DiffServ (DS), and Mobile MPLS network that can be implemented to facilitate interoperation of IS, DS capable mobile network over Mobile MPLS as a transit network. A four-queue system is proposed for packet scheduling which can be used at the output port of every modified label switched router(LSR) of Mobile MPLS in order to provide required QoS of the heterogeneous network and deriving analytically the performance of the proposed queuing system in terms of throughput, packet loss probability, and delay.

1.3 Major Contributions of the Thesis

(1) A novel technique namely- aggregate base resource reservation scheme combining with ISA is proposed. The proposed solution does not require per flow management at core router but can implement guaranteed services that can be provided with per flow mechanism. In this way, both required QoS and scalability can simultaneously be achieved. This is implemented by the combination of model developing, performance analysis, and computer simulation using MPEG video trace. A simplified network model for the proposed system has been built for consideration of performance of aggregation and developing a single-time scale demand model in which aggregate reservation is described by sinusoidal with random phase and additional white uniform noise.

The performance analysis of aggregate reservation is evaluated by implementing numerical experiments on three performance metric: overload probability, reserved resource utilization and the normalized available bandwidth. The numerical results indicate that system performance rapidly enhance as the control time scale τ is decreased from demand time scale T and the performance of aggregation can be degraded because of sufficient noise in the demand. The ability of the model base technique for performance of aggregation has been evaluated using the MPEG video trace for simulated experiments. The numerical and simulation experiments show that if the control time scale and demand time scale are separated by two orders of magnitude and variance of traffic is moderate, aggregation can provide performance close to IntServ.

(2) Mobility-Aware mobile access MPLS is presented as a desirable mobility solution. An MPLS architecture is considered for mobile access networks, since MPLS is a viable technology that will support a range of IP-based services with QoS. The proposed strategy provides efficient mobility in access networks while taking advantage of MPLS for IP QoS support. It is a scalable solution in supporting large number of mobile users since each LSP can be shared by many mobile nodes. It provides simple and fast handoff by updating the gateway (GW's) forwarding table while supporting the same QoS after handoff. Performance is evaluated by simulation experiments which are conducted using the same mobility model as the proposed LSP scheme. The evaluation has been devoted on network mobility in terms of new arrival call blocking probability and handoff call blocking probability. The simulation results clearly indicate the advantage that the proposed LSP scheme offers lower handoff call blocking probability (i.e., significantly improves probability of successful handover) under different traffic load scenarios.

A reliable handoff scheme is provided in order to satisfy the reliability and the efficiency of the handover process. Five important functional tasks are developed to implement the proposed scheme. The scheme uses quality parameter such as received signal strength (RSS) measurements of different BS at mobile node (MN) as the handover initiation

criterion due to the fact that signal level gives a direct indication of the actual energy collected by the MN. Resource sharing mechanism by each BS in the proposed scheme to provide priorities to handoff requests without significantly jeopardizing the new connection requests is demonstrated. The proposed novel scheme adopts Mobile Controlled Hand Off (MCHO) strategy with forward handover protocol. In MCHO strategy, handoff decision is made by MN and hence, highest degree of handoff decentralization is achieved. Fast handover is an important issue because, if handover does not occur quickly, QoS may degenerate below an acceptable level. By using MCHO, handoff time is approximately 100ms. Hence, MCHO would speed up the handover process and appropriate for microcells. The decision for handover should be as reliable as possible in order that unnecessary handovers to be prevented. In cellular system, where the existing communication link is likely to degrade quickly, thus forward handover protocol (handover initiation via new channel of target BS) could be more reliable in some microcell handover situations. Completion of handoff process includes changing to reserved MPLS label and pre-allocated LSP.

(3) An architectural model is provided to deliver end-to-end QoS for MPLS based mobile networks. Furthermore, a new QoS service mapping among IntServ (IS), DiffServ (DS) and MPLS based mobile network is also proposed. Therefore, the thesis focuses how incoming QoS service is mapped directly to the MPLS equivalent QoS service. Incorporating the MPLS header field including loss priority scheme and delay priority scheme is proposed to determine the QoS for heterogeneous traffic. In the proposed approach, modified LSR of Mobile MPLS is modeled as a tandem queuing system in which each output link consists of four parallel priority output queues representing different delay priorities for buffer management of different traffic classes to meet different QoS requirement. Note that the mobility is also included in the queuing model.

The Performance analysis of traffic with DiffServ is evaluated using multiplexed ON-OFF traffic model, while the analysis of IntServ (IS) traffic takes into account the per-flow property of the IS traffic. A number of key parameters are measured to determine the QoS performance in the proposed queuing model: They are throughput, packet loss probability, and delay. Performance analysis of the fluid model is established in order to analyze the proposed queuing model. The performance of high priority traffic is able to enhance by using priority queuing scheme. Numerical results demonstrate that the proposed queuing model is beneficial for provision of scalable QoS with multiple priority classes over MPLS based mobile network.

1.4 Organization of the Thesis

The rest of the thesis is organized as follows.

Chapter 2 describes the architectural issues for QoS in currents IP networks. The effective solutions for provision of scalable QoS over broadband Internet are presented.

Chapter 3 focuses the novel solution namely aggregate base resource reservation scheme coupling with IntServ architecture to provide scalable QoS over broadband Internet. The thesis first defines the proposed system and constructs the traffic demand models. Then, the impact of signaling, demand time scale, and demand variance on the performance tradeoffs of aggregate reservation are analyzed by developing the analytical method. The numerical experiments are performed to evaluate the performance metrics namely overload probability, reserved resource utilization and the normalized available bandwidth which plays important role for evaluating the effectiveness of aggregate base resource reservation in broadband Internet. Moreover, numerical experiments are carried out to investigate the performance impacts of periodic primary demand and additional secondary demand. Trace driven simulation experiments are implemented in order to evaluate the performance tradeoffs of aggregation.

Chapter 4 gives mechanisms, operations of MPLS and provision of QoS using traffic engineering techniques. QoS routing with MPLS is also discussed. For scalable QoS provisioning in core MPLS networks, a novel method called label swapping and traffic aggregation mechanisms of MPLS is provided and demonstrated by example network scenarios.

In Chapter 5, a new strategy called Mobility-Aware mobile access MPLS is developed for mobility solution with continuous QoS support in the mobility domain and fast handover performance. Framework, network architecture using MPLS and protocol description for mobility on MPLS based mobile network are presented together with network configurations and example forwarding tables. The simulation experiments are performed to evaluate the performance metrics namely new arrival call blocking probability and handoff call blocking probability which are the most important technical terms for performance evaluating in mobile networks. Trends of handover in mobile systems are discussed and reliable handoff scheme is elaborated with five functional tasks.

Chapter 6 first describes the framework combining the DiffServ (DS) ideas and Mobile MPLS in order to provide end-to-end QoS delivery between correspondent node (CN) of mobile user and mobile node (MN) and emphasizes on how to enhance QoS at IP packet level in MPLS based mobile networks. The multi-class QoS is implemented on different service basic using priority scheme in the MPLS packet header. An mobile enabled MPLS router is modeled as a tandem queuing system each output port consists of four space – priority output queues for delay priority control. Multiple thresholds are implemented in two output queues; DS high priority queue and DS low priority queue for packet loss priority control. The performance analysis is presented for heterogeneous traffic (e.g., signaling, voice, video and data) with multi-class QoS .

Conclusion of the thesis and suggestions for future research are presented in Chapter 7.

CHAPTER 2 COMPARATIVE STUDY OF DIFFERENT SCALABLE QOS OVER BROADBAND INTERNET

2.1 Introduction

Quality of Service (QoS) becomes one of the most important issues for multimedia networking which is perhaps the most exciting development in the Internet today. Several recent studies have focused on QoS as the Internet evolves into a global communication infrastructure. However, some Internet design principles discourage the use of the Internet as a universal communication platform. The difficulty of QoS guarantee becomes one of the typical problems over Internet. Since all Internet traffic shares the available resources based on forwarding in best-effort manner, it is impossible for Internet service providers (ISPs) to offer the service guarantee that some corporate communication applications (e.g., IP telephony, videoconferencing, stock information services) requirements. It is a growing need to support more sophisticated services such as the traffic management and QoS.

IP-based Internets have been able to provide a simple best-effort delivery service to all applications that they carry. Although the IPv4 header is associated with TOS (type of service field) however this information has generally been ignored by routers, both in the selection of routes and the treatment of individual packets. New real-time, multimedia, and multicasting applications are demanding requirements. Thus, there is a strong need to be able to support a variety of traffic, with a variety of QoS requirements, within the TCP/IP architecture. On the other hand, the fundamental requirement is to add new functionality to routers and a means for requesting QoS based service from internets. To meet such requirement, the IETF is developing two architectures to provide QoS transport over IP-based internets: the Integrated Services (IntServ) and the Differentiated Services (DiffServ).

Current IP networks provide the best effort datagram delivery. Such a simple service model allows IP routers to be stateless. By this model, routers do not keep any other fine information about traffic. Examples of stateless solutions are Differentiated Service (Diffserv) [1] for QoS and Random Early Detection (RED) for congestion control [11]. The corresponding examples of stateful solutions are Fair Queuing [12] for congestion control and Integrated Service (IntServ) [13] for QoS. This chapter first mainly studies these two QoS service models IntServ and DiffServ describing their mechanisms, differences and performance guarantees that can be provided.

Moreover, it has been observed that the link scheduling discipline plays an important role in providing QoS guarantees since real-time communication services such as video conferencing, scientific visualization and medical imaging require stringent performance guarantees expressed in terms of delay, delay jitter, throughput, and packet loss rate. An important issue in providing guaranteed performance service is the choice of the packet service discipline at switching nodes. Packets belonging to various network flows are multiplexed together and queued for transmission at the output buffers associated with a link. The manner in which queued packets are selected for transmission on the link is known as the link scheduling discipline which is one of the most important disciplines implemented for high speed networks. Therefore, the second focus of this chapter investigates three effective solutions for performance guarantees.

2.2 Architectural Issues for QoS in Current IP Networks

2.2.1 IntServ (IS) Architecture

IntServ (Integrated Services) Internet architecture is a framework which has been developed by the IETF to provide individualized quality of service guarantees to individual application sessions. A number of vendors have implemented portions of the ISA in routers and end-system software [14]. Two key features lie at the heart of Intserv architecture:

- *Reserved resources* . A router is required to know what amounts of its resources (buffers, link bandwidth) are already reserved for on going sessions.
- *Call Setup* . A session requiring QoS guarantees must first be able to reserve sufficient resources at each network router on its source to destination path to ensure that its end-to-end QoS requirement is met. This call setup (also known as call admission) process requires the participation of each router on the path.

Each router must determine the local resources required by the session while it must also consider the amount of its resources that are already committed to other ongoing sessions, and then it determines whether it has sufficient resources to satisfy the per-hop QoS requirement of the session at this router without violating local QoS guarantees made to an already admitted session.

Figure 2.1 illustrates the call setup process. The steps involved in call admission are considered as follows:

1. *Traffic characterization and specification of the desired QoS*. In order for a router to determine whether or not its resources are sufficient to meet the QoS requirements of a session, that session must first declare its QoS requirement as well as characterizes the traffic that it will be sending into network. In the IntServ architecture, the so-called Rspec (R for reserved) defines the specific QoS being requested by a connection :

the so-called Tspec (T for traffic) characterizes the traffic the sender will be sending into the network, or the receiver will be receiving from the network.

2. Signaling for call setup.

A session's Tspec and Rspec must be carried to the routers at which resources are reserved for the session. In the Internet, the RSVP protocol is currently the signaling protocol of choice. The use of the RSVP resource reservation protocol with the IntServ architecture is described in [15].

3. Per-element call admission.

Once a router receives the Tspec and Rspec for a session requesting a QoS guarantee, it can determine whether it can admit the call. The call admission depends on the traffic specification, the requested type of service and the existing resource commitments that already made by the router to ongoing sessions.

Figure 2.1 The Call Setup Process for IntServ

From the service definition point of view, the IntServ architecture defines two major classes of service: the guaranteed QoS service and the controlled load service.

Guaranteed QoS service

The Guaranteed Service (GS) includes guaranteed delay and guaranteed bandwidth. GS provides firm (mathematically provable) bounds on end-to-end datagram queuing delay. The applications require a firm guarantee that packets will arrive at destination no later than specified time after the transmission from source. Some audio and video playback applications as well as hard real-time applications fall into this category. The service does not minimize delay jitter rather it controls the maximum queuing delay experienced in the network. Network elements supporting guaranteed service within the guarantee of bandwidth that produces a delay-bounded service with no packet loss due to queuing overflow for all conformant traffic. Several scheduling algorithms and implementations exist that approximate the fluid model like, Weighted Fair Queuing (WFQ), Jitter EDD and Virtual Clock. The guaranteed service is the most demanding service provided by ISA. Since the delay bound is firm, the delay has to be set at a large value to cover rare cases of long queuing delays.

Controlled Load Service

Guaranteed QoS service provides hard, mathematical provable guarantees on the delay of each individual packets in a flow. By contrast, controlled load service does not provide any hard guarantees, but instead it ensures that most of an application's packets pass through a seemingly uncongested Internet. If the networks are lightly loaded (i.e., not experiencing heavy load or congestion) applications receiving controlled load service may assume that a high percentage of their packets are able to arrive successfully at the destinations.

The controlled load service targets real time multimedia applications that have been developed for today's Internet. The study points out that these applications perform quite well when the network is lightly loaded, but rapidly degrade in performance as the network becomes more loaded.

In order for a network to provide QoS guarantees, there must be a signaling protocol which allows applications running in hosts to reserve resources in the Internet. RSVP [16] is such a signaling protocol for the Internet. The RSVP protocol allows applications to reserve bandwidth for their data flows. It is used by a host, on behalf of an application data flow, to request a specific amount of bandwidth from the network. RSVP is also used by the routers to forward bandwidth reservation requests. To implement RSVP, RSVP software must be present in the receivers, senders and routers. The two principal characteristics of RSVP include

- It provides reservations for bandwidth in multicast trees (unicast is handled as a degenerate case of multicast.)
- It is receiver – oriented, that is, the receiver of a data flow initiates and maintains the resource reservation used for that flow.

- *Scalability*

Per-flow resource reservation using RSVP implies the need for a router to process resource reservation and to maintain per-flow state for each flow passing through the router. With recent measurements suggesting that even for an OC-3 (155Mbps) speed link, approximately 256,000 source destination pairs might be seen in one minute in a backbone router, per-flow reservation processing represents a considerable overhead in large networks [17].

- Flexible service models

The IntServ framework provides for a small number of pre specified services classes. This particular set of service classes does not allow for more qualitative or relative definitions of service distinctions. These more qualitative definitions might better fit our intuitive notion of service distinction. For example, first class versus coach class in air travel.

2.2.2 DiffServ (DS) Architecture

On the other hand, the other architecture namely, the Diffserv has been proposed for providing QoS in the Internet. Here the goal is to focus on the key aspects of Diffserv in order to clarify the differences between the IntServ. Diffserv architecture consists of two sets of functional elements:

- Edge function : *Packet classification and traffic conditioning* .

At the incoming “edge” of the network, arriving packets are marked. The Differentiated Service (DS) field of the packet header is set to some value. For example, in Figure 2.2, packets being sent from H1 to H3 might be marked at R1, while packets being sent from H2 to H4 might be marked at R2. The mark that a packet receives identifies the class of traffic which it belongs. Different classes of traffic then receive different service within the core network. After being marked, a packet may then be immediately forwarded into the network, delayed for some time before being forwarded, or it may be discarded. Many factors can influence how a packet is to be marked, whether it is to be forwarded immediately, delayed, or dropped.

- Core function : *forwarding*.

When a DS-marked packet arrives at a Diffserv capable router, the packet is forwarded onto its next hop according to the so-called per-hop behavior associated with packet's class. The per-hop behavior influences how router's buffers and link bandwidth are shared among the competing classes of traffic.

A basic principle of Diffserv is that a router's per-hop behavior will be based only on packet markings, that is, the class of traffic to which a packet belongs. Thus, if packets being sent from H1 to H3 in Figure 2.2 receive the same marking as packets from H2 to H4, then the network routers treat these packets as an aggregate, without distinguishing whether the packets originated at H1 or H2. For example, as shown in Figure 2.2, R3 does not distinguish between packets from H1 and H2 when forwarding these packets on to R4. Thus, Diffserv architecture obviates the need to keep router state for individual source-destination pairs – an important consideration in meeting the scalability requirement.

Several key characteristics of DS contribute to its efficiency and ease of deployment: IP packets are labeled for differing QoS treatment using the existing IPv4 Type of Service field or IPv6 Traffic Class field. Thus, no change is required to IP. DS provides a built-in aggregation mechanism. All traffic with the same DS field is treated the same by the network service. This provides for good scaling to large networks and traffic loads. DS is implemented in individual routers by queuing and forwarding packets based on the DS field. Routers do not have to save state information on packet flows [3].

Figure 2.2 A Simple Diffserv Network

The architecture relies on a model in which traffic entering a DS network is classified and conditioned at the network boundary. Then the packet is assigned to a traffic aggregate that determines its network treatment. Within the network, packets are forwarded according to the PHB associated with the aggregate class. Per-flow state does not need to be maintained in the interior network nodes, which leads to increased scalability. Traffic conditioning consists of various operations performed on traffic to ensure that it conforms to the rules of a specified traffic conditioning agreement. PHB in the network is typically implemented by a combination of buffer management and packet scheduling techniques. To date, two per-hop behaviors have been proposed in the IETF; namely they are

(1) Assure forwarding (AF) and (2) Expedited forwarding (EF)

(1) Assure Forwarding (AF)

Packets are marked with drop precedence values according to the class to which they belong. In the event of congestion, DS nodes try to protect packets with lower drop precedence marking from being lost by dropping packets with higher drop precedence values with higher probability. There are no delay requirements associated with AF.

(2) Expedited Forwarding (EF)

EF is designed to look to the endpoints like a dedicated “virtual leased line“ at some requested rate. Often referred to as “premium“ service. EF service requires queue management mechanism that can ensure that packets from a traffic aggregate see virtually empty (or very short) queues in transit.

2.2.3 Technical Comparisons of IS and DS

After technical surveying the two important architectures IntServ and Diffserv, it is desirable to make the following comparisons. The highlights of the comparisons include important differences in both service definitions and implementation architectures, advantages and disadvantages.

Integrated Service Architecture (ISA) [stateful Network]

- Provides end to end guaranteed QoS
- Controlled load service on a per flow basis
- Each router : process per flow signaling messages
- Maintain : per flow data forwarding & QoS state on the control path
- Perform per flow classification, scheduling and buffer management
- performing per flow management inside the network affects both the network Scalability and robustness.
- more powerful service than DS
- more flexible service than DS
- less scalable than DS
- less robust than DS

Differentiated Service Architecture (DS) [Stateless Network]

- simple and easy to implement
- Provides : service differentiation among a small number of traffic classes
- provides a built in aggregation mechanism
- Distinguishes between edge and core routers
- Edge routers: process packets on per flow basis
- Core routers :
 - not maintain state information on packet flows
 - process packets on small number of PHB
 - data plane : much more scalable than ISA
- need to address the problem of admission control
- services :
 - lower flexibility than ISA
 - lower utilization than ISA
 - lower assurance level than Intserv services

2.2.4 Multi-Protocol Label Switching (MPLS) Architecture

MPLS [5, 18] is a technology which combines high-speed switching with Layer 3 network routing aimed to address issues related to scalability, QoS management, and traffic engineering. MPLS allows packet-based networks to provide a range of QoS service capabilities to IP flows, such as resource reservation mechanism. These advantages lead to MPLS to match the current capabilities for ATM networks. When these mechanisms are applied to broadband Internet, this will allow for QoS guaranteed broadband networks. Further information on how to provide the scalable QoS will be illustrated in Chapter 4, Chapter 5, and Chapter 6 respectively.

2.3 Efficient Solutions for Performance Guarantee

2.3.1 Dynamic Packet State (DPS)

In recent years researchers have become increasingly interested in providing QoS guarantees in an efficient way. I.Stoica and H. Zhang [19] have proposed techniques to provide guaranteed service without per flow management at core routers. Their techniques provided high scalability and robustness. They have proposed scheduling and admission control algorithms for that purpose. Their scheme shares its intellectual roots with two pieces of related work : Diffserv and the Core-Stateless Fair Queuing.

Figure 2.3 The Illustration of the Dynamic Packet State (DPS) Technique used to Implement a Scalable Core SCORE Network: (a-b) upon a packet arrival the ingress node inserts some state into the packet header ; (b-c) a core node processes the packet based on this state, and eventually updates both its internal state and the packet state before forwarding it. (c-d) the egress node removes the state from the packet header.

The idea of implementing QoS services by using a core-stateless architecture is first proposed by Clark [20] and Jacobson [21], and is now being pursued by the IETF DiffServ working group [3]. The key technique they use to implement approximation algorithms is dynamic packet state (DPS) which encodes states in the packet header. Ingress node inserts scheduling parameters in each packet's header and core routers make scheduling decisions based on the parameters encoded in packet headers.

Figure 2.3 demonstrates the concept of DPS. In scalable core network (SCORE) only edge nodes perform per flow management; core nodes do not perform per flow management. Thus, this technique eliminates the need for maintaining per flow state at core nodes. Only per flow state can be stored at the ingress node, inserted into packet header by ingress node and retrieved later by core nodes.

Several studies focus on different algorithms in connection with scheduling policies that can provide QoS networking architectures. Fair queuing algorithm [12, 22] in which the deadline of a packet can depend on state variable of all active flows, in Virtual clock algorithm the deadline of each packet can be computed based on the state variable of the flow it belongs to. Therefore, the need of per flow state is eliminated at all nodes along the path. Jitter Virtual Clock (Jitter-VC) is a non-work conserving version of the Virtual Clock algorithm [25]. It uses combination of delay jitter rate controller [23, 24] and a virtual clock scheduler.

The algorithm works as follows: Each packet is assigned an eligible time and a deadline upon its arrival ($e_{i,j}^k$ & $d_{i,j}^k$). The packet is held at rate controller until it becomes eligible. The scheduler then orders the transmission of eligible according to their deadlines.

For the k^{th} packet of flow i , its eligible time $e_{i,j}^k$ and deadline $d_{i,j}^k$ at the j^{th} node on its path are computed as follows:

$$e_{i,j}^1 = a_{i,j}^1$$

$$e_{i,j}^k = \max(a_{i,j}^k + g_{i,j-1}^k, d_{i,j}^{k-1}), \quad i, j \geq 1, k \quad (1)$$

$$d_{i,j}^k = e_{i,j}^k + \frac{l_i^k}{r_i}, \quad i, j, k \geq 1 \quad (2)$$

where l_i^k is the length of the packet, r_i is the reserved rate for the flow, $a_{i,j}^k$ is the packet arrival time at the j^{th} node and $g_{i,j-1}^k$ stamped in to the packet header by the previous node, is the amount of time the packet is transmitted before its deadline, i.e., the difference between the packet's deadline and its actual departure time at the node before j node.

The purpose of having $g_{i,j-1}^k$ is to compensate at node j the variation of delay due to load fluctuation at the previous node $j-1$. Such regulations limit the traffic burstiness caused by network load fluctuations, and as a consequence, reduce both buffer space requirements and the scheduler complexity. Moreover, the eligible time and the deadline of p_i^k (k^{th} packet of flow i) depend on the deadline of the previous packet of the same flow, i.e., $d_{i,j}^{k-1}$.

It has been shown that if a flow's long term arrival rate is no greater than its reserved rate, a network of Virtual Clock servers can provide the same delay guarantee to the flow as a network of WFQ servers [26, 27, 28]. In addition, the study shows that that a network of Jitter-VC servers can provide the same delay guarantee as a network of Virtual Clock servers [29]. Therefore, a network of Jitter-VC servers can provide the same guaranteed service as network of WFQ servers.

I. Stoica and H. Zhang [19] have proposed a variant of Jitter-VC and this variant of JVC uses the DPS technique. DPS technique needs to pre-compute the eligible times and the deadlines of the packet at all nodes along its path and insert all of them in the header. This would eliminate the need for core nodes to maintain the deadline of the packet. From the study, it is found that by using one additional DPS variable δ^k_i which is the slack delay of P^k_i (the k th packet of flow i .) the need for maintaining $d^k_{i,j}$ at core nodes can be eliminated.

However, the main disadvantage of this approach is that the amount of information carried by the packet increases with the number of hops along the path. In addition, the biggest problem with using the IP header is to find enough space to insert the extra information since protocol header has limited space and most header bits have been allocated. The main challenges are to find space in the packet header for storing dynamic packet state variables, to fully compatible with current standards and protocols, to efficiently encode state variables so that they fit in the variable space without introducing too much inaccuracy.

The algorithms that used DPS technique do not maintain per flow state at core nodes, there is still a need for core nodes to perform regulation and packet scheduling based on eligible times and dead lines. By implementing the algorithms in Ipv4 networks, they have demonstrated that it is possible to apply DPS techniques and have minimum incompatibility with existing protocols.

Major concepts of DPS are characterized as follows: The state embedded in a packet is highly dynamic, ingress router inserts scheduling parameter and core router makes the scheduling decision. No per flow management is required at core routers but it can implement guaranteed service. It achieves high QoS, high scalability and robustness.

Using the key technique DPS for QoS scheduling and admission control in a scalable core network where core routers do not maintain per flow state, it is possible to significantly enhance the scalability of both the data and control plane mechanisms for implementing guaranteed services and at the same time, provide guaranteed services with flexibility, utilization, and assurance levels similar to those that can be provided with per flow mechanisms.

According to the technical survey, it has been observed that there are several scalability bottlenecks for solutions requiring per flow management. On the data path, the expensive operations are per flow classification and scheduling. On the control path, the complexity is the maintenance of consistent and dynamic state in a distributed environment. Among the three, it is easiest to reduce the complexity of the scheduling algorithm as there is a natural tradeoff between the complexity and the flexibility of the scheduler [30]. In fact, a number of techniques have already been proposed to reduce the scheduling complexity, including those requiring constant time complexity [31, 32, 33]. It has been proven that traffic regulating also reduces the scheduling complexity and in addition reduces the buffer space requirement [19].

2.3.2 Rate-Controlled Service Discipline (RCSD)

Various scheduling policies have been suggested in the literature for providing per connection end-to-end delay and throughput guarantees in high speed networks. Among them, policies based on fair queuing also known as Generalized Processor Sharing (GPS) [12], [34], [35] have attracted special attention since they guarantee throughput to individual connections and provide smaller end-to-end delay bounds than other policies for connections that cross several nodes.

GPS [34], [36] (also known as Weighted Fair Queuing (WFQ) [12]) and Earliest Deadline First (EDF) [23], [38] have emerged as among the most popular packet scheduling schemes for the provision of quality of service (QoS) guarantees to real time communication services in emerging broadband packet switched networks. It is generally accepted that real time services such as voice and video typically require some form of performance predictability in terms of end-to-end transfer delays.

GPS is an idealized fluid discipline with a number of very desirable properties such as the provision of minimum service guarantees to each flow and fair resource sharing among the flows. Due to its powerful properties, GPS has become the reference for an entire class of GPS related packet scheduling disciplines and relatively low cost implementations have started reaching the market. Nevertheless, it has been shown that the tight coupling between rate and delay under GPS in the deterministic setting leads to sub-optimal performance and reduced network utilizations [29].

EDF has long been known in the context of processor scheduling and has more recently been applied to broadband packet switches. In a single node setting, EDF is known to be the optimal scheduling policy [39] in terms of the schedulable region for a set of flows with given deterministic delay requirements. EDF scheduling coupling with per hop traffic shaping (together referred to as Rate Controlled EDF or RC-EDF) permits the provision of end-to-end delay guarantees [24] and work in [29] has shown that in the deterministic setting, RC –EDF can offer performance gains over GPS.

For this reason the analysis concentrates on rate-controlled service (RCS) disciplines, which have been proposed in the literature [24] to provide performance guarantees to individual connections. In that work it is assumed that connections whose traffic satisfies certain burstiness constraints enter the network at various nodes. A node can have several output links, each of which contains a scheduler that decides the order in which packets are transmitted .

At each node along the path of connection, traffic is reshaped to conform to its original envelope before it enters the appropriate scheduler. Based on the traffic envelope of the connection, upper bounds on the scheduling delays at each node can be guaranteed .

It may be beneficial for us to pay our attention on technical survey to RCS disciplines since it offers a powerful solution to provide end-to-end delay and throughput guarantees in high speed networks and it exhibits a number of important desirable properties regarding scheduling policies.

Figure 2.4 Connection Traffic Flow

Considering a network comprised of store and forward packet switches. In that switch a packet scheduler is available at each output link . Traffic from a particular connection entering the switch passes through a packetizer and traffic shaper before reaching the scheduler as shown in Figure 2.4. Packetizer reassembles the packets and traffic shaper regulates traffic (these packets) so that the output of the shaper satisfies certain pre specified traffic characteristics before reaching the link scheduler which arbitrates the transmission of packets on the link. Deterministic approach is used to specify the traffic characteristics of a connection. Modeling traffic as a fluid, $U [t, t+\tau]$ is used to denote the amount of traffic arriving at the network ingress in the interval $[t, t+\tau]$. It is assumed that nonnegative function $\bar{U}(\tau)$ is the envelope of $U [0, t]$. Also assuming that $\bar{U}(\tau)$ is right continuous, non decreasing and sub additive.

The packetizer spits out packets of maximum length L , instantaneously delivered to the shaper when the last bit of the packet is received. $I[t, t+\tau]$ is denoted the traffic at the output of the packetizer in the interval $[t, t+\tau]$. For any nonnegative t and τ

$$I[t, t+\tau] \leq U[t, t+\tau] + L \leq \bar{U}(\tau) + L =: \bar{I}(\tau) \quad (3)$$

Hence, $\bar{I}(\tau)$ is an envelope of the traffic input to the traffic shaper

Traffic shapers reshapes the incoming traffic by delaying the packets according to the rules and deliver them to the scheduler. The traffic shaper is characterized by traffic envelope $\bar{A}(\tau)$ which provides an upper bound on the amount of traffic that is output by the shaper in any interval of length τ . If $A[t, t+\tau]$ designates the traffic output from the shaper in the interval $[t, t+\tau]$ then the shaper ensures that $A[t, t+\tau] \leq \bar{A}(\tau)$.

Traffic shapers in this approach can be constructed from the simple (σ, ρ) traffic regulators. Operation of the (σ, ρ) traffic shaper can be described in terms of the backlog in a hypothetical queue. Assuming traffic $I[0, t]$ is fed to the queue that is served at the rate of ρ . Backlog in this queue at time t is given by

$$W_{\rho}(I)(t) := \max_{0 \leq s < t} \{ I[s, t] - \rho(t-s) \} \quad (4)$$

Operation of the (σ, ρ) traffic shaper can be described in terms of the backlog. The i th packet arriving at time s_i is released at the earliest time $f_i \geq s_i$ such that the shaper output traffic, $A[0, f_i]$, satisfies the condition

$$W_{\rho}(A)(f_i) \leq \sigma = L + \delta, \quad \delta \geq 0$$

The condition $\delta \geq 0$ is necessary in order to allow packets of size L to pass through the shaper. This shaper corresponds to the operation of a leaky bucket in a store and forward network.

$$A[t, t+\tau] \leq \sigma + \rho \tau$$

(σ, ρ) traffic shaper has also been described in the literature in terms of a token bucket (or a leaky bucket), with ρ being the rate of token accumulation and σ being the bucket depth.

Delay in the Traffic Shaper

Using shapers whose output is a concave, increasing, piecewise linear function with finite number of slopes (say K). These types of shapers are interesting since they are a generalization of the shapers adopted by the Internet and Asynchronous Transfer Mode (ATM) standards. It is assumed that passing the traffic through a series of K (σ_m, ρ_m) shapers, $m = 1, 2, \dots, K$. For the fluid traffic model described earlier as an input and A is a shaper with an envelope

$$\bar{A}(\tau) := \min_{m=1,2,\dots,K} \{ \sigma_m + \rho_m \tau \}$$

An upper bound on the delay can be developed by traffic with envelope $\bar{I}(\tau)$ through shaper A that has envelope $\bar{A}(\tau)$. This is denoted by $D(\bar{I} \parallel \bar{A})$

Figure 2.5 Graphical Representation of $D(\bar{I} \parallel \bar{A})$

An upper bound on the delay encountered in the shaper A by traffic with envelope $\bar{I}(\tau)$ is developed in [29]

$$D(\bar{I} \parallel \bar{A}) := \max_{\tau \geq 0} \left\{ \left(\max_{m=1,2,..K} \left\{ \frac{\bar{I}(\tau) - \sigma_m}{\rho_m} \right\} - \tau \right)^+ \right\}(5)$$

Figure 2.5 shows graphical illustration of the maximum horizontal distance between the envelopes $\bar{I}(\tau)$ and $\bar{A}(\tau)$. When the traffic entering shaper A2 with envelope A2 (τ) is the output of a shaper A1 with envelope A1 (τ), the bound on the delay in shaper A2 is denoted by any of the following.

$$D (A1 \parallel A2) \equiv D(\bar{A1} \parallel \bar{A2})$$

Researchers are interested in a generalized form of the class of RCS disciplines. Considering the scenario that the traffic of connection n having the envelope function $\bar{U}_n(\tau)$, enters the network. At node m , traffic of connection n is shaped by a traffic shaper A_n^m . Different traffic shapers can be used for the same connection at different nodes. Traffic exiting from shaper A_n^m enters a scheduler at the appropriate output link at node m . The traffic is scheduled for transmission to the next node or to its destination.

End-to-end delay bounds can be developed based on the scheduling policies at each node as well as the form of traffic shapers A_n^m . These bounds are used to provide delay guarantee to each connection. In this approach, service discipline refers to the operation of the system consisting of the traffic shaper as well as the scheduler.

(D_n^m) scheduling delay includes both queuing delay and transmission time of packet. Assuming that an upper bound on the propagation delay for link L is T^l . In order to provide bounds on the end-to-end packets delay, it is also necessary to consider any additional delay occurred in the traffic shapers.

Figure 2.6 The System S

The delay that a packet from connection n experiences between the time it exits shaper A_n^m and the time it exits A_n^{m+1} is bounded above by

$$D_n^{(m,m+1)} = D_n^m + T^{lm} + D(A_n^m \parallel A_n^{m+1}) \dots\dots\dots (6)$$

Delays in the traffic shaper

Considering RCS disciplines for any given connection if identical shapers are used at each node, i.e., $A_n^m = A_n$. Then, end-to-end delay guarantee for connection n becomes [29]

$$\overline{D}_n = D(\overline{I}_n \parallel A_n) + \sum_{m=1}^M D_n^m + \sum_{m=1}^M T^{lm} \dots\dots\dots (7)$$

The work in [29] shows that for the traffic shapers considered there, reshaping the traffic to its original envelope does not introduce extra delays. Therefore, an upper bound on the end-to-end packet delay is obtained as the sum of the scheduling and propagation delays. In this service disciplines, the traffic of each connection is reshaped at every node. Reshaping makes the traffic at each node more predictable and hence, simplifies the task of guaranteeing performance to individual connections.

Achieving Rate Proportional Processor Sharing (RPPS) Delay Guarantees

RPPS is a simpler version of GPS. It is assumed for comparison purposes that the traffic of connection n , entering the first node packetizer has envelope

$$\overline{U}_n(\pi) = \delta_n + \rho_n \tau$$

Therefore, the envelope of traffic that enters the first traffic shaper is

$$\overline{I}_n = L + \delta_n + \rho_n \tau$$

Assuming that connection n traverses nodes $1, 2, \dots, M$ and all the propagation delays are zero. Assuming the stability condition as

$$\sum_{n \in C^{m,l}} \rho_n \leq r^{m,l},$$

where $C^{m,l}$ is the set of connections that pass through the output link l of node m and $r^{m,l}$ denotes as the speed of link.

The end-to-end delay bound $\overline{D}_n^*$, obtained under nonpreemptive RPPS is given by [13],[19]

$$\overline{D}_n^* = \frac{\delta_n + ML}{\rho_n} + \sum_{m=1}^M \frac{L}{r^m} \quad (8)$$

From (8), it can be seen the weakness of RCS disciplines relative to RPPS, if the traffic shapers for connection n at every node have envelopes identical to the input envelope $\bar{I}_n$ (τ). In this case $D(\bar{I}_n \| A_n) = 0$. Since propagation delay are assumed to be zero, from (7), it obtains

$$\overline{D}_n = \sum_{m=1}^M D_n^m$$

The scheduler delay bound, D_n^m , can at best be, $(\delta_n + L)/r^m$. Thus, end-to-end delay bound guaranteed by the RCS discipline satisfies the following inequality.

$$\overline{D}_n \geq \sum_{m=1}^M \frac{\delta_n}{r^m} + \sum_{m=1}^M \frac{L}{r^m}$$

since δ_n can be much larger than L , the bounds provided by the RCS discipline under the scenario considered here can be much worse than those obtained under RPPS. RCS disciplines can be designed to provide the same delay guarantees as RPPS by employing traffic shapers with envelopes that are different from that of input traffic.

Design Consideration of RCS Discipline π

For each link, NPEDF scheduling policy is used. Traffic shaper A_n is chosen for connection n at each node along its path with its envelope being

$$\bar{A}_n(\tau) = L + \rho_n \tau$$

Assuming the connection n is routed through output link L at node m . Let the speed of this link is denoted by γ^m . For connection n , the delay bounds for the NPEDF scheduling policy, at node m is specified by

$$D_n^m = L / \rho_n + L / \gamma^m \dots\dots\dots(9)$$

Assuming zero propagation delays, from equation (7) and we obtain

$$\bar{D}_n = D(\bar{I}_n \parallel A_n) + \sum_{m=1}^M D_n^m$$

For the delay $D(\bar{I}_n \parallel A_n)$, using (5), we have

$$D(\bar{I}_n \parallel A_n) = \max_{\tau \geq 0} \left\{ \frac{\bar{I}_n(\tau) - L - \rho_n \tau}{\rho_n} \right\} = \frac{\delta_n}{\rho_n}$$

Therefore, taking into account (9) the end-to-end delay bound for the connections is obtained as

$$\begin{aligned} \bar{D}_n &= \frac{\delta_n}{\rho_n} + \sum_{m=1}^M \frac{L}{\rho_n} + \sum_{m=1}^M \frac{L}{r^m} \\ &= \frac{\delta_n + ML}{\rho_n} + \sum_{m=1}^M \frac{L}{r^m} \dots\dots\dots(10) \end{aligned}$$

Since (8) is identical to (10), the RCS discipline π can guarantee the same end-to-end delays as RPPS. If the delay bounds in (8) are required by the connections in the network, RCS discipline π above can be used.

Benefits of RCS discipline π

RCS discipline provides the flexibility of easily specifying other delay bounds, whereas the bounds in RPPS are tied to the rate ρ_n of a connection. Since reshaping is performed at each node, buffer requirements will be lower than those of RPPS, and its implementation may be simpler.

If the end-to-end delay requirements of connection n are smaller than (8), a more general version of RPPS can be used. Rather than providing a rate of ρ_n to connection n , better delay performance can be obtained by giving it a rate of $g_n \geq \rho_n$ at each node. The end-to-end delay bound is given by

$$\overline{D_n^*} = \frac{\delta_n + ML}{g_n} + \sum_{m=1}^M \frac{L}{r^m} \quad (11)$$

RCS discipline can guarantee the bounds in (11). For an RCS discipline it is beneficial to choose the “smallest” shaper at all the nodes, so that they can all take advantages of the smaller traffic envelope. In addition to being able to provide the same bound as RPPS, the RCS discipline also has the advantage of allowing additional connections to be accepted with looser delay requirements.

Achieving GPS Delay Guarantees

The delay guarantees provided by a pure GPS policy can also be achieved by an RCS discipline working with worst case delay at each node, where the scheduling policy at each node is GPS. If GPS is replaced with the (simpler) EDF scheduling policy at each node, it is not assured that it can still guarantee the GPS end-to-end delay bounds, but it also creates a service discipline that is better than GPS. Proper selection of the traffic shapers allow to construct an RCS discipline that outperforms GPS.

Here some interesting properties of traffic shapers are discussed in the context of RCS disciplines. It has been observed that considering RCS discipline for connection n uses shaper A at all nodes with envelope $\bar{A}_n(d)(\tau)$, where $d=D(\bar{I}_n \parallel A)$, can guarantee the same end-to-end delays to all connections [29].

As a result, to specify an RCS discipline, one has to determine the single parameter d as well as the scheduler delays, D_n^m for each node m along the path of connection n . The determination of these parameters is an interesting design problem. The use of traffic shapers at each hop can introduce extra delays for the traffic of connection n , even if there is no congestion in the network. This causes reduction of jitter and buffer requirement at each node in the network where the resulting increase in the average delay is undesirable.

However, using some simple modification to the RCS discipline that make it work conserving will reduce the average end-to-end delays. The study focuses on the underlying concepts regarding this modification from the work of [29]. In RCS disciplines [24] link schedulers are assumed to be work conserving but if the traffic shaper and the scheduler are considered as a single system this system is not work conserving since there may be instants in time when the link is idle even though there are packets waiting in the shaper. Modification to the system as work conserving will provide the same delay guarantees to all connections. Hence, output link will no longer be idle when there are packets in the system. Consequently, this will reduce the average end-to-end delay seen by the packets.

Using the model of [37] to represent both the shaper and the scheduler at an output link of node m . Instead of holding up the packets in the shaper, two queues are maintained in the system. One is denoted as Q_e^m which is a queue of packets that are eligible for scheduling i.e. have been reshaped and the other is designated as Q_r^m which is a queue of not yet eligible packets. Eligibility is determined by the shaper which stamps an eligibility time on the packets and enqueues them in Q_r^m .

The eligibility time is the earliest time the packets could have left the shaper to be in conformance with the pre-specified traffic envelope. Delay of the packet in the scheduler is calculated with respect to its eligible time. When a packet in Q_r^m becomes eligible for scheduling, it is promoted to Q_e^m . The scheduler in the nonwork conserving discipline, π_{NW} , selects packets in Q_e^m for transmission. After transmission, packet is removed from Q_e^m and the scheduler repeats the above process. π_{NW} is non work conserving since there can be packets queued in Q_r^m that are not considered for transmission by the scheduler, even though the link may be idle.

It is important to point out the knowledge of modifying the scheduler in π_{NW} for developing work conserving discipline π_W . In this approach, using nonpreemptive priority mechanism that arbitrates the service of Q_e^m and Q_r^m , with Q_e^m higher priority is considered. When Q_e^m is empty ineligible packets from Q_r^m are selected for transmission (in any order). When packets in Q_e^m same scheduling discipline as that used in π_{NW} is used to select packets for transmission.

Let S^m is the system consisting of the scheduler at node m . Figure 2.7 shows the modified system which is the same as the work conserving system operating under π_W except for a delay system inserted between S^m and shaper A_n^{m+1} . The work in [29] has shown that the end-to-end delay guarantees are not affected that is the guarantee upper bound on the end-to-end delays will not increase when the service discipline at each node is modified to be work conserving. Thus, if $\bar{D}_n$ is the end-to-end delay guarantee for connection n and suppose the nodes traversed by connection n be numbered $1, \dots, M$, then we have

$$\bar{D}_n \leq D(\bar{I}_n \| A_n^1) + \sum_{m=1}^{M-1} D(A_n^m \| A_n^{m+1}) + \sum_{m=1}^M D_n^m + \sum_{m=1}^M T_n^{l_m} \dots\dots\dots (12)$$

Figure 2.7 Original (work conserving) System and the Modified System

From the analysis it has been observed that the main advantages of an RCS discipline compared to GPS are offer flexibility (possible to use different schedulers and shapers at different nodes), lower buffer requirements at intermediate nodes and simpler implementation. The main disadvantage of RCS discipline is that end -to- end delay guarantees are obtained as the sum of the worse case delay at each node. This problem can be overcome by making “proper“ reshaping of the source traffic and by using” proper” selection of the traffic shapers .

The study has pointed out that for RCS discipline “good” shapers are required in order to outperform GPS. It has been proved by [20] that RCS disciplines can be designed so as to outperform GPS –based one even in a network environment. This is achieved by proper selection of the traffic reshaping performed at each node and used a simple algorithm to determine how to reshape the traffic and then specify worst-case delay bounds at each node.

The results have shown the following important properties:

- i. Any end-to-end delay bounds that can be guaranteed by the GPS discipline can also be achieved by an RCS discipline.
- ii. The sum of the worst-case delay bounds of this RCS discipline is then no longer than the delay guarantees provided by the GPS discipline.
- iii. Because of traffic reshaping, the network buffer requirements of RCS disciplines are significantly smaller than those of the GPS discipline.

Based on advantages and their potential implementation simplicity, the study evaluates that RCS disciplines are very effective solution for providing end-to-end performance guarantees to individual connections in broadband networks.

2.3.3 Aggregate Base Resource Reservation Scheme

Aggregate reservation is a mechanism that reserve network resources on behalf of multiple flows. IntServ has been designed as a flow base resource reservation scheme and provides QoS for each traffic flows. However, such kind of scheme needs to request reservation at core routers for each flow. This feature is not efficient in high speed networks, hence aggregation has been proposed for achieving scalability [1]. By using aggregation of resource reservation, the network can simultaneously achieve the profound benefits such as scalability, per –flow quality of service and high utilization.

As a result, a technique has been proposed in this thesis namely- aggregate base resource reservation scheme combining with Integrated Service Architecture (ISA) that does not require per flow management at core router but can implement guaranteed services that can be provided with per flow mechanism . In this approach, a flow signals its ingress router. The ingress router will make “ bulk” or aggregate resource reservation in the core router.

In addition, the ingress router may accept or reject the incoming flow requests base on the fact that whether there is sufficient available capacity in the bulk reservation. The ingress node will then periodically adjust the reservation of resources in the core node in accordance with its current demand. In this way, required QoS and high scalability can simultaneously be achieved. The proposed scheme will be analyzed in the following chapter.

2.4 Summary

The Integrated Services architecture (ISA) provides the Internet the ability of delivering end-to-end QoS to applications over heterogeneous networks. While ISA can provide good control granularity on QoS, the use of per-flow state and per-flow processing cause scalability concerns for large networks. The Differentiated Services (DS) proposes a scalable means to deliver IP QoS based on aggregate traffic handling. MPLS supports the label switching and resource reservation mechanism.

DPS technique can be implemented for scalable core network and provides performance guarantee on a per flow basis without per flow management at core routers. It provides a lightweight and robust means for routers to coordinate actions and implement distributed algorithms. Using the key technique DPS for QoS scheduling and admission control in a scalable core network, it is possible to significantly enhance the scalability for

implementing guaranteed services. The main challenges are to find space in the packet header for storing dynamic packet state (DPS) variables, to fully compatible with current standards and protocols, to efficiently encode state variables. As a final note, it has been observed that DPS is a powerful concept.

With a suitable choice of shaper envelopes the RCS discipline can provide the same end-to-end delay guarantees that the best delay bound for GPS can provide. The investigation evaluates that using RCS discipline can offer a powerful solution to provide end-to-end performance guarantees to individual connections in broadband networks.

Aggregate reservation scheme is beneficial for high speed networks since it provides scalability and high utilization. Based on such QoS mechanisms, customers are able to achieve guaranteed service quality. Therefore, the next chapter of this thesis will focus the performance analysis and evaluation of a novel technique: aggregate base resource reservation scheme combining with Integrated Service Architecture (ISA).

CHAPTER 3 MODELING AND PERFORMANCE ANALYSIS OF SCALABLE QoS OVER BROADBAND INTERNET

3.1 Introduction

The increasing need for providing QoS in broadband network is becoming important everyday. Integrated Services (IntServ) provides QoS on the level of application flows and thus it is considered not to be scalable to large networks like the Internet. The scalability problem is mainly due to the large number of flows in the core of the network. This can be burdensome, especially for backbone routers connecting to high speed links and may carry hundreds of thousands of flows simultaneously. For such network, effective flow management becomes a major concern because per-flow management is very costly proposition and not scalable. Hence, one obvious solution to this problem is the aggregation of IntServ flows in the core of the network. By aggregating flows inside the network, only the network edge needs to be aware of all individual flows. Aggregate base resource reservation scheme is useful for providing the QoS of the network in a scalable way. By using aggregation of resource reservation, the network can simultaneously be achieved scalability and high utilization.

In this chapter, a novel flow aggregation technique namely aggregate resource reservation coupling with IntServ Architecture (ISA) has been proposed with the major goal of providing scalable QoS for real-time traffic flows and is evaluated its performance by analyzing its blocking probability, reserved resource utilization and normalized available bandwidth. Moreover, the thesis discusses the problem of the aggregate resource reservation and analyzes the aggregation that is able to provide high performance and scalability. The work has been implemented by the combination of model development, performance analysis, and computer simulation using MPEG video trace.

3.2 Aggregate Base Resource Reservation Scheme

Aggregate reservation is a mechanism that reserve network resources on behalf of multiple flows. IntServ has been designed as a flow base resource reservation scheme and provides QoS for each traffic flow. However, such kind of scheme needs to request reservation at core routers for each flow. This feature is not efficient in high speed networks, hence aggregation has been proposed for achieving scalability [1]. By using aggregation of resource reservation, the network can simultaneously achieve the profound benefits such as scalability, per-flow quality of service and high utilization.

Assumptions and Terminology

It is assumed that an explicitly definable region called an aggregating region, will aggregate and implement supporting mechanisms at its boundary points. An ingress router is defined as a router placed at the edge of an aggregating region where the flows enter the region and egress as a router placed at the edge of aggregating region where the flow for a specific session leaves the region. Also, an interior (core) router is defined for a session as any router in the aggregating region that is on the data path for the session and is not the ingress or egress.

Overview of Mechanism Description

The goal of a scalable solution is to minimize the overall reservation state in the network, as well as the state that any router needs to maintain. In addition to the state needed, a scalable solution should also take into account processing and signaling requirements, ensuring that both are kept as low as possible. A related factor is the bandwidth efficiency of a solution, and in particular how often the bandwidth allocated to an aggregate reservation is updated.

Therefore, a solution has been presented in this thesis namely- aggregate base resource reservation scheme combining with Integrated Service Architecture (IntServ) that does not require per flow management at core router but can implement guaranteed services that can be provided with per flow mechanism. In this way, both required QoS and high scalability can simultaneously be achieved.

In the proposed scheme, the aggregate reservation is characterized by a sinusoidal with random phase and additive white uniform noise. In this approach, a flow signals its ingress router. The ingress router will make “bulk” or aggregate resource reservation in the core router. In addition, the ingress router may accept or reject the incoming flow requests based on the fact that whether there is sufficient available capacity in the bulk reservation. The ingress node will then periodically adjust the reservation of resources in the core node in accordance with its current demand. Thus, in proposed approach, ingress routers perform aggregation and interior routers keep a greatly reduced amount of state.

Issues Related to Performance of Aggregation

Performance of aggregation mainly relies on the traffic characteristic of flows. If traffic characteristic is constant over time, core reservation will be static i.e. the close match between the reservation and the actual traffic. If aggregate demanded bandwidth varies quickly, aggregation may result poor performance. This can be solved by readjusting the bulk reservation at the core node. The main disadvantage of this solving technique is losing the advantage of scalability in the network because of frequent signaling to core node.

3.3 System Modeling

Aggregation Scenario and Model Developing

Figure 3.1 A Simplified Network Model

Figure 3.1 (a) shows the simplified network model developed for consideration in proposed scheme. In this model, flows (indexed by j) are multiplexed on to a class or link (indexed by i). Bandwidth requirement for flow j in link i is indicated by ρ_i^j . Assuming that the network has a single bottleneck link with capacity C and all other links has infinite capacity. For simplicity, delay needs are ignored and consideration is focused only on bandwidth. The key job of the admission control is to ensure that the network resources are not over committed. Particularly, in the Intserv architecture, for providing guaranteed service any set of flows are admitted such that $\sum_i \sum_j \rho_i^j < C$. Furthermore, when the total reserved rate of the traffic flows exceed the capacity of the bottleneck link C , flows will be rejected in the network.

By implementing the aggregate resource reservation scheme, individual traffic flow does not require to signal the core routers. In this approach, a flow signals its ingress router. The ingress router will make “bulk” or aggregate resource reservation in the core router. In addition, the ingress router may accept or reject the incoming flow requests base on the fact that whether there is sufficient available capacity in the bulk reservation. Alternatively, the local admission control is performed at the ingress router. The ingress node will then periodically adjust the reservation of resources in the core node in accordance with its current demand. Aggregate demand of link i (from single ingress node to core router) is $\sum_j \rho_i^j$ that is defined as $r_i(t)$. From mathematical point of view, $r_i(t)$ can be seen as a time varying function since the traffic arrivals and departures rate may vary from time to time. Similarly, the aggregate reservation at time t is designated by $\tilde{r}_i(t)$.

When a new flow arrives at the ingress router at rate ρ_i^* and it requests admission to pass the network, the ingress router makes consideration for local admission control as follows:

The ingress node determines whether the total current aggregate demand exceeds or not the aggregate reservation which has assumed a certain value in the core router formerly.

If the ingress node has a current aggregate reservation such that $\sum_j \rho_i^j + \rho_i^* < \tilde{r}_i(t)$,

then the flow is admitted. Otherwise, when current aggregate reservation $\tilde{r}_i(t)$ is insufficient on account of new flow requesting resource, the ingress node will signal the core node for aggregate request $\hat{r}_i$, at time t denoted by $\hat{r}_i(t)$. The requested increment to the core node by the ingress node ($\hat{r}_i(t) - \tilde{r}_i(t)$) is referred to as the bulk reservation. Generally, bulk reservation is larger than ρ_i^* to avoid rapid subsequent requests to core routers.

Then, If $\sum_{l \neq i} \tilde{r}_l(t) + \hat{r}_i(t) < C$, the core node will grant the request $\hat{r}_i(t)$ to ingress node and the new aggregate reservation level $\tilde{r}_i(t) = \hat{r}_i(t)$ will be issued. As a result, the new flow will be accepted by ingress node.

If $\sum_{l \neq i} \tilde{r}_l(t) + \hat{r}_i(t) \geq C$ but $C - \sum_{l \neq i} \tilde{r}_l(t) > \rho_i^*$, then the new aggregate reservation level $\tilde{r}_i(t) = C - \sum_{l \neq i} \tilde{r}_l(t)$ will be established. Otherwise, the current reservation level is maintained at the core node and the flow is rejected at the ingress node.

In a similar way, if current demand $r_i(t)$ is too less than the current aggregate reservation $\tilde{r}_i(t)$, the ingress node will request the core node to decrease in reserved bandwidth in order to more efficiently utilize the network resources.

Demand and Aggregation Model

Aggregation of reservation provides a mechanism to reserve network resources on behalf of multiple traffic flows and aggregation introduces a tradeoff. If the aggregate reservation $\tilde{r}_i(t)$ is infrequently adjusted, the signaling overhead in the core network is minimal. However, if the aggregate demand $r_i(t)$ changes quickly because of large traffic variation, it will not match with aggregate reservation $\tilde{r}_i(t)$. In that situation, aggregation may cause either under-utilization of resource reservation or flows may be blocked unnecessarily. On the other hand, if the aggregate reservation is immediately adjusted so as to match the current demand level, the system will achieve high utilization but the loss of scalability may result because of increased number of signaling rate and in the limitation, reservation adjustment is made for each flow which are identical to that of IntServ.

In this section, a simple model has been introduced to study the relationship between system performance, control (or signaling) and demand time scales and demand variance. Specifically, considering a Basic Model, an aggregate demand of link i characterized by

$$r_i(t) = m_i + a_i \cos\left(\frac{2\pi}{T}t + \theta_i\right) + Z_i(t) \quad (1)$$

where m_i is the mean rate and a_i is the amplitude of sinusoidal with period T .

In addition, the random nature of the demand is modeled by additive white uniform noise $Z_i(t)$ (i.e., $E Z_i(t)Z_i(t+s) = 0$ for $s \neq 0$) having uniform distribution $\{Z_i(t) \sim U[-b_i, b_i]\}$. The sinusoids have random phase θ_i which is uniformly distributed with $\theta_i \sim U[0, 2\pi]$.

It has been designated that $p_i(t) = m_i + a_i \cos\left(\frac{2\pi}{T}t + \theta_i\right)$ as the primary demand and $Z_i(t)$ as the secondary demand such that $r_i(t)$ can also be expressed as the sum of $p_i(t)$ and $Z_i(t)$.

For simplicity the model serves to isolate the performance impact of two key factors: demand time scale and demand variance (via T, a_i, b_i). Furthermore, it is assumed that the model shows similar to some traces of traffic aggregates.

To quantify the aggregate reservation $\tilde{r}_i(t)$ the study focuses on periodic reservation adjustments at exactly intervals of τ seconds. Assuming that the requested reservation level for a bulk reservation at time t , $(k_i - 1)\tau \leq t < k_i\tau$ is given by

$$\hat{r}_i(t) = \max_{(k_i - 1)\tau \leq s < k_i\tau} r_i(s)$$

where $k_i = 1, \dots, \frac{T}{\tau}$. Assume $\frac{T}{\tau}$ is an integer.

Alternatively, aggregate bandwidth reservation is adjusted every τ seconds with a request rate sufficient for the future interval. In practical situation, the adjustments interval might be made adaptive and perfect. Prediction is impossible, the model serves to isolate the control time scale τ . Hence, under the above scenario, the study will examine the relative impact of demand and control time scales as well as demand variance on system performance using the performance measures defined in the following section.

3.4 Performance Evaluation

In this section, three performance measures: the overload probability, the reserved resource utilization and the normalized available bandwidth are described. The expression for overload probability is also derived. The theoretical model mentioned above characterizes the relationship among the time scale of demand, the demand variance, the control time scale and the performance of aggregation. Hence, numerical experiments are presented for these considerations. In particular, using the basic demand model, the study focuses on the role of the demand time scale and demand variance of the basic model. Furthermore, the investigation shows that alternate models of primary demand with different periodic functions and alternate models of secondary demand having temporal correlation, have little impact on system performance. Finally the ability of basic model is investigated for prediction of performance aggregation using MPEG video trace for trace driven simulation scenario.

3.4.1 Analysis on Proposed Aggregation Model

In this work the effectiveness of aggregate-based resource reservation is evaluated by considering three performance metrics, namely, overload probability, reserved resource utilization and the normalized available bandwidth. These three performance metrics are analyzed from the network performance point of view as follows.

Firstly, the overload probability is defined as the ratio of overload traffic(which can not be admitted) to the total aggregate demand . It is designated by P_{ol}

$$P_{ol} = \frac{E(\sum_{i=1}^N (r_i - \tilde{r}_i)^+)}{E(\sum_{i=1}^N r_i)} , \quad (2)$$

where

r_i denotes a random variable with the steady state distribution of $r_i(t)$.

the term $E(\sum_{i=1}^N (r_i - \tilde{r}_i)^+)$ represents the total amount of overloaded traffic and

the term $E(\sum_{i=1}^N r_i)$ represents total aggregate traffic demand

Secondly, reserved resource utilization is referred to the fraction of an aggregate resource reservation that has been utilized by the underlying traffic. It is denoted by U_r .

$$U_r = \frac{(1 - P_{ol}).E(\sum_{i=1}^N r_i)}{E(\sum_{i=1}^N \tilde{r}_i)} , \quad (3)$$

where

the term $(1 - P_{ol})$ represents the probability of traffic going through

the term $E(\sum_{i=1}^N r_i)$ represents total aggregate traffic demand and

the term $E(\sum_{i=1}^N \tilde{r}_i)$ represents total aggregate resource reservation

Finally, normalized available bandwidth, is described as the fraction of bandwidth available after accounting for all aggregate reservations. It is denoted by b_A which reflects the efficiency of aggregation.

$$b_A = \frac{C - E(\sum_{i=1}^N \tilde{r}_i)}{C} \quad , \quad (4)$$

where

the term C represents the bottleneck link capacity and

the term $E(\sum_{i=1}^N \tilde{r}_i)$ represents total aggregate resource reservation

Under the basic model of aggregate demand and the above performance measures, the overload probability of aggregate resource reservation may be computed as follows.

Aggregation Performance

Considering N aggregate demands which share a single bottleneck link with capacity C as described in the basic model. If the aggregate demand of class i is

$$r_i(t) = m_i + a_i \cos\left(\frac{2\pi}{T}t + \theta_i\right) + Z_i(t)$$

$$i = 1, 2, \dots, N.$$

where $Z_i(t)$ is white uniform noise with $Z_i(t) \sim U[-b_i, b_i]$

b_i is the range of white uniform noise

m_i is the mean rate of aggregate demand of link i and

a_i is the amplitude of sinusoidal with period T

Then, the overload probability is approximately given by

$$P_{ol} = \frac{\left(\frac{\tau}{T}\right)^N \cdot \sum_{k1=1}^{\tau} \dots \sum_{kN=1}^{\tau} \left[\sum_{i=1}^N f_{i,kd} - C \right]^+}{\sum_{i=1}^N \left(m_i + \frac{2\tau}{T} \cdot a_i + b_i \right)} \quad (5)$$

where the term $f_{i,kd} = \max_{(ki-1)\tau \leq s < ki\tau} [m_i + a_i \cos(\frac{2\pi}{T} S)]$,

the term $\sum_{i=1}^N (m_i + \frac{2\tau}{T} \cdot a_i + b_i)$ represents total aggregate traffic demand and the numerator represents the overloaded traffic.

In order to simplify the analysis, it first considers the phase θ_i to be discretely uniform in $[0, \tau, 2\tau, \dots, T]$. Secondly, it isolates the impact of primary demands and secondary demands to observe that over an interval τ , the secondary demand satisfies $P(\max_{0 \leq S \leq \tau} Z_i(s) = b_i) \approx 1$ such that, in order to ensure sufficient bandwidth is available over the entire window τ , an additional bandwidth b_i must be reserved.

Next, applying the odd symmetric characteristics of the cosine wave at points $\pi/2$ and $3\pi/2$ to compute the mean discrete primary demand i as

$$\frac{\tau}{T} \sum_{K_i=1}^{\tau} f_{i,K_i} = \frac{\tau}{T} \left(\frac{T}{\tau} \cdot m_i + 2a_i \right) \quad (6)$$

which simplifies to $m_i + \frac{2\tau}{T} a_i$.

where τ is the control time scale, T is the demand time scale

m_i is the mean rate of aggregate demand and

a_i is the amplitude of sinusoidal with period T

Finally, Equation (2) has been computed by conditioning on the relative phase of the different aggregates, and after some manipulation, Equation (5) follows.

3.5 Numerical Results and Discussions with respect to the Basic Model

In this section, the roles of control and demand time scales are isolated by exploring the performance of the basic model under the condition of $Z_i(t) = 0$ (no secondary demand). With this scenario, the investigation focuses on the frequency of reservation ($1/\tau$) which is required for aggregate based resource reservation to achieve performance similar to IntServ's flow based resource reservation.

Considering a simple case of $N = 2$ classes, a bottleneck link capacity of $C = 3$, for demand of both cases $m_i = 1$, and $a_i = 1$. This is demonstrated by making the performance tradeoffs of aggregate – based resource reservation as the control time scale τ varies from 0 to T .

Firstly, the impact of control time scale on overload probability is studied. By using the equation (5),

$$P_{ol} = \frac{\left(\frac{\tau}{T}\right)^N \cdot \sum_{k=1}^{\frac{T}{\tau}} \dots \sum_{kN=1}^{\frac{T}{\tau}} \left[\sum_{i=1}^N f_{i,k_i} - C \right]^+}{\sum_{i=1}^N \left(m_i + \frac{2\tau}{T} \cdot a_i + b_i \right)} \dots \dots \dots (5)$$

where the term $f_{i,k_i} = \max_{(k_i-1)\tau \leq s < k_i\tau} \left[m_i + a_i \cos \left(\frac{2\pi}{T} s \right) \right]$ and

$$k_i = 1, 2, 3, \dots, T/\tau$$

Applying the assumptions of $N = 2$, $C=3$, $m_i=1$, $a_i=1$, $b_i = 0$, the result is shown in Figure 3.2 for a fixed demand time scale of $T = 2\pi = 6.28$ (units of T as hours) and τ varies from 0.05 to T ($2\pi = 6.28$)

Figure 3.2 Impact of Control Time Scale upon Overload Probability

Then, the impact of control time scale over reserved resource utilization is observed as follows.

$$U_r = \frac{(1 - P_{ol}) \cdot E(\sum_{i=1}^N r_i)}{E(\sum_{i=1}^N \tilde{r}_i)} \dots\dots\dots(3)$$

where

$$P_{ol} = \frac{(\frac{\tau}{T})^N \cdot \sum_{k=1}^{\frac{T}{\tau}} \dots\dots\dots \sum_{kN=1}^{\frac{T}{\tau}} [\sum_{i=1}^N f_{i,ki} - C]^+}{\sum_{i=1}^N (m_i + \frac{2\tau}{T} \cdot a_i + b_i)} \dots\dots\dots(5)$$

in which the term $f_{i,ki} = \max_{(ki-1)\tau \leq s < ki\tau} [m_i + a_i \cos(\frac{2\pi}{T} S)]$ ($k_i = 1, 2, 3, \dots, T/\tau$)

the term $E(\sum_{i=1}^N r_i)$ is expressed as $\sum_{i=1}^N (m_i + \frac{2\tau}{T} a_i + b_i)$ and $E(\sum_{i=1}^N \tilde{r}_i)$ can be determined based on the characterization of $\tilde{r}_i$ and using the mean bulk size expression .

$$\text{Mean Bulk Size} = \frac{\sum_{i=1}^N \sum_{k=1}^{\frac{T}{\tau}-1} |f_{i,k+1} - f_{i,k}|}{N \times T / \tau} ,$$

where the term $f_{i,k} = \max_{(k-1)\tau \leq s < k\tau} [m_i + a_i \cos(\frac{2\pi}{T} s)]$

Assuming $N = 2$, $C = 3$, $m_i = 1$, $a_i = 1$, $b_i = 0$ The result is shown in Figure 3.3 for a fixed demand time scale of $T = 2\pi = 6.28$ (units of T as hours) and τ varies from 0.05 to $T(2\pi = 6.28)$

Figure 3.3 The Impact of Control Time Scale upon Reserved Resource Utilization

The study also focuses on the impact of control time scale(τ) over normalized available bandwidth (b_A). Using equation (4), it may write

$$b_A = \frac{C - E(\sum_{i=1}^N \tilde{r}_i)}{C}$$

Furthermore, $E(\sum_{i=1}^N \tilde{r}_i)$ can be determined numerically on the basic of the characterization of $\tilde{r}_i$ and using the mean bulk size expression.

$$\text{Mean Bulk Size} = \frac{\sum_{i=1}^N \sum_{k=1}^{\frac{T}{\tau}-1} |f_{i,k+1} - f_{i,k}|}{N \times T / \tau},$$

where the term $f_{i,k} = \max_{(k-1)\tau \leq s < k\tau} [m_i + a_i \cos(\frac{2\pi}{T} s)]$

Assuming $N = 2$, $C=3$, $m_i=1$, $a_i=1$, $b_i=0$. The result is shown in Figure 3.4 for a fixed demand time scale of $T = 2\pi=6.28$ (units of T as hours) and τ varies from 0.05 to $T(2\pi=6.28)$

Figure 3.4 The Impact of Control Time Scale upon Normalized Available Bandwidth

After achieving the three numerical results as illustrated in Figure 3.2, Figure 3.3 and Figure 3.4, the following discussions can be made on the results .

Firstly, two extreme cases of $\tau = 0$ and $\tau = T$ are studied, in which $\tau = 0$ corresponds to the case of no aggregation or IntServ . In other words, the core's requested reservation matched precisely to the flows ' total demanded bandwidth. This provided an upper bound to the efficacy of aggregation. Under the given work load the data for the three performance metric are obtained as follows.

an overload probability of 4.4% ,
a reserved resource utilization of 100% and
an available bandwidth of 36%.

At the other extreme case, when $\tau = T$, the aggregate reservation is static corresponding to the maximum total flow demand over the entire period .In this situation , the data for the performance measures are found as

the overload probability = 15.9% ,
the utilization = 56.1% and
the available bandwidth = 0

This establishes a lower bound for the performance of aggregation .

Furthermore, by comparing with static aggregate reservation, it is found that the system performance rapidly enhances as the control time scale τ is decreased from T . Two interpretations of this behavior are examined as follows.

Firstly, the curve indicates the signaling frequency required to achieve a certain level of performance. For example, the figure shows that when τ is less than 1% of T , aggregation achieves near ideal performance. In other words, if the control and demand time scales are separated by two orders of magnitude, the performance of aggregation is nearly indistinguishable from that of IntServ.

Secondly, the curves can be analyzed in terms of “ bulk size” i.e., the required increase or decrease in reserved bandwidth to acquire a desired performance level.

Mean bulk size can be described as $\frac{\sum_{i=1}^N \sum_{k=1}^{\frac{T}{\tau}} |f_{i,ki+1} - f_{i,ki}|}{N \times T / \tau}$. Hence, it can convert the conclusions with respect to control time scale into the conclusions with respect to the magnitude of the reservation updates.

Moving a step further, this section has been to analyze reserved resource utilization U_r as a function of demand time scale T for a fixed control time scale (τ) in order that the performance penalty for performing aggregation may be quantified.

From equation (3),

$$U_r = \frac{(1 - P_{ol}) \cdot E(\sum_{i=1}^N r_i)}{E(\sum_{i=1}^N \tilde{r}_i)} \dots\dots\dots(3)$$

$$P_{ol} = \frac{(\frac{\tau}{T})^N \cdot \sum_{k=1}^{\frac{T}{\tau}} \dots\dots \sum_{kN=1}^{\frac{T}{\tau}} [\sum_{i=1}^N f_{i,ki} - C]^+}{\sum_{i=1}^N (m_i + \frac{2\tau}{T} \cdot a_i + b_i)} \dots\dots\dots(5)$$

where the term $f_{i,ki} = \max_{(ki-1)\tau \leq s < ki\tau} [m_i + a_i \cos(\frac{2\pi}{T} S)]$, the term

$E(\sum_{i=1}^N r_i) = \sum_{i=1}^N (m_i + \frac{2\tau}{T} a_i + b_i)$ and $E(\sum_{i=1}^N \tilde{r}_i)$ can be determined based on the

characterization of $\tilde{r}_i$ and using the mean bulk size expression.

$$\text{Mean Bulk Size} = \frac{\sum_{i=1}^N \sum_{k=1}^{\frac{T}{\tau}} |f_{i,ki+1} - f_{i,ki}|}{N \times T / \tau}$$

where $f_{i,ki} = \max_{(ki-1)\tau \leq s < ki\tau} [m_i + a_i \cos(\frac{2\pi}{T} S)]$

Using the assumptions of $N = 2$, $C=3$, $m_i=1$, $a_i=1$, $b_i=0$ and $\tau=0.0983$ hr, the following expressions are obtained :

$$\text{Let } y1 = E\left(\sum_{i=1}^2 r_i\right) = \sum_{i=1}^2 \left(m_i + \frac{2\tau}{T} a_i + b_i\right)$$

$$\text{Let } P_{ol} = \frac{y2}{y1}$$

$$\text{Let } y2 = \left(\frac{0.0983}{T}\right)^2 \sum_{k1=1}^T \sum_{k2=1}^T \left[\sum_{i=1}^2 f_{i,k1} - 3\right]^+$$

U_r can be written in a simpler way as follows:

$$U_r = \frac{(1 - P_{ol})y1}{y3}$$

$$\text{where } y1 = \left(2 + \frac{0.3932}{T}\right)$$

$$\text{where } y2 = \frac{9.663 \times 10^{-3}}{T^2} \sum_{k1=1}^{10T} \sum_{k2=1}^{10T} \left[\sum_{i=1}^2 f_{i,k1} - 3\right]^+$$

where $y3 = E\left(\sum_{i=1}^N \tilde{r}_i\right)$ can be determined based on the characterization of $\tilde{r}_i$ and

applying the expression of mean bulk size.

$$\text{Mean Bulk Size} = \frac{\sum_{i=1}^2 \sum_{k1=1}^{10T-1} |f_{i,k1+1} - f_{i,k1}|}{20.3458T}$$

where the term $f_{i,k1} = \max_{(k1-1)0.0983 \leq s < 0.0983 k1} [1 + \cos\left(\frac{2\pi}{T} S\right)]$,

$$k_i = 1, 2, 3, \dots, \frac{T}{0.0983}$$

The result is illustrated in Figure 3.5 for a fixed $\tau = 5.9$ minutes $= 5.9/60 = 0.0983$ hour and T varies from 0.1hour to 10 hour.

Figure 3.5 Impact of the Demand Time Scale T

Figure 3.5 illustrates the reserved resource utilization as a function of the demand time scale T for a fixed control time scale τ of 5.9 minutes. The figure shows the performance limitations of core routers delivering a maximum signaling frequency of once per 5.9 minutes(per link, the total number of signaling messages increases with the number of links). Then, the curve characterizes the performance penalty for performing aggregation rather than IntServ as a function of the demand time scale. From the observation of the curve, for the aggregation of traffic, if one wishes to achieve performance within 10% of IntServ, the system demand period must not be smaller than 1.67 hr when the control time scale τ is 5.9 minutes.

Here, the study is broadened with regard to the influence of the variance of the secondary demand on the aggregation performance. Thus, the investigation includes secondary demand $Z_i(t) \sim U[-b_i, b_i]$, $b_i \leq m_i - a_i$, as described in the basic model. One bottleneck link is utilized with $C = 8$ and two aggregates traffic of $N = 2$ with $m_i = 2$ and $a_i = 1$ respectively. In addition, variance of the secondary demand is given by $\sigma_i^2 = b_i^2/3$.

The effect of σ_i^2 (variance of secondary demand) on Ur (reserved resource utilization) is observed. Using equation (3)

$$Ur = \frac{(1 - P_{ol})E(\sum_{i=1}^N r_i)}{E(\sum_{i=1}^N \tilde{r}_i)} \dots\dots\dots (3)$$

$$\text{where } P_{ol} = \frac{(\frac{\tau}{T})^N \cdot \sum_{k=1}^{\frac{T}{\tau}} \dots \sum_{kN=1}^{\frac{T}{\tau}} [\sum_{i=1}^N f_{i,ki} - C]^+}{\sum_{i=1}^N (m_i + \frac{2\tau}{T} \cdot a_i + b_i)} \dots\dots\dots (5)$$

in which the term $f_{i,ki} = \max_{(ki-1)\tau \leq t < ki\tau} [m_i + a_i \cos(\frac{2\pi}{T} S)]$

where the term $E(\sum_{i=1}^N r_i) = \sum_{i=1}^N (m_i + \frac{2\tau}{T} a_i + b_i)$ and

the term $E(\sum_{i=1}^N \tilde{r}_i) =$ mean value of aggregate reservation.

In the analysis, $E(\sum_{i=1}^2 \tilde{r}_i)$ can be evaluated numerically on the basic of characterization of

$\tilde{r}_i$ considering the impact of the variance of secondary demand ($\sigma_i^2 = \frac{b_i^2}{3}$) and using the expression for mean bulk size.

$$\text{Mean Bulk Size} = \frac{\sum_{i=1}^N \sum_{k=1}^{\frac{T-1}{\tau}} |f_{i,k+1} - f_{i,k}|}{N \times T / \tau}$$

Using the assumptions of $N=2$, $C=8$, $m_i = 2$, $a_i = 1$, $T = 2\pi = 6.28$ and σ^2_i varies from 0 to 0.34, the result is demonstrated in Figure 3.6 for two scenarios namely $\tau = T$ (static aggregate reservation) and $\tau = \frac{T}{16}$ (dynamic aggregate reservation)

Figure 3.6 Impact of Secondary Demand on Reserved Resource Utilization

Similarly, the effect of σ^2_i (variance of secondary demand) on normalized available bandwidth (b_A) is also examined. Using equation (4),

$$b_A = \frac{C - E(\sum_{i=1}^N \tilde{r}_i)}{C} \dots\dots\dots(4)$$

in which $E(\sum_{i=1}^N \tilde{r}_i)$ can be determined numerically on the basic of the characterization of $\tilde{r}_i$ considering the impact of the variance of secondary demand ($\sigma^2_i = \frac{b^2_i}{3}$) and using the mean bulk size expression.

Using the assumed parameters of $N = 2$, $C=8$, $m_i=2$, $a_i=1$, $T = 2\pi = 6.28$, σ^2_i varies from 0 to 0.34 and using the relation $\sigma^2_i = \frac{b^2_i}{3}$, obtained result is described in Figure 3.7

Figure 3.7 Impact of Secondary Demand on Normalized Available Bandwidth

After obtaining the encouraging numerical results as demonstrated in Figure 3.6 and Figure 3.7, the following discussions can be made on the results.

Numerical results obtained clearly show that the variance in secondary demand prevents the progress of the efficacy of aggregation. If the static aggregate reservation ($\tau=T$) is investigated, it is observed that the impact of secondary demand is severe since reserved resource utilization decreases from 100% to 53% when the variance of the secondary demand is 0.2. Moreover, if consideration focuses the aggregation with the control time scale of $\tau = T/16$, it can be observed that the effects are less severe. For example, reserved resource utilization decreases to 70% under the same variance of 0.2. For that reasons, it is determined that the performance of aggregation can be degraded because of sufficient “noise” in the demand. In other words, if the noise is moderate, performance close to IntServ can be achieved. For example, if wishing to achieve a reserved resource utilization within 20% of IntServ with aggregation and $\tau = T/16$, the variance of secondary demand must be limited to 0.05. In fact, the impact of such variance can be made less severe by using faster signaling and reduced τ .

Moreover, the following experiment is also carried out in order to investigate the impact of alternate primary demand models in addition to the sinusoid with random phase. Specifically, periodic sawtooth and square waves are analyzed. Random phase is assumed in both waves. Secondary noise $Z_i(t)$ is set to 0 in all cases. If the phase $\theta_i=0$, the three respective primary demands can be described as follows:

$$\text{Sinusoidal demand} = m_i + a_i \cos\left(\frac{2\pi t}{T}\right) \quad \text{for period } T$$

$$\text{Sawtooth demand} = (m_i - a_i) + \frac{4a_i}{T}t \quad \text{for } 0 \leq t \leq T/2$$

$$(m_i - a_i) - \frac{4a_i}{T}(t - T) \quad \text{for } T/2 \leq t \leq T$$

$$\begin{aligned} \text{Square wave's demand} &= (m_i + a_i) \quad \text{for } 0 \leq t \leq T/2 \\ &= (m_i - a_i) \quad \text{for } T/2 \leq t \leq T \end{aligned}$$

For three primary demand models, mean demand $m_i = 2$, Period $= 2 \pi$, a variance of 1.33 are used for analysis. To achieve a variance of 1.33, for sinusoid demand has amplitude $a_i = 1.63$, for sawtooth demand had amplitude $a_i = 2$ and for square wave demand has amplitude $a_i = 1.2$. It is assumed that bottleneck link capacity of $C = 6$ and number of classes $N = 2$. The results for the impact of alternate primary demand model on three performance measures : overload probability, reserved resource utilization and normalized available bandwidth are shown in Figure 3.8. Figure 3.9 and Figure 3.10 respectively. According to Figure 3.8, 3.9 and 3.10, such variations on the basic models of primary demand have little impact on performance. This indicates that the essential tradeoff of control and demand time scales is quite similar under different demand functions. That is why, consideration of alternate periodic demand functions may be of limited impact for characterizing the performance of aggregation. Thus, further investigation to the sinusoidal model is continued.

Figure 3.8 Impact of Alternate Primary Demand Model on Overload Probability

Figure 3.9 The Impact of Alternate Primary Demand Model on Reserved Resource Utilization

Figure 3.10 The Impact of Alternate Primary Demand Model on Available Bandwidth

The study also concentrates on the performance impact of an alternate secondary demand model comparing with the uniform white noise in the basic model. In particular, alternate secondary demand model $Z_i(t)$ is assumed as a sawtooth wave with random phase. Data used in the experiments are Sawtooth with mean = 0, Variance = 0.33, Maximum = 1, Minimum = -1 and Period = $T/4$. Under the same mean, variance, and range, the performance of sawtooth is compared with that of white noise. In addition, it is assumed that $C=6$, $N=2$, $m_i=2$, $a_i=1$, $b_i=1$ in the numerical study.

Figure 3.11 The Impact of Alternate Secondary Demand Model on Overload Probability

Figure 3.11 describes the impact of temporal correlation in secondary demand $Z_i(t)$ on overload probability. The numerical result show that for small control time scale τ , the performance of correlated secondary noise is not much difference to that of white noise whereas for large τ , it degrades performance a little more than that of white noise. Nevertheless, the figure indicates that the difference is minimal.

3.6 Trace Driven simulations Using MPEG video Trace

In this section, experimental investigation is explored to consider trace-driven simulations. The ability of the basic model is examined to predict the performance resulting from trace-driven scenarios and the usefulness of the simple model base technique for performance of aggregation is evaluated.

Simulation Source and Scenarios

In practice MPEG video traces with various contents such as Movies, Sports, News, several Episodes and Cartoons are traditionally chosen for evaluating the performance of the new proposed technique to verify its feasibility. The following MPEG video trace has been used for all simulated experiments reported in this section

Figure 3.12 MPEG Video Trace

The trace shown in Figure 3.12 indicates aggregate measurements taken from MPEG video trace “lambd_ipb”. The mean, variance and demand period T of the aggregate traffic are 1.46 Mbps, 5 and 26 minutes, respectively. Figure 3.12 shows the corresponding trace of the aggregate traffic. Measurements are reported over 2 second s intervals.

In simulations, above MPEG video trace is implemented to represent the aggregate demand $r_i(t)$. Consideration is focused on the perfect prediction of future demand such that a core reservation request at time t of aggregate i is for maximum bandwidth necessary over the next τ second interval, i.e., $\max_{t \leq s < t+\tau} r_i(s)$. For each scenario, 120 independent simulation runs are performed to empirically obtain the average of the performance parameters. For each run, it is simulated three demand periods, and discarded results from the first cycle as transient.

3.7 Comparison Between Trace Driven Simulations and Model Prediction

In the experiment, consideration is based on a single bottleneck link with capacity 5.4 Mbps for MPEG video trace and the comparison is made for the performance of the trace driven simulations with that predicted by the basic model. The mean, variance, and time scale are evaluated in order to compute the parameters of the basic model. Considering only the primary demand, the basic model produced parameters

$$r_i(t) = 1.46 + 1.2 \cos\left(\frac{2\pi t}{T} + \theta_i\right), \text{ where as considering both primary demand and}$$

secondary demand, it is described as $r_i(t) = 1.46 + 0.56 \cos\left(\frac{2\pi t}{T} + \theta_i\right) + Z_i(t)$,

$$Z_i(t) \sim U[-0.39, 0.39].$$

Trace driven simulation using MPEG video trace and Model Predictions for three performance measures are illustrated in Figure 3.13, Figure 3.14 and Figure 3.15 .

Figure 3.13 Overload Probability

Figure 3.14 Reserved Resource Utilization

Figure 3.15 Normalized Available Bandwidth

Figure 3.13, Figure 3.14 and Figure 3.15 illustrate the performance comparison between the trace driven simulations using MPEG video trace and the model predictions for the three performance measures such as overload probability, reserved resource utilization and normalized available bandwidth as described at the beginning of this chapter.

Trace driven simulation results show that the sinusoidal model with random phase is able to capture the basic behavior of the system. For example, for $\tau=48$ seconds, the predictions of overload probability, reserved resource utilization, and normalized available bandwidth are with 1.6%, 33%, and 25.% of the simulated values. It has been observed that considering only the primary demand and ignoring secondary demand may introduce a little prediction difference.

However, characterizing variance via additive random noise $Z_i(t)$ rather than purely through the sinusoid with random phase further improves the predictions. For example, for $\tau=48$ seconds, the predictions of overload probability, reserved resource utilization, and normalized available bandwidth are with 1.5%, 2%, and 3.6.% of the simulated values.

Finally, observation indicates that as predicted by the model, if demand and control time scales are separated by two orders of magnitude and the secondary demand is moderate, aggregation achieves performance near to IntServ. For example, under $\tau = 18$ seconds, $T=86\tau$, the overload probability is 3.7% for IntServ and 4.8% for aggregation. As a result, the simulation study shows that the simple model based approach is applicable for predicting the performance of aggregation.

Based on the numerical and simulation results obtained, major findings focus the following factors:

Aggregation reservation is greatly insensitive to the particular shape of the primary demand, as well as to temporal correlation to the secondary demand. The performance of aggregation is obviously related to the relationship between demand and control time scale as well as to the variance of the secondary demand. The simple model based technique is beneficial to characterize the performance of aggregation. A separation of time scales of two order of magnitude between demand and control ensure excellent performance of aggregation similar to those that of Intserv provided that additional noise (secondary demand in addition to the primary demand) is moderate. The results provide the basic performance tradeoffs of aggregation as well as a simple model based technique for performance prediction.

3.8 Summary

A novel scheme namely aggregate base resource reservation coupling with IntServ has been proposed. The proposed solution achieves both required QoS and scalability. The effectiveness of aggregate reservation has been evaluated by implementing numerical experiments on three performance metric: overload probability, reserved resource utilization and the normalized available bandwidth. The numerical results indicate that system performance rapidly enhance as the control time scale τ is decreased from demand time scale T and the performance of aggregation can be degraded because of sufficient noise in the demand. Moreover, it can be observed that alternate primary demand functions have little impact on performance of aggregation. In addition, the impact of temporal correlation in secondary demand on overload probability is investigated and it is found the difference is minimal.

Finally, the ability of the model base technique for performance of aggregation has been evaluated using the MPEG video trace for simulated experiments. The Trace driven simulation and model predictions for three performance measures are presented. The simulation results show that the basic model and the theoretical results could predict the performance of trace driven scenario. Hence, the simple model based technique is beneficial to predict the performance of aggregation. The numerical and simulation experiments show that if the control time scale and demand time scale are separated by two orders of magnitude and variance of traffic is moderate, aggregation can provide performance close to that of strong service model IntServ. The results provide the basic performance tradeoffs of aggregation as well as a simple model based technique for performance prediction.

One important goal of the MPLS architecture is to combine the scalability and flexibility of routing with the performance, quality of service (QoS), and traffic management . The next chapter will undertake the detail study with a broad overview of MPLS architecture and investigate traffic engineering and QoS routing using MPLS which offers the advantage of choosing paths that meet application QoS requirements and balance the load in the network and benefits come at the cost. Following this is a description of a novel technique namely traffic aggregation mechanism and label swapping algorithm with some illustrative examples for delivering scalable QoS. Later chapters will extend the implementation of MPLS architecture for delivering scalable QoS with multiple-priority classes over mobile access network using mobility aware scheme and mobile enabled MPLS network.

CHAPTER 4 SCALABLE QOS USING MPLS ARCHITECTURE

4.1 Overview of MPLS

The deployment of MPLS over the Internet has some profound consequences at the architectural level. Some of the immediate consequences are a more efficient transit core network, improved economy of scale, new connection oriented services and the ability for fast restoration of data traffic. As the Internet continues to grow, Internet service providers (ISP) are increasingly being challenged to deploy service differentiation in the network for providing competitive services. In order to provide such capabilities, the basic traffic forwarding paradigm of present-day IP networks must be enhanced to support guaranteed quality of service (QoS). The solution is based on MPLS.

Multiprotocol label switching (MPLS) [5,18] is a versatile solution to address the problems faced by present-day networks—speed, scalability, quality-of-service (QoS) management, and traffic engineering. MPLS addresses issues related to scalability and routing (based on QoS and service quality metrics) [66].

MPLS is a technology which combines high-speed switching with Layer 3 network routing aimed at greatly improving the packet forwarding performance of the backbone routers in the Internet. MPLS uses a label switching technique to forward data through the network. The swapping of label value and forwarding of the packet can be performed in hardware. This allows MPLS networks to be built on existing label switching hardware such as ATM and Frame Relay. Thus, a faster mechanism is provided because MPLS does not need to examine the full packet header to decide the next hop [8].

Routers supporting MPLS are known as label switching routers (LSRs). A path from the ingress node to the egress node of an MPLS domain followed by packets with the same label is referred to as a label switched path (LSP)[9]. A small fixed format label is encapsulated within each data packet on its entry into the MPLS network. Labels indicate both routes and service attributes of the network. High-speed switching of data is possible because the fixed-length labels are inserted at the very beginning of the packet and can be used by hardware to switch packets quickly between links [8]. Once a packet has been labeled, the rest of the journey of the packet through the backbone is based on label switching.

The Forward Equivalence Class (FEC) is a representation of a group of packets that are forwarded over the same path through a MPLS network even though they have different ultimate destinations. In MPLS, assignment of a particular packet to a particular FEC is done just once, as the packet enters the network. FECs are based on service requirements for a given set of packets or simply for an address prefix. One or more FECs may be mapped to a single LSP. LSP is created for each FEC through the MPLS domain. Once a packet has been classified as a new or existing FEC, a label is assigned to the packet.

In MPLS, data transmission occurs on label-switched paths (LSPs). LSPs are a sequence of labels at each and every node along the path from the source to the destination. LSPs are established either prior to data transmission (control-driven) or upon detection of a certain flow of data (data-driven). The labels, which are underlying protocol-specific identifiers, are distributed using Label Distribution Protocol (LDP) or RSVP or piggybacked on routing protocols like Border Gateway Protocol (BGP) and OSPF [8]. The major advantage of using label switching is that packets of different flows can be labeled differently and thus received different QoS [9]. This label swapping forwarding technique provides Internet service providers (ISPs) precise control over traffic flows and makes it possible to support traffic engineering and QoS guarantees.

By decoupling the forwarding mechanisms from routing, MPLS enables the deployment of specialized routing services without changes to the packet forwarding path [87]. In particular, MPLS contains support for setting up explicitly routed LSPs in which the entire path is determined at the source. This allows administrator, for example, to reduce certain classes of traffic to balance network load independent of the underlying conventional shortest-path routing protocol [83]. MPLS also specifies a routing framework called constraint-based routing (CBR) that enables demand-driven routing capable of satisfying resource and policy requirements. The network traffic engineer identifies classes of traffic and their endpoints, along with associated QoS metrics and policies. A constraint-based routing (CBR) protocol finds suitable paths that meet the requirements. CBR operates much the same as source-directed link-state QoS routing protocols. The value of CBR is in its ability to assist in traffic engineering while reducing the amount of manual route configuration.

This chapter studies the background materials of MPLS architecture related to switching and routing that can be applied in today's IP networks. Various label encoding methods are described and QoS routing protocols based on MPLS are discussed by its ability to find paths satisfying application QoS requirements. A novel method called label swapping and traffic aggregation mechanisms of MPLS is proposed to provide scalability and QoS guarantees over broadband Internet. This is demonstrated by example network scenarios. A summary of benefits is offered for the proposed method.

4.2 Basic Mechanisms of MPLS

4.2.1 Different Label Encapsulation Techniques

MPLS is a highly efficient scheme for forwarding data packets through a communication network. MPLS uses a technique known as label switching to forward data through the network. A small fixed format label is encapsulated within each data packet on its entry into the MPLS network. Labels indicate both routes and service attributes of the network. A label, in its simplest form, identifies the path a packet should traverse. The label values are of local significance only, meaning that they pertain only to hops between LSRs. Label assignment decisions may be based on forwarding criteria such as traffic engineering and QoS requirements [8][66] which are discussed elaborately at the last part of this chapter.

Each data packet encapsulates and carries the labels during their journey from source to destination. In router networks, the label is a separate, 32-bit header. Figure 4.1 shows the IP packet structure under MPLS.

Figure 4.1. IP Packet Structure under MPLS Switching

MPLS Header Format (Shim Header)

Each MPLS packet has a header. The MPLS header permits any link layer technology to carry an MPLS label so it can benefit from label swapping across an LSP [63]. The label values are derived from the underlying data link layer. In a non-asynchronous transfer mode, the MPLS header (Shim Header) contains the following fields as shown in Figure 4.2[8].

Figure 4.2 MPLS Shim Header

- ❖ **20 bit Label** : carrying the actual value of the MPLS Label
- ❖ **a 3 bit Experimental field (Class of Service field)** : affect queuing and discard algorithms applied to the packet as it is transmitted through network.
- ❖ **1 bit Label Stack indicator** : support hierarchical label stack for multiple labels
- ❖ **an 8 bit time-to-live (TTL) field** : provides conventional IP TTL functionality

Label Encapsulation

The goal of MPLS is to be Multi-Protocol at both Layer 2 (e.g., ATM , Frame Relay) and Layer 3 (e.g., IP). Depending on Layer 2 (DATA LINK LAYER) and Layer 3 (NETWORK LAYER) technologies involved, different label encoding schemes can be used. The generic label format is illustrated in Figure 4.3 [68].

Figure 4.3 MPLS Generic Label Format

Different label encoding schemes are demonstrated in Figure 4.4 and Figure 4.5 respectively.

Figure 4.4 Label Encoding of IPv6

In IPv6 standard, MPLS Label (Label Value) can be encapsulated in flow label field (20 bits) of IPv6 header. This is shown in Figure 4.4 [68].

Label Encapsulation in Non-ATM Environment

If the Data link layer technology does not support a label field, the MPLS label is encapsulated in a standardized MPLS header (Shim Header) that is inserted between the Layer 2 data link header and Layer 3 network layer header (IP header) as shown in Figure 4.5 .

(b) Label Encoding Scheme for Ethernet as the Data Link Layer

Figure 4.5 Label Encoding Schemes in non-ATM Environment

4.2.2 Functions and Characteristics of MPLS

MPLS provides for the efficient designation, routing, forwarding, and switching of traffic flows through the network. MPLS specifies mechanisms to manage traffic flows of various granularities, such as flows between different hardware, machines, or even flows between different applications and provides a means to map IP address to simple, fixed-length labels. In addition, MPLS interfaces to existing routing protocols such as resource reservation protocol (RSVP) and open shortest path first (OSPF). It supports the IP, ATM, and frame-relay Layer-2 protocols [66].

Label Switched Paths (LSPs) Set Up

Within an MPLS domain, a path is set up for a given packet to travel based on an FEC. The LSP is set up prior to data transmission. MPLS provides the following two options to set up an LSP [66].

(1) hop-by-hop routing

Each LSR independently selects the next hop for a given FEC. This methodology is similar to that currently used in IP networks. The LSR uses any available routing protocols, such as OSPF, ATM private network-to-network interface (PNNI), etc.

(2) explicit routing

Explicit routing is similar to source routing. The ingress LSR (i.e., the LSR where the data flow to the network first starts) specifies the list of nodes through which the ER-LSP traverses. The path specified could be non-optimal, as well. Along the path, the resources may be reserved to ensure QoS to the data traffic. This eases traffic engineering throughout the network, and differentiated services can be provided using flows based on policies or network management methods.

4.2.3 Signaling Mechanisms

Label Request

Using this mechanism, an LSR requests a label from its downstream neighbor so that it can bind to a specific FEC. This mechanism can be employed down the chain of LSRs up until the egress LER (i.e., the point at which the packet exits the MPLS domain).

Label Mapping

In response to a label request, a downstream LSR will send a label to the upstream initiator using the label mapping mechanism. The above concepts for label request and label mapping are explained in Figure 4. 6 and illustrated in Figure 4.7 [66].

Figure 4.6 Signaling Mechanisms

Figure 4.7 LSP Creation through an MPLS Domain

4.3 MPLS Operation

Figure 4.8 MPLS operation

Conventional forwarding is based on network layer information in the packet header and analyzed at each hop. In MPLS based network, packet header is analyzed at the entry and exit points of domain. In an MPLS network, LSP is first established using a signaling protocol such as LDP. Then, a label in the packet header is used for making forwarding decisions in the network. LSR examines the label and possibly the experimental field in forwarding the packet. The whole label switching process is shown in above Figure 4.8.

When entering the MPLS network, the packets are classified at the ingress node. Then, the ingress LSR maps the incoming traffic (IP packets) onto LSP using FEC. Thus, ingress LSR performs packet classification, assigns an initial label to each packet and forwards it to the next hop in the LSP. At the intermediate LSR (core LSR), incoming label is replaced by outgoing label and packet is switched to next LSR. This is faster than the process of normal IP routing. Finally, the egress LSR removes the label from the packet and forwards it as a regular IP packet.

4.3.1 Label Switching Scenario

- LIB : Label Information Base
- In Label : Incoming Label
- FEC : Forwarding Equivalence Class
- Out IF : Outgoing Interface
- Out Label : Outgoing Label
- IPH : IP Header

Figure 4.9 Label Switching Scenario

4.3.2 Label Stacking Concept

Figure 4.10 Label Stacking

- ❑ **20 bits Label** : actual value of MPLS Label
- ❑ **a 3 bit Experimental field (Class of Service field)** : affecting the Queuing & discarding algorithms
- ❑ **1 bit Label Stack indicator** : support hierarchical label stack
 - S bit is set to 1 for the last entry in the label stack (i.e., for the bottom of the stack) & zero for all other label stack entry
 - Forwarding Hierarchy via Label Stacking : Traffic Engineering & Scalable VPN
- ❑ **an 8 bit time-to-live (TTL) field** : provides conventional IP TTL functionality

IGP Label is used for routing packets. RSVP Label is used for TE Tunnels. If more than one service is used, multiple labels are required. Label stack is represented as a sequences of “Label Stack Entries “ (Shim Header) [6]. Each Label Stack Entry is represented by 4 octets. The label stack concept is illustrated in Figure 4.10.

4.4 Traffic Engineering and QoS Routing

Consumer applications such as streaming live video, packetized voice, multiplayer games, and World Wide Web-based shopping are now common place. The applications that deliver these new services impose new requirements on network performance and reliability. While providing QoS for customers is an important goal, equally important to the Internet service provider (ISP) is the efficient utilization of the network to reduce costs and improve manageability. To achieve this goal, ISP must engage in “traffic engineering” to optimize and maintain network resources.

Major goal of MPLS is to support traffic engineering and provide QoS. Most current network routing protocols such as Internet Gateway Protocol (IGP) are based on shortest path algorithm which implies that there is only one path between a given source and destination end system. The shortest paths from different sources overlap at some links, causing congestion on those links.

The purpose of traffic engineering is to enhance network utilization and to improve the architecture (topology and link capacity) of a network in a systematic way, so that the network is robust, adaptive and easy to operate. An efficient traffic engineering solution shares the data traffic load with the routers, nodes and switches in the network, making none of its individual components either over utilized or under utilized, thus assuring satisfactory service delivery and optimizing resource efficiency [6, 63].

For example, non-shortest paths can be chosen to forward traffic. An important result of traffic engineering is the avoidance of congestion on any path.

In MPLS, Traffic Engineering is provided using explicit routing. Based on the idea of label switching, MPLS provides explicit routing without requiring each IP packet to carry the explicit route [9]. Explicit route information is carried only at the time that LSP is set up; not with each packet. As a result of this technique, explicit routing has become the main building block of constraint-based routing (CBR), and hence, CR-LDP plays an important role in traffic engineering and QoS routing.

To accommodate diverse traffic characteristics and quality of service (QoS) requirements, emerging networks can employ a variety of mechanisms to control access to shared link, buffer and processing resources [58, 59, 138, 158]. These mechanisms include traffic shaping and flow control to regulate an individual or group of traffic flows, as well as link scheduling and buffer management to coordinate resource sharing at the packet or cell level.

In particular, QoS routing selects a path for each flow or connection to satisfy diverse performance requirements and optimize resources usage [84, 148]. Most current routing schemes are geared toward connectivity and reliability, and consider only a single metric such as hopcount or cost (e.g., Routing Information Protocol (RIP) as a distance vector protocol [155, 156] and Open-Shortest-Path-First (OSPF) as link-state routing [154,157]). QoS routing, on the other hand, must characterize network with multiple metrics, for example bandwidth, delay, and packet loss probability, to be useful for QoS- sensitive applications.

4.4.1 MPLS and Constrained-Based Routing (CBR)

CBR [6] computes routes that are subject to constraints such as link characteristics (bandwidth, delay), hop count and QoS requirement. CBR can be used in conjunction with MPLS to set up LSPs. The LSPs that are established could be CBR-LSPs, where the constraints could be explicit hops or QoS requirements. Explicit hops dictate which path is to be taken. QoS requirements dictate which links and queuing or scheduling mechanisms are to be employed for the flow. When using CBR, it may select a longer (in terms of cost) but lightly loaded path rather than a heavily loaded shortest path.

An illustrative example for CBR is shown in Figure 4.11. The shortest path between router A and C is through link A-C with IGP metric $m=1$. But because the reservable bandwidth on the shortest path is only $(622-600) = 22$ Mb/s, when CBR tries to find a path for an LSP of 40 Mb/s it will select path A-B-C instead because the shortest path does not meet the bandwidth constraint.

Although CBR increases network utilization, it adds more complexity to routing calculations, as the path selected must satisfy the QoS requirements of the LSP. The IETF has defined a CR-LDP component to facilitate constraint-based routes [66]. Traffic Engineering RSVP or TE-RSVP and Constrained-based LDP or CR-LDP are two possible approaches to supply dynamic traffic engineering and QoS in MPLS. TE-RSVP and CR-LDP follows setting ER-LSP. Therefore these two approaches are equipped to follow the route using predetermined CoS (class of service) and QoS (quality of service) parameters.

Using constraint-based routing (CBR), it is now possible to achieve measurable service interworking among today's most widely used networking technologies, including IP, ATM, Frame Relay, and Optical [72].

Figure 4.11 Constrained-Based Routing

4.4.2 Constrained-Based Routing Label Distribution Protocol (CR-LDP)

CR-LDP is a signaling protocol for the establishment of explicitly routed paths with quality of service attributes attached to them. In this section, the quality of service (QoS) parameters will be discussed.

4.4.2.1 QoS Parameters and Frame format

The CR-LDP QoS parameters enable users to customize traffic by several ways including resource reservation based on traffic engineering, frequency of service to enable the support of real-time applications such as voice over IP (VoIP), assignment of domain-specific priorities for LSPs and ability to renegotiate each of the traffic parameters and the LSP priorities. The network uses these parameters for local admission control at each of the LSRs along the label-switched path [72].

The CR-LDP frame format is shown in Figure 4.12 [76]. The call set-up procedure for CR-LDP is a very simple two steps process: Label Request and a Label Map as shown in Figure 4.13 [76]. CR-LDP is a hard state protocol which means that signaling messages are sent only once, and there is no refreshing of state information. The major advantage of a hard-state protocol is that it should be more scalable. CR-LDP signaling has been designed to support DiffServ and operator configurable QoS classes. The procedure for reporting failures in CR-LDP is designed with reliable TCP transport to ensure that the recovery process can be started in a timely manner based on the policies specified by the network operator

U	F	Traffic TLV	Length	
Flags		Frequency	Reserved	Weight
Peak Data Rate (PDR)				
Peak Burst Size (PBS)				
Committed Data Rate (CDR)				
Committed Burst Size (CBS)				
Excess Burst Size (EBS)				

Figure 4.12 CR-LDP Frame Format

Figure 4.13 CR-LDP Call Set Up

4.4.2.2 Advantages of CR-LDP Approach

The advantages of the CR-LDP service approach are customizable which means the network operator can define the services that fit his business and it is easy to change services since they are not “hardwired”. This approach facilitates service interworking between different domains using different technologies, such as ATM service categories, thus it can be used as standard implementation. In essence, it is fundamentally sound since changes to the basic paradigm are not required each time a new service is needed. In addition, it is important to point out that the CR-LDP service approach is flexible, therefore, it can easily interwork with other networking technologies such as DiffServ architecture, ATM quality of service, Frame Relay and RSVP traffic engineering. As a true open standard, CR-LDP is easy to enhance, easy to standardize, and has no backward compatibility issues.

4.4.3 RSVP-TE (Traffic Engineering) Approach

RSVP with features added to accommodate MPLS traffic engineering is called RSVP-TE. Traffic engineering functions allow for the management of MPLS labels.

Figure 4.14 RSVP-TE Path Request

Figure 4.15 RSVP-TE Reservation

Figure 4.14, Figure 4.15 and Figure 4.16 show how a path is setup between two endpoints. This signaling process is called “soft state” because the call will be torn down if it is not refreshed in accordance with the refresh timer [76].

Figure 4.16 RSVP-TE Path Set Up

In RSVP-TE, the sender first transmits a Path message to the receiver including traffic characteristic. The receiver returns a Resv message to request resources for the flow. Each node along the route has an opportunity to accept or reject the Resv message. If the request is rejected, the node will send an error message to the receiver to terminate the signaling process. If the request is accepted, the associated flow state information will be installed to reserve resources at each node. To work with MPLS, the first LSR inserts a Label_Request object into the path message to request a label binding. If an Explicit_Route object is added to the path message, the path message will be forwarded along a specific route. The last LSR will respond a Resv message including traffic descriptors, QoS parameters and a Label object traversing upstream to reserve resources at each node along the path and to provide label mapping. By piggybacking label binding information on RSVP messages, resources can be reserved for an LSP at the same time as label assignments [69].

4.4.4 Comparing CR-LDP to RSVP-TE

CR-LDP is an extended version of LDP specifically designed to facilitate constraint-based routing of LSPs. CR-LDP supports explicitly routed LSPs and allocation of resources. UDP is used for discovering MPLS peers and TCP is used for control, management, label requests and mapping. This allows it to assume reliable distribution of control messages. In strict CR-LDP, LSP has been established between ingress and egress LER in a MPLS network and the path is set constrained to two specific LSRs. The label requests have been passed down to each downstream device in a hop-by-hop fashion to the egress LER and mapping has been passed upstream in similar fashion to the ingress LER. This is advantageous, as a particular traffic type (such as voice) which can be matched to the optimal path to leverage bandwidth and prioritization. LDP uses ATM's traffic engineering terms to map QoS.

On the other hand, RSVP-TE uses the extensions to the existing RSVP protocol. An LER or LSR only requires the extensions to be able to support MPLS explicit routing. RSVP-TE is a "soft state" protocol and uses UDP or IP datagrams as the signaling mechanism for

LSP setup communications, including peer discovery, label requests and mapping and management [18, 63]. RSVP is typically paired with IntServ's detection of QoS. While in the industry today, it can be found that Cisco and Juniper favor the RSVP-TE model and Nortel favors the CR-LDP model. Both signaling protocols are supported by most vendors. The technical comparisons of these two protocols are provided in Table 4.1 shown below.

Table 4.1 CR-LDP vs RSVP

Comparison	CR-LDP	RSVP-TE
<i>Vendors</i>	Nortel	Cisco, Juniper , Foundry
<i>QoS Type</i>	DiffServ and ATM traffic classes	IntServ traffic classes
<i>Transport Mechanism</i>	LDP over TCP	RSVP on IP
<i>State Management</i>	Hard State	Soft State
<i>Modes of Label distribution and LSP Setup</i>	Easy to support all modes since CR-LDP is based on LDP	Only downstream on demand ; need to run both RSVP and LDP for other modes
<i>Messages required for LSP Setup and Maintenance</i>	Request and Mapping	Path, Resv, and ResvConf
<i>Base Architecture</i>	Based on LDP developed for MPLS	Based on RSVP, but may require major changes to the basic protocol to improve its scalability
<i>Failure Notification</i>	Reliable procedure	Unreliable procedure
<i>Recovery Time</i>	A little slower	Faster
<i>Chat Overhead</i>	Low	High
<i>Path Optimization and Rerouting</i>	LSP ID can be used to prevent double booking of Bandwidth for an LSP when doing “ make-before-break”	Shared explicit filter prevents double booking of bandwidth for an LSP when doing “ make-before-break”

4.5 IP QoS in MPLS Network

As the Internet continues to grow, Internet service providers (ISP) are increasingly being challenged to deploy service differentiation in the network for providing competitive services. In order to provide such capabilities, the basic traffic forwarding paradigm of current IP networks must be enhanced to support guaranteed quality of service (QoS). These issues can be solved by using MPLS. QoS is characterized by parameters such as bandwidth, delay, delay jitter and packet loss probability. The mechanisms for implementing a QoS solution include admission control, flow identification, traffic policing, and traffic scheduling.

With Diffserv, packets are classified at the edge of the network. The differentiated service field (DS fields) of the packets are set accordingly. In the middle of the network, packets are buffered and scheduled according to their DS fields. Important traffic such as network control traffic and traffic from premium customers will be forwarded preferentially.

With MPLS, QoS is provided in a slightly different way. Packets still have their DS field set at the edge of the network. Moreover, the experimental fields in the MPLS headers are set at the ingress LSR. In the middle of an LSR, packets are buffered and scheduled in accordance to the experimental fields [6].

One important goal of the MPLS architecture is to combine the scalability and flexibility of routing with the performance, quality of service (QoS), and traffic management. When MPLS is combined with differentiated services and constrained based routing, they become powerful and complementary abstraction for quality of service (QoS) provisioning in IP networks.

The Differentiated service (Diffserv) approach uses the differentiation model where packets are classified into a small number of classes at the edge of the network. This approach does not maintain any reservation state in intermediate LSR. Typically it involves no dynamic signaling. QoS guarantees are provided by offline reservation of resources and appropriate marking of packets by traffic conditioner at the label edge router. To provide performance guarantees in such environment, it is necessary to control the path of offered traffic. Using MPLS approach, virtual paths for traffic flows can be established. The Diffserv approach to QoS typically involves pre-configuration of resources for a specific class along the path and the use of admission control algorithms [6 7].

4.6 Traffic Aggregation Mechanism and Label Swapping

Algorithm of MPLS

By implementing label swapping algorithm of MPLS, the efficiency of packet forwarding is greatly improved. The swapping of label value and forwarding of the packet can be performed in hardware. This allows MPLS networks to be built on existing label switching hardware such as ATM and Frame Relay. Thus a faster mechanism is provided because MPLS does not require to examine the full packet header to decide the next hop. The main advantage of using label switching is that packets of different flows can be labeled differently and thus received different QoS [9]. This label swapping forwarding technique provides Internet Service Providers (ISPs) precise control over traffic flows and make it possible to support traffic engineering and QoS guarantees. Therefore, the thesis presents the example network scenarios for the proposed novel method: traffic aggregation mechanism and label swapping algorithm of MPLS to provide scalability and QoS in the core network.

Concepts of Flows, LSP and Links

In the context of MPLS, a traffic trunk is an aggregation of traffic flows of the same class, which are placed inside an LSP. Hence, all packets on a traffic trunk have the same label and the same 3-bit class of service (currently experimental) field in the MPLS header. Traffic trunks are routable objects like virtual circuits in ATM and Frame Relay networks. These trunks can be established either statically or dynamically (on demand) between any two nodes in an MPLS domain. In other words, an LSP can carry any aggregate of flows where each flow consists of packets belonging to a same class of QoS service. Figure 4.17 illustrates the relationships between the various abstractions that have been described above.

Figure 4.17 Relationships among Flows, LSPs and Links

PREMIUM FLOW represents Highest Priority Traffic

GOLD FLOW represents High Priority Key Traffic

Application Scenario

Figure 4.18 . Workstations Communicating under MPLS based Backbone Network

An example real-life application scenario is shown in Figure 4.18 and the key mechanisms and algorithm are discussed in the framework of this scenario, Figure 4.17 and example extended label tables. Two workstations (ws1& ws2) provide network control information such as signaling and routing through MPLS backbone network to possibly ws7 and ws8. Similarly, another two workstations (ws3 & ws4) send audio streams to ws9 and ws10. Assuming different traffic classes are defined at the sending workstations by marking packets. Two LSP between LER1 and LER4, one (LSP1) of a bandwidth (560kbps) and the other (LSP2) of a bandwidth (313 kbps) are considered for illustrative purpose.

The simple example of the label mapping table with the proposed extension to the LIB is shown in Table 4.2, Table 4.3 and Table 4.4. These tables shows the binding of two highest priority flows to the same LSP (LSP1) with LSP_ID 5 and two high priority key flows to the same LSP (LSP2) with LSP_ID 3.

Table 4.2 Example of a Modified Label Table of an FTN with Extension for MPLS Support at Ingress LER

FEC	INCOMING LABEL	LSP_ID	MPLS EXP VALUE	OUTGOING LABEL	END-to-END FLOW BW
Flow1 (Highest Priority)	7	5 (LSP 1)	111	21	560 kbps
Flow1 (Highest Priority)	7	5 (LSP 1)	111	21	560 kbps
Flow2 (High Priority)	9	3 (LSP 2)	110	36	313 kbps
Flow2 (High Priority)	9	3 (LSP 2)	110	36	313 kbps

Table 4.3 Example of a Modified Label Table of an ILM (Incoming Label Map) at Core LSR

INCOMING LABEL	FEC	LSP_ID	MPLS EXP VALUE	OUTGOING LABEL
21	Flow1	5 (LSP 1)	111	29
21	Flow 1	5 (LSP 1)	111	29
36	Flow 2	3 (LSP 2)	110	39
36	Flow 2	3 (LSP 2)	110	39

Table 4.4 Example of Label Table at Egress LER

INCOMING LABEL	FEC	LSP_ID	MPLS EXP VALUE	OUTGOING LABEL
29	Flow1	5 (LSP 1)	111	–
29	Flow 1	5 (LSP 1)	111	–
39	Flow 2	3 (LSP 2)	110	–
39	Flow 2	3 (LSP 2)	110	–

FTN : FEC to Next Hop Label Forwarding Entry (NHLFE)

FTN is a mapping from FEC to NHLFE , is implemented at ingress LER

It is interesting to consider the example of four flows of data packets entering an MPLS domain requiring the traffic engineering parameters of QoS as illustrated in Figure 4.18. Two packet flows from ws1 and ws2 are highest priority traffic (e.g., signaling and routing update). The other two flows from ws3 and ws4 are high priority key traffic (e.g, voice). These packets flows are classified into 2 separate FECs at the ingress LSR (LER1). Referring to Table 4.2, the label mapping associated with the flows are 7 and 9 respectively. The corresponding EXP values are 111 and 110. Label swapping must also be done, and the previous labels must be exchanged for 21 and 36. The ingress LER “encapsulates“ the packet in an MPLS header with the appropriate label and EXP value . Packets are treated according to EXP values and then forwarded along corresponding LSP. (eg., LSP1 and LSP2). Label swapping at the core LSR and egress LSR is shown in Table 4.3 and Table 4.4 respectively.

Another encouraging example scenario for proposed traffic aggregation and label swapping algorithm for providing scalable QoS is shown in Figure 4.19.

If 2 flows having the Same Traffic Class => Same FEC
=> Same LABEL
=> Same EXP value
=> Same LSP
=> AGGREGATION OF TRAFFIC FLOWS
OF THE SAME CLASS

Figure 4.19 Example Scenario of Scalable IP QoS using Traffic Aggregation and Label Swapping in an MPLS Domain

Benefits of label swapping algorithm and traffic aggregation mechanism of MPLS can be briefly summarized as follows:

The label swapping algorithm can be performed in hardware. Providing a faster mechanism and high-speed switching is achievable because the fixed length labels are inserted at the very beginning of the packet. Therefore, network performance is increased. Traditionally, router performance is limited by the need to look up the routing table for every IP packet. Label swapping algorithm of MPLS is simple and a highly efficient scheme which is based on a short label carried in the MPLS shim header for forwarding data packets, it also increases the performance of Internet Protocol (IP) routers.

By using label switching technique, packets of different flows can be labeled differently and thus received different QoS. Depending on the traffic characteristics, different LSPs could be created for packets with different CoS requirements. Network scalability is obtained by implementing aggregation of traffic flows of the same class inside LSP. The investigation suggests that in order to guarantee the quality of service (QoS) to a given flow, different types of flows should be isolated in different LSP so that even if a given flow increases its source rate, the different flows are unaffected by this. For the above scheme to be effective, the trunks or LSP need to be end-to-end, otherwise the advantages of isolation in other parts of the network are reduced or eliminated significantly. This label swapping and traffic aggregation technique provides Internet service providers (ISPs) precise control over traffic flows and make it possible to support traffic engineering and QoS guarantees. The advanced forwarding scheme (an enhancement to MPLS) support IP mobility and enables building IP based scalable VPN which allows ISP to offer VPN services by providing a simple, flexible, and powerful tunneling mechanism. The following section demonstrates the implementation of aggregation of flows, label swapping and tunneling concepts in multiple MPLS network segments for providing scalable IPQoS.

Tunneling in Two MPLS Network Segments

Figure 4.20 Concept of Tunneling

A unique feature of MPLS is that it can control the entire path of a packet without explicitly specifying the intermediate routers. It does this by creating tunnels through the intermediary routers that can span multiple segments. This concept is used in provisioning MPLS-based VPNs.

Consider the scenario in Figure 4.20. LERs (LER1, LER2, LER3, and LER4) all use BGP and create an LSP between them (LSP 1). LER1 is aware that its next destination is LER2, as it is transporting data for the source, which must go through two segments of the network. In turn, LER2 is aware that LER3 is its next destination, and so on. These LERs will use the LDP to receive and store labels from the egress LER (LER4 in this scenario) all the way to the ingress LER (LER1).

However, for LER1 to send its data to LER2, it must go through several (in this case two) LSRs. Therefore, a separate LSP (LSP 2) is created between the two LERs (LER1 and LER2) that spans LSR1 and LSR2. This, in effect, represents a tunnel between the two LERs. The labels in this path are different from the labels that the LERs created for LSP1. This holds true for LER3 and LER4, as well as for the LSRs in between them. LSP 3 is created for this segment.

To achieve this, the concept of a label stack is used when transporting the packet through two network segments. As a packet must travel through LSP 1, LSP 2, and LSP 3, it will carry two complete labels at a time. The pair used for each segment is (1) first segment, label for LSP 1 and LSP 2 and (2) second segment, label for LSP 1 and LSP 3.

When the packet exits the first network and is received by LER3, it will remove the label for LSP 2 and replace it with LSP 3 label, while swapping LSP 1 label within the packet with the next hop label. LER4 will remove both labels before sending the packet to the destination. Example scenario of Figure 4.21 demonstrates the implementation of aggregation of flows, label swapping and tunneling concepts in multiple MPLS network segments for providing scalable IPQoS.

Scalable IP QoS in Multiple MPLS Network Segments

Figure 4.21 Example Scenario for Aggregation of Flows and Label Swapping in Multiple MPLS Network Segments

Consider the example scenario of Figure 4.21. The source sends its data to the destination. Depending on the traffic characteristics, different LSPs are created for packets with different QoS requirements. Assuming that LERs (LER1, LER2, LER3 and LER4) all use CR-LDP and create an LSP between them (LSP1). CR-LDP follows setting explicit route ER-LSP. Thus equipped to follow the route using predetermined CoS (class of service) and QoS (quality of service) parameters.

LER1 is aware that its next destination is LER2, as it is transporting data for the source, which must go through two segments of the network. In turn, LER2 is aware that LER3 is its next destination, and so on. Using CR-LDP, these LERs will receive and store labels from the egress LER (LER4) all the way to the ingress LER (LER1). The LSPs that are established could be CBR-LSPs, where the constraints could be QoS requirements. Two LSP namely LSP2 and LSP3 are created between LER1 and LER2. LSP2 extends across LSR1 and LSR2. LSP3 consists of LSR3 and LSR4. These represent the tunnels between the two LERs. The labels in LSP2 and LSP3 are different from the labels that the LERs created for LSP1. This holds true for LER3 and LER4, as well as for the LSRs between them. LSP4 and LSP5 are developed for second MPLS network segment.

In Figure 4.21, ws1, ws2, ws3 and ws4 are sources and ws5, ws6, ws7 and ws8 are destinations. Source ws1 and ws2 are sending Premium traffic (highest priority) packets to destination ws6 and ws5 through two MPLS network segments. Source ws3 and ws4 are sending Gold traffic (high priority key) packets to destination ws8 and ws7 through two MPLS network segments.

To achieve this, the concept of a label stack is applied when transporting the packets through two MPLS network segments. In the scenario, highest priority packets are designated as flow1 and high priority packets are denoted as flow2. As a flow 1 packets, they must travel through LSP1, LSP2 and LSP4, it will carry two complete labels at a time. The pair used for each segment is (1) first segment, label for LSP1 and LSP2 and (2) second segment, label for LSP1 and LSP4. For two packet flows of the same traffic types, they would have the same label values, same LSP and thus implementing aggregation of the traffic flows. These concepts are illustrated in Figure 4.22. When the packets exit the first network and are received by LER3, it will remove the label for LSP 2 and replace it with LSP 4 label, while swapping LSP 1 label within the packet with the next hop label. LER4 will remove both labels before sending the respective packets to the corresponding destination. Similar situation is true for flow2 (high priority packets).

*If 2 flows having the Same Traffic Class => Same FEC
=> Same LABEL
=> Same EXP value
=> Same LSP
=> AGGREGATION OF TRAFFIC FLOWS
OF THE SAME CLASS*

For FLOW 1 (Premium : Highest Priority) Traffic

- ❖ **PATH : LSP 1 → LSP2 → LSP4**
- ❖ **FEC to 2 Complete Labels Mapping**

1 st NETWORK SEGMENT	LSP1 LABEL	LSP2 LABEL	Aggregate Packets of the same traffic flow
2nd NETWORK SEGMENT	LSP1 LABEL	LSP4 LABEL	Aggregate Packets of the same traffic flow

For FLOW 2 (Gold : High Priority Key) Traffic

- ❖ **PATH : LSP 1 → LSP3 → LSP5**
- ❖ **FEC to 2 Complete Labels Mapping**

1 st NETWORK SEGMENT	LSP1 LABEL	LSP3 LABEL	Aggregate Packets of the same traffic flow
2nd NETWORK SEGMENT	LSP1 LABEL	LSP5 LABEL	Aggregate Packets of the same traffic flow

Figure 4.22 Flows and Multiple MPLS Network Segments

4.7 Summary

This section summarizes the key observations regarding the profound capabilities of MPLS at architectural level to present-day IP networks. MPLS addresses today's network backbone requirements effectively by providing a standards-based solution that accomplishes the following:

MPLS improves packet-forwarding performance in the network since MPLS enhances and simplifies packet forwarding through routers using Layer-2 switching paradigms. It is simple, which allows for easy implementation. MPLS increases network performance because it enables routing by switching at wireline speeds. MPLS supports QoS and CoS for service differentiation since MPLS uses traffic-engineered path setup and helps achieve service-level guarantees. It incorporates provisions for constraint-based and explicit path setup. MPLS supports network scalability and integrates IP and ATM in the network. It provides a bridge between access IP and core ATM and it can reuse existing router/ATM switch hardware, effectively joining the two disparate networks. MPLS builds interoperable networks. It also facilitates IP-over-synchronous optical network (SONET) integration in optical switching. MPLS helps build scalable VPNs with traffic-engineering capability.

After all, MPLS enables the industry to offer an advanced forwarding scheme that provides the important capabilities to today's best effort IP networks such as traffic engineering (TE), classes of service (CoS), quality of service (QoS) provisioning, IP based virtual private networks (VPN's), IP mobility [enhancement to MPLS] and resource management [enhancement to MPLS].

In this chapter, the study evaluates that CR-LDP is regarded as a practical and powerful tool that can be used for traffic engineering and QoS routing. The novel method has been presented for providing guaranteed QoS and enhancing the network scalability. It has been observed that packets of different flows can be labeled differently and thus receive different QoS. Implementing aggregation of traffic flows of the same class inside LSP supports network scalability and establishing of end-to-end trunk inside LSP benefit for traffic engineering. Investigation shows that guaranteed QoS is achieved by isolating different types of flows in different LSP. Moreover, an enhancement to MPLS supports IP mobility. As a result, a significant number of Internet service providers and network operators are migrating towards MPLS as the transport option for IP services.

The next chapter of the thesis will present network architecture and protocol for mobility with continuous QoS support and handoff performance using MPLS based mobile access network. Efficient handoff schemes are cost effective of enhancing capacity and QoS of mobile access network. Therefore, the reliable handoff scheme is also established by developing five functional tasks.

CHAPTER 5 MOBILITY MANAGEMENT AND PROVISION OF QOS GUARANTEE IN IP-BASED MOBILE ACCESS NETWORKS USING MOBILITY AWARE MPLS SCHEME

5.1 Introduction

Due to the advent of high performance portable devices and the user demand for Internet, providing IP-based multimedia services over mobile environment becomes attractive. Provision of QoS in mobile networks is more complex than in fixed networks due to user mobility. Recently, there has been much interest on the provision of QoS guarantees in wireless and mobile networks. This increased interest has been fueled by major advances such as the increase in the number of portable devices, the increased performance of the portable devices enabling multimedia applications and the emergence of wireless access technologies that provide high data rates.

Of increasing importance is supporting QoS for mobile users accessing real time services over Internet. Next generation mobile networks offer the promise of high speed access as well as IP based data services to mobile hosts. Mobile IP [177] provides scalable wide-area mobility support. However, Mobile IP is not suited for localized mobility between pico-cells, especially for real time applications due to signaling overhead, handover latency and transient packet loss. These factors collectively called fast handover issue [162]. A number of micro-mobility schemes have been proposed over the last couple of years. These schemes address the fast handover issues with a good degree of success but lacking in the capabilities for QoS. A significant number of Internet service providers and network operators are migrating towards MPLS [178] as the transport option for IP services. Therefore, in this chapter, the thesis presents an efficient mobility in IP based mobile access networks using Mobility-Aware MPLS scheme which aims to provide desirable properties for mobility with capabilities of providing QoS.

Proposed framework presents efficient mobility with continuous QoS support and handover performance. The scheme is developed by combining the advantage of MPLS such as IP QoS support with the advantage of host-based micro-mobility schemes such as low-latency handoffs. Moreover, the scheme provides flexibility to support IPQoS model such as DiffServ model. An MPLS architecture is considered for mobile access network. Access MPLS administrative domain is connected to the core network. Mobility –Aware MPLS scheme is applied within the mobile access MPLS administrative domain for a mobility solution.

The main requirements specific to an IP based mobility management comprises hierarchical architecture to localize scope of location updates, supporting fast handoffs- packet redirection with minimal latency, paging functionality to minimize network signaling and conserve battery power and providing compatibility with QoS.

In mobile networking, a mobile host is provided continuous network connectivity through either a wireless or conventional wired network interface when the mobile user changes the computer's point of attachment to the Internet. All the needed reconnection occurs automatically. Mobile IP provides a simple and global mobility solution, but it has some deficiencies. In Mobile IP, a mobile user has to send an update message to its home agent for every subnet change (possibly every base station change in all-IP networks evolved from cellular networks.) Thus, it introduces significantly latency when used in a cellular environment simply because handoffs occur frequently and registration messages may travel large distances before packet redirection occurs. If the mobile node's visiting network is far away from its home network, the update signaling delay may be long and this results in disruption to user data during handoff. A number of mobility solutions have been proposed to overcome the limitations of Mobile IP [179,180,181]. The common approach of these schemes can be broadly classified into two types: Tunnel-based and Routing-based schemes [162]. This section briefly surveys key characteristic of these two types.

Tunnel-based Scheme

Examples of tunnel-based scheme include Mobile IP Regional Registration [179] and General Packet Radio System (GPRS) tunneling protocol (GTP) used in GSM networks. Figure 5.1 shows tunnel-based scheme. Tunneling based schemes use the idea of a gateway foreign agent (GFA) to create a hierarchical mobility structure. Mobile host registers the GFA as the foreign agent. GFA tunnels the packets destined to the host to the currently attached access router (AR). Mobility within the subnet of GFA does not require re-registration with remote home agent and is achieved by local registration with GFA. Thus, it may result the decreased in signaling overhead and latency. A forwarding entry for the mobile host is necessary only at the GFA and AR. Such schemes address the fast handover issues. They lack in flexibility because of rigidly imposed hierarchy and may create a single point of failure.

Routing-based Scheme

HAWAII [180] and Cellular IP (CIP) [181] are the examples of routing-based scheme. Robustness is provided by maintaining regular IP forwarding within the micro-mobility domain and also eliminate the tunnel overhead. Figure 5.2 shows a generic example. Each domain has a root node. Root node provides routable address and acts as a foreign agent for all the mobiles in its scope. Mobility between AR is handled by changing routing table entries on all the nodes in the path between the root and AR. Limiting scalability is that nodes need to maintain forwarding entries for all the mobiles. Moreover, every node in the domain has to implement the protocol.

Figure 5.1 Tunnel-based Scheme

Figure 5.2 Routing-based Scheme

All of the protocols described above extend Mobile IP and propose some form of hierarchy with gateway routers. These micro-mobility protocols reduce the traffic volume of signaling messages to a home agent provide fast handoff by limiting updates within a domain. While many of those proposals address the fast handover issues but lacking in flexibility and the capabilities for QoS provisioning even though some protocols that adopt a host-based routing mechanisms can be integrated with a resource reservation protocol. Therefore, there is a need for a specific mobility protocol with the ability of continuous QoS provision within the administrative domain.

The objectives for proposing Mobility-Aware MPLS are to adopt MPLS in these IP-based mobile access networks and to integrate mobility management with MPLS mobile access networks. Thus, the proposed strategy can take advantage of MPLS features such as IP QoS support while providing an efficient mobility paradigm.

5.2 Framework Overview

Figure 5.3 Hierarchical Mobility Management Framework of Mobile Access MPLS

Mobile hosts such as notebook and palmtop computers and portable workstations are now widely available and affordable, and the distinction between desktop workstations and portable computers is beginning to disappear in terms of both features and computational power. A mobile host may move from one network to another within the mobility domain at any time.

Figure 5.3 shows the hierarchical mobility management framework of mobile access MPLS. A domain-based approach is used to address hierarchical mobility management. Domain is defined as mobile access MPLS domain and it can be a campus network, a company network, a set of LANs. Each mobile access MPLS domain has one access gateway (AGW) which is connected to node visiting gateway (VGW). The administrative domain has one VGW through which the access MPLS network is connected to a core network. The proposed Mobility-Aware scheme is applied within mobile access MPLS administrative domain for a mobility solution.

For mobility between mobile access MPLS administrative domain, assume the core network is like Mobile MPLS [142] mechanism. When a mobile host visits a new mobile access foreign MPLS administrative domain, it sends a registration message with the IP address of the visited GW to the home GW. The home GW updates its forwarding table so that all the packets destined to the mobile host are rerouted to the visited GW using the LSP between the home GW(ingress LSR) and the visited GW(egress LSR).

5.2.1 Design Goals and Constraints

The ultimate goal of this chapter is to design a mobility management framework to integrate mobility management with MPLS based mobile access networks, fast reliable handovers by reducing signaling overhead and latency (caused by wide-area mobility protocol) and also transient packet loss associated with localized movement. In addition, consideration primarily aims to provide the capability for scalable QoS in the mobility domain.

5.2.2 Label Distribution Mechanism

Within a mobile access MPLS domain, LSP are formed between each base station (BS) and access GW (AGW) by provisioning. Multiple LSPs are created between a base station and access GW to support multiple FECs that can be mapped to signaling traffic and different classes of user data traffic to support QoS. Distributing label-binding information among LSRs can be accomplished using downstream label allocation as described in Chapter 4.

5.2.3 Deploying LSP Configuration in Mobile Access MPLS Domain

LSPs are created by a network operator based on provisioning policies. It is a scalable solution in supporting large number of mobile users since each LSP can be shared by many mobile hosts. The total number of LSPs in a mobile access MPLS domain is related to the number of traffic classes supported, since each LSP is shared by multiple mobile nodes whose data traffic needs to be treated with the same QoS policy. In order to make the proposed scheme be scalable to match of growth for the future mobile IP network with MPLS, the MPLS label switched path (LSP) is designed as multiple dimensions as shown in Figure 5.4.

This approach provides simple and fast handoff by updating the gate way's forwarding table while supporting the same QoS after handoff. The proposed LSP configuration is used to support the IP diff-serv (DS) model for supporting IP QoS services. The DS model divides traffic into a small number of classes and allocates resources on a per-class basis. Different per-hop behavior (PHB) groups are defined for the different classes of services. A Differentiated Services Code Point (DSCP) is carried in the Type of Service (ToS) byte of an IP packet header and used to select a PHB for scheduling and discarding at a router.

Figure 5.4 Multiple Dimension LSP Configuration with Mobility Aware MPLS on Scalable Access Network

To support the DS model, an MPLS network has to provide some way to ensure that packets marked with various DSCP receive the appropriate PHB at each LSR. The label distribution mechanism described above can be enhanced so that a label can be bound to both an FEC and PHB. At each edge switched router, mapping from a DS (diff-serv) class to an appropriate MPLS LSP is performed.

In a mobile access MPLS domain, a gate way (GW) performs the mapping for downlink traffic and each base station performs the mapping for uplink traffic. The thesis implements the above LSP configuration and therefore the detailed protocol description of Mobility Aware MPLS for such LSP configuration is demonstrated in the following section.

5.3 Network Architecture using MPLS

The proposed framework is currently limited to two levels of hierarchy in an administrative domain. The first level of the hierarchy is called the mobility across different domains or access gateway (AGW) and the second level is created for the mobility within a domain or an AGW. The approach is based on the deployment of one AGW in each domain and these AGW are connected to visiting gateway (VGW) for mobility management purpose. VGW performs the mapping for downlink traffic and each base station (BS) performs the mapping for uplink traffic. LSPs are setup between each AGW and VGW to support multiple FEC that can be mapped to signaling traffic and different classes of user data traffic to provide different QoS. Similarly, within a mobile access MPLS domain, multiple LSP are created between each base station and AGW to support different QoS for different traffic classes.

Access gateway (AGW) is a router that supports the mobile node (MN) roaming within its mobile access domain supporting IP QoS. The scalable LSP scheme is used to support the DiffServ model for providing IP QoS services. Example network configuration between end-to-end connections using Mobility Aware MPLS is shown in Figure 5.5. The proposed strategy uses scalable LSP configuration for the mobility management support as illustrated in Figure 5.6.

Figure 5.5. Example Network Configuration between End-to-End Connections using MPLS with Mobility Aware

Figure 5.6. Mobility Aware Scheme in Access MPLS Network

5.4 Protocol for Mobility on MPLS based Mobile Network

When mobile node (MN) powers up in a mobile access MPLS domain, it listens to beacon signals transmitted from base stations (BSs). MN sends registration message to the BS of strongest signal it detects. BS sends registration message to access GW (AGW) using pre-established LSP between BS and AGW. When AGW receives the registration message, it updates its forwarding table by associating MN's IP address with registration received interface. Then, AGW forwards the registration message to visiting GW (VGW) using pre-established LSP between A GW and VGW. VGW also updates its forwarding table by associating MN's IP address of mobile access MPLS domain with registration received interface.

Referring to Figure 5.3, if mobile access MPLS administrative domain is not the MN's home administrative domain, VGW assigns a virtual care-off address (VCoA) for the MN and registers it to the MN's home GW on its behalf using hop by hop normal IP routing as in Mobile MPLS. An LSP using CR-LDP [76] is established between home GW and visiting GW to satisfy the QoS requirements of the traffic. The home GW updates its forwarding table so that packets destined for the MN can be forwarded from the home GW to the current visited GW using CR-LSP between the home GW and the current visited GW. The registration process is illustrated in Figure 5.7. A virtual care-off address (VCoA) is an IP address that is assigned to the MN. This address stays constant and valid as long as the MN is roaming within the access administrative domain. This VCoA can be the address of the visiting GW or any address on the VGW sub-network. In terms of transport efficiency, IP-in-IP tunnels are not used between home GW and visiting GW. Instead, MPLS labels are employed for the same purpose.

Figure 5.7 Mobile Node Registration in Mobility Aware Access MPLS

Figure 5.8 Registration for Mobility between Different AGW

As illustrated in Figure 5.3 and Figure 5.6, when MN enters the coverage area of new BS under different domain (AGW), MN first sends the registration request to new BS. New BS (BS4) forwards the request to new AGW (AGW2). Then, AGW forwards this request to an VGW in the administrative domain. Figure 5.8 shows the registration procedure when MN handovers between different domain. In this case, MN does not require to reregister the home agent (HA) since MN is under the coverage area of administrative domain. Therefore, the benefits of such procedure are reducing delay, packet loss during handoff and eliminating registration between mobile hosts and HA when mobile hosts remain inside their local coverage area. Eliminating registration reduces the signaling load experienced by core network in support of mobility. Reducing signaling in this manner is necessary for the Mobile Internet to scale to support very large volumes of mobile users.

Example LSP Configuration for Illustration

Figure 5.9 shows a mobility with example LSP configuration. Assuming access MPLS network supports Diff-Serv (DS) Model, LSP are pre-established between each base station (BS) and access gateway (AGW) as well as each AGW and visiting gateway (VGW). VGW performs the mapping for downlink traffic and each base station performs the mapping for uplink traffic. In Figure 5.9, for illustrative purpose, six LSP are pre-established between BS1 and AGW1. LSP1, LSP2 and LSP3 are designated for uplink and LSP4, LSP5 and LSP6 are assigned for downlink. For BS1, LSP1 is used for common uplink signaling and LSP4 is used for common downlink signaling. LSP2 and LSP3 are used for uplink user traffic where as LSP5 and LSP6 are used for downlink user traffic.

Each LSP is mapped to one of the DS classes or used as a signaling LSP. For uplink, LSP2 is mapped to Assured Forwarding (AF) [94] and LSP3 is mapped to Expedited Forwarding (EF)[95]. For downlink, LSP5 is mapped to Assured Forwarding (AF) and LSP6 is mapped to Expedited Forwarding (EF).

Figure 5.9 Example of LSP Configuration in Mobility Aware MPLS Access Network

Table 5.1 shows the forwarding table of the AGW1 when the MN registers its IP address with the AGW1. It is assumed that MN enters the access MPLS domain and sends registration message to BS1. BS1 updates its forwarding table with MN's IP address (3.3.3.3). BS1 uses common uplink signaling LSP1 for forwarding the registration message to AGW1. AGW1 updates its routing table with MN's IP address and sends a registration message to VGW. VGW also updates its routing table and sends the registration message to MN's home GW if the access MPLS domain is not the MN's home domain.

By keeping MN's IP address with downlink signaling LSP (LSP4) of BS 1 as shown in Table 5.1 and LSP 59 as in Table 5.2, VGW knows AGW level information and also AGW knows BS level location information of the MN so that packets destined for MN can be forwarded to the corresponding BS (here BS1) correctly. MN's IP address in the forwarding tables at the VGW, AGW and BS1 is maintained as soft-state and it is refreshed by a periodic route update message from the MN.

Table 5.1 Forwarding Table of AGW1 after MN's Registration

Input		FEC	Output		Per-hop behavior (PHB)	Associated BS (Base Station)
I/F	Label		I/F	Label		
b1	1	-	a1		-	BS1 (LSP1)
b1	2	-	a1		AF	BS1 (LSP2)
b1	3	-	a1		EF	BS1 (LSP3)
a1	-	3.3.3.3	b1	4	-	BS1 (LSP4)
a1	-	-	b1	5	AF	BS1 (LSP5)
a1	-	-	b1	6	EF	BS1 (LSP6)
...	...	...	...	...	...	...
c1	7	-	a1		-	BS2 (LSP 7)
c1	8	-	a1		AF	BS2 (LSP8)
c1	9	-	a1		EF	BS2 (LSP 9)
a1	...	...	c1	10	-	BS2 (LSP10)
a1			c1	11	AF	BS2 (LSP 11)
a1			c1	12	EF	BS2 (LSP 12)
.....						
d1	13	--	a1		-	BS3 (LSP 13)
d1	14	--	a1		AF	BS3 (LSP14)
d1	15	--	a1		EF	BS3 (LSP 15)
a1			d1	16	--	BS3 (LSP16)
a1			d1	17	AF	BS3 (LSP 17)
a1			d1	18	EF	BS3 (LSP 18)

MN's IP address : 3.3.3.3

AGW : Access Gate Way

AF : Assured Forwarding (Loss sensitive & Delay Insensitive)

EF : Expedited Forwarding (Loss sensitive & Delay sensitive)

I/F : Interface

LSP : Label Switched Path

Table 5.2 Forwarding Table of VGW after MN’s Registration from AGW1

Input		FEC	Output		Per-hop behavior (PHB)	Associated AGW (access gateway)
I/F	Label		I/F	Label		
g1	56	-	a		-	AGW1 (LSP56)
g1	57	-	a		AF	AGW1 (LSP57)
g1	58	-	a		EF	AGW1 (LSP58)
a	-	3.3.3.3	g1	59	-	AGW1 (LSP59)
a	-	-	g1	60	AF	AGW1 (LSP60)
a	-	-	g1	61	EF	AGW1 (LSP61)
...	...	...	...	...	...	...
g2	62	-	a		-	AGW2 (LSP62)
g2	63	-	a		AF	AGW2 (LSP63)
g2	64	-	a		EF	AGW2 (LSP64)
a	...	...	g2	65	-	AGW2 (LSP65)
a			g2	66	AF	AGW2 (LSP66)
a			g2	67	EF	AGW2 (LSP67)
.....						
g3	68	--	a		-	AGW3 (LSP68)
g3	69	--	a		AF	AGW3 (LSP69)
g3	70	--	a		EF	AGW3 (LSP70)
a			g3	71	--	AGW3 (LSP71)
a			g3	72	AF	AGW3 (LSP72)
a			g3	73	EF	AGW3 (LSP73)

MN’s IP address : 3.3.3.3

VGW : Visiting GateWay

AF : Assured Forwarding (Loss sensitive & Delay Insensitive)

EF : Expedited Forwarding (Loss sensitive & Delay sensitive)

I/F : Interface

LSP : Label Switched Path

Packets Delivery

Packets destined for MN arrive at the VGW from MN's correspondent node CN. Assuming that DiffServ module is located at the CN and DiffServ Code Point (DSCP) of the packet has been set at the DiffServ module. When packets arrive VGW, VGW knows the PHB identified by the DSCP of the packet and assigns an appropriate label to the packet before putting it onto the corresponding LSP. At the same time, VGW updates its forwarding table.

Table 5.3 illustrates the example of the forwarding table at the VGW when it receives the first packet marked AF from the CN of mobile user. After the update, MN's IP address is associated with AGW1 and corresponding LSP 60. For all other packets marked AF coming from the CN to the MN, VGW assign the label "60" to them and put the packets onto the LSP 60.

When labeled packets arrive AGW1, it updates its forwarding table and label switching is performed and swaps with label "5" and place on "LSP5" as shown in Table 5. 4. All LSR on LSP5 examine the label header of each packet and PHB (AF) associated with the label determine how the packet is treated to support QoS. BS1, egress LSR for LSP5, removes the label of each packet and sends it to the MN. Similarly, when MN sends a packet to the CN, BS1 performs mapping from the DiffServ class of the packet to appropriate LSP. It is assumed that DS module is located at the MN and DSCP of the packet has been set at the DS module.

Table 5.3 Forwarding Table of VGW after Packet Arrival from CN

Input		FEC	Output		Per-hop behavior (PHB)	Associated AGW (access gateway)
I/F	Label		I/F	Label		
g1	56	-	a		-	AGW1 (LSP56)
g1	57	-	a		AF	AGW1 (LSP57)
g1	58	-	a		EF	AGW1 (LSP58)
a	-		g1	59	-	AGW1 (LSP59)
a	-	3.3.3.3	g1	60	AF	AGW1 (LSP60)
a	-	-	g1	61	EF	AGW1 (LSP61)
...	...	...	...	...	...	...
g2	62	-	a		-	AGW2 (LSP62)
g2	63	-	a		AF	AGW2 (LSP63)
g2	64	-	a		EF	AGW2 (LSP64)
a	...	...	g2	65	-	AGW2 (LSP65)
a			g2	66	AF	AGW2 (LSP66)
a			g2	67	EF	AGW2 (LSP67)
.....						
g3	68	--	a		-	AGW3 (LSP68)
g3	69	--	a		AF	AGW3 (LSP69)
g3	70	--	a		EF	AGW3 (LSP70)
a			g3	71	--	AGW3 (LSP71)
a			g3	72	AF	AGW3 (LSP72)
a			g3	73	EF	AGW3 (LSP73)

MN's IP address : 3.3.3.3

CN : Correspondent Node

AGW : Access GateWay

VGW : Visiting GateWay

AF : Assured Forwarding (Loss sensitive & Delay Insensitive)

EF : Expedited Forwarding (Loss sensitive & Delay sensitive)

I/F : Interface

LSP : Label Switched Path

Table 5.4 Forwarding Table of AGW1 after Packet Arrival from VGW

Input		FEC	Output		Per-hop behavior (PHB)	Associated BS (base station)
I/F	Label		I/F	Label		
b1	1	-	a1	-----	-	BS1 (LSP1)
b1	2	-	a1	-----	AF	BS1 (LSP2)
b1	3	-	a1	-----	EF	BS1 (LSP3)
a1	-		b1	4	-	BS1 (LSP4)
a1	60	3.3.3.3	b1	5	AF	BS1 (LSP5)
a1	-	-	b1	6	EF	BS1 (LSP6)
...	...	...	...	...	...	...
c1	7	-	a1		-	BS2 (LSP 7)
c1	8	-	a1		AF	BS2 (LSP8)
c1	9	-	a1		EF	BS2 (LSP 9)
a1	...	...	c1	10	-	BS2 (LSP10)
a1			c1	11	AF	BS2 (LSP 11)
a1			c1	12	EF	BS2 (LSP 12)
.....						
d1	13	--	a1		-	BS3 (LSP 13)
d1	14	--	a1		AF	BS3 (LSP14)
d1	15	--	a1		EF	BS3 (LSP 15)
a1			d1	16	--	BS3 (LSP16)
a1			d1	17	AF	BS3 (LSP 17)
a1			d1	18	EF	BS3 (LSP 18)

MN's IP address : 3.3.3.3

AGW : Access GateWay

VGW : Visiting GateWay

AF : Assured Forwarding (Loss sensitive & Delay Insensitive)

EF : Expedited Forwarding (Loss sensitive & Delay sensitive)

I/F : Interface

LSP : Label Switched Path

Handoff Technique

When MN moves within the same Mobile MPLS domain, no location update messages are sent to the remote home GW (gate way) via VGW. Hence, the proposed handoff procedure is able to provide lower handoff delay to the traffic heading to the MN during the handoff period. This technique leads to much better scalability than an approach based on Mobile IP. Maintaining the IP address of the MN unchanged across movements within the same mobile access domain results in straight forward support for QoS.

Figure 5.10 illustrates the handoff procedure within mobile access MPLS domain (AGW1). When the MN moves to the BS2's cell area, it initiates handoff by sending a route update message to BS2. BS2 forwards the route update message to AGW1 through LSP7 (common uplink signaling LSP between BS2 and the AGW1). AGW1 updates its forwarding table so that MN's IP address can be associated with LSP12 as shown in Table 5.5. After handoff, packets coming from CN are forwarded to BS2 along LSP12. In all LSR on LSP12, packets are treated with same PHB. Thus, the same QoS is maintained after handoff. Similarly, the same procedure is performed in AGW1 when MN moves from BS2 to BS3 and the forwarding table after binding update is shown in Table 5.6.

Figure 5.10 MN Handoff Procedure for Mobility between BS under same AGW

Table 5.5 Forwarding Table of AGW1 after Handoff from BS1 to BS2

Input		FEC	Output		Per-hop behavior (PHB)	Associated BS (base station)
I/F	Label		I/F	Label		
b1	1	-	a1	-----	-	BS1 (LSP1)
b1	2	-	a1	-----	AF	BS1 (LSP2)
b1	3	-	a1	-----	EF	BS1 (LSP3)
a1	-		b1	4	-	BS1 (LSP4)
a1	60	---	b1	5	AF	BS1 (LSP5) (before handoff to BS2)
a1	-	-	b1	6	EF	BS1 (LSP6)
...	...	...	...	...	...	...
c1	7	-	a1		-	BS2 (LSP 7)
c1	8	-	a1		AF	BS2 (LSP8)
c1	9	-	a1		EF	BS2 (LSP 9)
a1	...	...	c1	10	-	BS2 (LSP10)
a1	60	3.3.3.3	c1	11	AF	BS2 (LSP 11) (after handoff to BS2)
a1			c1	12	EF	BS2 (LSP 12)
.....						
d1	13	--	a1		-	BS3 (LSP 13)
d1	14	--	a1		AF	BS3 (LSP14)
d1	15	--	a1		EF	BS3 (LSP 15)
a1			d1	16	--	BS3 (LSP16)
a1			d1	17	AF	BS3 (LSP 17)
a1			d1	18	EF	BS3 (LSP 18)

MN's IP address : 3.3.3.3

AGW : Access GateWay

AF : Assured Forwarding (Loss sensitive & Delay Insensitive)

EF : Expedited Forwarding (Loss sensitive & Delay sensitive)

I/F : Interface

LSP : Label Switched Path

Table 5.6 Forwarding Table of AGW1 after Handoff from BS2 to BS3

Input		FEC	Output		Per-hop behavior (PHB)	Associated BS (base station)
I/F	Label		I/F	Label		
b1	1	-	a1	-----	-	BS1 (LSP1)
b1	2	-	a1	-----	AF	BS1 (LSP2)
b1	3	-	a1	-----	EF	BS1 (LSP3)
a1	-		b1	4	-	BS1 (LSP4)
a1	60	----	b1	5	AF	BS1 (LSP5)
a1	-	-	b1	6	EF	BS1 (LSP6)
...	...	...	...	...	...	...
c1	7	-	a1		-	BS2 (LSP 7)
c1	8	-	a1		AF	BS2 (LSP8)
c1	9	-	a1		EF	BS2 (LSP 9)
a1	...	...	c1	10	-	BS2 (LSP10)
a1	60	---	c1	11	AF	BS2 (LSP 11) (before handoff to BS3)
a1			c1	12	EF	BS2 (LSP 12)
.....						
d1	13	--	a1		-	BS3 (LSP 13)
d1	14	--	a1		AF	BS3 (LSP14)
d1	15	--	a1		EF	BS3 (LSP 15)
a1			d1	16	--	BS3 (LSP16)
a1	60	3.3.3.3	d1	17	AF	BS3 (LSP 17) (after handoff to BS3)
a1			d1	18	EF	BS3 (LSP 18)

MN's IP address : 3.3.3.3

AGW : Access GateWay

AF : Assured Forwarding (Loss sensitive & Delay Insensitive)

EF : Expedited Forwarding (Loss sensitive & Delay sensitive)

I/F : Interface

5.5 Performance Evaluation of LSP Configuration Scheme

In essence of mobility management, handoff performance is one of primary important issues. Handoff is a process of transferring the existing connectivity between mobile node (MN) and base station (BS) to its neighboring BS, when the MN moves across the cell boundary and enters into another neighboring BS. However, since the cell size in the next generation mobile networks will become even small, the handover between cells will become unavoidable. In the performance evaluation of mobility in mobile networks, the handoff call blocking probability is an important performance measurement, especially when the mobile cell is in small size such as micro-cell. In this section, the major performance metrics are investigated in terms of new arrival call blocking probability and handoff call blocking probability, which are denoted by Pr_N and Pr_H , respectively. The new arrival call blocking probability, Pr_N , is defined as the probability that a new arrival call cannot enter service due to unavailability of mobile channels. The handoff call blocking probability, Pr_H , is defined as the probability that an existing call is eventually not completed due to lack of mobile channel when handoff attempt is requested. However, in the proposed LSP configuration scheme, the handoff connection request is given higher priority than the new arrival call connection request to improve handoff related system performance that a certain number of mobile channels are reserved for handover exclusive use since it is widely believed that a mobile network must give higher priority to handoff requests.

The performance of the proposed LSP configuration scheme based on MPLS protocol is evaluated by simulation experiments. The simulation system is shown in Figure 5.4 (page 121), where each BS is able to provide sufficient mobile channels for the existing mobile connections and the size of packets generated by each mobile node is identical. Either uplink or downlink between each BS and access gateway (AGW) can simultaneously accumulates the bandwidth in support of up to C_N mobile connections, among which the bandwidth for C_H mobile connections is exclusively assigned for handoff calls in a cell,

and the remaining bandwidth for $C_N - C_H$ mobile connections are shared by both new arrival calls and handoff calls. When a new call arrives, if the number of existing mobile connections in this cell is greater than $C_N - C_H$, then this new arrival call is blocked. On the other hand, a handoff attempt is unsuccessful if there is no bandwidth available between the targeted BS and AGW.

Table 5.7 Simulation Parameters

Items	Value
C_N	30
C_H	1~5
T_C	120s
T_R	100~400s
λ_N	0.2~0.5 per sec

A discrete simulator is developed using the well-known handoff simulation model [56], where new call arrival is implemented as a Poisson process with an average rate λ_N . The call holding time is exponentially distributed with mean value of T_C , and cell residence time of each mobile node is also implemented as an independent random process with exponential distribution and a mean value of T_R , which can be used to indicate the mobility degree of mobile nodes in a cell. The simulation parameters are shown in Table 5.7.

All statistics of new arrival call blocking probability and handoff call blocking probability are estimated using the data obtained from M independent simulation runs. Since each simulation run depends on a particular stream of pseudo-random numbers to drive the simulation process, where the pseudo-random stream is generated by computer with a random seed number, the obtained results may typically vary from one simulation run to another.

To ensure the accuracy of the simulation results, the confidence intervals are calculated using independent replication method [119] as follows:

1. The simulation is independently repeated $M = 30$ times, and M groups of data are thus obtained. Each simulation run has a length of 10^8 slots excluding a warm up period of 1000 slots, which is setup to ensure that the results are obtained on the basis of a stable simulation process, because of each simulation run starts with an empty network environment. Actually, before the 1000 slots are decided as the warm up period, it has compared the results, which were obtained from the simulations runs using different warm up periods of 100, 500, 1000, 2000 and 5000 slots, respectively. The comparison has shown that a warm up period of 1000 slots or more is reasonable adequate.
2. The confidence interval δ is calculated as that at least 90 percent of estimated values for ρ obtained from these M simulation runs fall in the interval $(\rho - \delta, \rho + \delta)$, where $(\delta \ll \rho)$.
3. It should note that this performance evaluation is mainly devoted to the statistics of network mobility under high traffic density condition during the busiest hours of the day. This is feasible for relatively high new arrival call blocking probability and high handoff call blocking probability. In this case, the confidence intervals of the simulations are not difficult to be implemented. However, when these blocking probabilities are low, the simulations require an excessive amount of computing resources and the confidence interval becomes more difficult. The simulation experiments conducted for this research has not addressed that region.

Figure 5.11 and Figure 5.12 show the performance statistics in terms of new arrival call blocking probability and handoff call blocking probability for the proposed LSP configuration scheme under different traffic load conditions.

(a) $\lambda_N = 0.2$ per sec

(b) $\lambda_N = 0.5$ per sec

Figure 5.11 New Call Blocking Probability versus Number of Channels Reserved for Handoff Call

Figure 5.11 shows new call blocking probability Pr_N versus C_H , which is the number of bandwidth exclusively assigned for handoff calls in a cell. It can be seen that Pr_N increases with C_H when the average cell residence time T_R has a fixed value. This is because that the available bandwidth for new call decreases with the increase of C_H . Moreover, the offered traffic load changes with the new call arrival rate λ_N changes.

As shown in Figure 5.11(a), when the traffic load is $\lambda_N = 0.2$ per second, Pr_N increases with C_H following an approximately linear rate. When $C_H \geq 2$, Pr_N increases with the **decrease** of cell residence time T_R . On the other hand, for the traffic load case of $\lambda_N = 0.5$ per second as shown in Figure 5.11(b), when C_H increases, the increase rate of Pr_N is much less than that for the traffic load case $\lambda_N = 0.2$ per second. However, Pr_N increases with the **increase** of T_R . The difference between the values of Pr_N for variable T_R decreases with the increase of C_H . For example, when $C_H = 0$, Pr_N for $T_R = 400s$ is 10% higher than that for $T_R = 100s$; when $C_H = 5$, the values of Pr_N for $T_R = 400s$ and $T_R = 100s$ are almost same. Comparing Figure 5.11(b) to Figure 5.11(a), it can also be found that new call blocking probability significantly increases with λ_N when both T_R and C_H are kept as fixed values.

Figure 5.12 Handoff Call Blocking Probability versus Number of Channels Reserved for Handoff Call

Table 5.8 Simulation Results of Handoff Call Blocking Probability (Pr_H) for Load

Traffic $\lambda_N = 0.2$ per sec

number of BW assigned for handoff in the cell	cell residence time of mobile node (T_R)			
	$T_R = 100s$	$T_R = 200s$	$T_R = 300s$	$T_R = 400s$
0	3.08E-2±2.10E-3	3.46E-2±1.58E-3	3.62E-2±1.88E-3	3.71E-2±1.63E-3
1	1.59E-2±2.34E-3	1.24E-2±1.19E-3	9.95E-3±9.25E-4	8.27E-3±7.41E-4
2	8.03E-3±7.41E-4	4.36E-3±3.24E-4	2.685E-3±1.59E-4	1.811E-3±1.27E-4
3	3.932E-3±2.52E-4	1.49E-3±2.41E-4	7.07E-4±6.50E-5	3.87E-4±2.28E-5
4	1.867E-3±1.46E-4	4.96E-4±3.85E-5	1.81E-4±2.84E-5	8E-5±7.52E-6
5	8.56E-4±6.83E-5	1.59E-4±2.64.E-5	4.5E-5±3.35E-5	1.6E-5±1.24E-6

Table 5.9 Simulation Results of Handoff Call Blocking Probability (Pr_H) for Load

Traffic $\lambda_N = 0.5$ per sec

number of BW assigned for handoff in the cell	cell residence time of mobile node (T_R)			
	$T_R = 100s$	$T_R = 200s$	$T_R = 300s$	$T_R = 400s$
0	3.38E-1±1.91E-2	4.06E-1±3.18E-2	4.36E-1±3.13E-2	4.53E-1±3.92E-2
1	1.52E-1±1.32E-2	1.29E-1±1.51E-2	1.08E-1±1.22E-2	9.22E-2±9.5E-3
2	7.26E-2±6.82E-3	4.35E-1±3.74E-2	2.82E-2±1.52E-3	1.96E-2±2.11E-3
3	3.49E-2±2.55E-3	1.46E-2±2.21E-3	7.22E-3±6.62E-4	4.06E-3±3.35E-4
4	1.65E-2±1.45E-3	4.74E-3±3.53E-4	1.79E-3±2.79E-4	8.11E-4±7.46E-5
5	7.57E-3±6.33E-4	1.485E-3±2.75E-4	4.25E-4±3.64E-5	1.55E-4±1.41E-5

Figure 5.12 shows handoff call blocking probability Pr_H versus C_H for different values of cell residence time T_R . The curves of Pr_H shown in Figure 5.12(a) and Figure 5.12(b) have the similar trend with the change of C_H . In other words, for both the traffic case of $\lambda_N = 0.2$ per second and the traffic load case of $\lambda_N = 0.5$ per second, Pr_H always decreases with the increase of C_H . This is because that the bandwidth reserved for handoff calls increases with C_H . Comparing Figure 5.12 with Figure 5.11, it can be found that for fixed values of T_R and λ_N , Pr_H has the same value of Pr_N when $C_H = 0$, since both handoff calls and new calls are treated same in this case. However, when $C_H \geq 1$, Pr_H is much smaller than Pr_N . For example, when $T_R = 100s$, $C_H = 4$ and $\lambda_N = 0.5$ per sec, the value of Pr_N is more than 25 times of Pr_H . Moreover, it is obvious that there is tradeoff between handoff call blocking probability and new call blocking probability. In order to decrease Pr_H , we can increase the value of C_H , but Pr_N increases accordingly. From Figure 5.12, the study shows that Pr_H increases with the decrease of T_R when $C_H \geq 1$, i.e., the increase of mobile node moving rate will increase handoff call blocking probability and the number of forced call terminations.

The simulation results clearly indicate the advantage that the proposed LSP scheme offers lower handoff call blocking probability (i.e., significantly improves probability of successful handover) under different traffic load scenarios. Therefore, the system can reduce packet loss during handoff. After handoff, continuous QoS can be provided to mobile users by using the pre-established LSP between base station (BS) and access gateway (AGW). The scheme does not require to make re-registration between mobile node (MN) and access GW (AGW) when MN moves to another BS as long as the MN remains inside their local coverage area. Eliminating registration reduces the signaling load experienced by core network in support of mobility. Reducing signaling in this manner is necessary for the mobile network to scale to support very large volumes of mobile users, thus resulting in scalability improvement.

The simulation experiments show that under the traffic load case of $\lambda_N = 0.2$ per second, when $C_H \geq 2$, new call blocking probability increases with the decrease of cell residence time T_R . On the other hand, for the traffic load case of $\lambda_N = 0.5$ per second, when C_H increases, the increase rate of Pr_N is much less than that for the traffic load case of $\lambda_N = 0.2$ per second. However, Pr_N increases with the increase of T_R . Moreover, the performance study points out that the new call blocking probability largely increases with λ_N when both T_R (cell residence time of mobile node) and C_H (the number of bandwidth exclusively assigned for handoff calls in a cell) are kept as fixed values.

The dropout performance of new call can be enhanced by reserving dynamically adjustable number of channels exclusively for handoff requests. Such approach requires the determination of an optimum number of guard channels, knowledge of the traffic pattern in the area, the existing connections in the neighboring cells, and estimation of the channel occupancy time distributions. This will enable the BS to approximately reserve the actual amount of resources for handoff requests and thereby accept more new connection requests as compared to a fixed scheme. Another alternative solution to improve the dropout performance significantly is the combination of reserving channels for handover and queuing of handover requests but it is possible in larger cells because of queuing new calls result in increased handoff blocking probability and the effectiveness of queuing decreases for small cell size.

Overall, based on the investigation of major performance measures conducted in the simulation experiments, the scheme is well suited for supporting good degree of success on mobility with continuous QoS provision in IP based mobile access networks.

5.6 Trends of Handover in Cellular Systems

5.6.1 Handover Basics

Handoff is an essential mechanism of cellular communication. Handover function transfers an ongoing traffic from one cell to another as a user moves through the coverage area of a cellular system. If a mobile unit moves out of range of one cell and into the range of another during a connection, the traffic channel has to change to one assigned to the base station (BS) in the new cell. The system makes this change without either interrupting the traffic flow or alerting the user. As small cells are deployed to meet the demands of increased capacity, the number of cell boundary increases. Each handover requires network resources to reroute the call to the new BS. If handover does not occur quickly, QoS may degenerate below an acceptable level. Efficient handoff schemes are cost effective of enhancing the capacity and QoS of cellular system.

5.6.2 Handover Criteria and System Impact

The handover process would ideally be triggered by deteriorating communication quality to user. Variables in the handover criteria process can be classified into three spheres [185] :

- (i) Basic channel quality and mobile location assessment criteria. For example, received signal strength, distance and bit error ratio .
- (ii) Control mechanisms to limit the numbers of handovers, such as threshold level and hysteresis window.
- (iii) Parameters influencing the time for a handover decision are sample interval size, averaging period and detection time

Current systems employ channel quality assessment based on received signal strength indicator (RSSI) level and signal to noise ratio (S/N). The handover criteria used in current cellular systems are based on received signal strength combined with a basic measure of channel quality.

5.6.3 Handover Initiation Control

Handover Initiation

Figure 5.13 [186] shows a mobile node (MN) moving from one base station (BS1) to another base station (BS2). The average signal strength of BS1 decreases as the MN moves away from it. However, the averages signal strength of BS2 increases as the MN approaches it.

Figure 5.13 Trends in Handover Process

Using the Figure 5.13 shown, the following discussions explain some interesting approaches.

Relative signal strength Method

The relative signal strength method chooses the strongest received BS at all times. The decision is based on averaged measurement of the received signal. In Figure 5.13, the handoff will occur at position X.

Relative signal strength with threshold

This approach allows a user to handover only if the current signal is sufficiently weak (less than a threshold). The effect of the threshold depends on its value compared to the signal strengths of two BS at the point at which they are equal (say T). If the threshold T1 is higher than this value T in Figure.5.10, handover occurs at position X. If the threshold (T2) is lower than this value (T), the MN will delay handover until the current signal level crosses the threshold at position Y. The lower the threshold than T, the longer the delay may result in a dropped call. Its effectiveness depends on prior knowledge of the crossover signal strength between the current and target BS.

Relative signal strength with hysteresis

This method allows the user to handover only if the new BS is sufficiently stronger (by a hysteresis, h, in Figure.5.13) than the current one. In this case, handover will occur at point Z. Actual handover times from 250 to 500ms, with the total time to complete the handover sequence typically approximately 20 seconds [185].

Therefore, to efficiently use the frequency resources and to minimize the percentage of the connection time wasted by the handover process, both handover numbers and the time required to action each one must be minimized. To manage the expected increase in subscriber density in future mobile system, microcells must be used. The microcellular base station is mounted on the tops of small buildings or the sides of large buildings.

Table 5.10 Typical Parameters for Macrocells and Microcells

Parameters	Macrocell	Microcell
Cell radius	1 to 20 km	0.1 to 1 km
Transmission power	1 to 10 W	0.1 to 1 W
Average delay spread	0.1 to 10 μ s	10 to 100 ns
Maximum bit rate	0.3 Mbps	1Mbps

Microcells are useful in city streets in congested areas, along highways, and inside large public buildings. Table 5.10 [203] suggests typical parameters for traditional cells, called macrocells and microcells with current technology . The average delay spread refers to multipath delay spread (i.e., the same signal follows different paths and there is a time delay between the earliest and latest arrival of the signal at the receiver). As indicated, the use of small cells enable the use of lower power and provides superior propagation conditions . It has been observed that the handover request probability in city microcells differed significantly with respect to locations. This implies that the radiowave propagation characteristics are highly dependent on the local terrain features and environments.

Initiation Control

Control parameters include hysteresis margin, threshold level, averaging interval, and window shape. With the averaging interval fixed, results report on the tradeoff between delay and the expected number of handovers. Figure 5.14 shows the method of reporting these results [186]. The y-axis shows the average number of handovers that occur while a MN is moving from BS1 to BS2 in Figure.5.13. The x-axis shows the location of the first handover between BS1 and BS2, where 0.50 indicates that it occurs at the halfway point and 0.75 means the MN is 75 percent of the way to BS2.

For a particular set of values for the hysteresis margin, h , and the threshold level, T , the system will operate at the point marked A. The horizontal distance from $x=0.5$ and the location of A is a measure of the handover delay.

Figure 5.14 shows that as either the hysteresis margin is increased or the threshold level lowered, the mean number of handover decreases as the expected delay increases. When the hysteresis level is small for the considered parameters, the threshold has a greater influence on the expected number of handovers and crossover point.

Figure 5.14 The Trade-off between the Mean Number of Handovers and the Location of the First Handover (fraction of the distance between two identical base stations) is Controlled Jointly by the Hysteresis Margin, h , and the Threshold level, T .

5.6.4 Handoff Prioritization

One of the ways to reduce the handoff failure rate is to prioritize handoff [189]. Channel assignment strategies with handoff prioritization have been proposed to reduce the probability of forced termination [190, 191]. Two generic handover prioritization schemes are described as guard channels (reserving a number of channels exclusively for handovers) and queuing of handoffs (queuing handover requests).

Guard Channels

Such scheme improves the probability of successful handoff by reserving fixed or dynamically adjustable number of channels exclusively for handoffs. For example, priority can be given to handoff by reserving H channels for handoff among C channels in the cell [192]. The remaining $(C-H)$ channels are shared by both new calls and handoff calls. Handoff fails if no channel is available in the target cell.

An adaptive number of guard channels can reduce the problem of underutilizing spectrum. Efficient usage of guard channels require the determination of an optimum number of guard channels, knowledge of the traffic pattern of the area, and estimation of the channel occupancy time distributions.

Queuing of Handoff

Queuing is a way of delaying handoff [195]. Queuing new calls result in increased handoff blocking probability. The effectiveness of queuing decreases for microcells (relatively small cell size) due to stricter time requirements. The combination of queuing and channel reservation can be employed to improve dropout performance significantly [193].

Prioritization of Handover Requests Scheme

Queuing handover requests, with or without reserving channels for handovers, is another method of reducing the probability of forced termination at the expense of an increased call blocking probability and decrease in the ratio of carried-to-admitted traffic. The reason is that no originating call is granted a channel before the handover requests in the queue are served. Queuing of handover requests is made possible by the existence of the time interval the MN spends in the handover area, where it is physically capable of communicating with both the current and the next base station (Figure 5.15). The fact that a successful handover can take place any where during this interval marks a certain amount of tolerance in the delay for the actual channel assignment to the handover request [189].

Figure 5.15 Queuing of the Handovers during the “Degradation Interval” the Period in which the MN spends in the Handover Area where the Received Power Level is between the Handover and Receiver Thresholds.

5.6.5 Handoff Protocols

Table 5.11 Characterization of Different Handoff Protocols

TYPE	HANDOFF PROCESS	HANDOFF TIME	EXAMPLE APPLICATION	REMARK
Network-Controlled Handoff (NCHO)	Network makes a handoff decision based on measurements of the RSS of the MS(mobile station) at a number of BS (base station)	According to [198], overall delay can be the order of 5-10s	1 st generation analog system AMPS	This type of handoff is not suitable for a rapidly changing environment & high density of users due to associated delay.
Mobile-Assisted Handoff (MAHO)	MS makes measurement & BS makes decision.	According to [198], delay can be 1s	GSM	This delay may be too much to counteract the corner affect.
Mobile-Controlled Handoff (MCHO)	MS completely controls the handoff process & decision is made by MS.	Handoff time is approximately 100ms.	Cellular IP, GPRS, European standard for cordless telephones, DECT.	MCHO is the highest degree of handoff decentralization . Advantage is that handoff decision can be made fast.

Handover protocols describe how handoff procedure is implemented. Decision making process of handoff may be centralized or decentralized (i.e., handoff decision may be made at the MN or BS). Different systems use different approaches for handoff process and handoff protocol characterize these approaches. These concepts are shown in Table 5.11. In the Mobile Assisted Handoff and Mobile Controlled Handoff category, it can be further identified as backward handover (handover initiation via the existing channel) and forward handover (handover initiation via the target channel).

Backward Handover

Backward handover is to initiate the handover process via the existing communication link .Advantage is the signaling information may be transmitted through the existing radio channel .Thus, “new“ channel for handover signaling is not required during the initial stage of the handover process. This technique has been adopted by the 2nd generation pan-European GSM system. Disadvantage of this technique is in situations where the existing channel is rapidly deteriorating, the channel may degrade to the point that it can not maintain sufficient quality (BER) for the signaling channel. The worst case scenario will be the complete loss of both the signaling link and the traffic channel. In a cellular radio environment where the existing communication link is likely to degrade quickly, a backward handover protocol is usually not the best option [189].

Forward Handover

In contrast to backward handover, forward handover is to initiate the handover process via an optimum channel of the target base station. Advantage is in the event that the quality of the existing communication link deteriorates rapidly, the signaling channel and the traffic channel can be re-established (with a high success rate) with the target base station. In other words, the overall reliability of the handover process could be better than using a backward handover protocol under certain circumstances [189].

5.7 The Reliable Handoff Scheme

The handoff of IP packet flow from one cell to the other in a cellular system enables mobile communications to continue across the coverage areas of different base stations and hence the reliable handoff scheme is an extremely important system feature for mobile environment. In this section, the reliable handoff scheme is provided based on the following five functional tasks.

STEP 1 : Detection of Trend of Handoff

- (1) Handover initiation criterion*
(detection of received signal strength, RSS and bit error ratio, BER)
- (2) Threshold level for handover*

STEP 2 : Preparation of Handoff is performed by the following procedure

- (1) Channel allocation strategy & handoff prioritization*
- (2) LSP & Label reservation prior to the handover initiating process*

STEP 3 : Implementation of Handoff

- (1) Handoff decentralization with Forward Handover Protocol*
- (2) Handoff Procedure described in section 5.4*

STEP 4 : Completion of Handoff Process

- (1) Changing to reserved MPLS Label*
- (2) Changing to pre-allocated LSP*

STEP 5 : Ensure the handoff is completed (Position Location)

STEP 1 : Detection of Trend of Handoff

A handoff process is usually initiated either by the deteriorating communication quality of the current channel or a “new” channel with better communication quality being identified. This quality is assessed by one or more quality parameters such as received signal strength (RSS) and bit error ratio (BER)[185]. The handover decision has to be taken based on detected RSS and BER measurements taken at regular intervals. In any intercell handover, the received signal level is the most fundamental handover initiation criterion.

Theoretically, the best threshold level is the average signal level at the mid-point between two cell BS as this would offer equal chance for handover initiation and completion in both directions. However, as the path loss differs among cell base sites in a city, only an estimated threshold could be used [188].

The investigation shows that the appropriate initiation threshold for a specific handover criteria is location dependent. Due to the variations in terrain features, the shape of the signal profiles will vary from location to location. All these signal profiles will respond differently to a specific handover initiation threshold [188].

The handover process is initiated by the issuing of a handover request when the power received by the MN from the BS of neighboring cell exceeds the power received from the BS of the current cell by a certain amount. This is a fixed value called the handover threshold. For successful handover, a channel must be granted to the handover request before the power received by the MN reaches the receiver threshold. The handover area is the area where the ratio of received power level from the current and the target BS's is between the handover and the receiver threshold. Such scenario is shown in Figure 5.15. If the power level from the current BS falls below the receiver threshold prior to the MN being assigned a channel by the target BS, the call is terminated; therefore the handover attempt fails.

Figure 5.15 Handover during the “ degradation interval “ , the period in which the MN spends in the handover area where the received power level is between the handover and receiver thresholds.

This section discusses the mechanism for triggering off a handover request in order that appropriate handover criteria and threshold values may be selected.

RSS Handover Criteria & Threshold Value

The received signal for the current BS decreases monotonically as the MN moves away from the BS. As shown in Figure 5.16 [188], suppose that both BS are symmetrical with respect to the BS characteristics and transmit power, the mobile unit will experience a similar signal level, T_s , at a location half-way between the two BS. If the handover threshold is set at T_s (optimum threshold level), handover request will be uniquely controlled by the two signal profiles. If the threshold level is set above optimum threshold level T_s , handover will be delayed. Distance delayed will be dependent on the gradient of the signal profile and the difference between the chosen threshold level and the optimum level T_s .

Figure 5.16 The Threshold Assignment for Handover for Symmetrical Signal Profiles

Furthermore, the handover location will be dependent of the signal level received from the target BS but independent of the corresponding level from the current BS. This is because the threshold level of the current BS will already be exceeded before the signal level of the target BS is satisfied. If the threshold level is set below optimum threshold level T_S , handover initiation will be determined by the RSS from the current BS. This is because the threshold of the target BS will be satisfied before the current threshold level is attained. Thus, handover will be delayed beyond the optimum – half way handover initiation point.

BER Handover Criterion and Threshold Level

By contrast, if BER is used as a handover initiation criterion, a high received signal level will correspond to a low BER and vice-versa. If the BER profile are symmetrical, the optimum threshold level will be the cross-over BER level between the profiles. Specifically, setting a threshold level above or below the optimal threshold will delay the handover initiation. For the assignment of BER threshold level, independent BER threshold levels can be set for the current and the target BS. The effect will depend entirely on their relative magnitude and the shape of the profiles.

Hysteresis and Pre-mature Handover

Implementing hysteresis could delay the handover initiation process . However, there are some circumstance where a delay of the handover from the cross-over point of the signal profile is desirable. For example, in the situation of asymmetrical signal level profiles as shown in Figure 5.17a [188], if the cross-over point is too pre-mature, the setting of hysteresis window and the threshold level appropriately could defer the handover request to a more optimal position. Another example is for the case when the transmit power for the two BS are asymmetrical as illustrated in Figure 5.17b [188]. In this circumstance, the setting of a threshold level above or below the cross-over value will be more appropriate according to the actual circumstance.

Figure 5.17 Premature Handover due to Asymmetrical Signal Profiles

The Figure 5.18 demonstrates the handover request flow for the implementation of Step 1 in the proposed reliable handoff scheme.

Figure 5.18 Handover Request Flow Diagram

BS : Base Station
P : Power
RSS : Received Signal Strength
Ts : Threshold level

STEP 2: Preparation of Handoff is performed by the following procedure*(1) Channel allocation strategy & Handoff Prioritization*

The mobile nodes (MNs) in a cell are served by a base station. Before a MN can communicate with other MN in the network, a connection must usually be established between the MN and BS. The establishment and maintenance of a connection in a mobile network is the responsibility of the BS. The resource sharing algorithm is used by each BS to allocate wireless bandwidth to every connection based on its QoS requirements and the QoS requirements of other connections sharing the same BS. The scheduling algorithm is performed in a decentralized fashion and ensures that BS capacity is efficiently shared among different classes of connections. It is assumed that QoS to different classes of traffic is provided using a highly reliable radio link between the mobile terminals and its access point. This can be done for example by using array antennas and optimal combining [199, 200]. When a mobile user moves from one cell to another, the BS in the new cell must take responsibility for all the previously established connections. A significant part of this responsibility involves allocating sufficient resources in the cell to maintain the QoS needs of the established connections. If sufficient resources are not allocated, the QoS needs may not be met, which in turn may result in premature termination of the connection. Since premature termination of established connections is usually more objectionable than rejection of a new connection request, it is widely believed that a wireless network must give higher priority to the handoff connection requests as compared to new connection requests [205]. Hence, the system must provide rapid handoff, and signaling for resource management purposes [201].

In order to improve the successful handoff probability, a number of channels need to be reserved for handover exclusive use. Figure 5.19 shows the resource sharing mechanism used by each base station (BS) in the proposed scheme in order to provide priorities to handoff requests without significantly jeopardizing the new connection requests.

Figure 5.19 Resource Sharing Mechanism by each BS

Figure 5.20 Call Flow Diagram for Channel Reservation based Handoff Prioritization

The strategy designates the total base station wireless capacity by C channels in the cell. Priority is given to handoff by exclusively reserving H guard channels for handoff among C channels in the cell. Thus, handoff (voice or data) can access H channels. The remaining $C - H$ channels are shared by new traffic and handoff traffic flows.

Target BS provides priority of handover requests over new call attempts. Hence, originating calls (new calls) are not assigned channels until the requested handovers are served. A new call is blocked if the number of available channels is at most equal to H channels. Reserving channels H for handovers means less channels are being granted to new call attempts so that total carried traffic reduced. In order to overcome this disadvantage and increase the carried traffic load, new traffic attempts, which are considerably less sensitive to delays than handovers, are allowed for queuing in the priority queue with finite buffer length, L in the target BS if $C-H$ channels of a cell are occupied. These concepts are demonstrated in Figure 5.20.

Moreover, upon arrival of handoff requests to target BS, priority is given to voice handoff over data handoff to avoid forced termination for voice users. Resources are allocated in such a way that when MN handoffs from one BS to another, resources of BS are available with an acceptably high probability for requested handoff traffic. Assuming that control functions of the network are simple and high rate of handoff events can be accommodated.

In the light of this scenario, the investigation suggests the combination of reserving channels for handover and queuing of handover requests to improve the dropout performance significantly. Queuing of handovers is possible in larger cell because the MN spends some period of time in the handover area, during which its communication with the current BS degrades at a rate depending on its velocity. This degradation is easily monitored by means of radio channel measurements, usually taken by the MN and submitted to the network. Such a handoff prioritization scheme is proposed and evaluated in Reference [208, 209]. Another desirable strategy is for each BS to dynamically adapt the capacity reserved for dealing with handoff requests based on the current connections in the neighboring cells. This will enable the BS to approximately reserve the actual amount of resources for handoff requests and thereby accept more new connection requests as compared to in a fixed scheme. Such dynamic strategies are proposed and evaluated in [206, 207].

(2) LSP & Label reservation prior to the handover initiating process

Figure 5.4. Multiple Dimension LSP Configuration with Mobility Aware MPLS on Scalable Access Network

Revisiting to Figure 5.4 and Label Distribution Mechanism described in 5.2.2, the scheme briefly explains how LSP configurations are created and Labels are reserved as follows.

Using CR-LDP, multiple LSP are pre-established between each base station (BS) and access gateway (AGW) to support different QoS (quality of service) for different classes of traffic. Each LSP is mapped to one of the DiffServ classes or used as a signaling LSP. Moreover, LSPs are reserved between each AGW and visiting GW(VGW). When correspondent node (CN) of mobile user sends packets to mobile node (MN), VGW knows the PHB identified by the DiffServ Code Point (DSCP) of the packet and assigns an appropriate label reserved for the packet and then places onto the corresponding reserved LSP. Similarly, when MN sends packet to CN of mobile user, BS1 performs mapping from the DiffServ class of packet to corresponding label and LSP.

Table 5.12 Example LSP & Label Reservation between BS & AGW prior to Handoff

Associated BS (base station)	Reserved LSP (label switched path)	Type of Traffic	PHB (per-hop- behavior)	Reserved Label
BS1	LSP(1, 1)	Uplink Signaling	-	1
	LSP(1, 2)	Uplink Traffic1	AF	2
	LSP(1, 3)	Uplink Traffic 2	EF	3
	LSP(1, 4)	Downlink Signaling	-	4
	LSP(1, 5)	Downlink Traffic 1	AF	5
	LSP(1, 6)	Downlink Traffic 2	EF	6
BS2	LSP(2, 1)	Uplink Signaling	-	7
	LSP(2, 2)	Uplink Traffic1	AF	8
	LSP(2, 3)	Uplink Traffic 2	EF	9
	LSP(2, 4)	Downlink Signaling	-	10
	LSP(2, 5)	Downlink Traffic 1	AF	11
	LSP(2, 6)	Downlink Traffic 2	EF	12
BS3	LSP(3, 1)	Uplink Signaling	-	13
	LSP(3, 2)	Uplink Traffic1	AF	14
	LSP(3, 3)	Uplink Traffic 2	EF	15
	LSP(3, 4)	Downlink Signaling	-	16
	LSP(3, 5)	Downlink Traffic 1	AF	17
	LSP(3, 6)	Downlink Traffic 2	EF	18

- AF : Assured Forwarding (Loss sensitive & Delay Insensitive) (e.g., email)
- EF : Expedited Forwarding (Loss sensitive & Delay sensitive) (e.g., voice)

STEP 3: Implementation of Handoff

- (1) *Handoff de-centralization with Forward Handover Protocol*
- (2) *Handoff Procedure described in section 5.4*

The decision making process of the handoff may be Mobile Assisted Handoff (MAHO) (centralized) or Mobile Controlled Handoff (MCHO) (decentralized). For example, GSM uses MAHO since mobile station makes signal measurements and the network makes the decision. On the other hand, Cellular IP explicitly states that handoff is initiated by the mobile host [175], hence supports mainly MCHO. In GPRS, a mobile station autonomously initiates a cell reselection process. The mobile station informs the network when it re-selects another cell [200]. Thus, GPRS also supports MCHO.

From handoff decision point of view, Mobile Controlled Handoff (MCHO) is also applied in the proposed reliable handoff scheme due to the reasons explained below. Fast handover is an important issue because, if handover does not occur quickly, QoS may degenerate below an acceptable level. MCHO would speed up the handover process and appropriate for microcells. In MCHO protocol, mobile node (MN) measures the signal strength (SS) from surrounding BS. Handoff is initiated if the SS (signal strength) of serving BS is lower than that of another BS by a certain threshold. MN requests the target BS for a channel with lowest interference. Handoff decision is made by MN and hence, highest degree of handoff decentralization is achieved. The advantage is that handoff decision can be made fast. By using MCHO, handoff time is approximately 100ms. Therefore, Mobile Controlled Handoff (MCHO) is used in the proposed scheme. In cellular mobile system, where the existing communication link is likely to degrade quickly, thus forward handover protocol (handover initiation via new channel of target BS) could be more reliable in some microcell handover situations. Therefore, the reliable scheme adopts MCHO with forward handover protocol and handoff procedure described in section 5.4 to ensure the reliability within the mobile access network.

STEP 4 : Completion of Handoff Process

- (1) Changing to reserved MPLS Label
- (2) Changing to pre-allocated LSP

Figure 5.21 An Illustrative Example of Changing to Reserved Label & LSP after Handoff to Base Station 2 (BS2)

Table 5.13 Forwarding Table of AGW1 after MN Handoffs from BS1 to BS2

<i>Input Label</i>	<i>FEC</i>	<i>Output Label</i>	<i>PHB</i>	<i>LSP</i>	<i>BS</i>	<i>Handoff</i>
60	-----	5	AF	LSP1	BS1	Before handoff
60	3.3.3.3	11	AF	LSP11	BS2	After handoff

STEP 5: *Ensure the handoff is completed (Position Location)*

Information of position location may help to make ensure the completeness of handoff. Since handover is one of the major parameters that usually influence the stability of a cellular network, the maintenance of accurate location-related information can lead to a more effective adjustment of the handover parameters and the scheme itself. As a result, some working assumptions on the available location accuracy will have to be made. These assumptions will be established on a location techniques evaluation. Standard methods, such as GPS-assisted may be considered. In our scheme, handoff request is issued at the optimum position (i.e., halfway between two base stations, BS1 & BS2).

For successful handover, channel must be granted to the handover request before the power received by the MN reaches the receiver threshold. Hence, after performing the proposed procedures and reliable scheme mentioned above, MN receives the same IP QoS packet from BS2 after handoff. The completeness of successful handoff process should occur within the handoff area, some distance threshold beyond optimal distance and before crossing the Cell 1 boundary at which signal transmission range of BS1 is minimum. MN ensures the handoff completeness by informing BS1 that it has already made the connection setup with BS2 and receiving the packets from CN of mobile user via BS2.

5.8 Summary

In this chapter, the thesis presents the Mobility-Aware MPLS scheme for supporting mobility in IP based mobile access networks. An MPLS architecture is considered in mobile access networks. The proposed scheme provides an efficient mobility while taking advantages of MPLS, such as IP QoS support and better scalability to a large numbers of mobile hosts. The scheme uses the scalable LSP configuration approach to support the IP DiffServ (DS) model. The total number of LSPs in a mobile access MPLS domain is related to the number of traffic classes supported, since each LSP is shared by multiple mobile nodes whose data traffic needs to be treated with the same QoS policy. In order to make the scheme be scalable to match of growth for the future mobile IP network with MPLS, the MPLS label switched path (LSP) is designed as multiple dimensions.

The major attention focuses on the performance evaluation of the proposed LSP configuration on MPLS in mobile network. The performance is evaluated by simulation experiment which is conducted using the same mobility model as the proposed LSP scheme. The evaluation has been focusing network mobility in terms of new arrival call blocking probability and handover call blocking probability which are most important technical terms being commonly used for the performance evaluation in mobile networks. Since handoff prioritization can improve handoff related system performance, handoff prioritization such as guard channels are implemented. The simulation results clearly indicate the advantage that the proposed LSP scheme offers lower handoff call blocking probability (i.e., significantly improves probability of successful handover) under different traffic load scenarios. Therefore, the system can reduce packet loss during handoff. After handoff, continuous QoS can be provided to mobile users by using the pre-established LSP between base station (BS) and access GW (AGW).

The scheme does not require to make re-registration between mobile node (MN) and access gateway (AGW) when MN moves to another BS as long as the MN remains inside their local coverage area. Eliminating registration reduces the signaling load experienced by core network in support of mobility. Reducing signaling in this manner is necessary for the mobile network to scale to support very large volumes of mobile users, thus resulting in scalability improvement.

It is obvious from the study that a backward handover protocol is more suitable for the situation where the “new” channel establishment at the target base station is not reliable. This is in fact the majority of the cases in conventional cellular radio environment. However, when the possibility of losing a rapidly deteriorating communication link is taken into account, a forward handover protocol may be more suitable in some small cell handover situations. It is important that both the forward and backward handover protocol could be implemented as a mobile control or assisted protocol. De-centralization (mobile controlled handover) would help to speedup the handover process and would, therefore, be appropriate for microcells.

A reliable handoff scheme is established by developing five functional tasks. The scheme uses received signal strength (RSS) measurements of different BS at MN as the handover initiation criterion. The investigations shows that if the gradient of the signal degradation is steep, the handover region will be less sensitive to a small variation in the threshold level and vice versa. The selection of the threshold levels for the received signal level criterion can be performed by using predicted profiles. The only effect of introducing a hysteresis window is to delay the handover request. Hysteresis may be useful in preventing premature handover requests if the signal level are fluctuating. This is particularly useful in small cell size. By contrast, for a larger cell size, hysteresis will delay the handover initiation process. Handovers are only permitted when absolutely necessary, while at the same time maximizing the quality of the communication link.

The effectiveness of RSS with threshold depends on prior knowledge of the crossover signal strength between the current BS and target BS. Based on investigation of handover criteria, handover request flow diagram is provided. Resource sharing mechanism by each BS is demonstrated and call flow diagram for channel reservation based handoff prioritization is developed for enhancing the successful handoff probability and spectrum efficiency.

The proposed novel approach adopts Mobile Controlled Hand Off (MCHO) strategy with forward handover protocol due to fast handover and reliability. Completion of handoff process is described by using block diagram and forwarding table.

Based on the investigation of major performance measures undertaken in the simulation experiments, the LSP configuration scheme for mobile access network using MPLS presented in this thesis is well suited and can provide mobility with the capability for QoS. Therefore, the proposed LSP scheme provides the beneficial solution for supporting mobility with continuous QoS provision in IP based mobile access networks.

The next chapter of the thesis will mainly illustrate how incoming IP QoS service is mapped directly to the MPLS equivalent QoS service. Furthermore, queuing models are developed and performance analysis of mixed IP traffic with multiclass QoS over MPLS based mobile network are evaluated.

CHAPTER 6 QUEUING MODELS AND PERFORMANCE ANALYSIS OF SCALABLE QOS WITH MULTIPLE PRIORITY CLASSES OVER MPLS BASED MOBILE NETWORKS

6.1 Introduction

Multimedia applications necessitate support of different classes of traffic with diverse quality-of-service requirements which need to be guaranteed by the transmission network. By examining the characteristic of MPLS, one can see that MPLS is a very good candidate for providing differentiated services. When MPLS is combined with DiffServ (DS) and constrained based routing (CBR), they become powerful and complementary abstraction for QoS provisioning in IP networks .

As portable computers become more powerful and the accessibility of a fixed network from a mobile host becomes easier, the number of mobile users will grow and additionally the mobile users will demand the same real time services available to fixed hosts. This chapter first describes the framework combining the DiffServ (DS) ideas and enhanced mobile enabled MPLS as a backbone network in order to provide end-to-end QoS delivery between correspondent node (CN) of mobile user and mobile node (MN). The proposed QoS model considers that CN of mobile user from IS(Intserv)/DS(DiffServ) capable mobile network needs to communicate with other IS/DS capable MN through Mobile MPLS transit network. This is consequent to the capability of MPLS to provide virtual private network (VPN) to connect organizations to their multiple sites with compatible end-to-end QoS requirements.

The research work also highlights the modifications and extensions to MPLS including proposed QoS mapping in order to perform the mapping from incoming IP QoS requests into equivalent services in MPLS network in mobile environment. Queuing models are developed for prioritized packet scheduling to provide required QoS .

Priority queues are used in this thesis because of the following reasons. The thesis aims to treat different traffic flows differently. For example, some applications such as voice and video, are delay sensitive but loss insensitive. Others, such as file transfer and electronic mail are delay insensitive but loss sensitive. Still others, such as interactive graphic are delay sensitive and loss sensitive. Also, different traffic flows have different priorities. For example, network management traffic, particularly during times of congestion or failure, is more important than application traffic. It is particularly important during periods of congestion that traffic flows with different requirements are treated differently and provided a different quality of service (QoS). Therefore, a router might maintain different queues for different QoS levels and give preferential treatment to the higher levels. The performance analysis has been presented for heterogeneous traffic (e.g., signaling, voice, video and data) with multi-class QoS in terms of packet loss probability, traffic throughput and mean delay in the proposed model using fluid flow technique [136, 114].

Mobile IP is a newly defined Internet Protocol designed for providing host mobility. Mobile IP is able to track a mobile host without necessary to change the mobile host's permanent IP address. Network can be accessed at all times from all locations. It supports mobile users and also compatible with current IP but still many items need to be worked on and enhanced (e.g., security issues and the routing issues) [177] [183]. The following entities include in Mobile IP [183][184].

Mobile Node (MN) :

A host or router that may change its point of attachment from one network to another through the Internet. Pre-assigned a fixed home address on a home network. A mobile node may change its location without changing its IP.

Home Agent (HA) :

A router maintaining a list of registered mobile nodes. By checking the current mobility binding, home agent (HA) encapsulates datagrams and sends to the mobile node using care-of-address when the MN is away from home network.

Foreign Agent (FA):

A router on an mobile node's visited network which is delivering information between the mobile node and the home agent. The FA detunnels and delivers datagrams to the mobile node that were tunneled by the mobile node's home agent. The connection of a mobile host to its current foreign agent may be either wired or wireless.

Care-of-address (COA) :

An address identifying the MN's current location. It can be viewed as the end of tunnel directed towards a mobile node. It can be either assigned dynamically or associated with its foreign agent.

Correspondent Node (CN):

This node send packets which are addressed to the mobile node (MN) .

Home Address :

A permanent IP address that is assigned to a MN. It remains unchanged regardless of MN's attachment to the Internet.

Home Network :

A network having a network prefix matching that of a mobile node's home address. Within administrative domain, network administrators usually reserve one or more subnetworks for MN. The home address of a MN is allocated from the address space of one of these subnetworks, referred to as the home network.

Foreign Network:

Any connected segment of an Internet, other than the home network, to which the MN is allowed to attach is referred to as a foreign network.

Figure 6.1 The Mobile IP Architecture

Figure 6.1 shows the Mobile IP Architecture and brief operations is described as follows. mobile node (MN) uses two IP addresses. Home address is static (fixed). Care-of-address (COA) changes at each new point of attachment. Whenever the MN moves, it registers new care-of-address with HA (home agent). To get a packet to a MN from home network, HA delivers the packet from the home network to the care-of-address by constructing a new IP header containing COA. New header encapsulates the original packet. Such encapsulation is called tunneling. Cooperation of three mechanisms for Mobile IP [182] are discovering the care-of-address, registering the care-of-address and tunneling to the care-of-address

Discovering the care-of-address

When a MN is away from home, it finds agents not to loose Internet access. Two ways of finding agents: (i) by selecting an agent from among those periodically advertised or (ii) by sending out a periodic solicitation until it receives a response from a mobility agent. Thus, MN gets its care-of-address associated with its foreign agent.

Registering the care-of-address

MN registers its care-of-address with its home agent to obtain service. Performed directly from the MN or relayed by the foreign agent to the home agent depending on whether the care-of-address was dynamically assigned or associated with its foreign agent.

Tunneling to the care-of-address

The process of enclosing an IP datagram within another IP header which contains the care-of-address of the mobile node is called encapsulation. The default encapsulation supported by all mobility agents using Mobile IP is IP-within-IP as shown in Figure 6.2. Reverse process of encapsulation is called decapsulation.

Figure 6. 2 Encapsulation

Mobility Support in IPv6

Host mobility is becoming important because of strong wish to get continuous network connectivity regardless of location. Mobile IP is an Internet protocol designed for providing host mobility. Figure 6.3 shows the typical scenario of the mobile IPv6. IPv6 will support mobility to a greater degree than IPv4 because IPv6 includes many features for mobility support that are missing in IP version 4.

Mobile node (MN) obtains its care-of-address (COA) by using Stateless Address Autoconfiguration and Neighbor Discovery, thus no need foreign agent. In IPv6, MN is always responsible for delivering binding update message to corresponding node (CN). If, in case, CN has no cache binding, the CN sends the packets normally (with no routing header) to home agent (HA). Then, each packet is tunneled to the MN by HA using encapsulation as shown in Figure 6.2.

Figure 6.3 IPv6 Mobility

Moreover, triangle routing will not be a factor for IPv6 mobility. Route optimization can be deployed on a global scale between all MN and CN. In route optimization technique, when CN knows the MN's current care-of-address, a CN can deliver the packets directly to the MN without assistance of the Home Agent. In other words, if cache binding containing the mobility binding of mobile node is provided for the CN, the CN can deliver the packets to MN using IPv6 routing header analogous to Loose Source Route in IPv4. This is important for scalability and reliability for minimizing overall network load. Therefore, route optimization improves performance of IPv6 mobile nodes [161][184]. Overall, IPv6 looks very promising as an efficient and natural protocol for supporting mobility.

The technical survey shows that Mobile IP should be the basis for the mobility protocol in wide-area wireless data networks and several limitations when applied to next-generation wireless / mobile networks with high mobility users that may require QoS. For example, frequent movement between two base stations may cause significant disruption (loss and delay) to user traffic as a handoff occurs and mobile host is expected to update the network on every move resulting in excessive battery power consumption which is unacceptable for wide-area wireless devices. In addition, a mobile host using Mobile IP acquires a new care-of-address on every handoff from one base station to another. This would trigger the establishment of new QoS reservations from the home agent to the mobile node even though most of the path between the home agent and the mobile host is unchanged.

Mobile IP Scalability Issues

In the data forwarding process of Mobile IP, home agent (HA) will perform IP tunneling of the packet resulting in non optimal routing and header overheads. If large number of mobile node (MN) involves, forwarding process is very long. This poses a scalability concern that affects the use of the Mobile IP protocol in future wireless system.

6.2 An Overview of Mobile MPLS

MPLS is a packet forwarding technology that combines the simple IP routing with high-speed ATM switching. Mobile IP provides host mobility solution. Hence, the integration improves the scalability of Mobile IP data forwarding process. This scheme does not need for IP-in-IP tunneling from home agent (HA) to foreign agent (FA).

Figure 6.4 Mobile MPLS Architecture

Table 6.1 Example Label Table of HA after Registration

Incoming Port	In Label	FEC	Outgoing Port	Out Label
2	-	w.x.y.z	1	5
1	9	a.b.c.d	1	5
...	...	...	...	...

Assuming MN home address : a.b.c.d
HA address : a.b.c.e
FA care-of-address COA : w.x.y.z

As shown in Figure 6.4, HA and FA are edge LSRs and belong to the same MPLS domain. They support both MPLS and Mobile IP functionality. Mobile node (MN) determines either it is in home network or foreign network. If MN is in foreign network, MN gets COA (Care-of-Address) from FA and sends registration request to FA. Then, FA updates its routing table. FA forwards registration request to HA hop-by-hop using IP routing. HA searches label table to find the row with MN home address as FEC (2nd row in Table 6.1). LSP setup between HA and FA using LDP (1st row in Table 6.1 is created). Then, HA sends registration reply to FA along the LSP. FA records the incoming port number and in label value of the reply message.

Packets from a CN to the MN are addressed to MN home address and intercepted by HA. HA uses incoming label value as an index to look up label table. By Table 6.1, HA inserts out Label value of 2nd row into the packet and sent. If MN is in home network, out label and out going port entries are empty. Packet will be sent based on routing table entry to MN. Packet is delivered from HA to FA along the LSP by label swapping. FA receives the packet and removes the label and forwards the packet to MN based on newly added host specific row of the routing table [142].

Benefits from Integrating MPLS and Mobile IP

By integrating MPLS and Mobile IP, IP-in-IP tunneling is unnecessary in data forwarding process. MPLS is used to switch the packet to foreign network. Switching is much faster than conventional IP forwarding. The whole forwarding process is done at the MPLS layer and HA doesn't need to involve the IP layer. This improves the scalability of the Mobile IP. Traffic overhead from HA to FA is also reduced because label header is much smaller than IP header. With constraint-based label distribution protocol (CR-LDP) [98], we can set up an LSP satisfying the QoS requirements of the traffic[83].

The technical survey shows that provisioning of QoS guarantees is an important requirement in the Mobile MPLS scheme. The thesis fills up this gap and hence first focus the implementation of the framework combining the DiffServ (DS) ideas and enhanced Mobile MPLS as a backbone network in order to provide end-to-end QoS delivery between correspondent node (CN) of mobile user and mobile node (MN) in mobile environment.

6.3 Multiple QoS Classes in MPLS –based Mobile Network

6.3.1 Implementation of multi-class QoS based on Mixed IP Traffic

Figure 6.5 End-to-End QoS Network Model using MPLS based Backbone in Mobile Environment

This section focuses on the implementation of multiple QoS classes at IP level because IP-level QoS mechanism is able to provide guaranteed performance to the heterogeneous traffic. The framework described in Figure 6.5 proposes the desirable mechanism to deliver end-to-end service guarantees between correspondent node (CN) of mobile user and mobile node (MN) roaming in home network via MPLS core network. For illustrative purpose, Figure 6.5, CN1 and CN2 would send the flow based traffic packets to MN1 and MN2. Similarly, CN3 and CN4 would send the DiffServ traffic packets to MN3 and MN4. It is assumed that all MN are in home network. Mobility with capabilities of providing QoS and handoff performance on access MPLS mobile network has already studied in the previous chapter.

Mobile MPLS scheme [142] improves the scalability of the Mobile IP data forwarding because in the backbone MPLS network, the whole forwarding process is done at the MPLS layer. The studies from previous chapters point out that depending on traffic characteristics from end host, different LSP could be created for packets with different CoS (Class of Service) requirements. Packets of different flows can be labeled differently and thus received different QoS. Network scalability is achieved by traffic aggregation of same flows of the same class packets inside same LSP. The proposed framework of Figure 6.5 uses this concept and the relationship among flows, LSP and Link is as illustrated in Figure 4.20.

Assuming that all CN as well as all MN are running IPv6. It is desirable because IPv6 is not only able to interoperate with the current IPv4 but also able to run well in broadband networks to provide new internet functionality, including the guarantee of QoS based on application priority levels. A 4-bit priority field is designed in the IPv6 header [50, 96] for originating nodes and/or forwarding routers to identify and distinguish between different classes of priority of IPv6 packets so that it provides various forms of “differentiated service” for the IP packets other than through the use of explicit flow set up.

Home agent (HA) and visiting GW are assumed as edge LSR and belong to the same MPLS domain. Assuming that edge LSR designated as HA and visiting GW support DS functionality as well as MPLS and Mobile IP functionality.

In order for the end-user to receive service from the network, he must have a service level agreement (SLA) with the provider. SLA can be established either statically or dynamically. In order to support both types of SLA and to minimize the LSP setup delay, the MPLS network is assumed to have already been provisioned by statically allocating the necessary resources and pre established all the CR-LSP between MPLS endpoints, based on end-to-end user needs and incoming traffic pattern through the SLA.

Once the LSP is setup, desired requested bandwidth (BW) will be available end-to-end on the explicit route for the “sum” of all aggregate traffic in all the classes. At the ingress, the mobile user traffic entering MPLS network will be classified and conditioned. Then, DSCP value / PHB or IP packets (flow) is mapped to an LSP and the flow is assigned to a particular MPLS service class.

Proposed QoS mapping provides how the EXP field in the MPLS header of packets can be implemented to obtain the equivalent service from IS and DS capable mobile network. At the output scheduler, by using the proposed four space- priority queuing system, MPLS packets are provided service according to EXP field and forwarded to appropriate queue.

The premium queue is designed to carry the highest priority packets required to provide guaranteed bandwidth and delay. Gold queue (DS high priority queue) is designed to carry delay sensitive traffic and bronze queue (DS low priority queue) is designed to carry delay insensitive traffic. Multiple thresholds are implemented in two queues (gold queue & bronze queue) for packet loss priority control [135]. Silver queue (normal priority queue) is used to carry low loss rate traffic. Then, labeled packets are switched to neighboring LSR along pre-allocated LSP until to reach edge LSR.

In the core MPLS, scalable end-to-end QoS is obtained by aggregation of traffic flow. The primary benefit of DS mechanism is scalability. Thus, DS mechanisms are good candidate to provide QoS within MPLS network.

Packets from CN to the MN are addressed to MN home address along pre-configured LSP (here LSP1 and LSP2) and intercepted by HA (LER3). Then, HA uses incoming label value and MN home address (FEC) as an index to look up FTN label table. When HA receives the packets, it removes the label since it is the egress of LSP, out label and output port entries are empty. Thus, packets will be sent based on routing table entry to respective MN since MN is in home network. In this scenario, MN1 receives the packets sent by CN1. MN2 also receives the packets from CN2. Similarly, MN3 and MN4 receive the packets sending from CN3 and CN4 respectively.

6.3.2 Modifications and Extensions to MPLS Network

In order for MPLS to be able to support DiffServ / IntServ services and mobility, a number of modifications/extensions are required to MPLS . These are described below.

6.3.2.1 Service Class Differentiation

For service differentiation in the MPLS network, labeled packets are divided into separate traffic classes. 3bit EXP field of the MPLS packet header can be used to distinguish the service class of each packet. Eight behavior aggregates (BA) can be defined within the MPLS network. The mapping between BA and EXP field values are defined by the network operator specifically for MPLS.

The service classes classified are

- I. Premium/Gold class consists of (highest priority/ high priority key service) for delay sensitive traffic such as voice and videoconferencing.
- II. A Silver class and a Bronze class comprises (normal priority / low priority service) for controlled delay and throughput sensitive traffic such as VPN outsourcing, e-commerce and e mail.
- III. A Default Best Effort (BE) service class for packets without any guarantee from the network similar to Internet today.

6.3.2.2 LDP Extension

The next important extension is the addition of the appropriate messages to LDP. There are two basic requirements. One is downstream on demand label allocation and the other is addition of the Behavior Aggregate attributes in label binding messages. The first requirement is obvious. In order to set-up end-to-end LSP with differential QoS, it is necessary to ensure that all LSR in the same LSP perform the label binding in an ordered manner. This can be done by using downstream-on demand label allocation. The implementation of second requirement depends on how required QoS will be utilized. This thesis presents that incoming QoS service is mapped directly to the MPLS equivalent QoS service. The PHB to MPLS Label mapper will be responsible for that service.

6.3.2.3 Architectural Modification to LSR

Figure 6.6 and Figure 6.7 illustrate the architectural model of an edge LSR and core LSR used in MPLS network to provide end-to-end QoS in mobile environment. Functional modules such as constrained based LSP setup with link admission control, flow MIB controller, MPLS module need to be supported in both types of LSR as shown in Figure 6.6 and Figure 6.7.

Figure 6.6 Architectural Components of Edge LSR

Figure 6.7 Architectural Components of Core LSR

In Figure 6.6, edge LSR is responsible for DiffServ module (classification, policing and shaping), admission control, interworking function with DiffServ/ Intserv traffic and supporting Mobile IP functionality. The interaction between these functional modules are outlined below.

MPLS Daemon

Referring to Figure 6.6 and Figure 6.7, the main process of LSR and core of the MPLS protocol is located. For edge LSR in Figure 6.6, the PHB to MPLS label mapper is responsible for mapping incoming QoS service directly to the MPLS equivalent QoS parameters. For core LSR as shown in Figure 6.7, label swapping engine is responsible for label switching and forwarding process.

Service Provider Managing Daemon

In Figure 6.6, the management daemon at each edge LSR is used by the network manager to setup the constrained based routing LSPs by specifying the explicit routes (ER) and the associated traffic parameters, which, for the sake of simplicity, are considered in the form of token bucket values (r,b) . These parameters should be chosen sufficient to accommodate the traffic of all classes to be forwarded on that LSP, i.e., must reflect at least the “sum” of all the traffic parameters of the flows to be reserved traversing the LSP.

CR-LSP Setup Daemon

As shown in Figure 6.6 and Figure 6.7, the CR-LSP setup module uses either LDP (Label Distribution Protocol)[98] or an extension of RSVP [28]. Labels allocations are performed using control driven, downstream-on-demand approach. To set-up end-to-end LSPs with differential QoS, all LSRs belonging to the same LSP need to perform the label binding in an ordered manner and better scalability in resource conservation.

LSP Set up and Admission Control

The CR-LSP configuration is explained by referring to Figure 6.8. It is desirable to setup LSP = LSR1-LSR2-LSR3-LSR5-LSR7 with a particular QoS. The first thing to be done is to reserve the necessary bandwidth (BW) to accommodate the stream that will be admitted and also allocate the corresponding labels. If LDP is used, the CR-LSP setup daemon as shown in Figure 6.6 and Figure 6.7 first determines the **link admission control daemon** of its outgoing interface to the next hop in the explicit route (ER) to reserve the required bandwidth. If successful, label request message is sent to the next hop in the ER of that LSP. The next hop LSR (a core LSR) also checks its link admission control to set up a reservation on its outgoing interface and so forth until the egress LSR of the ER is reached. The egress LSR then sends a label mapping message to the originating LSR (OLSR) following the reverse path with the label information that is stored in the incoming label map (ILM) table within each core LSR.

The exchange of LDP messages is illustrated in Figure 6.9. In case one of the LSR on the followed path does not have sufficient resources, it will send an error message back to the originating LSR to indicate unavailability of resources. Hence, the LSP path will not be completed and the administrator will try another path. Once the LSP is set up, the desired requested bandwidth would then be available end-to-end on the explicit route for the “sum” of all aggregate traffic in all the classes.

For example, Figure 6.8 shows an LSP between LSR1 and LSR6 of a reserved 313 kbps aggregate end-to-end capacity, and an LSP between LSR1 and LSR7 of a reserved 560 kbps aggregate end-to-end capacity. The remaining 2.127 Mbps would still be available from the 3 M bps link on the outgoing interface of LSR1. At the ingress LSR (LSR1), Table 6.2 demonstrates the bandwidth characteristics associated with each LSP. The network administrator can now start allocating bandwidth statically for each service class within these LSPs.

Figure 6.8 Mobile MPLS with End-to-End Link Admission Control

- CN : correspondent node of mobile users
- MN : mobile node
- HA : home agent
- GW : gateway

Figure 6.9 LDP Message Exchange for Requesting and Confirming Label Mapping and Bandwidth Allocation

Table 6.2 Example of LSPs and Associated Amount of Reserved Bandwidth

<i>LSP_ID</i>	<i>Label out</i>	<i>Bandwidth (BW) reserved</i>
3	36	313 kbps < aggregate end-to-end >
5	21	560 kbps < aggregate end-to-end >

DiffServ (DS) Module

MPLS being a core technology, the focus of QoS support in MPLS networks is scalability, which is achieved by flow aggregation that ensures individual end-to-end QoS guarantees without maintaining individual flow on each segment of their path. DiffServ mechanisms are a good candidate to provide QoS within MPLS core networks because services are based on a per-hop model and aggregate forwarding resources (buffer space, bandwidth, scheduling policy) that are pre-allocated in the LSR for each service type.

Functions such as classification, shaping and policing are needed at the edge LSR of the network while core LSRs need only to have PHB classification, hence the scalability is achieved at the core. In Figure 6.6, the DiffServ (DS) module includes packet classifier, traffic conditioner and per-hop forwarding behavior (PHB). DS module is responsible for identifying the flow of packets at the ingress LSR in order to associate it with the appropriate label. IP packets belonging to a particular traffic stream arrive at LSR1, having already been marked at the source end-host to indicate the level of service they expect. At LSR1, the classification and traffic conditioning functions on the specified traffic are performed by the service provider managing the core network according to service level agreement (SLA).

Packet Classifiers

Packet classification is a significant function required at the edge of the MPLS network. Its aim is to distinguish among packets belonging to different classes of traffic. The classifier can be multifield (MF) classifier which performs the selection based on the combination of one or more header fields (destination IP address/ port, Type of Service (TOS) field of the IPv4 packet headers or the Traffic Class field of the IPv6 packet headers, referred to as the DS field or Code Point (DSCP)). Once the CR-LSP have been set up, the next task is to configure the classifier to bind a particular flow and its traffic parameters to an LSP and assign the flow to a particular service class.

An LSP Admission Control module at the ingress LSR of Figure 6.6 provides control-load support on that LSP at the ingress. This module measures whether bandwidth is still available on that LSP for the traffic parameters requested by the new flow being added to that path and stores the information in the FEC to Next hop label forwarding entry (FTN) of the edge LSR. Various admission control algorithms [140] can be used in the architectural model of Figure 6.6. When a marked packet is received from CN (correspondent node) of mobile user which is connected to a neighboring DS capable network, the edge LSR "encapsulates" the packet in an MPLS header with the appropriate label and EXP value.

Table 6.3 illustrates the binding of two high priority flows to the same LSP that are each using 25 kbps of the bandwidth of the LSP with LSP_ID 3.

Table 6.3 Example of a Modified FTN Table at an Ingress LSR for LSP_ID 3 and its Remaining Capacity of 263 kbps

FEC	INCOMING LABEL	LSP_ID	MPLS EXP VALUE	OUTGOING LABEL	FLOW CAPACITY
Flow1 (high priority)	9	3	110	36	25 kbps
Flow2 (high priority)	9	3	110	36	25 kbps

Traffic Conditioners

Traffic conditioners form a vital part of a DiffServ module as shown in Figure 6.6. Their goal is to perform the metering, shaping and policing functions on previously classified packets at the edge of the network. A traffic conditioner illustrated in Figure 6.10 consists of meter, shapper / policer and their interaction is described by referring to Figure 6.11.

Figure 6.10. Traffic Conditioner

Meter

A device measuring the temporal properties (e.g., rate) of a traffic stream of packets selected by a classifier. The meter determines whether a given packet stream class is within or exceeds the service level guaranteed for that class. A meter passes state information to other conditioning functions such as shaping or policing to trigger a particular action for each packet which is either in- or out-of-profile.

Figure 6.11. A Simplified View of a Packet Classifier and Traffic Conditioner including Meter, Shaper and Policer

Shaper

Shapers delay the packets as necessary in order that the packet stream in a given class does not exceed the traffic rate specified in a profile for that class. A shaper usually has a finite-size buffer, and packets may be discarded if there is not sufficient buffer space to hold the delayed packets. In this case, the shaper delays incoming Premium and Gold traffic packets to conform to Leaky bucket meter and also delays Silver and Bronze traffic packets to conform to Token bucket meter.

Dropper / Policer

A device that discards packets based on specified rules. Incoming packets that are out of profile are either discarded (in the Premium, Gold and Silver Class), or given a high drop precedence (in the Bronze /BE class). In-profile packets are allowed to enter the network.

Flow MIB and Controller

Flow MIB is a database for maintaining flow related information such as per-flow traffic statistics and path information for aggregated flows. This information is needed for resource management. Flow MIB Controller monitors the LSR and its flows and collect statistics useful for evaluating the local resources.

Interworking with DS (DiffServ) and IS (IntServ) capable mobile networks

This section discusses how the enhanced MPLS based mobile network can support QoS for IS and DS capable mobile networks and provide interoperability at the boundary. In order for the end-user to receive service from the network, he must have a service level agreement (SLA) with the provider. SLA can be established either statically or dynamically. Static SLA are negotiated on a regular (e.g., monthly) basis, while dynamic SLA require the use of some signaling protocol (e.g., RSVP) to request service on demand. These dynamic requests are considered by the IWF (Interworking Function) Module at the ingress LSR as shown in Figure 6.6. The ingress LSR “ tunnels” them through the core MPLS network to the destination (egress), thus ensuring that flow reservation occurs end-to-end.

In order to support both types of SLA and to minimize the LSP setup delay, the MPLS network is assumed to have already been provisioned by statically allocating the necessary resources and pre-established all the CR-LSP between MPLS endpoints, based on end-to-end user needs and incoming traffic pattern through the SLA.

Ingress LSR maps the incoming QoS requests from the neighboring DS and IS capable mobile networks to the corresponding service class within the mobile enabled MPLS network. In this framework, each incoming flow is assigned to one of the available classes for the duration of the flow and traverses the MPLS backbone network in this class.

The service mappings given in Table 6.4 follow from the service definitions: Guaranteed Service (GS) [93] and Expedited Forwarding (EF)[95] must be mapped onto Premium /Gold MPLS class and Controlled-Load service (CL) [203] and Assured Forwarding (AF) [94] must be mapped onto Silver/Bronze classes. Mapping element located in the ingress LSR will perform mapping incoming QoS requests from DiffServ classes / IntServ classes to the associated service class within the MPLS network.

Table 6.4 Service Map from IntServ/DiffServ to MPLS

Incoming QoS Service Requests		MPLS Equivalent QoS Service
IntServ (IS)	DiffServ (DS)	
Guaranteed Service (GS)	Expedited Forwarding (EF)	Premium / Gold
Controlled Load (CL)	Assured Forwarding (AF)	Silver/Bronze
Best Effort	Default Forwarding (DF)	Best Effort

Table 6.5 proposes possible mapping for the EXP to MPLS service class combination and identify how the EXP field of MPLS packet header across the MPLS network to achieve the equivalent service. The network administrator can choose to group together traffic flows requiring similar service into single MPLS service class and configure the mapping of QoS request from DS as well as the incoming IS service request, to the EXP value for output port scheduling and congestion avoidance mechanisms.

Table 6.5 Proposed QoS Mapping

IntServ/DiffServ Traffic Class (Incoming QoS Request)	MPLS EXP Field	Equivalent Priority Mapping	MPLS Service Class Designation	Example Applications
IntServ GS	111	Guaranteed Bandwidth & Delay	Premium (Highest Priority Service)	Network Control, Signaling, Routing Update
DiffServ EF	110	Loss Sensitive & Delay Sensitive	Gold (High Priority Key Service)	Voice
DiffServ AF1X	101	Delay Sensitive		Real time Video
IntServ CLS	100	Low Loss Rate	Silver (Normal Priority Service)	Adaptive Real time Applications
DiffServ AF2X	011	Loss Sensitive & Delay Insensitive (Low Drop Precedence)	Bronze (Low Priority Service)	Email , FTP
DiffServ AF3X	010	Loss Sensitive & Delay Insensitive (High Drop Precedence)		<i>Mission – Critical Data</i>
DiffServ AF4X	001	Loss Insensitive & Delay Insensitive		Other
Best Effort	000	No performance Guarantee (Best Effort Delivery)	Best Effort	Similar to current Internet

Output Scheduler

Figure 6.12 Output Scheduling Model in LSR

Figure 6.12 shows the output scheduler which includes in both types of LSR (i.e., edge LSR & core LSR). The output scheduler contains four queues, one for each traffic class. Considering the service requirement for each traffic class, Premium queue is designed to serve the IntServ Guaranteed Service (GS), the Gold queue is used to treat the DiffServ high priority traffic, Silver queue receives IntServ Controlled Load Service (CLS) and Bronze queue is used to carry the DS low priority traffic. The queues are served by using the simple priority scheduling scheme. The service differentiation is achieved by classifying the MPLS packets according to the EXP field and forwarded to the appropriate queue. Traffic of the same class from different LSPs are statically multiplexed in the same queue.

Mobile IP Functionality Daemon

In next generation mobile networks, the mobile users will demand the real-time services such as Video-on-Demand, Internet online audio, or Internet interactive games, the senders can be located on either mobile networks or wired network with powerful computing capability and plenty of disk storage. Receivers are mobile nodes to access the service on Internet in convenient way at anytime and anywhere. In Figure 6.13, the sender correspondent node (CN) of mobile user is a server located on either mobile network or wired network and the receiver MN (mobile node) is freely roaming in mobile network. In order to deliver end-to-end service guarantee between the CN which is connected to DifServ (DS) or IntServ (IS) capable mobile network and MN roaming in home network or foreign network, edge LSRs need to support MPLS as well as Mobile IP functionality. These requirements are considered by **Mobile IP Functionality daemon** at the ingress LSR as shown in Figure 6.6. The interaction of Mobile IP Functionality daemon can be explained by referring to Figure 6.13 as shown below.

Figure 6.13 End-to-End QoS between CN of Mobile User and MN over Mobile Enabled MPLS Network

Consider the application scenario demonstrated in Figure 6.13. Assuming that DiffServ(DS) module is located at the MN and the DS Code Point (DSCP) of the packet has been set at the DS module and for CN of mobile user, the DSCP of the packet has been set at the DS capable mobile network. Home agent (HA) and visiting GW are assumed as edge LSR and belong to the same MPLS domain. Assuming that edge LSR designated as HA and visiting GW include Mobile IP functionality Daemon as shown in Figure 6.6. Integration of the Mobile IP and MPLS also improves the scalability of Mobile IP data forwarding process.

In order to minimize the LSP setup delay, the MPLS network is assumed to have already been provisioned by statically allocating the necessary resources and pre established all the CR-LSP between MPLS endpoints, based on end-user needs and incoming traffic pattern through the service level agreement (SLA). Once the LSP is setup, desired requested bandwidth will be available end-to-end on the explicit route for the “sum” of all aggregate traffic in all the classes. At the ingress, the user traffic entering MPLS Network will be classified, conditioned. Then, DSCP value / PHB or IP packets (flow) is mapped to an LSP and the flow is assigned to a particular MPLS service class.

Firstly, MN determines either it is in home network or visiting network. When MN arrives in visiting network, MN obtains care of address (COA) according to the methods of IPv6 neighbor discovery. MN sends registration request to visiting GW which updates its routing table. Then, visiting GW forwards registration request to HA using IP routing. Then, LSP satisfying the QoS requirements of the traffic will be configured between HA and visiting GW using constrained based label distribution protocol (CR-LDP). The association between a MN's home address and COA is known as a “binding”. Delivery of MPLS packets, in case, CN has no cache binding, can be considered as follows :

Packets from a CN to the MN are addressed to MN home address along pre-configured LSP and intercepted by HA. Then, HA uses incoming label value and MN home address (FEC) as an index to look up FTN label table. If MN is in Home network, out label and out going port entries are empty. Packet will be sent based on routing table entry to MN. Otherwise, packet is delivered from HA to visiting GW along the LSP by label swapping based on label table as in Mobile MPLS mechanism [142]. When visiting GW receives the packet, it removes the label since it is the egress of LSP from HA, out label and output port entries are empty. Therefore, visiting GW forwards the packet to MN based on newly added host specific row of the routing table.

Another alternative packet delivery mechanism is that when visiting GW receives the labeled packet from HA, it will also notify to MN as an indication that CN has no binding for the MN. Therefore, MN can establish a binding update message with that CN. Afterwards, CN will know where the MN is. When the CN wishes to send the packets to MN, it can deliver the packets directly to visiting GW using MN's care of address (COA) over MPLS pre-established LSP. By this way, route optimization can be achieved and it improves the performance of mobile nodes. In this manner, scalable end-to-end QoS can be obtained in mobile environment using DiffServ (DS) like mechanism and enhanced MPLS network.

6.3.3 Modeling of LSR in Mobile MPLS Architecture

Figure 6.14 demonstrates how packets forward from the ingress LSR to egress LSR via the core LSR in the mobile MPLS network. Each LSR consists of several input links and output links.

Figure 6.14 Packets Forwarding from Ingress LSR to Egress LSR

Figure 6.15. Mobile MPLS Network Node

The modified LSR in mobile MPLS architecture can be modeled as shown in Figure 6.15. Each router has several different input links and output links. Each input link may contain a number of IP packet streams with different priorities. These packet streams are multiplexed when they are transmitted over the transmission link. Before the packets are processed, the IP packet streams on the same input link are de-multiplexed by the DEMUX. Then, different IP packet streams with different priorities are routed to corresponding output ports according to the Mobile MPLS routing table. At the output port, those IP packet streams are multiplexed again before they are transmitted onto the output link. The multiplexer at the output port consists of four parallel output queues. Each output queue operates on a First-In-First-Out (FIFO) non-preemptive priority queuing discipline while the queue is being served.

Queue discipline refers to the manner in which packets are selected for service when queue has formed. The most common discipline that can be observed in every day life is First Come, First Served (FCFS). Some others in common usage are Last Come, First Served (LCFS), which is applicable to many inventory when there is no obsolescence of stored units, as it is easier to reach the nearest items, which are the last in; selection of service in random order independent of the time of arrival to the queue (RSS); and a variety of priority schemes, where packets are given priorities upon entering the system, the ones with higher priorities to be selected for service ahead of those with lower priorities, regardless of their time of arrival to the system.

There are two general situations in priority disciplines. In the first, which is called preemptive, the packet with the highest priority is allowed to enter service immediately even if a customer with lower priority is already in service when the higher priority packet enters the system; that is, the lower priority packet in service is preempted, its service stopped, to be resumed again after the higher priority packet is served.

In the second general priority situation, called the nonpreemptive case, the highest priority customer goes to the head of the queue but can not get into service until the packet presently in service is completed, even though this packet has a lower priority [160].

The queue model used in this thesis is the fluid flow model, which is explained in [116]. The fluid flow technique is used in the following performance analysis with assumptions that (1) the population of IP sources is large ($N \gg 1$), (2) the output transmission link is divided into equal time slots and each slot is equivalent to the maximum transmission time of IP packet such that the transmission time of IP packet is assumed to be uniformly distributed. This assumption is reasonable, because that when the population of IP sources is large, the bit rate allocated to each IP source comparing to the huge capacity of the broadband transmission link is negligible. Therefore, the variation of the transmission time of IP packet is also negligible.

Figure 6.16 . Buffer Management of Multiclass QoS by Four Queue System

Referring to Table 6.4 and Table 6.5, a service mapping among IntServ (IS) service class, DiffServ (DS) service class, MPLS EXP field have been proposed. Figure 6.16 demonstrates the modeling of four-queue system at the output port of the modified LSR of mobile MPLS network shown in Figure 6.15. Premium Queue is used for the IP packet stream with highest priority. Thus, traffic from IS Guaranteed Service (GS) are received and treated in this queue. Multiple loss priority [159] is implemented using threshold control mechanism over the partitioned buffer of the DS high priority queue and DS low priority queue.

When the Premium Queue is empty, the Gold Queue will provide the service. DS High Priority traffic is fed to the Gold Queue. The Gold Queue consists of $K1$ Loss priority classes corresponding to the different thresholds $U_1, U_2, \dots, U_{K1}$ for packet loss control.

When the buffer occupancy of the priority queue exceeds the threshold U_n , where n is the priority index ($1 \leq n \leq K1$), only the IP packet with loss priority ranging from 1 to $n-1$ are permitted to input into the queue while packets with priorities equal to or higher than n are discarded. When the buffer occupancy of the DS high priority queue exceeds the threshold U_1 , all the packets are lost.

When both the Premium Queue and Gold Queue are empty, the Silver Queue can serve. The Silver Queue gives the service to the packets from IS Controlled Load Service (CLS). The Bronze Queue is designed to perform the Lowest Priority and serves when all the other queues have no input traffic. The DS Low Priority traffic are input to the Bronze Queue and it consists of $U_{K1+1}, U_{K1+2}, \dots, U_{K2}$ different thresholds for packet loss control over the partitioned buffer.

The modeling proposed favors IntServ (IS) traffic over DiffServ (DS) traffic for the similar level of service requirement. This is because the IS and DS sources generate traffic in different manner and we have to use separate queues to contain them for ease of performance analysis. Otherwise, the performance analysis may result in significant and analytical complexity.

6.4 Performance Evaluation

The performance analysis of mixed IP traffic with multiclass QoS over enhanced Mobile MPLS network is evaluated using the Fluid Flow Traffic Model [114], [128] in which each individual source is modeled as a Markov Modulated Fluid Source consisting of the ON state and OFF state. The technical approach of Markov Modulated Process and fluid flow model are briefly described as follows.

A simple example of Markov process is the Discrete-State, Continuous-Time, Markov Process. Because of its modeling simplicity with a clear definition of correlation and burstiness and the simplicity involved in queuing analysis, it is very successful in modeling diverse individual ON-OFF traffic sources. Figure 6.17 illustrates an ON-OFF traffic source alternating between ON period and OFF period. During the ON period, information is generated at a constant peak access rate, while in the OFF period, no information is transmitted. The two state Markov chain is characterized by three parameters: the transition probability from the OFF to the ON state α ; the transition probability from the ON state to the OFF state β ; the packet transmission rate in the ON state.

Figure 6.17 Two-state Markov Chain

The most commonly used Markov-Modulated Process is the Markov-Modulated Poisson Process (MMPP) [105], [106] in which the packet generated by each source is a Poisson process with exponentially distributed in length. The transition between the ON state and the OFF state are controlled by an underlying continuous-time Markov chain [112] which determines the rate of fluid generating. The transport of packets over a transmission link is operated in the same manner.

The fluid flow traffic model is used in the following performance analysis. In the fluid flow technique, the arrivals of packets are treated as a continuous arrival of a liquid flow, characterized by a flow rate. Fluid models are effective at the burst level in its accuracy and capacity to solve large systems. The studies have shown that fluid models are well matched to analyzing the performance of the traffic in emerging high speed broadband network environment. An important advantage of the fluid flow model is the analytical simplification of the performance analysis

In high speed networks, fluid flow modeling is especially applicable because interarrival times of data units transmitted during the ON period are small compared to the time between arrival rate changes. Therefore, the impact of individual arrivals on the queue performance is negligible. For an ON-OFF fluid source, transmission of data unit during an ON period is seen at the burst level as a single arrival event with certain flow rate.

Thus, the arrivals of a series of ON period form a rate process. If the distribution of the interarrival time is exponentially distributed, the rate process is then a Markov process. This is similar to MMPP model whose arrivals of each ON period at the burst level from a rate process which is a Markov process. In our analysis, each DS user source is modeled as Markov Modulated ON-OFF fluid traffic source. The fluid flow model is established through a set of differential equations describing the equilibrium buffer distribution. The key approximation used for this model is to assume Poisson Process for the ON period event and OFF period event.

6.4.1 Analysis on Proposed Model

6.4.1.1 Premium Queuing Model

The Premium queue accepts traffic from IS Guaranteed Service (GS) which is on per-flow basis. Assuming that the arrival rate λ_{PQ} of GS traffic is of Poisson distribution and the service time $1/C$ is of exponential distribution. As long as Arrival rate $\lambda_{PQ} <$ channel capacity C , which is of the normal case, and the buffer is big enough, therefore Premium queue can be modeled as an M/M/1 queue system shown in Figure 6.18 in which μ is the average service rate and $\mu = C$.

Figure 6.18. Modeling of Premium Queue

The Probability of n packets in the system :

$$P_n = P_0 \left[\frac{\lambda_{PQ}}{C} \right]^n \quad (n = 0, 1, 2, \dots) \dots\dots\dots(1)$$

The Probability of no packet (empty) in the system :

$$P_0 = 1 - \rho$$
$$P_0 = 1 - \frac{\lambda_{PQ}}{C} \dots\dots\dots(2)$$

The Loss Probability, γ , that the incoming traffic will be blocked and discarded is zero.

Thus, the throughput of the Premium Queue

$$T_{PQ} = \lambda_{PQ} (1 - \gamma) = \lambda_{PQ} \dots\dots\dots(3)$$

Average Delay of the Premium Queue

$$D_{PQ} = \frac{1}{\lambda_{PQ}} \sum_{n=0}^{\infty} n.P_n$$
$$= \frac{1}{C - \lambda_{PQ}} \quad (\mu = C) \dots\dots\dots(4)$$

where $\sum_{n=0}^{\infty} n.P_n$ is the average number of packets in the system.

6.4.1.2 Fluid Model and Analysis of Gold Queue

Assuming DS sources can generate arbitrary number $K2$ of the priority classes. Packets of priorities $\ell \in [1, K1]$ belongs to the Gold Queue while packets of priority $\ell \in [K1 + 1, K2]$ belong to the Bronze Queue. Each DS source has two states with generator G and the traffic streams are mutually correlated. Loss Priority is implemented by partitioning the buffer into as many levels of occupancy as there are priority classes. Packets of a particular priority class are either accepted or rejected depending on the prevailing level of buffer occupancy.

Each source when in state j ($j=1, 2$) generates traffic of priority class ℓ ($\ell = 1, 2, \dots, K2$) at rate $\lambda_j^{(\ell)}$. Thus, each of the N sources is characterized by $(G; \lambda^{(1)}, \dots, \lambda^{(K2)})$ where $\lambda^{(\ell)} = [\lambda_1^{(\ell)}, \lambda_2^{(\ell)}]$. $\lambda_1^{(\ell)}$ designates the arrival rate of the OFF state of class ℓ for each source. $\lambda_2^{(\ell)}$ represents the arrival rate of the ON state of class ℓ for each source. Hence, the rate of generation of class ℓ traffic $R_n^{(\ell)} (n \in [0, N])$ is defined to be $n\lambda_2^{(\ell)} + (N - n)\lambda_1^{(\ell)}$.

Assume that the arrival rates of the OFF state of any of the priority classes are zero, therefore, $\lambda_1^{(\ell)} = 0$. In this case, the source is modeled as a simple ON-OFF process, with $R_n^{(\ell)} = n\lambda_2^{(\ell)}$. For the purpose of convenience in representation, λ_ℓ is used to represent $\lambda_2^{(\ell)}$.

Each DS source can be modeled as an Discrete-state, continuous time, ON-OFF Markov Modulated Fluid Source [136]. As illustrated in Figure 6.19, the mini source turns on with exponential rate α and OFF with rate β . Then, the probability that the process is in the ON state is $\frac{\alpha}{\alpha + \beta}$.

Figure 6.19. Mini Source Model

When N of such ON-OFF packet sources are multiplexed, the aggregated stream can be represented by a $(N+1)$ state. Markov Modulated process as shown in Figure 6.20. In this case, state n represents n ($n = 0, 1, 2, \dots, N$) sources are in the ON state. The transition rate from the state n to the state $(n-1)$ is $n\beta$ and the transition rate from the state n to the state $(n+1)$ is $(N-n)\alpha$.

Figure 6.20 . Modeling of N DiffServ (DS) sources

Gold Queue gives service when Premium Queue is empty. The service rate for the Gold Queue is reduced from C to C_{GQ} and it is given by

$$C_{GQ} = C - \lambda_{PQ} \dots\dots\dots(5)$$

where C is the channel capacity

where λ_{PQ} is the arrival rate of Premium Queue

Probability that Premium Queue empty is $P_{epq} = 1 - \frac{\lambda_{PQ}}{C} \dots\dots\dots(6)$

Fluid Flow Queuing Model [135] is implemented for the modeling of Gold Queue. Given the average OFF period of the source is denoted by $1/\alpha$, if at time t the number of ON sources equal n , two elementary events can occur during the next interval Δt , i.e., a new source can start or a source can turn off. Since ON and OFF periods are exponentially distributed, the corresponding probabilities are $(N - n)\alpha\Delta t$ and $n\beta\Delta t$. Compound events have probabilities $O(\Delta t^2)$.

The Probability of no change is $1 - \{(N - n)\alpha + n\beta\} \Delta t + O(\Delta t^2)$.

$P_n^\ell(t, x)$, ($n = 0, 1, 2, \dots, N$, $\ell = 1, 2, \dots, K2$, $t \geq 0, x \geq 0$) is defined as the probability that at a time t , n sources are ON and the queuing buffer content does not exceed x so that only the traffic with priorities higher or equal to ℓ can be admitted into the buffer. The traffic with priorities lower than ℓ will be discarded and lost.

Now,

$$P_n^\ell(t + \Delta t, x) = [(N - (n - 1))\alpha \Delta t P_n^\ell(t + \Delta t, x) + (n + 1)\beta \Delta t P_{n+1}^\ell(t + \Delta t, x) + [1 - \{(N - n)\alpha + n\beta\} \Delta t] \times P_n^\ell(t, x - (n\lambda^\ell - C_{GQ})\Delta t + 0(\Delta t^2)) \quad (7)$$

where $\lambda^\ell = \sum_{r=1}^{\ell} \lambda_r$ (λ_r is the arrival rate of r th class DS Traffic at ON state) per source of traffic that can be admitted to the buffer.

On the right-hand side of Equation (7) , the first term is the probability that the number of ON state sources transfers from $(n-1)$ state to n during the time interval of Δt , the second term denotes the probability of the number of ON state sources transfer from $(n+1)$ to n , while the third term represents the probability that the number of ON state sources does not change during the time interval of Δt . The term $0(\Delta t^2)$ is the higher order terms which go to zero much more rapidly than Δt when Δt is tending to zero. Hence, the effects of $0(\Delta t^2)$ is negligible when Δt is small enough.

Since we are interested only in the time-independent, equilibrium probabilities, hence, taking partial derivation of Equation (7) to t , pass to the limit $\Delta t \rightarrow 0$ and setting

$$\frac{\partial P_n^\ell}{\partial t} = 0 \text{ and obtain the following expression , for } n \in [0, N] ,$$

$$(n\lambda^\ell - C_{GQ}) \frac{dE_n^\ell}{dx} = (N - n + 1)\alpha E_{n-1}^\ell - [(N - n)\alpha + n\beta] E_n^\ell + (n + 1)\beta E_{n+1}^\ell \quad (8)$$

In (8), $E_n^\ell(x)$ denotes the equilibrium probability that n sources are in the ON state and buffer occupancy does not exceed x of class ℓ . Note that $E_n^\ell = 0$ if n is not in the stated interval.

Equation (8) can be expressed in the matrix notation as shown in (9),

$$D^\ell \frac{d}{dx} E^\ell(x) = M \cdot E^\ell(x), x \geq 0 \quad \dots\dots\dots (9)$$

where $D^\ell = \text{diag} \{-C_{GQ}, \lambda^\ell - C_{GQ}, 2\lambda^\ell - C_{GQ}, \dots, N\lambda^\ell - C_{GQ}\}$ and

M is the $(N+1) \times (N+1)$ tri-diagonal matrix as shown below :

$$M = \begin{bmatrix} -N\alpha & \beta & & & & \\ N\alpha & \{(N-1)\alpha + \beta\} & 2\beta & & & \\ & (N-1)\alpha & -\{(N-2)\alpha + 2\beta\} & 3\beta & & \\ & \cdot & \cdot & \cdot & & \\ & \cdot & & \cdot & & \\ & & & & 2\alpha - \{\alpha + (N-1)\beta\} & N\beta \\ & & & & \alpha & -N\beta \end{bmatrix}$$

Assuming $n\lambda^\ell - C_{GQ}$ is not equal to zero for any n , $0 \leq n \leq N$, the general solution of Equation (9) is :

$$E^\ell(x) = \sum_{n=0}^N a_n^\ell V_n^\ell e^{z_n^\ell x} \quad \dots\dots\dots (10)$$

where $E^\ell(x) = [E_0^\ell(x), E_1^\ell(x), \dots, E_N^\ell(x)]$, the elements in vector

$z^\ell = [z_0^\ell, z_1^\ell, \dots, z_N^\ell]$ are the eigenvalues of matrix $(D^\ell)^{-1} M$ and V_n^ℓ is the eigenvector of matrix $(D^\ell)^{-1} M$.

In (10), the coefficients $\{a_n^\ell\}$ can be obtained from boundary conditions by defining

$$Q_D^\ell = \{n \mid n\lambda^\ell < C_{GQ}\} \quad \text{and} \quad Q_H^\ell = \{n \mid n\lambda^\ell > C_{SQ}\}$$

Boundary Conditions of Gold Queue can be obtained :

$$E_n^1(U_1) = P(n) \quad \text{if } n \in Q_D^1 \quad \dots\dots\dots (11)$$

$$E_n^\ell(U_\ell) = E_{n-1}^\ell(U_\ell) \quad \text{if } (n \in Q_H^{\ell-1} \cup Q_D^\ell, 2 \leq \ell \leq K1) \dots\dots\dots (12)$$

$$E_n^{K1}(0) = 0 \quad \text{if } n \in Q_H^{K1} \dots\dots\dots (13)$$

where $P(n)$ is the probability that n DS sources are in the ON state and

P_{ON} is the probability of 1 DS source is in the ON state.

$$\left. \begin{aligned} P(n) &= \binom{N}{n} P_{ON}^n (1 - P_{ON})^{N-n} \quad (0 \leq n \leq N) \\ P_{ON} &= \frac{\alpha}{\alpha + \beta} \end{aligned} \right\} \dots\dots\dots (14)$$

Throughput Computation for Gold Queue

Arrival Rate of the ℓ th priority class traffic

$$A_{GQ}^\ell = \sum_{n=0}^N \lambda_\ell n P(n), \quad (1 \leq \ell \leq K1) \dots\dots\dots (15)$$

Where λ_ℓ represents the arrival rate of the ON state of priority class ℓ ($\ell = 1, 2, \dots, K1$) for each source.

In order to calculate the throughput of the ℓ th class packet in the gold Queue, it is necessary to consider two scenarios since the Premium Queue may or may not have input traffic. The performance analysis of Gold Queue under these two scenarios vary.

Scenario I. *Premium(IS Guaranteed Service) Queue has no Input traffic*

The throughputs of different priority classes in this scenario are evaluated below.

Throughputs of ℓ th priority class traffic $T_1^\ell = (\text{Arrival Rate of } \ell \text{ th priority class traffic}) - (\text{Loss Rate of the } \ell \text{ th priority class traffic})$

For class 1 traffic in this Queue,

$$\begin{aligned} T_1^1 &= \text{Arrival rate of Class 1 Traffic} - \text{Loss rate of Class 1 Traffic} \\ &= \sum_{n=0}^N \lambda_1 n E_n^1(U_1) + C_{GQ} \left[1 - \sum_{n=0}^N E_n^1(U_1) \right] \dots\dots\dots(16) \end{aligned}$$

For other priority classes other than class1, that is $\ell \in [2, K1]$:

$$\begin{aligned} T_1^\ell &= (\text{Arrival Rate of class } \ell \text{ traffic}) - (\text{Loss Rate of the class } \ell \text{ traffic}) \\ &= \sum_{n=0}^N \lambda_\ell n E_n^{\ell-1}(U_\ell) - \sum_{n=Q_H^{\ell-1} \cap Q_D^\ell} \{ [E_n^{\ell-1}(U_{\ell-1}) - E_n^\ell(U_\ell)] \times [n\lambda^\ell - C_{GQ}] \} \dots\dots\dots(17) \end{aligned}$$

Scenario II. *Premium Queue has Input traffic*

$$\text{Throughputs } T_2^\ell = \sum_{n=0}^N \lambda_\ell n E_n^\ell(U_1) \quad , \quad (1 \leq \ell \leq K1) \dots\dots\dots(18)$$

Therefore, the Average Throughput of DS high priority queue is given by

$$T_{GQ}^\ell = P_{epq} T_1^\ell + (1 - P_{epq}) T_2^\ell \quad , \quad (1 \leq \ell \leq K1) \dots\dots\dots(19)$$

The packet loss rate probability due to buffer overflow of the ℓ th priority class traffic is given by

$$R_{GQ}^\ell = 1 - \frac{T_{GQ}^\ell}{A_{GQ}^\ell} \quad , \quad (1 \leq \ell \leq K1) \dots\dots\dots(20)$$

Calculation of Delay Statistic for Gold Queue

When considering the Gold Queue, delay is also an important QoS parameter because the traffic inputs to the Gold Queue are delay sensitive traffic. Then, the equations for the delay of the different priority classes in the Gold Queue will be derived. Delay distribution of the ℓ th priority class traffic is obtained from the buffer content distribution seen by the arriving ℓ th priority class packets.

Let $\pi^\ell(t)$ be the probability that the delay is less than t for the ℓ th priority class packets.

$$\pi_k^\ell(t) = \frac{1}{T_{GQ}^\ell} \sum_{n=0}^N n \lambda_\ell E_n^k(C_{GQ}t) \quad \text{where } \frac{U_{k+1}}{C_{GQ}} < t < \frac{U_k}{C_{GQ}}, k \in (\ell, K1-1) \dots (21)$$

$$\pi_{K1}^\ell(t) = \frac{1}{T_{GQ}^\ell} \sum_{n=0}^N n \lambda_\ell E_n^{K1}(C_{GQ}t) \quad \text{where } 0 \leq t < \frac{U_{K1}}{C_{GQ}} \dots \dots \dots (22)$$

For traffic with the highest packet loss priority, the probability that the delay is less than t is

$$\Pr(\text{delay} = \frac{U_1}{C_{GQ}}) = \frac{1}{T_{GQ}^\ell} \left[1 - \sum_{n=0}^N n \lambda_\ell E_n^1(U_1) \right] C_{GQ} \dots \dots \dots (23)$$

In the case of $t = \frac{U_k}{C_{GQ}}$, the probability that the delay is less than t for the ℓ th priority

class traffic is given by

$$\Pr(\text{delay} = \frac{U_k}{C_{GQ}}) = \frac{1}{T_{GQ}^\ell} \sum_{n=0}^N n \lambda_\ell \left[E_n^{k-1}(U_k) - E_n^k(U_k) \right], k \in [\ell, K1] \dots (24)$$

Average delay for the ℓ th priority class traffic ($\ell \in [2, K1]$) is obtained from the delay distribution function

$$D_{GQ}^{\ell} = \int_0^{\left(\frac{U_{K1}}{C_{GQ}}\right)} td(D_{K1}^{\ell}(t)) + \sum_{k=1}^{K1-1} \int_{\left(\frac{U_{k+1}}{C_{GQ}}\right)}^{\left(\frac{U_k}{C_{GQ}}\right)} td(D_k^{\ell}(t)) + \sum_{k=1}^{K1} \frac{U_k}{C_{GQ}} \Pr(\text{delay} = \frac{U_k}{C_{GQ}}) \dots\dots\dots(25)$$

Therefore, the average delays for different priority classes are calculated below. Average delay when $\ell = 1$ is given by

$$D_{GQ}^1 = \int_0^{\left(\frac{U_{K1}}{C_{GQ}}\right)} td(D_{K1}^1(t)) + \sum_{k=1}^{K1-1} \int_{\left(\frac{U_{k+1}}{C_{GQ}}\right)}^{\left(\frac{U_k}{C_{GQ}}\right)} td(D_k^1(t)) + \sum_{k=2}^{K1} \frac{U_k}{C_{GQ}} \Pr(\text{delay} = \frac{U_k}{C_{GQ}}) + \frac{U_1}{C_{GQ}} \Pr(\text{delay} = \frac{U_1}{C_{GQ}}) \dots\dots\dots(26)$$

6.4.1.3 Silver Queue Modeling

Traffic from IS Controlled-Load Service (CLS) goes into Silver Queue. Traffic through the Silver queue is also per flow basic similar to Premium Queue. The Silver Queue provides service when Premium Queue and Gold Queue have no input traffic. Thus, the service rate of the Silver Queue reduces further to C_{SQ} ,

where $C_{SQ} = C_{GQ} - \sum_{\ell=1}^{K1} \sum_{n=0}^N n\lambda_{\ell}P(n)$. Let P_{eq} be the probability that G old

Queue is empty and it is written as

$$P_{eq} = \sum_{n=0}^N E_n^{K1}(0) \dots\dots\dots(27)$$

Assume the arrival rate λ_{SQ} of the IS Controlled-Load Service (CLS) is of Poisson distribution and the service time $\frac{1}{C_{SQ}}$ is of exponential distribution. As long as $\lambda_{SQ} < C_{SQ}$, which is of the normal case, and the buffer is sufficiently large in size. Hence, the Silver Queue can be modeled as M/M/1 queuing system.

Figure 6.21 . Modeling of Silver Queue

The probability of n packets in the system

$$P_n = P_0 \left[\frac{\lambda_{SQ}}{C_{SQ}} \right]^n \dots\dots\dots(28)$$

The probability that the system is empty

$$P_0 = \left[1 - \frac{\lambda_{SQ}}{C_{SQ}} \right] \dots\dots\dots (29)$$

Thus, the throughput of the Silver Queue

$$T_{SQ} = \sum_{n=0}^{\infty} \lambda_{SQ} P_n = \lambda_{SQ} \quad \text{if } \lambda_{SQ} < C_{SQ} \quad \dots\dots\dots(30)$$

Number of packets in the system

$$N_{SQ} = \sum_{n=0}^{\infty} n P_n = \frac{\lambda_{SQ}}{C_{SQ} - \lambda_{SQ}} \quad \dots\dots\dots(31)$$

By Little’s Theorem, the average delay of the Silver Queue

$$D_{SQ} = \frac{N_{SQ}}{T_{SQ}} = \frac{1}{C_{SQ} - \lambda_{SQ}} \quad \dots\dots\dots(32)$$

6.4.1.4 Fluid Model and Analysis of Bronze Queue

DS Low Priority Queue provides the service when above three queues have no input traffic since it is of the lowest priority. Let P_{esq} be the probability that Silver Queue is empty and it is expressed as

$$P_{esq} = \left[1 - \frac{\lambda_{SQ}}{C_{SQ}} \right] \quad \dots\dots\dots(33)$$

Multiple priorities control system with different thresholds is implemented. The detailed Modeling of Bronze Queue is similar to the Gold Queue and consists of $U_{K1+1}, U_{K1+2}, \dots, U_{K2}$ different thresholds for packet loss control over the partitioned buffer. With these multiple threshold setting, hence, DS Low Priority traffic with priority level $\ell \in [K1+1, K2]$ can input to the Bronze Queue.

Similar to the Gold Queue, two scenarios are needed for consideration since the three higher priority queues may or may not have input traffic and the performance analysis of the Bronze Queue under these two situations are different.

Scenario I. *Premium Queue, Gold Queue and Silver Queue have no Input traffic*

Assuming the input traffic to the DS High Priority Queue (Gold Queue) consists of totally i sources in the ON state.

Defining $C_{BQi} = C_{SQ} - i\lambda^{K1}$. Hence, Equation (8) can be modified as shown in (34)

, with $E_{in}^\ell(x)$ indicating Equilibrium Probability that buffer contents of Bronze Queue (DS Low Priority Queue) does not exceed x when there are i sources at ON state for Gold Queue and n sources are at ON state for Bronze Queue.

$$(n\lambda^\ell - C_{BQi}) \frac{dE_{in}^\ell}{dx} = (N - n + 1)\alpha E_{i(n-1)}^\ell - [(N - n)\alpha + n\beta] E_{in}^\ell + (n + 1)\beta E_{j(n+1)}^\ell$$

.....(34)

where $\ell \in [K1+1, K2], n, j \in [0, N], x \in [0, U_{N+1}]$

The solution to (34) is ,

$$E_{in}^\ell(x) = \sum_{n=0}^N a_{in}^\ell V_{in}^\ell e^{z_{in}^\ell x}, x \in [0, U_{N+1}] \dots\dots\dots (35)$$

In (35), the coefficients $\{a_{in}^\ell\}$ can be obtained from boundary conditions by defining

$$Q_{iD}^\ell = \{n | n\lambda^\ell < C_{BQi}\} \text{ and } Q_{iH}^\ell = \{n | n\lambda^\ell > C_{BQi}\}$$

With $P(n)$ previously defined in (14), the boundary conditions of the Bronze Queue are

$$E_{in}^{K1+1}(U_{N+1}) = P(n) \quad \text{if } n \in Q_{iD}^{K1+1} \dots\dots\dots (36)$$

$$E_{in}^\ell(U_\ell) = E_{in}^{\ell-1}(U_\ell) \quad \text{if } (n \in Q_{iH}^{\ell-1} \cup Q_{iD}^\ell, K1+1 \leq \ell \leq K2) \dots\dots\dots (37)$$

$$E_n^{K2}(0) = 0 \quad \text{if } n \in Q_{iH}^{K2} \dots\dots\dots (38)$$

Throughput Calculation of Bronze Queue (DS Low Priority Buffer)

Now we are at the stage to calculate the throughput of the Bronze Queue traffic. First and foremost, for the priority class $K1+1$, throughput is given by

$$\begin{aligned} T_{i1}^{K1+1} &= (\text{Arrival Rate of } K1+1 \text{ priority class traffic}) - \\ &\quad (\text{Loss Rate of } K1+1 \text{ priority class traffic}) \\ &= \sum_{n=0}^N \lambda_{K1+1} n E_{in}^{K1+1}(U_{K1+1}) + C_{BQ1} \left[1 - \sum_{n=0}^N E_{in}^{K1+1}(U_{K1+1}) \right] \dots\dots\dots (39) \end{aligned}$$

The throughputs of the classes other than class 1 ($\ell = K1+2, K1+3, \dots, K2$) are given by

$$\begin{aligned} T_{i1}^{\ell} &= (\text{Arrival Rate of } \ell \text{ th priority class traffic}) - \\ &\quad (\text{Loss Rate of the } \ell \text{ th priority class traffic}) \\ &= \sum_{n=0}^N \lambda_{\ell} n E_{in}^{\ell-1}(U_{\ell}) - \sum_{n=Q_{H}^{\ell-1} \cap Q_{L}^{\ell}} \left\{ \left[E_{in}^{\ell-1}(U_{\ell-1}) - E_{in}^{\ell}(U_{\ell}) \right] \times \left[n \lambda^{\ell} - C_{BQ\ell} \right] \right\} \dots\dots\dots (40) \end{aligned}$$

Therefore, Throughput of Bronze Queue for scenario I is
(Premium Queue, Gold Queue and Silver Queues are empty)

$$T_1^{\ell} = \sum_{i=0}^{\lambda_{K1} \frac{C_{SQ}}{\lambda_{K1}}} T_{li}^{\ell} E_i^{K1}(0) \quad \text{for } \ell \in [K1+1, K2] \quad \dots\dots\dots (41)$$

Next, consideration is given in the second case where the higher priority queues have input traffic.

Scenario II. *Premium Queue, Gold Queue and Silver Queue are not empty*

$$\text{Hence , } C_{BQi} = 0$$

Under such circumstances, the Boundary conditions are

$$E_n^{K2}(0) = 0 \dots\dots\dots (42)$$

$$E_{in}^\ell(U_\ell) = E_{in}^{\ell-1}(U_\ell) \dots\dots\dots (43)$$

The Throughput of Bronze Queue for scenario II is

$$T_2^\ell = \sum_{n=0}^N \lambda_\ell n E_n^\ell(U_\ell) \quad \text{where } (K1+1 \leq \ell \leq K2) \dots\dots\dots(44)$$

In conclusion , the Average Throughput for DS Low Priority Queue (Bronze Queue) is given by

$$T_{BQ}^\ell = P_{epq} P_{esq} T_1^\ell + \left[(1 - P_{epq}) + P_{epq} (1 - P_{egq}) + P_{epq} P_{egq} (1 - P_{esq}) \right] T_2^\ell, \\ \text{where } (K1+1 \leq \ell \leq K2) \dots\dots\dots(45)$$

Arrival Rate of the ℓ th priority class traffic in the Bronze Queue :

$$A_{BQ}^\ell = \sum_{n=0}^N \lambda_\ell n P(n), \quad \text{where } (K1+1 \leq \ell \leq K2) \dots\dots\dots(46)$$

The packet loss probability due to buffer overflow of the ℓ th priority class traffic in the Bronze Queue is given by

$$R_{BQ}^\ell = 1 - \frac{T_{BQ}^\ell}{A_{BQ}^\ell}, \quad \text{where } (K1+1 \leq \ell \leq K2) \dots\dots\dots(47)$$

6.5 Numerical Results and Investigations

The following numerical results emphasize on the effect of priorities on the steady-state performance including the different IntServ (IS) traffic and different DiffServ (DS) traffic. For the illustrative purposes only, as shown in Figure 6-22, the queues are classified according to service mapping proposed in Table 6.6.

Figure 6.22. Priority Queuing Model for Numerical Analysis

For the purpose of Performance Analysis

CLASS 1 is designated for **Loss Sensitive & Delay Sensitive** Traffic Class

CLASS 2 is designated for **Delay Sensitive** Traffic Class

CLASS 3 is designated for **Loss Sensitive & Delay Insensitive**
(*Low Drop Precedence*) Traffic Class

CLASS 4 is designated for **Loss Sensitive & Delay Insensitive**
(*High Drop Precedence*) Traffic Class

CLASS 5 is designated for **Best effort** Traffic Class

Table 6.6 Simplified Proposed QoS Mapping

Incoming QoS Request	MPLS EXP Field	Equivalent Priority Mapping	MPLS Service Class Designation
IntServ GS	111	Guaranteed Bandwidth & Delay	Premium (Highest Priority Service)
DiffServ EF	110	Loss Sensitive & Delay Sensitive	Gold (High Priority Key Service)
DiffServ AF1X	101	Delay Sensitive	Silver (Normal Priority Service)
IntServ CLS	100	Low Loss Rate	
DiffServ AF2X	011	Loss Sensitive & Delay Insensitive (Low Drop Precedence)	
DiffServ AF3X	010	Loss Sensitive & Delay Insensitive (High Drop Precedence)	Bronze (Low Priority Service)
DiffServ AF4X	001	Loss Insensitive & Delay Insensitive	
Best Effort	000	No performance Guarantee (Best Effort Delivery)	Best Effort

The thesis propose 4 parallel queues at the output port of modified LSR. LSR is modeled as a tandem queuing system. Gold Queue (DS High Priority Queue) is designed for two classes of packet loss priority, namely Class 1 and Class 2. Class 1 (DS: Expedited Forwarding) higher priority than Class 2 (DS : Assured Forwarding AF1x). Threshold U_2 is used for the packet loss priority control in the Gold Queue. Loss Priority mechanism discards Class 2 packets when the buffer occupancy exceeds the threshold U_2 . All incoming packets to Gold Queue are accepted if the buffer occupancy is less than U_2 . Higher priority packets (Class 1) traffic are lost only if the buffer is full. When the buffer occupancy is exactly U_2 , the rate at which class 1 packets are accepted equals the residual capacity. Service to packets already in the buffer is provided on a FCFS basis. The presence of low priority packets (Class 2) in the buffer influence the delay seen by the higher priority packets. This influence is controlled by the choice of threshold U_2 . Thus, threshold U_2 controls the important tradeoffs between delay of the higher priority packets and the loss probability of the Class 1 packets. Bronze Queue accepts DS traffic but lower priority than Gold Queue input. Bronze Queue of size U_3 includes three classes of packet loss priority, Class 3, Class 4 and Class 5 respectively. Class 3 traffic has higher priority than class 4 traffic. Similarly, Class 4 traffic has higher priority than class 5 traffic. A proposed scheme is that DiffServ (DS) AF2x, AF3x, AF4x traffic are assigned to Class 3, Class 4 and Class 5.

Since the behavior of two queues storing per-flow IntServ (IS) traffic can be analyzed easily, therefore, the thesis focuses on the two queues comprising the DiffServ (DS) traffic which are designated as Gold Queue and Bronze Queue. The numerical results for performance evaluation are presented in this section.

Figure 6.23 The Effect of Class 1 Traffic on Loss Probability of Class 2

The effects of higher priority traffic on the performance of different lower priority traffic classes has been investigated. Figure 6.23 demonstrates the effect of Class 1 traffic on performance of Class 2 packet loss probability with and without threshold control. Numerical experiments are performed by varying percentage of Class 1 traffic load from 10% to 30 % with and without threshold control for Class 2 packets, while traffic load for Class 2 is fixed to be 40% of the total capacity. Results show that packet loss probability of Class 2 increases significantly when increasing the traffic load of Class 1. For example, when the total offered load is 0.65, the loss probability of Class 2 increases from 10^{-14} to 10^{-4} as the Class 1 traffic load increases from 10% to 30% respectively. Furthermore, while the solid lines represent the scenarios with threshold control, the dotted lines show the scenarios without threshold control. In this scenario, it can be found that with threshold control , the loss probability of Class 2 traffic increases.

Figure 6.24 The Effect of Class 1 Traffic on Class 3 Loss Probability

Figure 6.25 The Effect of Class 1 Traffic on Class 4 Loss Probability

Figure 6.26 The Effect of Class 1 Traffic on Class 5 Loss Probability

The effect of Class 1 traffic on DS Low Priority Class Traffic is also evaluated. In Figure 6.24, Figure 6.25 and Figure 6.26, plots are obtained by fixing the percentage of traffic load for each class such as Class 2, Class 3, Class 4 and Class 5 to be 20% of the total capacity while the percentage of Class 1 traffic load is increased from 5% to 15 % respectively. Numerical results from Figure 6.24, Figure 6.25 and Figure 6.26 indicate that priority scheme is effective on the performance of packet loss probability for the delay sensitive and insensitive traffic such as voice and e-mail. The investigation shows that traffic load offered by Class 1 has significant effect on the performance of packet loss probability for all classes in Bronze Queue (DS Low Priority Queue). When the traffic load of Class 1 decreases, the throughput of Class 1 decreases. By contrast, the throughput of lower priority traffic increases.

Figure 6.27. The Effect of Class 2 Traffic on Class 1 Loss Probability

Figure 6.27 demonstrates the effect of Class 2 traffic on performance of Class 1 packet loss probability with and without buffer threshold control. In this numerical experiment, results are obtained by varying the percentage of Class 2 traffic load from 10% to 30% of the total capacity with and without threshold control. On the other hand, the traffic load from Class 1 is fixed to be 40% of the total capacity. Moreover, while the solid lines represent the scenarios with threshold control, the dotted lines show the scenarios without threshold control. In this scenario, it can be found that with threshold control as the Class 2 traffic load increases from 10% to 30% of the total capacity, the packet loss probability of Class 1 only increases by a very small amount. However, without threshold control, the packet loss rate of Class 1 increases (i.e., the performance of Class 1 degrades) significantly when the Class 2 traffic load increases from 10% to 30% of the total capacity.

Figure 6.28. The Effect of Class 1 & Class 2 Traffic on Loss Probability of Class 3

Figure 6.29. The Effect of Class 1 & Class 2 Traffic on Loss Probability of Class 4

Figure 6.30. The Effect of Class 1 & Class 2 Traffic on Loss Probability of Class 5

Effect of Class 1 and Class 2 traffic on DS Low Priority Class Traffic are also examined. The numerical experiments are performed in order to investigate the effect of Gold Queue (DS High Priority) traffic load on the performance of Bronze Queue (DS Low Priority) traffic. The results are shown in Figure 6.28, Figure 6.29 and Figure 6.30 respectively. In these plots, Bronze Queue traffic (Class 3, Class 4 and Class 5) are fixed at 20% of total capacity while Class 1 and Class 2 traffic are varied from 10% to 15%. In this scenario, it can be observed that the Class 1 and Class 2 traffic has also large effect on packet loss probability of DS low priority class traffic. Furthermore, on account of buffer threshold control, it is found that Class 3 has lower packet loss rate than Class4. Similarly, Class 4 has lower packet loss rate than Class 5. As the lowest Class traffic, the performance of Class 5 will be affected by all other higher classes. An illustrative example referring to Figure 6.28, Figure 6.29 and Figure 6.30 is shown in Table 6.7.

Table 6.7 Packet Loss Probability Evaluation

CASE	OFFERED LOAD	AMOUNT OF CLASS 1& CLASS 2 TRAFFIC	TRAFFIC CLASS FOR EVALUATION	PACKET LOSS PROBABILITY
1.	0.7	0.15	CLASS 3	10^{-11}
2.	0.7	0.15	CLASS 4	$10^{-3.8}$
3.	0.7	0.15	CLASS 5	$10^{-1.9}$

As a result, in order to achieve the desired packet loss rate for different traffic classes, both the total traffic load and the traffic load offered by other higher classes (eg. Class 1) need to be properly controlled.

Figure 6.31. The Effect of Buffer Threshold on Packet Loss Probability of Class 1

Figure 6.32 The Effect of Buffer Threshold on Packet Loss Probability of Class 2

The effect of buffer threshold on performance of packet loss probability on Class 1 traffic and Class 2 traffic is also studied. The numerical results of Figure 6.31 and Figure 6.32 demonstrate the packet loss probability of Class 1 and Class 2 for different threshold values in the DS high priority Queue (Gold Queue). In this case, the traffic load from Class 2 is fixed to be 40% of the total capacity and the traffic load from Class 1 is 40% of the total capacity while varying threshold values from 0.2 to 0.6 respectively. Moreover, the solid lines represent the scenarios with different threshold controls, the dotted lines show the scenarios without threshold control. Figure 6.31 and Figure 6.32 indicate that the effect of different threshold values on the performance of packet loss probability is significant for both Class 1 and Class 2 traffic. An illustrative example is shown in the following Table 6.8.

Table 6.8 Threshold Evaluation

OFFERED LOAD	TRAFFIC CLASS	THRESHOLD VALUE	PACKET LOSS PROBABILITY
0.9 of the total capacity	Class 1	0.2	10^{-14}
	Class 1	0.4	$10^{-11.63}$
	Class 1	0.6	10^{-9}
0.45 of the total capacity	Class 2	0.2	10^{-6}
	Class 2	0.4	$10^{-8.8}$
	Class 2	0.6	10^{-11}

Using the buffer threshold control in the DS high priority Queue (Gold Queue), it can be found that packet loss probability of Class 1 traffic decreases significantly compared with the case of without threshold control. Alternatively, packet loss probability of Class 2 becomes higher with threshold control. Therefore, significant decrease of Class 1 packet loss rate occurs at the moderate increment of Class 2 packet loss rate. The evaluation has shown that the size of the buffer threshold will affect the performance of each traffic class. In addition, within the same queue, there is trade-off between the packet loss rate of the lower priority class traffic and the mean delay of the higher priority class traffic. In order to achieve the desired performance for each class of traffic, it is important to control the trade-off.

Figure 6.33 Throughput of Class 1 and Class 2

Figure 6.34 . Throughput of Class 3 , Class 4 and Class 5

A set of performance on throughput for each class is illustrated in Figure 6.33, Figure 6.34 . Figure 6.33 presents throughput of Class 1 and Class 2 versus offered load in which Class 1 and Class 2 traffic load are fixed 40% of the total capacity. In Figure 6.33, it can be seen that Class 1 traffic has higher throughput than Class 2. This is because there is a threshold for Class 2 and Loss Priority mechanism discards Class 2 packets when the buffer occupancy exceeds this threshold . All incoming packets to Gold Queue (DS High Priority Queue) are accepted if the buffer occupancy is less than this threshold. Higher priority packets (Class 1) traffic are lost only if the buffer is full. When traffic in the DS High Priority Queue exceeds this threshold , the traffic for Class 2 will be dropped.

Similarly, Figure 6.34 shows the throughput of Class 3, Class 4 and Class 5 versus offered load in which the traffic load for each of these three classes are fixed at 25% of the total offered load. The result of Figure 6.34 indicates that Class 3 traffic has better throughput than that of Class 4 and Class 5. This is also because there are two threshold for Class 4 and Class 5 respectively. When traffic in the DS Low Priority Queue exceeds these thresholds, the traffic for Class 5 and Class 4 will be dropped. Higher priority packets (Class 3) traffic are lost only if the buffer is full.

Figure 6.35. The Effect of Class 1 Traffic on Class 2 Throughput

In addition, the effect of Class 1 traffic on Class 2 throughput is illustrated in Figure 6.35 in which traffic load for Class 2 is fixed to be 40% of the total capacity while the percentage of Class 1 traffic is varied from 10% to 40% with and without threshold control. In Figure 6.35, the solid lines represent the scenarios with threshold controls, the dotted lines show the scenarios without threshold control.

From the plot of Figure 6.35, it can be observed that without threshold control, when the traffic load of Class 1 increases from 10% to 40% of the total capacity, the throughput of Class 2 increases. With threshold control, the throughput of Class 2 reduces significantly compared with the scenario without threshold control. On the contrary, the throughput of Class 1 becomes higher with implementation of threshold control.

According to the results of Figure 6.33, Figure 6.34 and Figure 6.35, the investigation shows that DS high priority queue has better performance on throughput than DS low priority queue. On the other hand, within the same queue, the traffic with higher priority class traffic has better throughput than that with lower priority class. That is, Class 1 traffic has better performance than Class 2 traffic and Class 3 traffic has better performance than Class 4 and Class 5 traffic respectively.

Figure 6.36 . Mean Delay of Class 2 and Class 3

Numerical result of Figure 6.36 illustrates the average delay for Class 2 of Gold Queue (DS High Priority Queue) and Class 3 of Bronze Queue (DS Low Priority Queue). In this case, traffic load for Class 2 and Class 3 are fixed at 80% of their total capacity. The result indicates that Class 2 has much better performance than Class 3 with respect to mean delay. As a result, it can infer that Class 1 traffic also has excellent performance on delay because Class 1 is higher priority than Class 2 in the same Queue. In similar way, Class 4 and 5 have poor delay performance as compare to Class 3 since Class 3 traffic is higher priority than Class 4 and Class 5 in the same queue (DS Low Priority Queue).

6.6 Summary

In this chapter, an architectural model is described to deliver end-to-end QoS in mobile environment using MPLS approach. The thesis focuses how incoming QoS service is mapped directly to the MPLS equivalent QoS service. Various functional elements required for the model are presented and a number of modifications have been proposed to be implemented to the MPLS architecture and its associated signaling protocol CR-LDP.

As the second major part of the thesis, the QoS mapping from IS/DS capable mobile user traffic into Mobile MPLS has been proposed and architectural modifications of LSR have been elaborated. Queuing model for providing scalable QoS with multiple priority class over Mobile MPLS has been proposed. Modified LSR of Mobile MPLS is modeled as a tandem queuing system in which each output link consists of four parallel priority output queues for buffer management of different traffic classes to meet different QoS requirement. A number of key parameters are measured to determine the QoS performance in the proposed model : They are throughput, packet loss probability and average delay . A scheme of multi-threshold has been implemented in each DS Queue for packet loss priority control. The loss priority is implemented by selectively discarding low priority packets when the buffer contents exceeds a threshold level in a buffer. Performance analysis of the fluid model is established in order to analyze the proposed queuing model.

The implementation of multi-class QoS on priority basis makes the QoS provision in Mobile MPLS more flexible. The numerical results obtained clearly demonstrate that performance of high priority traffic class is enhanced by using priority queuing scheme. The behavior of multi-class priority scheme is studied with a variety of traffic conditions. The performance evaluation is mainly focused on the loss probability and throughput of DS traffic. Investigation of DS Queue shows that the traffic in the high priority DS Queue is capable of providing better performance than DS low priority queue .

In connection with loss probability, the higher class traffic has better performance than lower class traffic in the same queue. Loss probability of lower class traffic is affected by all higher class traffic. In order to achieve the desired performance, not only the overall traffic load but also high priority traffic load should be properly controlled. Moreover, the investigation shows that packets of different flows can be labeled differently and thus received different QoS along the path and service differentiation can greatly benefit from traffic engineering associated with simple priority scheduling at the LSR without the use of complex per class weight scheduling .

Network scalability is achieved by the aggregation of same flows of the same class packet inside same LSP. Depending on traffic characteristics from end host, different LSP could be created for packets with different class of service (CoS) requirements. Isolating various flows in the network yields in a better quality of service to the various classes. Numerical results prove that the proposed scheme is beneficial for provision of scalable QoS with multiple priority classes over Mobile MPLS. Therefore, customers are able to achieve the high priority classes from the Internet service provider (ISP) of MPLS based mobile network to support the guaranteed QoS .

CHAPTER 7 CONCLUSIONS AND RECOMMENDATIONS

7.1 Conclusions

This thesis focuses on providing of scalable QoS over broadband Internet and mobile networks. It includes three major parts. The first focus is to investigate beneficial solutions for providing guaranteed QoS and the modeling and performance analysis of aggregate reservation to provide required QoS and scalability. The second focus is to study the characteristics analysis of MPLS, QoS routing and the Mobility Aware mobile access MPLS as a mobility solution with continuous QoS support in the mobility domain including the performance evaluation of the proposed LSP configuration on MPLS in mobile network and the reliable handoff scheme for fast handover performance. The last focus is the modeling and performance analysis of IntServ (IS) and DiffServ (DS) traffic with multiple priority classes over MPLS based mobile network.

First of all, the thesis studies the two IP QoS models, namely IntServ (IS) and DiffServ (DS) and then provides technical comparisons of these two approaches. The choice of the packet service discipline at the routers is one of the most important issues in providing guaranteed performance services for new applications such as video conferencing and medical imaging which require strict performance requirements in terms of throughput, delay and packet loss rate.

The thesis investigates three beneficial solutions for providing guaranteed QoS. The first solution is the dynamic packet state (DPS) technique, which is able to provide the scalable and robust network since edge routers perform per-flow management and core routers do not perform per-flow management. Using the key technique DPS for QoS scheduling and admission control in scalable core network, it is possible to significantly enhance the scalability for implementing guaranteed services.

The second solution is the rate controlled service discipline (RCSD) relating to the provision of per connection end-to-end delay guarantees in a high speed network. In such discipline, traffic from each connection is reshaped at every hop. The main disadvantage of this service discipline is that the end to end delay guarantees are obtained as the sum of the worst case delays at each node. But the problem can be eliminated by using good traffic shapers. With a suitable choice of shaper envelopes the RCS discipline can provide the same end-to-end delay guarantees in which the best delay bound for GPS can be provided. The investigation evaluates that using RCS discipline, it can offer a powerful solution to provide end-to-end performance guarantees to individual connections over broadband networks. The third solution is the aggregate base resource reservation scheme which is useful for providing the QoS of the network in a scalable way. By using aggregation of resource reservation, the network can simultaneously be achieved scalability and high utilization.

In chapter 3, the modeling and performance analysis of aggregate reservation combining with IntServ has been presented. This novel solution does not requires per flow management at core router but it can implement guaranteed services that can be provided with per flow mechanism. In this way, required QoS and scalability can simultaneously be achieved. The performance analysis of aggregate reservation has been evaluated by implementing numerical experiments on three performance metric: overload probability, reserved resource utilization and the normalized available bandwidth. The numerical results indicate that system performance rapidly enhance as the control time scale τ is decreased from demand time scale T and the performance of aggregation can be degraded because of sufficient noise in the demand. The ability of the model base technique for performance of aggregation has been evaluated using the MPEG video trace for simulated experiments.

The Trace driven simulation and model predictions for three performance measures are presented. The numerical and simulation experiments show that if the control time scale and demand time scale are separated by two orders of magnitude and variance of traffic is moderate, aggregation can provide performance close to that of strong service model IntServ.

Recently, there are several network technologies such as IPv6, IntServ (IS), DiffServ (DS) and MPLS have been proposed to meet the demand for QoS over IP. In DS mode, IP streams are marked and classified into different QoS level for different types of services. On the other hand, MPLS based on label switching offers an ability to build highly scalable networks. Based on the above techniques, network providers are able to support different class of QoS for heterogeneous IP traffic.

The core value and characteristics analysis of MPLS including various label encapsulation techniques have been described and two MPLS signaling protocols: Resource Reservation Protocol- Traffic Engineering (RSVP-TE) and Constraint based Routing Label Distribution Protocol (CR-LDP) dealing with QoS routing are discussed. The study points out that CR-LDP is a practical and powerful tool that can be used for traffic engineering and QoS routing. In order to improve the performance of IP routers, the thesis presents label swapping algorithm and traffic aggregation mechanism as a highly efficient scheme for forwarding data packets since packets of different flows can be labeled differently and received different QoS and hence achieving network scalability by implementing aggregation of traffic flows of the same class inside LSP.

There is much emergent interest among both service providers and equipment manufacturers in a marriage of multimedia and mobile networking technologies. It is important that the mobile Internet environment in the near future provides QoS guarantees due to mobility. To overcome mobility impact on service guarantee, Mobility Aware mobile access MPLS has been provided as a mobility solution. An MPLS architecture is considered for mobile access networks, since MPLS is a viable technology that will support a range of IP-based services with QoS. The proposed strategy implements scalable LSP configuration in mobile access MPLS domain.

In the proposed LSP configuration, LSP are created by a network operator based on provisioning policies. It supports the IP diff-serv model for supporting IP QoS services. In order to make the proposed LSP scheme be scalable to match of growth for future mobile IP network, the proposed LSP scheme has been designed to use multiple dimension label architecture rather than flat static architecture. The LSP scheme in this thesis is suitable and scalable network requirement. It provides simple and fast handoff by updating the GW's (gateway) forwarding table while supporting the same QoS after handoff. In a mobile access MPLS domain, GW performs the mapping for downlink traffic and each base station (BS) performs the mapping for uplink traffic. The performance is evaluated by simulation experiment which is conducted using the same mobility model as the proposed LSP scheme. The evaluation has been focusing on network mobility in terms of new arrival call blocking probability and handover call blocking probability which are most important and essential technical terms in mobile networks. The simulation results clearly demonstrate the advantage that the proposed LSP scheme offers lower handoff call blocking probability (i.e., significantly improves probability of successful handover) under different traffic load scenarios. The investigation of major performance measures conducted in the simulation experiments show that the proposed LSP scheme using MPLS is well suited for supporting good degree of success on mobility with continuous QoS provision in IP based mobile access networks.

Handoff is an essential mechanism of IP based mobile access networks since if handover does not occur quickly, QoS may degenerate below an acceptable level. Efficient handoff schemes are cost effective of enhancing the capacity and QoS of cellular system. Therefore, a reliable handoff scheme is provided by developing five functional tasks. The scheme uses received signal strength (RSS) measurements of different BS at MN as the handover initiation criterion. The selection of the threshold levels for the received signal level criterion can be performed by using predicted profiles. The effectiveness of RSS with threshold depends on prior knowledge of the crossover signal strength between the current BS and target BS.

Based on investigation of these handover criteria, the thesis provides handover request flow diagram. Resource sharing mechanism by each BS in the proposed scheme to provide priorities to handoff requests without significantly jeopardizing the new connection requests is demonstrated. Call flow diagram for channel reservation based handoff prioritization is developed in order to improve the successful handoff probability and overcome the problem of underutilizing spectrum efficiency. The proposed novel approach adopts MCHO (Mobile Controlled Hand Off) strategy with forward handover protocol due to reliable and fast handover performance. It has been shown, by construction, that the reliable handoff scheme presented in this thesis is scalable and provides the capability for QoS.

In chapter 6, the thesis first highlights the framework combining the DiffServ (DS) ideas and modified mobile enabled MPLS in order to provide end-to-end QoS delivery between correspondent node (CN) of mobile user and mobile node (MN). By such a way, mobile users are able to access services according to QoS contracts negotiated with service providers. No per flow state information is required. The traffic aggregation in the MPLS leads to increased scalability of the service provisioning architecture.

Afterwards, the thesis presents proposed QoS mapping among IntServ, DiffServ and MPLS network, queuing model developing and the performance analysis for heterogeneous traffic with multi-class QoS in mobile MPLS. Multi-class QoS is implemented on differentiated service basis using loss priority scheme and delay priority scheme incorporated in the packet header of MPLS. Modified LSR of mobile enabled MPLS is modeled as a tandem queuing system in which each output link consists of four parallel priority output queues representing different delay priorities for buffer management of different traffic classes to meet different QoS requirement .

This thesis provides traffic classification and prioritized scheduling for performance guarantee by implementing multiple thresholds in two queues (DS high priority queue & DS low priority queue) for packet loss priority control. A number of key parameters such as throughput, packet loss probability and delay are measured to determine the QoS performance in the proposed queuing model by using fluid flow technique. The obtained numerical results from the performance analysis demonstrate that the proposed queuing model and the differentiated service based on the priority schemes defined in MPLS packet header are able to provide the multi-class QoS over MPLS based mobile network.

The fundamental theme of this research uses modeling and analysis techniques for scalable QoS in support of multimedia services over broadband Internet and MPLS based mobile networks. Throughout this thesis, the proposed mechanisms and algorithms such as aggregate resource reservation coupling with IntServ (IS) and queuing models with multi-class QoS over MPLS based mobile network are evaluated using numerical experiments and simulations based on real-life video trace.

7.2 Recommendation and Future Research

This section indicates several related issues and promising areas need to be investigated more thoroughly in the future work that follow from the work in this thesis.

The effect of characteristics of wireless transmission channel, including bit error rate, fading, multi-path effects on the performance of QoS in terms of packet error probability and throughput. In Mobility-Aware mobility access MPLS scheme, the transmission is through wireless channels. The quality of the wireless link is apt to change due to fading, multi-path effects and Doppler effects. How these features affect on the performance of QoS should be considered in future research .

One interesting and important part of future work is to conduct more sophisticated usage of queuing system for provision of multi-class QoS over mobile enabled MPLS network. The four –queue system presented in this research implements strict priority queuing scheme. For more sophisticated implementation purpose, a combination of priority queuing and Weighted Fair Queuing (WFQ) or Weighted Random Early Drop (WRED) could be employed in such system. Moreover, in this thesis four-queue system analysis only considers a single network node and numerical results primarily demonstrate the packet loss probabilities in various scenarios. Therefore, in the future research, the end-to-end performance parameters should be derived in terms of packet loss rate, delay and throughput.

AUTHOR'S PUBLICATIONS

- (1) M. N. Zaw, L. Zhang and H. Fu, "Comparison of Scalable QoS Schemes over Broadband Internet, " Vol. 27, No. 10, pp. 998 - 1007, *Computer Communications*, Elsevier Science, 2004.
- (2) M. N. Zaw, L. Zhang and H. Fu, "Scalable QoS over Broadband Internet," Accepted by *IEEE/ACM Transaction on Networking*, October 2003.
- (3) M. N. Zaw, L. Zhang and H. Fu, "MPLS Based Scalable QoS Provisioning in Mobile Networks," Accepted for publication on *Computer Communications*, Elsevier Science, December 2004

BIBLIOGRAPHY

- [1] F. Baker, C. Iturradle, F. Le Fuacheur, and B. Davie. "Aggregation of RSVP for IP4 and IP6 Reservations". Internet Draft, draft-ietf-issll-rsvp-aggr-02.txt, March, 2000.
- [2] Y. Bernet et. Al. "A framework for Integrated Services Operation Over DiffServ Networks," FRC 2998, November 2000
- [3] Y. Bernet et. Al. "A framework for differentiated services," November 1998 Internet Draft, draft-ietf-diffserv-framework-01.txt.
- [4] D. Black, S. Blake, M. Carlson, E. Davies, Z. Wang, and W. Weiss. "An Architecture for Differentiated Services," 1998. Internet RFC 2475.
- [5] Rosen, E., Viswanathan, A. and R. Callon, "Multiprotocol Label Switching Architecture," *IETF RFC* 3031.
- [6] Xipeng Xiao, Hannan, A; Bailey, B; Ni, L.M; "Traffic Engineering with MPLS in the Internet," *IEEE Network Magazine*, volume 14 Issue : 2, Mar/Apr 2000
- [7] A. Ghanwani, B. Jamoussi, and D. Fedyk, "Traffic Engineering Standards in IP Networks Using MPLS" *IEEE Communication Magazine*, Dec 1999, vol. 37
- [8] Vasu Jolly and Shabram Latifi "An Overview of MPLS and Constrained Based Routing," Department of Electrical and Computer Engineering, University of Nevada, Las Vegas, USA, <http://www.ee.unlv.edu/~venkim/opnet>
- [9] Wei Sun Praveen Bhaniramka, Raj Jain, "Quality of Service using Traffic Engineering over MPLS : An Analysis" 2000 *IEEE*
- [10] Zhong Ren, Chen-Khong tham, Chun-Choong Foo, and Chi-Chung Ko, "Integration of Mobile IP and Multi – Protocol Label Switching," ICC2001, June 2001.
- [11] S. Floyd and V. Jacobson. "Random early detection for congestion avoidance". *IEEE / ACM Transactions on Networking*, 1(4): 397-413, July 1993.

- [12] A. Demers, S. Keshav, and S. Shenker. "Analysis and simulation of a fair queueing algorithm," In *Journal of Internet working Research and experience*, page 3-26 ,October 1990.
- [13] S. Shenker R . Barden , D. Clark. " Integrated services in the Internet architecture: an overview," June 1994. Internet RFC 1633.
- [14] William Stalling " High-Speed Networks and Internets – Performance and QoS "Prentice Hall, 2nd edition, 2000.
- [15] J. Wraclawski, "The Use of RSVP with IETF Integrated Services," RFC 2210, September 1997
- [16] B. Barden, L. Zhang, S. Berson, S. Herzag and S. Jamin, "Resource Reservation Protocol (RSVP)-version 1 functional specification," RFC 2205, Internet Engineering Task Force, Oct 1997.
- [17] Kursoe, Jame F, "computer networking: a top-down approach featuring the Internet" Addison-Wesley, 2nd edition, 2003.
- [18] Paul Brittain Adrian Farrel, "MPLS Traffic Engineering: A Choice Of Signaling Protocols," *Data connection*, white paper 2000.
- [19] I. Stoica and H. Zhang. "Providing guaranteed services without per flow management," May 1999 Technical Report CMU-CS -99-133.
- [20] D. Clark and J. Wroclawski. "An approach to service allocation in the internet " Internet Draft, draftclark-diff-svc-alloc-00.txt, July 1997. ([www-rrp.lip6.fr/pub/documentaire/ietf QoS/draft/draft-clark-diffsvc -alloc-00.txt](http://www-rrp.lip6.fr/pub/documentaire/ietf/QoS/draft/draft-clark-diffsvc-alloc-00.txt))
- [21] K. Nichols, V. Jacobson , and L. Zhang. "An approach to service allocation in the Internet," November 1997, Internet Draft.
- [22] A. Parekh and R. Gallager . "A generalized processor sharing approach to flow control – the single node case". In *Proceedings of the INFOCOM'92*, 1992.

- [23] D. Verma , H. Zhang , and D. Ferrari . “Guaranteeing delay jitter bounds in packet switching networks “. In Proceedings of Tricomm ’91, pages 35-46, Chapel Hill, North Carolina , April 1991.
- [24] H. Zhang and D. Ferrari. “Rate-controlled service disciplines “. Journal of High Speed Networks, 3(4):389-412, 1994.
- [25] L. Zhang. “Virtual Clock : A new traffic control algorithm for packet switching networks “. In Proceedings of *ACM SIG-COM*’90, pages 19-29, Philadelphia, PA , September 1990.
- [26] N. Figueira and J. Pasquale . “An upper bound on delay for the Virtual Clock service discipline“. *IEEE / ACM Transactions on Networking*, 3(4), April 1995.
- [27] P.Goyal, S. Lam, and H. Vin . “Determining end-to-end delay bounds in heterogeneous networks “. In Proceedings of NOSS-DAV’ 95, pages 287-298, Durham, New Hampshire , April 1995.
- [28] D. Stilliadis and A. Verma . “ Latency-rate servers : A general model for analysis of traffic scheduling algorithms “. *IEEE/ACM Transactions on Networking*, 6(2) : 164-174, April 1998.
- [29] L. Georgiadis , R. Guerin, V. Peris, and K. Sivarajan . “ Efficient network QoS provisioning based on per node traffic shaping “. *IEEE/ACM Transactions on Networking*, 4(4), 482-501, August 1996.
- [30] D. Wrege, E. Knightly, H. Zhang, and J. Liebeherr. “Deterministic delay bounds for vbr video packet switching networks : Fundamental limits and practical trade-offs”. 4(3):352-362, June 1996.
- [31] D. C . Stephens, J.C.R. Bennett , and H. Zhang “Implementing scheduling algorithms in high speed networks”. *IEEE JSAC Special Issue on High Performance Switches/Routers*, Sept, 1999.
- [32] D.E. Wrege and J. Liebeherr. “A Near-Optimal Packet Scheduler for QoS Networks”. *IEEE INFOCOM 1997*, p.p. 576, Japan, 1997.

- [33] H. Zhang and D. Ferrari. "Rate-controlled static priority queueing". In Proceedings of *IEEE INFOCOM'93*, PAGES 227-236, San Fransisco, California, April 1993.
- [34] A. K. Parekh and R. G. Gallager, "A generalized processor sharing approach to flow control in integrated service networks : The single node case," *IEEE/ACM Trans. Networking* , vol.1, n0. 3, pp. 344-357, June 1993.
- [35] A. K. J. Parekh, "A generalized processor sharing approach to flow control in integrated service networks," Ph.D. dissertation , Laboratory for Information and Decision Systems, Massachusetts Institute of Technology, Cambridge, MA 02139, Feb. 1992.
- [36] A. K. Parekh and R. G. Gallager, "A generalized processor sharing approach to flow control in integrated service networks: The multiple node case," *IEEE/ACM Trans. Networking* , vol.2, n0. 2, pp. 137-150, Apr. 1994.
- [37] D. D. Kandlur, K. G. Shin, and D. Ferriari, "Real-time communication in multi-hop networks," Proc. *INFOCOM'91*, 1991, pp. 300-307.
- [38] D. Ferrari and D. Verma . "A scheme for real-time channel establishment in wide-area networks," *IEEE Journal on Selected Areas in communications*, 8(3) : 368-379, April 1990.
- [39] L. Georgiadis, R. Guerin, and A. Parekh, "Optimal multiplexing on a single node link : Delay and buffer requirements," *IEEE Transaction on Information Theory*, 43(5) : 1518-1535, September 1997.
- [40] R. L. Cruz, " Service burstiness and dynamic burstiness measures: a framework," *J. High Speed Networks* , vol. 1 , no. 2 , pp. 105-127, 1992.
- [41] P. Y Wang, Y. Yemini, D. Florissi, J. Zinky, P. Florissi, "Experimental QoS Performances of multimedia applications," Proceedings of *Nineteenth Annual Joint Conference of the IEEE computer and Communications Societies, INFOCOM 2000*, 2: 970-979, 2000

- [42] P. Ferguson and G. Huston, "Quality of Service," *John Wiley & Sons*, 1998.
- [43] T. Ferrari, W. Almesberger, and J. Le Boudec. "SRP : a Scalable Resource Reservation Protocol for the Internet". In Proceedings of *IWQOS '98*, Napa, CA, May 1998.
- [44] O. Schelen and S. Pink . "Aggregating resource reservation over multiple routing domains" In Proceedings of *IWQOS '98*, Napa, CA, May 1998.
- [45] J. Schmitt, M. Karsten, L. Wolf , and R. Steinmetz. "Aggregation of guaranteed service flows" In *IWQOS'99*, London , UK , May 1999.
- [46] A. Terzis, L. Zhang, and E. Hahne. "Making reservations for aggregate flows : Experiences from an RSVP tunnels implementation." In Proceedings of *IWQOS '98*, Napa, CA, May 1998.
- [47] P. Pan , E. Hahne, and H. Schulzrinne. "BGRP : A tree-based aggregation protocol for inter-domain reservations". *Journal of Communication and Networks* , 2(2): 157-167, June 2000.
- [48] P. Ferguson and G. Huston , "Quality of Service : Delivering QoS in the Internet and the Corporate Network," *Wiley Computer Books*, New York, 1998.
- [49] D. Awduche et al. "Extension to RSVP for LSP Tunnels". Internet Draft, draft-ietf-mpls-rsvp-lsp-tunnel-07.txt, August 2000.
- [50] H. Zhang, "Service disciplines for guaranteed performance service in packet – switching networks," *Proc. IEEE*, vol. 83, no. 10, pp. 1374-1396, Oct. 1995.
- [51] D. Clark, S. Shenker, and L. Zhang . " Supporting real-time applications in an integrated services packet networks : Architecture and mechanism". In Proceedings of *ACM SIGCOMM '92*, Baltimore, Maryland, August 1992.
- [52] Rayes, K. Sage, "Integrated management architecture for IP-based networks," *IEEE Communication Magazine*, 38(4) : 48-53, April 2000.
- [53] O. Yaron and M. Sidi, "Calculating performance bounds in communication networks," in *Proc. IEEE INFOCOM '93*, San Francisco, CA, Apr. 1993, pp. 539-546.

- [54] L. Kleinrock, *Queuing Systems* New York : Wiley, 1975.
- [55] B. Maglaris, et al., "Performance Models of Statistical Multiplexing in Packet Video Communications," in *IEEE Trans. Commun.*, Vol. 36, pp. 834-844, July 1998.
- [56] Y. Lin, S. Mohan, and A. Noerpel, "Queuing priority channel assignment strategies for PCS hand-off and initial access," *IEEE Trans. on Vehicular Technology*, vol. 43, no. 3, pp. 704-712, Aug. 1994
- [57] D. Verma, "Guaranteed performance communication in high speed network," Ph.D. dissertation , Univ, Calif , Berkeley, Nov. 1991.
- [58] R. Cruz, "A Calculus for network delay, Part I : Network elements in isolation," *IEEE Trans. Inform. Theory*, vol. 37, pp. 114-121, Jan. 1991.
- [59] C. Kalmanek, H. Kanakia, and S. Keshav, " Rate controlled servers for very high-speed networks," in *IEEE Global Telecommun. Conf.*, San Diego, CA Dec. 1990.
- [60] S. Golestani, "Congestion-free transmission of real-time traffic in packet networks," in *Proc. IEEE INFOCOM '90*, Sand Francisco, CA , June 1990, pp. 527-542, IEEE Computer and Commun . Societies.
- [61] E. Knightly, D. Wrege, J. Liebeherr, and H. Zhang, "Fundamental limits and tradeoffs for providing deterministic guarantees to VBR video traffic," in *Proc. ACM Sigmetric '95*, Ottawa, CA , May 1995, pp-275-286
- [62] E. Knightly, and H. Zhang, "Traffic characterization and switch utilization using deterministic bounding interval dependent traffic models," in *Proc. IEEE INFOCOM '95*, Boston , MA, Apr. 1995
- [63] "Multiprotocol Label Switching: Enhancing Routing in the New Public Network" White paper (Juniper Networks), <http://www.juniper.net>, part number: 200001-002, 27 Sept 1999.

- [64] B. Maglaris, et al., "Performance Models of Statistical Multiplexing in Packet Video Communications," in *IEEE Trans. Commun.*, Vol. 36, pp. 834-844, July 1998.
- [65] J. Liebeherr, D. Wrege, and D. Ferrari, "Exact admission control for networks with bounded delay services," Tech. Rep. CS-94-29, Univ. Virginia, Dept. Computer Science, July 1994.
- [66] "Tutorials on MPLS Architecture," <http://www.iec.org/online/tutorials/mpls>
- [67] Archan Misra, Subir Das, Anthony Mcauley, Ashutosh Dutta, Sajal K. Das "Integrating QoS Support in TeleMIP's Mobility Architecture" ICPWC'2000
- [68] I. Andrikopoulos, G. Pavlou, "Supporting Differentiated Services in MPLS Networks," Proc.IWQoS '99, Seventh International Workshop on QoS, 1999
- [69] Chen, T. M; T. H, "Reliable Services in MPLS" *IEEE Communication Magazine*, Dec 1999, vol.37, No. 12
- [70] H. Sariowan, "A service curve approach to performance guarantees in integrated-service networks," Ph.D. dissertation, Dept. Elect. Comput. Eng., Univ. California at Sandiego, 1996.
- [71] Wei Sun Praveen Bhaniramka, Raj Jain, "Quality of Service using Traffic Engineering over MPLS : An Analysis" 2000 *IEEE*
- [72] "MPLS – Making IP Quality of Service" web resources
- [73] "MPLS using LDP and ATM VC Switching" RFC 3035
- [74] "MPLS Label Stack Encoding" RFC 3032
- [75] Bruce Davie and Yakov Rekhter, "MPLS Technology and Applications," Morgan Kaufmann Publishers, 2000
- [76] Traffic Control in MPLS Network, <http://www.convergedigest.com/tutorials/mpls3>
- [77] Jong-Moon Chung, "Analysis of MPLS traffic engineering," Circuits and Systems, 2000. *Proceedings of the 43th IEEE Midwest Symposium on*, Volume: 2 2000 Page(s): 550-553 vol.2.

- [78] Rick Gallaher, online series, Global Knowledge and President of *Telecommunications Technical Services*, Inc. web resources:
www.telecommgazine.com/default.asp?journalid=3&func=MEDIA&option=p10
- [79] Netplane, Cisco, Juniper networks, Nortel networks, MPLSRC web resources
- [80] K. Nichols, et al., "Definition of the Differentiated Services field (DS field) in the IPv4 and IPv6 Headers," RFC 2474, standard tracks, December 1998.
- [81] L. Andersson, et al ., "LDP Specification," *IETF* Internet Draft, work in progress, MPLS WG, January 1999.
- [82] T. Li, Y. Rekhter, "A Provider Architecture for Differentiated Services and Traffic Engineering (PASTE)," RFC 2430, Oct. 1998.
- [83] D. Awduche, J. Malcolm, M. O'Dell, J. McManus, "Requirements for Traffic Engineering Over MPLS," RFC 2702, Sep. 1999.
- [84] E. Crawley et al., "A framework for QoS -based Routing in the Internet," RFC 2386, Aug. 1998.
- [85] X. Xiao and L. Ni, "Internet QoS: A Big Picture," *IEEE Network*, Mar./ Apr. 1999, pp. 8-18
- [86] R. L. Cruz, "Quality of Service Guarantees in Virtual Circuit Switched Networks," *IEEE J. Select. Areas Commun.*, vol. 13, 1995
- [87] Viswanathan, A.; Feldman, N.; Wang, Z.; Callon, R., " Evolution of Multiprotocol label switching," *IEEE Communications Magazine* , Volume : 36 Issue : 5, May 1998 Page(s) : 165-173
- [88] R. Rigault, "MPLS-VPN Solution Development by France Telecom," proceeding of *MPLS Forum 2000*, Apr. 2000 (FT)
- [89] R. Polley, "Implementing VPNs Using MPLS," Proceeding of *MPLS Forum 2000*, Apr. 2000 (Harris and Jeffries)
- [90] Talor, "Practical Development of Provider Based IP-VPNs with MPLS," Proceeding of *MPLS Forum 2000*, Apr. 2000 (Juniper)

- [91] S.Blake et al. "An Architecture for Differentiated Services," RFC 2475, informational, December 1998.
- [93] S. Shenker, et al., "Specification of the Guaranteed Quality of Service," RFC 2212 , September 1997
- [94] Heinanen et al., "Assured Forwarding PHB Group, " RFC 2597, June 1999
- [95] Jacobson et al., "An Expedited forwarding PHB," RFC 2598, June 1999
- [96] S. Deering, R. Hinden, " Internet Protocol, Version 6 (IPv6) specification," RFC 2460 , December, 1998.
- [97] Le Faucheur et al., "MPLS Support of Differentiated Services," IETF Internet Draft , March 2000
- [98] Jamoussi et al., "Constrained Based LSP Setup Using LDP," IETF Internet Draft, October 1999
- [99] R. Guerin et al., "QoS Routing Mechanisms and OSPF Extensions," draft-guerin-QoS-routing-ospf-05.txt, 1999
- [100] D. Clark, et al., "An Approach to Service Allocation in the Internet," draft-clark-different-svc-alloc-00.txt, July 1997
- [101] Sally Floyd, Van Jacobson, "Link-sharing and Resource Management Models for Packet Networks," *IEEE/ACM Transactions on Networking*, Vol. 3, No. 4, Aug. 1995
- [102] Nicolas Rouhana & Eric Horlait , "Differentiated Services and Integrated Services use of MPLS" Fifth *IEEE Symposium on Computers and Communications*, Jul.2000
- [103] S. Lavenberg, *Computer Performance Modeling Handbook*. New York: Academic, 1983
- [104] Franco Tommasi, Simone Molendini & Andrea Tricoo, "Mapping of IntServ/RSVP reservations into MPLS domains," University of Lecce, ITALY

- [105] S. Shah-Heydari, L. N. Tho, "MMPP Modeling of aggregated ATM traffic," *IEEE Canadian conference on Electrical and Computer Engineering*, 1: 129-132, 1998.
- [106] X. Chen, "Modeling connection admission control," *Proceedings of Twelfth Annual Joint Conference of the IEEE Computer and Communications Societies, INFOCOM'93*, 1: 274-281, 1993
- [107] W. Feng, et al., "Blue : A New Class of Active Queue Management Algorithms," *University of Michigan CSE-TR-387-99*, Apr. 1999
- [108] J. N. Daigle and D. M. Lucantoni, "Queueing systems having phase-dependent arrival and service rates," in *Numerical Solutions of Markov Chains* (Ed. W. J. Stewart) Marcel Dekker, 1991
- [109] G. A. Awater and F. C. Schoute, "Optimal queueing policies for fast packet switching of mixed traffic," *IEEE JSAC*, 9, No. 3, 1991, 458-467
- [110] R. Wolff, *Stochastic Modeling and the theory of Queues*. Englewood Cliffs, NJ: Prentice Hall, 1989.
- [111] A. I. Elwaid, D. Mitra and T. E. Stern, "Statistical multiplexing of Markov modulated sources: theory and computational algorithms," *Proc. ITC-13*, Copenhagen, June 1991, 495-499.
- [112] H. Heffes and D. Lucantoni, "A Markov modulated characterization of packet voice and data traffic and related statistical multiplexer performance," *IEEE JSAC*, 4, No. 6, 1986, 856-868
- [113] T. E. Stern and A. I. Elwalid, "Analysis of separable Markov-modulated rate models for information handling systems," *Adv, Appl. Prob.*, 23, 1991, 105-139
- [114] I. Kosten, "Stochastic theory of data-handling systems with groups of multiple sources," in *Performances of Computer-Communication Systems*, Elsevier, Amsterdam, 1984, 321-331
- [115] M. F. Neuts, "Matrix Geometric Solutions in Stochastic Models," John Hopkins University Press, Baltimore, 1981.

- [116] D. Mitra , "Stochastic Theory of A Fluid Model of Producers And Consumers Coupled By A Buffer," *Adv. Appl. Prob.*20, pp. 646-676, 1998.
- [117] M. Markaki & I.S. Venieris, "A novel buffer management scheme for CBQ-based IP routers in a combined IntServ and DiffServ architecture," *Proceedings. ISCC 2000. Fifth IEEE Symposium on* , pp : 347-352, 2000.
- [118] J.-Y. Le Boudec, "An efficient solution method for Markov models of ATM links with loss priorities," *Selected Areas in Communications, IEEE Journal on* Volume : 9 Issue : 3 , April 1997
- [119] T.G. Robertazzi, "Computer Networks and Systems: Queueing theory and Performance Evaluation," Springer-Verlag, New York Inc., 1990
- [120] H. Kroner, G. Hebuterne, P. Boyer and A. Gravery, "Priority management in ATM switching nodes," *IEEE JSAC* , 9, No. 3, 1991, 418-427
- [121] S. Q. Li, "Study of packet loss in a packet switched voice system," in *Proc. ICC'88*, June 1988.
- [122] R. Nagarajan, J. F. Kurose and D. Towsley, "Approximation techniques for computing packet loss in finite-buffered voice multiplexers," *IEEE JSAC*, 9, No. 3, 1991, 368-377
- [123] P. Van Mieghem, G. H Petit, B. Steyaert, "Throughput optimality of single queue priority schemes," *Proceedings of Second IEEE Symposium on computers and Communications*, pp. 240-248, 1997.
- [124] R. C. F. Tucker, "Accurate method for analysis of a packet speech multiplexer with limited delay," *IEEE Trans. Commun.*, vol. 36, pp. 479-483, apr. 1988.
- [125] C. Yuan and J. A. Silvester, " Queueing analysis of delay constrained voice traffic in a packet switching system," *IEEE J. Select, Areas Commun.*, vol. 7, pp. 729-738, June 1989.
- [126] L. Georgiadis, R. Guerin, and A. Parekh, "Optimal multiplexing on a single link : Delay and buffer requirements," *IEEE Trans. Inform. Theory*, vol. 43, pp. 1518-1535, Sept. 1997.

- [127] P. Goyal, S. Lam, and H. Vin, "Determining end-to-end delay bounds in heterogeneous networks," in Proc. 5th Int. workshop Network and Operating System Support for Digital Video (NOSSDAV), 1995, pp. 287-298
- [128] G. Mao, D. Habibi, "Heterogeneous on-off sources in the bufferless fluid flow model," Proceedings of *IEEE International Conference on Networks, 2000 (ICON 2000)*, pp. 307-312, 2000.
- [129] K. Sohraby, "Design and analysis of ATM networks with fluid flow sources," *IEEE Global Telecommunications Conference, 1997 (GLOBECOM' 97)*, 2 : 1108-1113, 1997
- [130] N. Arayanan, V.G. Kulkarni, "First passage times in fluid models with an application to two priority fluid systems," Proceedings of *IEEE International Computer Performance and Dependability Symposium*, pp 166-175, 1996.
- [131] P. Fonseca, J. M. Pitts, L. G. Cuthbert, "Exact fluid –flow analysis of single ON/OFF source feeding an ATM buffer with space priority," *Electronics Letters*, 31(13): 1028-1029, 22 June 1995.
- [132] S. Blaabjerg, H. Andersson, H. andersson, "Approximating the heterogeneous fluid queue with a birth-death fluid queue," *IEEE Transactions on communications* 43(5) : 1884-1887, May 1995.
- [133] P. Fonseca, L.G. Cuthbert, J.M. Pitts, "Study of a space priority mechanism for ATM networks using both analytical and simulation techniques," *IEEE Twelfth UK Teletraffic Symposium on Performance Engineering in Telecommunications Networks*, (Digest No. 1995/054), pp3/1-3/8, 1995.
- [134] C.H. Lam, T.T.Lee, "Fluid flow models with state –dependent service rate for data handling systems," *IEEE Global Telecommunications conference*, 1995 (GLOBECOM '95), 2: 1412-1416, 1995.

- [135] A. I. Elwalid & D. Mitra, "Fluid Models For The Analysis And Design Of Statistical Multiplexing With Loss Priorities On Multiple Classes Of Bursty Traffic," in : *IEEE INFOCOM'92*, 1992.
- [136] B. Maglaris, et al., "Performance Models of Statistical Multiplexing in Packet Video Communications," in *IEEE Trans. Commun.*, Vol. 36, pp. 834-844, July 1988.
- [137] K. Iida, K. Kawahara, T. Takine, Y. Oie, "Performance evaluation of the architecture for end-to-end quality-of-service provisioning," *IEEE Communications Magazine*, 38(4): 76-81, April 2000.
- [138] C. M. Aras, J. F. Kurose, D. S. Reeves, and H. Schulzrinne, "Real-time communication in packet-switched networks," *Proceedings of the IEEE*, vol. 82, no. 1, pp. 122-139, January 1994.
- [139] A.Y.M. Lin, J.A. Silvester, "Priority queuing strategies and buffer allocation protocols for traffic control at an ATM integrated broadband switching system," *IEEE JSAC* 9(9) (1991)1524-1536
- [140] Jamin, S., et al., "Comparison of measurement-based admission control algorithms for controlled-load service," *Proc. IEEE INFOCOM 97*, April 1997.
- [141] Ying-Dar Lin, Nai-Bin Hsu and Ren-hung Hwang "Granularity of QoS Routing in MPLS Networks " *IWQoS 2001, LNCS 2092*
- [142] Zhong Ren, Chen-Khong tham, Chun-Choong Foo, and Chi-Chung Ko, "Integration of Mobile IP and Multi – Protocol Label Switching," Internet Draft, July 2000.
- [143] G. Apostolopoulos, R. Guerin, S. Kamat and S.Tripathi, "Quality of service based routing : A performance perspective," in proceedings of *ACM SIGCOMM*, September 1998
- [144] G. Apostolopoulos and S.K. Tripathi, "On the effectiveness of path pre-computation in reducing the processing cost of on-demand QoS path computation," in *proceedings of IEEE Symposium on computers and communication*, June 1998.

- [145] S. Chen and K. Nahrstedt, "An overview of quality of service routing for next-generation high-speed networks: Problems and solutions," *IEEE Network Magazine*, pp.64-79, November/December 1998
- [146] F. Hao and E. W. Zegura, "Scalability techniques in QoS routing," Technical Report GIT-CC-99-16, College of Computing, Georgia Institute of Technology, Atlanta, GA, 1999
- [147] J. Y. Le Boudec and T. Przygienda, "A route pre-computation algorithm for integrated services networks," *Journal of Network and Systems Management*, vol.3, no.4, pp.427-449, 1995
- [148] W. C. Lee, M. G. Hluchyj, and P. A. Humblet, "Routing subject to quality of service constraints in integrated communication networks," *IEEE Network Magazine*, pp. 46-55, July/August 1995
- [149] Q. Ma and P. Steenkiste, "On path selection for traffic with bandwidth guarantees," in *Proceeding of IEEE International Conference on Network Protocols*, Atlanta, GA, October 1997
- [150] Q. Ma and P. Steenkiste, "Quality-of-service routing for traffic with performance guarantees," in *Proc. IFIP International Workshop on Quality of Service*, pp. 115-126, Columbia University, New York, May 1997.
- [151] M. Peyravian and A. D. Kshemkalyani, "Network path caching : Issues, algorithms and a simulation study," *Computer Communication*, vol. 20, pp. 605-614, 1997
- [152] C. Pornavalai, G. Chakraborty, and N. Shiratori, "QoS based routing in integrated services packet networks," in *Proceedings of IEEE International Conference on Network Protocols*, Atlanta, GA, October 1997
- [153] A. Shaikh, J. Rexford, and K. Shin, "Dynamics of quality-of-service routing with inaccurate link-state information," Technical Report CSE-TR-350-97, *Computer Science and Engineering Division, University of Michigan*, Ann Arbor, MI, November 1997.

- [154] J. T. Moy, "OSPF : Anatomy of an Interent Routing Protocol," Addison-Wesley, Reading, MA, 1998.
- [155] C. Hedrick, "Routing Information Protocol," Internet Request of Comments (RFC 1058) June 1988.
- [156] G. Malkin, "RIP version 2," Internet Request of Comments (RFC 2453) November 1998.
- [157] J. Moy, "OSPF Version 2," Internet Request of Comments (RFC 2328) April 1998
- [158] D. Towsley, "Providing quality of service in packet switched networks," in *Performance Evaluation of computer and Communication Systems*, pp. 560-586, Springer Verlag, 1993.
- [159] C. N. Chauh, Randy H. Katz, "Network Provisioning and Resource Management for IP Telephony," University of California, Berkely, Report No. *UCB/CSD-99-1061*, September 1, 1999.
- [160] Donald Gross , Carl M. Harris "*fundamentals of Queuing Theory*" 3rd edition, Wiley Series in Probability and Statistics, 1998.
- [161] D.B Johnson and C.Perkins, "Mobility Support in IP v6," Internet Draft, draft-ietf-mobileip-ipv6-18.txt
- [162] Chiussi, F.A.; Khotimsky, D.A.; Krishnan, S.; "A network architecture for MPLS-based micro-mobility," Wireless Communications and Networking Conference, 2002. WCNC2002. 2002 IEEE , Volume: 2 , 17-21 March 2002 Page(s): 549 -555 vol.2
- [163] Talukdar, A.K.; Badrinath, B.R.; Acharya, A.; "On accommodating mobile hosts in an integrated services packet network," *INFOCOM '97*. Sixteenth Annual Joint Conference of the IEEE Computer and Communications Societies. Proceedings IEEE, Volume: 3 , 7-11 April 1997 Page(s): 1046 -1053 vol.3

- [164] M. Naghshineh, A. Acampora, "QoS Provisioning in Micro-Cellular Networks Supporting Multimedia Traffic," *INFOCOM* '95, 1995 .
- [165] Terzis, M. Srivastava , and L. Zhang, "A simple QoS Signaling Protocol for Mobile Hosts in the Integrated Services Internet," *INFOCOM '99*. Eighteenth Annual Joint Conference of the IEEE Computer and Communications Societies. Proceedings. IEEE , Volume: 3 , 21-25 March 1999 Page(s): 1011 –101 vol.3
- [166] Jain, R.; Knightly, E.W.; "A framework for design and evaluation of admission control algorithms in multi-service mobile networks," *INFOCOM '99*. Eighteenth Annual Joint Conference of the IEEE Computer and Communications Societies. Proceedings. IEEE , Volume: 3 , 21-25 March 1999 Page(s): 1027 -1035 vol.3
- [167] A.; Znati , T.F . "A Framework for Call Admission Control and QoS Support in Wireless Environments," *INFOCOM 1999*.
- [168] W.T. Chen and L.c. Huang , "RSVP Mobility Support : A Signaling Protocol for Integrated Services Internet With Mobile Hosts," *INFOCOM 200* Nineteenth Annual Joint Conference of the IEEE Computer and Communications Societies. Proceedings. IEEE , Volume: 3 , 26-30 March 2000 , Page(s): 1283 –1292 vol.3
- [169] Tingzhou Yang ; Dimitrios Markrakis "Hierarchical Mobile MPLS: Supporting Delay Sensitive Applications Over Wireless Internet ," 2001 IEEE.
- [170] Castelluccia, C.; Bellier, L.; "A hierarchical mobility management framework for the Internet," *Mobile Multimedia Communications, 1999. (MoMuC '99) 1999 IEEE International Workshop on* , 15-17 Nov. 1999 Page(s): 149 –154 March 1999 Page(s): 3 -11 vol.1
- [171] Chiussi, F.A.; Khotimsky, D.A.; Krishnan, S.; *Wireless Communications and Networking Conference, 2002. WCNC2002. 2002 IEEE*, Volume: 2, 17-21 March 2002 Page(s): 549 -555 vol.2

- [172] *Fei Yu; Leung, V.C.M.*; "Mobility-based predictive call admission control and bandwidth reservation in wireless cellular networks," INFOCOM 2001. Twentieth Annual Joint Conference of the IEEE Computer and Communications Societies. Proceedings. IEEE, Volume: 1, 22-26 April 2001 Page(s): 518 -526 vol.1
- [173] *Chun-Ting Chou; Shin, K.G.*; "Analysis of combined adaptive bandwidth allocation and admission control in wireless networks," INFOCOM 2002. Twenty-First Annual Joint Conference of the IEEE Computer and Communications Societies. Proceedings. IEEE , Volume: 2, 23-27 June 2002 Page(s): 676 -684 vol.2
- [174] *Mitchell, K.; Sohraby, K.*; "An analysis of the effects of mobility on bandwidth allocation strategies in multi-class cellular wireless networks," INFOCOM 2001. Twentieth Annual Joint Conference of the IEEE Computer and Communications Societies. Proceedings. IEEE , Volume: 2 , 22-26 April 2001 Page(s): 1005 -1011 vol.2
- [175] A. Campbel et al, "Cellular IP," draft-ietf-mobileip-cellularip-00.txt, January 2000.
- [176] Jan, R.; Raleigh, T.; Yang, D.; Li Fung Chang; Graff, C.; Bereschinsky, M.; Patel, M.; "Enhancing survivability of mobile Internet access using mobile IP with location registers," *INFOCOM '99*. Eighteenth Annual Joint Conference of the IEEE Computer and Communications Societies. Proceedings. IEEE, Volume: 1 , 21-25
- [177] C. Perkins, ed., "IP Mobility Support," IETF RFC 2002, Oct 1996.
- [178] E. Rosen, A. Viswanathan, R. Callon, "Multiprotocol Label Switching Architecture," IETF RFC 3031, Jan 2001.
- [179] E. Gustafsson , A. Jonsson, C. Perkins, "Mobile IP Regional Registration," Internet Draft, draft-ietf-mobileip-reg-tunnel-04.txt, Mar 2001, Work in Progress

- [180] R. Ramjee, T. La Porta , S. Thuel, K. Varadhan , S. Y. Wang, "HAWAII :A Domain-based Approach for Supporting Mobility in Wide-Area Wireless Networks," in *Proc. IEEE International Conference on Network Protocols*, 1999.
- [181] A. T. Campbell , et. Al., "Design, Implementation and Evaluation of Cellular IP," in *IEEE Personal Communications*, pp. 42-49, Aug 2000.
- [182] Charles E. Perkins, "Mobile Networking Through Mobile IP" web resources :<http://www.computer.org/internet/v2n1/perkins.htm>
- [183] Yi-an Chen, "A Survey on Mobile IP " web resources: http://www.cse.ohio-state.edu/~jain/cis788-95/mobile_ip
- [184] P.Bhagwat, C.Perkins, and S.Tripathi," Network Layer Mobility : an Architecture and Survey," *IEEE Pers, commun*, vol.3, no.3, June 1996 , pp 54-64
- [185] Bye, K.J.; "Handover criteria and control in cellular and microcellular systems" *Mobile Radio and Personal Communications*, 1989., Fifth International Conference on , 11-14 Dec 1989 Page(s): 94 -98
- [186] Pollini, G.P.; "Trends in handover design," *Communications Magazine, IEEE*, Volume: 34 Issue: 3 , March 1996 Page(s): 82 -90
- [187] N. Zhang and J. M. Holtzman , "Analysis of Handoff Algorithms using Both Absolute and Relative Measurements," *Proc. Vehicular Tech. Conf. 94'*, Stockholm, Sweden , June 8-10 , 1994, pp. 82-86.
- [188] Chia, S.T.S.; "The control of handover initiation in microcells" *Vehicular Technology Conference*, 1991. 'Gateway to the Future Technology in Motion', *41st IEEE* , 19-22 May 1991 Page(s): 531 -536
- [189] Tripathi, N.D.; Reed, J.H.; VanLandingham, H.F.; "Handoff in cellular systems " *Personal Communications, IEEE* [see also *IEEE Wireless Communications*], Volume: 5 Issue: 6, Dec. 1998 Page(s): 26 -37
- [190] S. Tekinay and B. Jabbari, "Handover and Channel Assignment in Mobile Cellular Networks," *IEEE Commun. Mag*, Nov. 1991, pp. 42-46

- [191] S. Tekinay and B. Jabbari, "A measurement-based Prioritization scheme for handovers in mobile cellular networks," *IEEE JSAC*, vol. 10, no. 8, Oct. 1992
- [192] P. O. Gassvik, M. Cornefjord, and V. Svensson, "Different methods of giving priority to handoff traffic in a mobile telephone system with directed retry," *Proc. 41st IEEE VTC*, May 1991, pp. 549-53
- [193] D. Giancristofaro, M. Ruggieri and F. Santucci, "Queuing of Handover Requests in Microcellular Network Architectures," *Proc. 44th IEEE VTC*, May 1994, pp. 1846-49
- [194] W. C. Y. Lee, *Mobile Communications Design Fundamentals*, Wiley, 2nd ed., 1993
- [195] W. C. Y. Lee, *Mobile Cellular Telecommunications*, McGraw-Hill, 2nd ed., 1995
- [196] C. N. Chuah, R. D. Yates, and D. J. Goodman, "Integrated dynamic radio resource management," *Proc. 45th IEEE VTC*, May 1995, pp. 584-88.
- [197] R. Rezaiifar, A. M. Makowski and S. Kumar, "Optimal control of handoffs in wireless networks," *Proc. 45th IEEE VTC*, May 1995, pp. 887-91.
- [198] P. E. Ostling, "High Performance Handoff Schemes for Modern Cellular Systems," Ph.D. dissertation, Royal Inst. Tech., Sept. 1995.
- [199] W. C. Jakes, *Microwave mobile communications*, New York, by IEEE Press 1994.
- [200] 3rd Generation Partnership Project; Technical Specification Groups Services and Systems Aspects; General Packet Radio Service (GPRS); Service description; stage 2 (3G TS 23.060 version 3.2.1), January 2000.
- [201] A. S. Acampora, M. Naghshineh, *Control and Quality-of-Service Provisioning in High Speed Microcellular Networks*, IEEE Personal Communications, Vol. 1, No. 2, 1994.

- [202] Akyildiz, I.F.; Ho, S.M.; "On location management for personal communications networks," *Communications Magazine, IEEE* , Volume: 34 Issue: 9 , Sept. 1996, Page(s): 138 –145
- [203] J. Wroclawski, et al., "Specification of the controlled load network element service," RFC 2211, September 1997.
- [204] P. Ramanathan, K. M. Sivalingam, P. Agrawal and S. Kishore "Resource allocation during handoff through dynamic schemes for mobile multimedia wireless networks "1999 IEEE
- [205] P. Agrawal , D. K. Anvekar, and B. Narendran, "Channel management policies for handovers in cellular networks," *Bell Labs Technical Journal*, vol.1, no. 2, pp. 97-110, 1996.
- [206] M. Naghshineh and M. Schwartz , " Distributed call admission control in mobile / wireless networks," *IEEE Journal on Selected Areas in Communications*, vol. 14, pp. 711-717, May 1996.
- [207] O. T. W. Yu and V. C. M. Leung, "Adaptive resource allocation for prioritized call admission over an ATM-based wireless PCN," *IEEE Journal on Selected Areas in Communications*, vol. 15, pp. 1280-1225, Sept. 1997.
- [208] D. Calin and D. Zeghlache , "Performance and Handoff Analysis of an Integrated Voice-Data Cellular System " 1997 IEEE
- [209] Sirin Tekinay and Bijian Jabbari, "A Measurement-Based Prioritization Scheme for Handovers in Mobile Cellular Networks," *IEEE Journal on Selected Areas in Communications* , Vol. 10, No. 8 , October 1992.