ROUTING IN HIGH-SPEED NETWORKS

A Dissertation Presented

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Dedicated to

My wife, Pey-Lin Su

and

My parents, Yu-In Yang and Tsung-Hsiung Hwang

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ABSTRACT

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In the 1990s, significant advances in fiber optic and switching technology have precipitated the current explosion in the amount of research in future high-speed networks. The challenges facing high-speed network researchers result not just from the high data transmission rate but also from other new characteristics not previously encountered in traditional circuit-switched or packet-switched networks. For example, features such as large propagation delay as compared to transmission delay, diverse application demands, constraints on call processing capacity, and quality-of-service (QOS) support for different applications all present new challenges which arise from the new technology and new applications. Thus, much research is needed not just to improve existing technologies, but to seek a fundamentally different approach toward network architectures and protocols. In particular, new flow control and routing algorithms need to be developed to meet these new challenges.

The goal of this dissertation is to solve routing problems in the context of high-speed networks taking into account the new network environment. In order to better understand the influence of these new network characteristics on the routing problem, we first study the computational delays associated with call admission,

routing and call setup in future high-speed networks. This study also leads us to explore the similarity between call setup in high-speed networks and circuit-switched networks and to study how approaches toward routing in circuit-switched networks can be adapted for routing in high-speed networks. Specifically, based on the Markov Decision Process (MDP) framework, we next design and evaluate state-dependent routing algorithms for high-speed networks which account for these new network characteristics. Finally, we consider a Virtual Path (VP) concept that has been proposed to simplify traffic control and resource management in future high-speed networks. Thus, in the last part of this dissertation, we also demonstrate the efficiency of MDP-based routing algorithms in virtual path-based networks.

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CHAPTER 1

INTRODUCTION

In computer networking, major research activities have always been preceded by significant advances in the underlying communication technology. For example, research in packet-switching networks during the 1970s and 1980s was preceded by major advances in processors (store-and-forward switches) and communication technologies. In the 1990s, significant advances in fiber optic and switching technology have precipitated the current explosion in the amount of research in future high-speed networks.

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Decision Process (MDP) framework, we next design and evaluate state-dependent routing algorithms for high-speed networks which account for these new network characteristics. Finally, we consider a Virtual Path (VP) concept that has been proposed to simplify traffic control and resource management in future high-speed networks. Thus, in the last part of this dissertation, we also demonstrate the efficiency of MDP-based routing algorithms in virtual path-based networks.

1.1 Problem Formulation

In this dissertation, we will first examine the influence of the characteristics of high-speed networks on the routing problem. We then design and evaluate routing algorithms to account for these new features.

In the first part of this dissertation, we focus on the computational delays associated with call admission, routing and call setup in future high-speed networks. Although the CCITT specifications on ATM [17] specify a cell-based, packet-like transport mode for information within the network, the need to provide a guaranteed QOS has resulted in the need for a *call-level* admission control mechanism and the reservation of resources (e.g., bandwidth [2]) by a call on a link-by-link basis. Thus when a call is "offered" to a route, computations will be required at each node on the route in order to determine whether the selected route can indeed support the additional call while meeting the QOS guarantees already made to existing calls. In future high-speed networks, significant burdens will therefore be placed on the processing elements in the network and the bottlenecks are likely to shift from the communication links to the processing elements. The processing delays at these processing elements are influenced by network parameters such as the routing algorithm used, propagation delays, admission control functions (due to QOS requirements), path lengths (or network topology), and processing capacities at these elements. The goal of this research is to characterize the influence of these network characteristics on the call setup time and accepted call throughput. In chapter 3, we will develop analytic models for evaluating the average call setup time and accepted call throughput for various routing algorithms.

The need to reserve resources such as buffer space and bandwidth for each individual connection in order to guarantee its QOS requirement has made connection establishment in high-speed networks quite similar to call setup in circuit-switched networks. Thus in the second part of this dissertation, we develop adaptive routing algorithms for high-speed networks based on routing algorithms previously developed for circuit-switched networks. Adaptive routing algorithms in circuit-switched networks can be classified into two categories: Least-Loaded Path-based (LLP) and Markov Decision Process-based (MDP). Three problems can be identified in applying these routing algorithms to high-speed networks. First, most routing algorithms assume the existence of a "*direct link*" between every O-D pair and use that link as the primary path. This may not be a reasonable assumption in high speed networks. Second, while MDP routing algorithms yield very promising performance in single class circuit-switched networks, they become too computationally expensive in multi-class networks. An approach is thus needed which can reduce this computational complexity if we are to apply Markov decision theory to routing problem in high-speed networks. Finally, in order to make the analytical model tractable, MDP routing algorithms make a link independence assumption. Such an assumption is acceptable in circuit-switched networks because of their densely connected topologies. However, as the network topology becomes increasingly sparse, the states of the links in the network become increasingly correlated. Since we expect that future high speed networks will indeed be very sparse, it is questionable whether applying MDP routing algorithms from circuit-switched networks to future high-speed networks can still yield good performance. We will address these issues in chapter 4.

In the last part of the dissertation, we study the routing problem in Virtual Path networks. In order to facilitate traffic control and resource management in future high-speed networks, a Virtual Path (VP) concept has been proposed [18, 20, 72]. A VP can be viewed as a logical direct link between a source node and a destination node which physically consists of a set of adjacent links. By reserving "capacity", e.g., bandwidth and buffer space, on VPs, the call establishment processing can be simplified significantly because a call acceptance decision can immediately be made at the source node (without the need for a hop-by-hop call set up). However, this

advantage is offset by the increase in network blocking probability because of the statistical multiplexing gains lost by the preallocation of capacity to the virtual paths. Thus, there exists a trade-off between the call setup cost and the call blocking probability. Two approaches can be used to reduce the network blocking probability when bandwidth is reserved for virtual paths. The first approach is to logically view the network as a fully connected network with virtual paths serving as links in the virtual network. Adaptive routing algorithms based on Markov decision theory can then be used to reduce the blocking probability. The second approach is to have a dynamic virtual path bandwidth allocation algorithm which can reduce the call set up delay significantly while maintaining a competitive call blocking probability. In chapter 5, we will focus on the first approach and develop and evaluate adaptive MDP routing algorithms and virtual path bandwidth reservation schemes for networks with reserved virtual path bandwidth.

1.2 Research Contributions

The goal of this dissertation is to characterize the influence of new network characteristics introduced by high-speed networks on the routing problem and to develop and evaluate appropriate approaches toward routing in such networks which account for these new features. Below, we list our important contributions in the order in which they appear in this dissertation.

We first study the influence of new network characteristics introduced by high-speed networks on the routing problem. Our contributions in this study include:

- The development of analytic models for evaluating the average call setup delay and accepted call throughput for three sequential routing algorithms and two flooding routing algorithms. These analytic models are more comprehensive than those introduced in previous works on routing in the literature in that propagation delay, call setup delay, blocked call clear time and processing delay are explicitly modeled.

- The examination, through analytic models, of the effect of call processing delay, propagation delay, admission control function, and routing algorithms on the average call setup delay and call throughput.

- The observation that the average call setup delay and call throughput are very sensitive to that call processing service time, admission control function, and routing algorithm, but are relatively insensitive to the propagation delay.

The great similarity between routing in circuit-switched networks and high-speed networks has inspired us to examine and apply adaptive routing algorithms in circuit-switched networks to high-speed networks. We focus on designing and evaluating adaptive routing algorithms for future high-speed networks based on Markov decision theory. The contributions of this study are listed below:

- We propose and evaluate an LLP-based routing algorithm for high-speed networks.

- We propose an approach for computing approximate link shadow prices for multi-class networks which is computationally feasible while still providing quite accurate routing information.

- Based on the approximate link shadow prices, we design and evaluate various MDP-based routing algorithms, and show that they outperform the LLP-based routing algorithm.

- From a simple network model, we have found that the performance of MDP-based routing algorithms is insensitive to the validity of the link independence assumption.

In the last part of this dissertation, we study the routing problem in VP networks. Our contributions in this work are listed below:

- We explore two strategies for reserving VP capacity: a deterministic strategy and a statistical strategy. We also examine advantages and disadvantages of these two strategies. We show that routing policies are able to yield significantly lower blocking probability when the statistical reservation strategy is adopted.

- We design and evaluate MDP routing algorithms for homogeneous VP networks and show that MDP routing algorithms are able to provide significantly lower network blocking probability with slightly increase in control cost.

- We propose a VP capacity sharing concept in which some VPs are allowed to borrow capacity from other VPs. We also design MDP routing algorithms which account for this concept and show that routing algorithms which account for the VP capacity sharing concept are able to further reduce the network blocking probability.

1.3 Overview of the Dissertation

The remainder of the dissertation is organized as follows. We first briefly review, in chapter 2, existing routing algorithms for both circuit-switched and packet-switched networks. We also examine a commonly-used analytic technique for evaluating call blocking probability in circuit-switched networks. We then discuss some new network characteristics introduced by new technologies and applications and challenges posed on the routing problem in future high-speed networks.

In chapter 3, we examine the influence of these new network characteristics on routing. This influence is examined, through analytic models, with three sequential routing schemes and two flooding routing schemes under various network parameters and different forms of admission control.

Chapter 4 applies adaptive routing algorithms from circuit-switched networks to high-speed networks. In particular, we focus on developing computationally feasible MDP routing algorithms for high-speed networks. We also examine the effect of the link independence assumption on the performance of MDP routing algorithms.

While chapter 4 deals with networks without the VP concept, chapter 5 deals with VP networks. In this chapter, we design and evaluate MDP routing algorithms under different virtual path capacity reservation strategies for "homogeneous" VP networks, i.e., VP networks with homogeneous traffic sources.

Finally, chapter 6 summaries this dissertation and suggests directions for future work.

CHAPTER 2

AN OVERVIEW OF ROUTING

In this chapter, we first briefly review existing routing algorithms used in both circuit-switched and packet-switched networks. There are three categories of routing algorithms: static routing, adaptive routing and dynamic routing [38]. In static routing, all routing decisions are made at network setup time and are fixed, independent of the network state. Dynamic routing allows routing decisions to vary over time, but not necessarily depend on the network state. Finally, in adaptive routing, routing decisions are based on a function of some estimate of the network state and thus may also vary over time. The difference between dynamic routing and adaptive routing is that an adaptive routing algorithm usually is also dynamic unless the network is so quiet that the network state is not changing at all. However, the converse is not necessary true. Especially in circuit-switched networks, some algorithms, e.g., Dynamic Non-Hierarchical Routing in AT&T's long distance network, change their routing decisions at fixed time intervals, but do not use network state measurements.

This chapter is structured as follows. The first section reviews existing routing algorithms used in circuit-switched networks. It also examines a commonly used analytic technique for evaluating the performance of a circuit-switched network. Routing algorithms used in wide-area packet-switched networks are briefly discussed in the second section. We are not interested in routing algorithms in LAN's and MAN's because they are relatively simple due to the special structure of network topology, e.g., bus, loop, ring and tree. In the last section, we discuss some new characteristics introduced by new technologies and applications and the resulting challenges posed by the routing problem in future high-speed networks. This section concludes with a list of the characteristics of the routing problem in high-speed networks.

2.1 Routing in Circuit-Switched Networks

In a circuit-switched network, a call setup procedure is used to set up a dedicate, end-to-end, connection between users who desire to communicate. The routing algorithm provides the rules which a call setup procedure follows for each connection request. When a call arrives, the call setup procedure first chooses a path (a sequence of links from the origin node to the destination node) from a set of possible paths according to the routing algorithm and then proceeds as follows. First, the origin node passes the call request to its downstream neighbor on the chosen path. If the neighbor, "successfully" receives the request, it then passes it to its downstream neighbor on the path. This transfer of the call request from one node to another is "successful" if there is at least one free circuit in the link connecting these nodes that can be reserved for the call. This process continues until either the request is successfully passed to the destination node or the request cannot be passed further towards the destination node by an intermediate node on the path. In the first case, the call is established and the circuits reserved along the path by the call setup procedure provide a private, end-to-end connection. In the second case, the call cannot be established on the path and, depending on the routing algorithm, a "blocking" message needs to be sent back either to the upstream node or all the way back to the source node.

Three methods have been proposed for responding to a blocking message. The first method is called originating-office control (OOC). Under OOC, the blocking message is transmitted all the way back to the origin node and all circuits that have been reserved are released. When the blocking message is received, the origin node chooses the next path from the set of all possible paths and repeats the procedure described above. The second method is called sequential-office control (SOC), also called progressive control, sequential control, or spill forward without crankback, in which the intermediate node chooses the next outgoing link to try when a blocked link is encountered. In this case, no blocking message is fed back to upstream nodes and a call is blocked only when all the possible outgoing links of an intermediate node (or the origin node) are blocked. If this occurs, the circuits reserved on the path to the node are released and the call is aborted. The third method is called crankback, or spill forward with crankback. The only difference between the SOC

method and crankback method is that with crankback, a blocking message is fed back to the upstream node when the downstream node is blocked. A node is blocked if all of its possible outgoing links (for the O-D pair) are blocked and a call is blocked if the origin node is blocked. If a blocking message is fed back to the upstream node, the circuit reserved on the link from the upstream node to the node is released.

A detailed survey of existing routing algorithms for circuit-switched networks is given in [48]. We briefly list below some well known routing algorithms. The oldest and most extensively used *static* routing algorithm in telephone networks is the hierarchical alternate routing algorithm [9, 75]. Because of the inefficiency of static routing, a *dynamic* routing algorithm, *Dynamic Non-Hierarchical Routing* (DNHR), was introduced in the long distance AT&T network in the United States in the mid 1980s to replace the old hierarchical alternate routing algorithm [7]. Another dynamic routing algorithm is *Dynamic Alternate Routing* (DAR) which has been proposed for the British Telecom's domestic network. Both DNHR and DAR offer a new arriving call to the direct link first. If it is blocked, alternate paths are tried. In DNHR, a day is typically divided into 10 periods of time and a near-optimal alternate routing sequence is computed off-line for each period. As each time period starts, the appropriate routing table is selected. Calls that are blocked on the direct link are offered to the alternate paths according to the predefined order in the routing table. In DAR, the origin switch keeps a recommended alternate path. A new arriving call that is blocked on the direct link is offered to the recommended alternate path only. If the recommended alternate path is busy, the call is blocked and a new recommended alternate path is selected randomly from the set of possible alternate paths. Otherwise, the call is set up on the recommended alternate path and the recommended alternate path remains unchanged.

Since the mid 1980s, considerable research has focused on the design of *adaptive* routing algorithms for circuit-switched networks. Existing adaptive routing algorithms can be classified into two categories: *Least-Load Path*-based (LLP-based) and *Markov Decision Process*-based (MDP-based). The LLP approach tries to route calls to the "least busy" path, i.e., the path that has the maximum free circuits. The MDP approach, on the other hand, formulates the routing problem as a Markov

decision process. LLP-based routing algorithms include AT&T's Trunk Status Map Routing (TSMR) [6] and Real-Time Network Routing (RTNR) [8], and Bell-Northern Research's Dynamically Controlled Routing (DCR) [70, 80]. MDP-based routing algorithms include Bellcore's State-Dependent Routing (SDR) [55] and Forward-Looking Routing (FLR) [57], and routing algorithms based on INRS-Telecommunications' Markov Decision Process Decomposition (MDPD) approach [28, 30, 31, 32].

2.1.1 Performance Evaluation

The most important performance measure for circuit-switched networks is the call blocking (or loss) probability, either the loss probability of an origin-destination pair or the loss probability of the network over all O-D pairs. Thus, minimizing the call blocking probability is the most common objective of routing algorithms in circuit-switched networks.

A rather simple, explicit method for computing call blocking probability is to make appropriate assumptions, model the system as a Markov process, and compute the blocking probability. However, this quickly yields a very large state space as the complexity of the network and the capacity of the links grow. On the other hand, with the assumption of Poisson arrivals at each link, the individual link blocking probability can be computed easily and quite accurately by the well-known *Erlang's B formula* provided that the link-offered traffic (i.e., the mean call arrival rate of the arrival process at the link) and link capacity are known. After the blocking probability of each link is computed, the overall network performance can be computed based on these measures. This idea yields the link-decomposition method [38], a simple and commonly used method for evaluating the call blocking probability for circuit-switched networks.

The link-decomposition method is based on the following assumptions:

1. External call arrivals are assumed to be independent, stationary Poisson processes.

2. There is no blocking at the switches.

3. Calls are blocked independently at all links.

4. Calls that cannot be routed on a path are cleared in zero time, and are returned to the responsible switches in zero time, if necessary.

5. The network topology is fixed, routing decisions are given, and are not allowed to change during the calculation.

6. Call holding times are independent and identically distributed, and are usually assumed to be exponentially distributed, but not always. (e.g. in [52, 81], call holding times are arbitrarily distributed.)

The difficulty with this method arises from the fact that the link-offered traffic is unknown and coupled with the link-blocking probability. That is, on one hand, we need to use the link-blocking probability to compute the link-offered traffic. On the other hand, we need the link-offered traffic to compute the link-blocking probability. This yields a fixed point problem which in most cases, is solved by an iterative method consisting of the following two stages:

1. Given the link-blocking probabilities, one can compute the link-offered traffic and call-blocking probabilities. This computation depends on the routing algorithm used.

2. The resulting link-offered traffic in turn yields, via the Erlang-B formula, a new set of values for the link-blocking probabilities.

When an iterative method is used to find a solution, one may ask if a unique solution exists. Since the two stages can be viewed as a continuous mapping from $(0,1)^\ell$ to itself, where ℓ is the number of links in the network, and the possible solution set is bounded by $(0,1)^\ell$, one can use the classical Brouwer fixed-point theorem to prove that there exists at least one fixed point. Then, under some sufficient conditions, this mapping can be proven to be a contraction mapping, and therefore yielding a unique solution. However, the sufficient conditions are usually very strong (e.g., [81]). Furthermore, there is no guarantee that the iterative method will find a solution, even if one exists [38]. We will discuss the convergence problem in more detail after we review some related previous works.

In [81], Whitt proposes a model for calculating the blocking probability in setting up virtual circuits in packet-switched networks. The model, called reduced-load approximation, is very similar to the link-decomposition method. In [81], alternate path routing is not considered, i.e., every virtual circuit has only one path to try, and is lost if it is blocked at any link of the path. All resources required by a virtual circuit are seized and freed together. Poisson arrival processes and general service distribution are assumed (M/G/s/loss), although Whitt indicates how to apply the reduced-load approximation to a single facility (link) with non-Poisson arrival processes. Whitt has proved that the approximate link-blocking probabilities can be bounded above and below, and very often these upper and lower bounds will converge. However, the exact link-blocking probabilities are not necessarily bounded by these bounds. He also shows that under some very strong conditions the iterative method has a unique solution. Kelly in [52] applies this model to circuit-switched networks and proves that this model always has a unique solution, the so called Erlang fixed point. Kelly also shows that the approximate solution is asymptotically exact in heavy traffic.

Yee [83] develops a model in which SOC routing is used and the procedure for choosing the next node through which to set up a call is rather complicated. In his model, each node has multiple permutations of downstream adjacent nodes for each call set up request to be selected as the next node. The choice of a permutation is determined by a predefined probability. The link-decomposition method is used to transform the model into a set of nonlinear equations. He proves that, under certain conditions, the system has a unique solution by showing that the nonlinear transformation is a contraction mapping.

The nonlinear equations formulated by Whitt, Kelly and Yee are rather simple because they only consider fixed routing or SOC routing. When OOC routing and crankback routing are used, a complication arises, due to the fact that some links can appear on more than one path (place) in the same routing tree. This fact complicates the computation of the probability that a call will be routed through a particular path. The reason is that if some link on the path appears on some other path before the one considered, the probability that the path is free and the probability that the

call overflows to this path are not independent. This is true even if we assume that the individual link-blocking probabilities are independent. The problem arises from the assumption that blocked calls have zero holding time, so if a given link blocks a call on one path that link blocks all calls on all paths to which it belongs. As a consequence, the probability that a call will be routed through a particular path is a conditional probability.

Butto, Colombo, and Tonietti [15] first propose a complete analysis for OOC routing for the case in which a link may appear more than once in a given routing tree. Links that appear more than once in a tree are called *repeated links*. The probability of routing a call through a particular path is computed by explicitly considering all of the possible combinations of states of all the repeated links, where the state of a repeated link is either free or busy. It is easy to see that the state space grows exponentially in the number of repeated links. They also formulate a model for SOC routing and observe that it is not necessary to compute conditional probabilities when calculating the probability of using a particular path. This is because the probability of using a particular path is independent of whether a link on the path appears elsewhere in the routing tree when SOC routing is used.

In parallel with [15], Lin, Leon, and Stewart [58] propose an analysis method for computing the end-to-end blocking probability for OOC, SOC, and OOC with spill-forward routing strategies. (OOC with spill-forward is a combination of OOC and SOC routing.) Repeated links are also considered. The probability of using a particular path is calculated by examining the sequence of all paths specified by the routing strategy, the so called path-loss sequence (which is generated by hand). An iterative procedure is used to solve the non-linear equations. However nothing is mentioned regarding whether the procedure will find a solution, or even if a solution exists.

The difficulty of generating the path-loss sequence in [58] is solved by Chan [21]. Chan develops a recursive algorithm for computing the end-to-end blocking probability for an arbitrary routing tree by recognizing the underlying recursive nature of the problem. The main idea is to compute the *link set*, a set containing links that appear in previous paths but not in current path, recursively. The importance of this

work is the use of recurrence relations which makes the calculation applicable to all static routing strategies with alternate routing.

The complexity of the recursive algorithm in [21] cannot be polynomially bounded. This is due to the nature of the problem. Girard and Ouimet [39] propose a recursive algorithm that can be polynomially-bounded for a routing tree in which there are no repeated links. The result is not surprising because the complexity of the problem in [21] comes from the fact that they consider arbitrary trees with repeated links.

In parallel with Chan's work, Gaudreau [35] also proposes a recursive formula for calculating the end-to-end blocking probability for SOC routing with a fixed sequence route selection at each node. The computation is based on the assumption that the individual link blocking probabilities are known and independent. The repeated links cause no complication in the computation, as explained above.

Algorithms presented in [15, 58, 21, 39, 35] describe how the end-to-end blocking probability can be computed, given the individual link blocking probabilities. Only [39] and [58] show how to compute the individual link blocking probability. Girard and Ouimet [39] further show that the computation of the appropriate link blocking probabilities is best done by a fixed point method, rather than a Newton-type algorithm, because the amount of computation required by the latter algorithm is too high. Unfortunately, there is no guarantee that the fixed point method will find a solution, even if one exists [38]. Moreover, Akinpelu's work in [4] indicates that the existence of more than one solution arises in connection with real instabilities in networks operating with nonhierarchical alternative routing. Recall that this undesirable phenomenon does not happen in Kelly and Yee's work [52, 83]. It therefore seems that this phenomenon is closely related to the nature of OOC routing (crankback routing has never been explored by this kind of analysis). Theoretically, some very strong sufficient conditions can be imposed that guarantee a unique solution.

2.2 Routing in Packet-Switched Networks

In a packet-switched network, the function of the routing algorithm is to select the best set of available links to transfer a packet between a source and destination. In the broadest sense, two approaches towards routing may be identified. They

are distinguished by the times at which routing decisions are made. In a virtual circuit network, a routing decision is made when each virtual circuit (VC) is set up. The routing algorithm is used to choose an end-to-end path for the VC. All packets belonging to the VC subsequently traverse the network following this path and arrive at the destination node in the sequence in which they were transmitted. This VC routing approach embodies the so-called "connection-oriented" approach towards packet-switching. The second approach towards routing is a connectionless one, with packets (datagrams) moving through the network on an individual basis, with a routing decision being made for each individual packet. The issue of whether a datagram or VC approach is a better choice for a packet switching network has been debated for a long time. The advantages of datagram routing include the provision of robust service in the face of network component failures and the flexibility of building different types of services on top of it. But congestion control and the requirement of resequencing for some applications are serious problems for datagram networks. On the other hand, virtual circuit networks provide easier congestion control and no need for resequencing; these are provided at the cost of explicit call set up overhead and a vulnerability to network component failures. In terms of performance, Rudin in [71] notes that the delay in a network that assign routes on a session basis is less than that in a system that adapts the route more frequently when the network is protected by flow control and the network load is reasonable high. Finally, it should be noted that most of the high-speed network designs for the future are clearly directed toward connection-oriented or partially connection-oriented approaches.

The objective of routing in packet-switched networks is to obtain a high network throughput while keeping the average packet delay as low as possible. (Unfortunately, operating any queueing system near capacity implies a long queueing delay.) Throughput and average packet delay are the two important performance measures that are substantially affected by the routing algorithm. Routing interacts with flow control in determining these performance measures [12]. The effect of good routing is either to (a) increase the throughput as high as possible for the same value of average delay per packet under high network load or (b) decrease the average packet delay for the same level of network throughput under low network load.

During the last three decades, many routing algorithms for wide area networks have been proposed. A survey of those routing algorithms is given in [48]. In this survey, due to the great diversity in existing sophisticated adaptive routing algorithms, adaptive routing algorithms are further classified according to the following two ways. First, according to the route selection mechanism, adaptive routing algorithms are further classified into shortest-path-based algorithms and gradient-based algorithms. Second, according to the method of information dissemination, adaptive algorithms are further classified into centralized, isolated, and distributed algorithms. Although research in adaptive routing in packet-switched networks is quite successful, in the next section, we will show that these adaptive algorithms are not suitable for high-speed networks.

2.3 Routing in High-Speed Networks

In this section, we first examine four new network characteristics introduced by the new technology and new applications of future high-speed networks and discuss how they affect the routing problem. We then discuss how these characteristics make routing in future high-speed networks similar to, as well as different from, the routing in both circuit-switched and packet-switched networks. Finally, we conclude this section by listing characteristics of routing in high-speed networks.

2.3.1 New Characteristics of High-Speed Networks

- **Very limited processing capacity:**

  In the last decade, network transmission rates have increased by at least four orders of magnitude while processing capacities have only increased by two orders of magnitude. The result of this change is that the time available to process a packet at an intermediate node is very limited [59]. Thus implementing complicated datagram routing algorithms in high-speed networks may not be feasible. In other words, virtual circuit routing will be better suited for future high-speed networks. For example, CCITT has recommended that the ATM networks be connection-oriented [17].

The very limited processing time for each packet leads to a need for developing a dedicated hardware switch that has the ability to route packets very quickly, without consuming the processing capacity [25]. According to [25, 24], this can be achieved by building a network node in two parts: the switching subsystem (SS) and the Network Control Unit (NCU). The SS is a fast hardware switch with relatively limited functionality. The NCU is a slower but more sophisticated processor. Packets that are only relayed through the node, e.g., control packets in a virtual circuit, are handled by the SS directly without involvement of the NCU. Packets that require more complex processing, e.g. packets that carry call set up and routing information, are forwarded from the SS to the NCU. Source routing, where every packet carries its own path information, is also recommended in [25, 24] so that packets can be quickly switched by the SS at transit nodes.

Although processing requirements can be decreased by doing virtual circuit routing and switching normal packets via a switching subsystem, significant burdens will still be placed on the processing elements in future high-speed networks since call routing and admission control will be computationally intensive. Thus the processing elements in high-speed networks are very likely to become a new bottleneck.

- **Long relative propagation delay:**

  When transmission rates and processing capacities both increase by at least two orders of magnitude, the propagation delay does not change. It has been shown that delays at the switches are less than 1% of the total end-to-end delay [10, 84]. It is thus obvious that propagation delay forms a major component of average packet delay in a high-speed wide area network. This has a big impact on current existing adaptive routing algorithms in the following two ways:

  1. For shortest-path based routing algorithms, the link lengths are mostly determined by the measured delay on the link. Since the propagation delay contributes 99% of the delay, the measured link delay is almost always equivalent to the propagation delay. In other words, the delay on the

link is not significantly affected by traffic changes unless the load is very high. For gradient-based routing algorithms [12, 34], the cost function for optimization is also a function of propagation delay and queueing delay. Similar to the shortest-path based routing algorithms, the gradient-based routing algorithms are also very likely to route all virtual circuits to the path with shortest propagation delay.

2. For adaptive routing algorithms (centralized or distributed), information collected at individual nodes needs to be passed to decision point(s). The quicker the information is received, the better the performance of the routing algorithm. When the propagation delay is much larger than the packet transmission time, the information received by the decision point(s) may be too old to use [36].

Another consequence of long propagation delays is that the number of packets in transmission media becomes very large so that the ability to do source-based (or feedback-based) flow control decreases. The validity of traditional window-based flow control mechanisms in high-speed future networks is questionable. Rate-based flow control mechanisms are believed to be more appropriate for the future network [27, 63, 85].

- **Support of diverse services:**

  Another challenge of high-speed networks is the need to support a wide variety of applications with stringent and diverse service requirements. Examples are file transfer, real-time packet voice, teleconferencing, high-resolution graphics, computer visualization, and remote procedure calls. Applications such as conventional file transfer require high reliability, while applications such as real-time packet voice, teleconferencing, and remote procedure calls require short network delay as a primary consideration. Applications such as high-resolution graphics and computer visualization require a significant amount of bandwidth while others, such as remote procedure calls, require only a very small amount of bandwidth.

- **Support of quality of service:**

  One of many interesting challenges facing future high-speed networks is how to provide Quality-Of-Service (QOS) support for different classes of applications. The QOS requirement of a connection with respect to loss probability, end-to-end delay, delay jitter, has to be negotiated with the network at call set up time. The QOS requirements clearly affect (and are affected by) both routing and flow control [85]. In routing, an admission control function is invoked at each link on the route being examined to decide whether the new connection can be accepted or not. The decision has to be made based on the incoming call's traffic characteristics, the QOS requirements of the new connection, and all the existing connections sharing part or all of the same network resources. Acceptance occurs only if the QOS requirements can be achieved for all connections concerned (i.e., the newly arriving call and the already accepted calls). The objective of the admission control function as well as the routing algorithm is to accept as many new connections as possible while still guaranteeing the QOS requirements for every existing call. Due to the statistical multiplexing nature of packet-switched networks, call admission may not be sufficient for providing QOS guarantees to all accepted calls because the actual traffic of a connection may deviate from its negotiated traffic characteristics. Therefore, it is necessary to introduce a new congestion control function, the so called "policing" function, to restrict the behavior of each traffic source to the characteristics negotiated in order to prevent the degradation of the QOS for all connections.

  The influence of this characteristic on routing is that some connection-oriented performance metrics, such as average packet delay or packet loss probability, which used to be part of the cost function to be optimized by routing in traditional packet-switched networks now becomes a constraint to be satisfied when routing in high-speed networks. Other design goals, such as minimizing processing requirements, average call set up time, or maximizing the number of accepted calls, will become the new objective of routing [3, 10].

2.3.2 Comparison of Routing in High-Speed Networks with Routing in Traditional Networks

- **Similarities and differences between call set up in high-speed networks and circuit-switched networks**

  The need for admission control to support QOS requirements has made connection establishment in high-speed networks very similar to call set up in circuit-switched networks. In circuit-switched networks, a call is successfully established on a route if each link on the route has at least one nonbusy circuit. On the other hand, the need to provide a guaranteed QOS in high-speed networks has resulted in the need for a *call-level* admission control mechanism and the reservation of resources (e.g., bandwidth [2]) by a call on a link-by-link basis. Thus in high-speed networks, a new connection is accepted if and only if, for each link on the route, adding the new connection will not violate the QOS requirements of all existing connections sharing the same link. In the case that a new call (connection) is rejected along the route under investigation, the routing functions in both types of network need to select, if possible, another route between the same source and destination node to try to set up the call.

  The fundamental difference between high-speed networks and circuit-switched networks is that high-speed networks are expected to be packet-switched networks. For example, the Asynchronous Transfer Mode (ATM) standard that has been designated for the future high-speed networks is packet-oriented [61]. Packet switching has many advantages, such as providing higher bandwidth utilization, more flexible multiplexing of diverse services with different bandwidth requirements, and taking advantage of statistical multiplexing gains among services. Given that high-speed networks will be packet-switched, the call (connection) set up in circuit-switched networks and high-speed networks differ at least in two aspects:

  1. The criteria (admission control function) to determine whether a link can accept a new call is very different, i.e., no free circuits (in the circuit-switched

case) vs. some complicated function of traffic characteristics of existing and new connections (in the packet-switched case).

2. In circuit-switched networks, a dedicated (unshared), end-to-end connection is provided to each call, while this is not necessarily true in high-speed networks. High-speed networks may be engineered to take advantage of statistical multiplexing gain among different services while still guaranteeing QOS requirements for every service.

- **Similarities and differences between call set up in high-speed networks and packet-switched networks**

  Although high-speed networks are expected to be packet-switched, the method for establishing a new connection (virtual circuit) will be very different from that in traditional packet-switched networks. In traditional packet-switched networks, QOS is not provided and no admission control is used in setting up a virtual circuit. In most cases, congestion control has not been a concern of the routing algorithm. That is, a virtual circuit will not be rejected by the routing algorithm in a traditional packet-switched network unless there is no physical path from source to destination. The objective of routing is to route every virtual circuit (or datagram) on the best path according to current available information. (This is the so called "best-effort" delivery strategy.)

  With QOS requirements, routing in high-speed networks cannot use such a "best-effort" delivery strategy. A new connection needs to be rejected if its acceptance will prohibit the network from providing the negotiated QOS for all existing calls. Moreover, among the paths that can accept the new connection, the path that has the expected minimum packet delay or loss probability need not be the best path to choose under a new optimization criteria such as maximizing the total number of connections accepted by the network. Still another difference is, as mentioned above, that routing in high-speed networks needs to attempt call setup on other possible routes when a call is rejected by the route under investigation; there is no such "blocked route" in traditional packet-switched networks.

2.3.3 Characteristics of Routing in High-Speed Networks

It should be clear that routing in high-speed networks has elements of routing in both circuit-switched and packet-switched networks, but that it also significantly differs from both. From our previous discussion, we can conclude that the future high-speed networks are expected to be connection-oriented networks, supporting multiple classes of QOS requirements. Thus routing in future high-speed networks is expected to be virtual circuit-oriented with the following constraints:

1. A connection is accepted on a route only if for each link on the route, adding the new connection will not violate the QOS requirements for all existing connections sharing the same link.

2. Due to the long propagation delay, global network status information is either unavailable or obsolete.

Furthermore, the objective of routing is to:

1. Maximize the total number of connections admitted to the network under some form of fairness constraint.

2. Minimize the average call set up time which includes processing time and queueing delay at each node (switch) visited, and propagation delay at each link traversed.

Maximizing the network throughput (in this case, the total number of connections admitted) has always been the primary objective in designing a routing algorithm in the past. It will remain the primary objective for routing algorithms in future high-speed networks. In traditional packet-switched networks, another objective would be to minimize the end-to-end packet delay. In future high-speed networks, end-to-end packet delay may be not the concern of all applications and in the case that it is the main concern of the application, the end-to-end packet delay for each packet (or certain percentage of the packets) can be guaranteed according to the application's requirement by appropriate network resources management (e.g., scheduling policy, buffer management, etc). On the other hand, because of the long propagation delay

and processing delay (as compared to the transmission delay), the call setup delay will become an issue. Thus minimizing the call setup time will be a desirable objective in designing routing algorithms for future high-speed networks.

CHAPTER 3

THE INTERACTION OF CALL PROCESSING DELAYS AND ROUTING IN HIGH-SPEED NETWORKS

In order to ensure that accepted calls will be provided with a guaranteed QOS, significant burdens will be placed on the processing elements in future high-speed networks. The need to provide a guaranteed QOS has resulted in the need for a *call-level* admission control mechanism and the reservation of resources (e.g., bandwidth [2]) by a call on a link-by-link basis. In this latter case, when a call is "offered" to a route, computation is required to determine whether the selected route can indeed support the additional call while continuing to meet the QOS guarantees of existing calls. The manner in which calls are offered to the various routes (e.g., the order in which routes are attempted and the decision as to whether multiple call setup paths will be attempted in parallel) will clearly influence the call setup time as well as the maximum call arrival rate that can be supported by the network call processing elements. Network parameters such as call processing capacities, propagation delays, and path lengths will also influence these performance measures. We seek to characterize the influence of these routing, call setup, and network characteristics on the call setup time and the accepted call throughput in this chapter.

This chapter is organized as follows: in section 3.1, we discuss our model of the network. In section 3.2, we specify the routing mechanisms examined. Section 3.3 contains the analytical models for different routing mechanisms. The results of our investigation on the influence of processing delay, admission control function, and routing algorithms on the average call setup delay and accepted call throughput are presented in section 3.4. The influence of propagation delay on routing, presented in [50], is summarized in section 3.5. Section 3.5 also discusses the issue of whether it is important to model resources held by blocked calls. Finally, section 3.6 summarizes this chapter.

3.1 Network Model

We consider a connection-oriented high-speed network consisting of N nodes, $N \geq 2$, with each node having some number of incoming and outgoing links. We adopt the network node structure described in [24, 25], in which a node consists of two components: the switching subsystem (SS) and the network control unit (NCU). The SS is a fast hardware switch with relatively limited functionality. The NCU is a slower, but more sophisticated, processor. Packets (or cells) that need only be relayed through the node are handled by the SS directly without the involvement of the NCU. We further assume that control packets associated with routing are the only control packets processed by the NCU. Each NCU is assumed to have a sufficiently large buffer to avoid the loss of routing control packets due to buffer overflow.

External connection requests, each having an associated QOS requirement, can arrive at any node in the network. When a call request arrives, the routing algorithm (which resides in the NCU's of the network nodes) chooses a path for the call from a set of possible paths (according to the rules specified in next section) and then proceeds as follows. First, the source node invokes the admission control function to check if the new call can be accepted on the first link on the path. The call is accepted on the link if sufficient bandwidth is available on the link to meet the new call's QOS requirement, while maintaining the agreed-upon QOS for existing calls. If the call is accepted on the link, a certain amount of the bandwidth, determined by the QOS requirement, is reserved for the call. The source node then passes the call request to its downstream neighbor on the chosen path. This neighbor then passes the call request to its downstream neighbor if the QOS can also be guaranteed at the next link. This process continues until either the request is successfully passed to the destination node or the request cannot be passed further toward the destination node by an intermediate node on the path. In the first case, the call is established and the resources reserved along the path during the call set-up phase provide the guaranteed QOS required by the call. In the second case, the call cannot be established on the path, bandwidth that has been reserved for this call is released, and another path, if available, may then be tried.

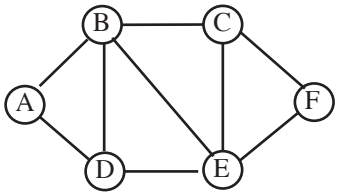
3.2 Routing Algorithms

We examine the effects that five routing algorithms, three sequential algorithms and two controlled-flooding algorithms, have on the call setup time and accepted call throughput. We assume that for each source-destination pair, there is a predefined *routing tree* available at the source node [38]. A routing tree can be viewed as a predefined set of possible paths connecting the source and destination node. These paths will be tried in some order defined by the routing rule when a new call setup arrives. The three sequential routing algorithms are based on the three well-known routing rules in the circuit-switched literature, i.e., OOC, SOC, and crankback. Specifications of these three rules are already presented in section 2.1. Two controlled flooding algorithms based on these three rules will also be investigated. They are referred to as parallel algorithms.

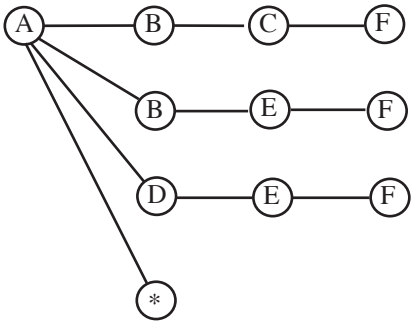
We first describe the three sequential routing algorithms using the *augmented route tree* [58] as shown in Figure 3.1. A call is blocked if a so-called *loss node*, a node labeled by an asterisk, is reached. Paths are tried sequentially, from top to bottom, left to right, with the following control rules.

1. **OOC routing:** According to this rule, the choice of the next path to try is always made by the source node. When a call request is rejected on a link on the chosen path, a blocking message is returned along the path to the source node, bandwidth reserved previously is released, and the source node sends another call request on the path to be tried next. For example, as shown in Figure 3.1(b), path $A-B-C-F$ is tried first. If it is blocked, path $A-B-E-F$ is then tried. If this path is also blocked, path $A-D-E-F$ then is tried. The call is blocked if a blocked message is received on the last path, i.e., $A-D-E-F$. Note that this rule does not use any information about *which* link resulted in the call being blocked on a path.

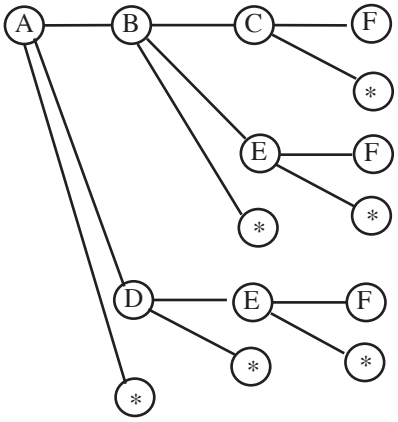
2. **SOC routing:** The SOC control rule is also called progressive control, sequential control, or spill-forward without crankback. This rule tries to improve upon the OOC control rule by allowing the intermediate nodes to react to link blocking. Each intermediate node is given a set of possible outgoing links to try. When



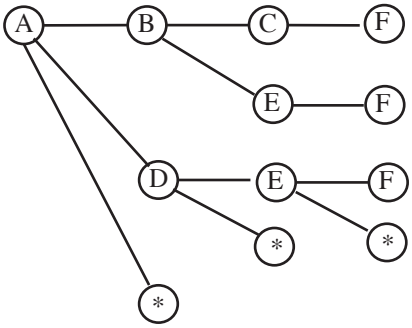
(a) An example network.



(b) The augmented route tree for OOC control.



(c) The augmented route tree for SOC control.



(d) The augmented route tree for crankback control.

Figure 3.1 Augmented Route Tree for OOC, SOC and Crankback Control Rules

a call request is blocked on one of the links, instead of immediately returning a blocking message to the source node, the intermediate node tries to set up the call on another link from the set. If none of the possible outgoing links at the intermediate node is able to provide the required QOS, the call is blocked. For example, as shown in Figure 3.1(c), if the call request is accepted on link $A-B$ then node B is responsible for selecting the next link. A block on link $B-C$ results in another trial on link $B-E$. The call request is blocked at node B if link $B-E$ rejects the request.

3. **Crankback routing:** Crankback routing is also called spill-forward with crankback. It improves upon the SOC rule by allowing a blocked message to be sent back to upstream nodes in the routing tree. Recall the situation described in the SOC rule; in the case of Crankback, when a call request is blocked on link

$B-E$, a blocking message is sent back to node A, the upstream node of node B. The call request is then tried on link $A-D$. The call is blocked only when none of the paths defined in the routing tree are available. In the example shown in Figure 3.1(d), the call is blocked if it is blocked on link $A-D$, $D-E$, or $E-F$.

Instead of trying downstream neighbors one at a time, the controlled-flooding routing algorithms try all downstream neighbors simultaneously. They are referred to as controlled-flooding algorithms because call requests are sent only to downstream neighbors that appear in the routing graph. For the OOC control rule, only the source node sends out multiple call request messages. Controlled flooding versions of SOC and Crankback collapse into one identical algorithm in which nodes with more than one downstream neighbor send out simultaneous multiple call request messages.

3.3 Analytical Models

In studying the influence of network characteristics such as routing algorithms, the network performance measures in which we are interested are end-to-end call set up delay (which includes call processing delays and propagation delays) and the call blocking probability. The influence of network characteristics on these performance metrics will be examined through analytical models. In this section, we present the analytical models for homogeneous networks. With the homogeneity assumption, the network topology is symmetric, every node is treated independently, and behaves in a statistically identical manner. The same is true for the link as well. Appendix B discusses how this methodology can be extended to heterogeneous networks. In our analytical models, we make the following assumptions and notations.

1. External call arrivals at each node are assumed to be governed by a stationary Poisson process with parameter λ.

2. Call set-up requests (either originating at the node or being received from "upstream" nodes) to a node are also assumed to arrive according to a Poisson process.

3. The call holding time, denoted by τ, for each call is assumed to be arbitrarily distributed with mean $\bar{\tau}$.

4. Call request processing (service) times, denoted by T, are assumed to be exponentially distributed with parameter $1/\bar{T}$.

5. Propagation delays, denoted by D_p, are assumed to be constants.

6. We assume that call requests are transmitted on dedicated channels and do not contend for transmission media. We also assume that they are not lost in the switches (or NCU's) and their transmission delays are negligible (due to the extremely high transmission rate).

7. Calls are blocked independently at all links [15, 21, 81].

8. The release of resources, either due to call termination or call abortion, consumes zero processing time.

9. The network topology and routing trees are fixed.

10. Each node has M statistically identical incoming, as well as outgoing, links.

11. Let $\mathcal{L}$ be the steady state blocking probability on each link. Note that $\mathcal{L}$ is identical for each link due to the homogeneity assumption.

12. Let P_{rej} be the end-to-end call blocking probability.

13. Let $\bar{T}_{setup}$ be the average end-to-end call set up delay.

Resources on the link are reserved either for the entire call holding time or a short *blocked-call-clear time*, the time from when the resource is first reserved by a call until it is released, due to the call being blocked downstream. This blocked-call-clear time includes the downstream call processing delays and propagation delays. Unlike [15, 21, 35, 39, 58, 83], we do not assume that call setup time and the blocked-call-clear time are negligible. Indeed, modeling these non-negligible delays and determining their effect on call setup times is one of the goals of this research.

The design of algorithms for deciding whether to admit/reject the call on a given link is beyond the scope of this research. However, we model the effects of such algorithms in the following manner. From the standpoint of admission control, the

number of existing connections together with their QOS requirements is the minimal information needed to make new connection acceptance decisions. Note that these existing connections should include not only connections already established but also connections that are in the process of being set up, having already reserved resources on a link. In this study, we assume that the total number of existing connections on the link is the only parameter that affects admission control. Define $B_\ell(x)$ as the probability that link ℓ cannot guarantee the QOS for a new call or for some existing call given the addition of the new call, where x is a measure of the current number of existing calls at link ℓ. The rationale for such a nondeterministic "call admission function" is that external calls may require different amounts of resources. Thus $B_\ell(x)$ gives the probability that the resource requirements of an arriving call exceeds the remaining available capacity of link ℓ, causing the call to be blocked. Modeling the call admission control in this manner enables us to decouple specific QOS mechanisms from the routing issues which are the main focus of this research. Throughout the remainder of this chapter, we will assume that all links are alike and drop the dependence on the link identity. In our numerical examples, we examine the call setup delays and accepted call throughput associated with various routing algorithms and network parameters when $B(x)$ is concave, convex, or linear.

Our analytic models are based on the link-decomposition method. Similar to the original link-decomposition method, we first decompose the overall network problem into a set of independent link and node problems. After the performance of each link and node is computed, the overall network performance such as end-to-end call setup delay and call blocking probability can be recovered from the individual link- and node-performance measures such as nodal processing delay and link blocking probability.

As stated in section 2.1, the coupling between the link- and node-offered traffic and the link blocking probability and the mean nodal processing delay yields a fixed point problem. That is, to compute the link blocking probability and the mean nodal processing delay we need to solve a set of nonlinear equations that have the following forms:

$$\mathcal{L} = F_1(\mathcal{L}, \bar{T})$$

$$\bar{T} \;=\; F_2(\mathcal{L}, \bar{T})$$

Like most sets of nonlinear equations, solutions to this set of nonlinear equations can be obtained only by numerical methods. The numerical method used most frequently is the so called relaxation method. By using the relaxation method, the network-level and link/node-level performance measures are obtained by using an iterative procedure which, at each iteration, does the following:

Algorithm A:

1. Given the individual link-blocking probabilities (due to the admission control), compute the traffic offered to each link and node.

2. The resulting link- and node-offered traffic in turn yield, via simple link and node models, a new set of values for the link-blocking probabilities.

We first discuss how to calculate the nodal processing delays and link-blocking probabilities and then show how the offered traffic to each link and node are computed for the routing algorithms specified in the previous section. Finally, we show how the end-to-end call set up delay and call blocking probabilities are calculated for different routing policies.

3.3.1 Computing Nodal Processing Delay

As discussed earlier, when a call setup request arrives at a node, a certain amount of computation must be performed in order to determine whether to admit/reject the new call. We refer to the delay associated with this computation (both waiting for processing by the NCU as well as NCU processing itself) as the *nodal processing delay*, denoted by T. In addition to assuming that exogenous call arrivals are Poisson, we further assume that the arrival process of call setup requests to each node (NCU) is a Poisson process [15, 21, 81]. Define X to be the time required by an NCU to process a call request (hereafter, we refer this as the call processing time). Recall that X is an

exponential random variable. Therefore, the average call processing delay (queuing delay plus service time), $\bar{T}$, is given by [54],

$$\bar{T} = \frac{\bar{X}}{1 - \lambda \bar{X}},$$

where λ is the arrival rate and $\bar{X}$ is the mean call processing (service) time at a node. (The notation $\bar{Y}$ will be used to represent the mean of a random variable Y throughout this chapter.)

Besides the processing delay, the end-to-end call set up delay also includes the propagation delay on links traversed by a call request. The propagation delay will be modeled as a delay center with a deterministic service time.

3.3.2 Computing Link Blocking Probability

Before we compute the link blocking probability (i.e., the probability that an arriving call cannot obtain the link resources it needs to satisfy its QOS requirement), it is very important to clearly make the distinction between the node-offered call-requests and link-offered call requests. The link-offered call requests are the calls offered by call setups to the *single* link under consideration. The node-offered call requests are the calls offered to *any* of the outgoing links of this node. In the case of OOC and Crankback control, these call requests include the overflow call requests from previous paths.

Let us consider a single link in isolation, as shown in Figure 3.2. (Note that the call setup requests arriving at the node shown in the figure are only the requests that are intended to be sent to the link under consideration; the call requests that come to the node to be sent out on other outgoing links of this node are not shown.) As we will see shortly, the incoming requests to a link can be divided into a number of distinct classes (e.g., in the case of OOC control, each class will be distinguished by being on the ith path attempted between a source/destination pair and being the jth link on that path). We refer to the arrival rate of class i calls as γ_i and define $\Gamma = \sum_{i=1}^{n} \gamma_i$. Each class of calls can be further divided into two types. The first type of calls, which will eventually be accepted if not blocked on the link under investigation, has an arrival rate of γ_i^s. The second type of calls, which has an arrival rate of γ_i^f, will be

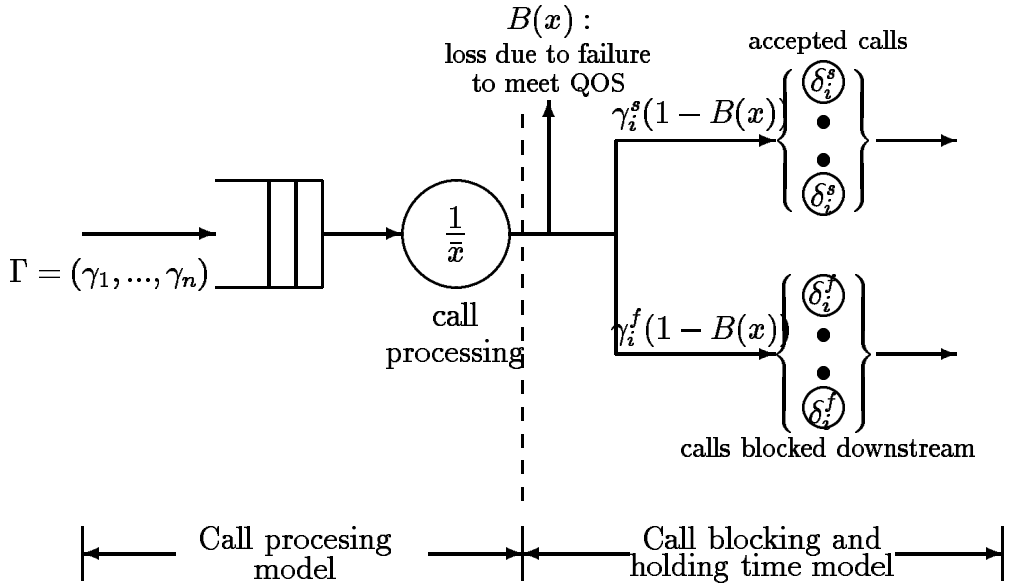


Figure 3.2 Queueing model for processing delay and call holding times

blocked downstream if not blocked on this link. Here, the superscripts s and f refer to whether a call request is eventually successful on this path or fails (is blocked) downstream. Let δ_i^s and δ_i^f denote the mean resource holding times for the two classes of calls. Note that a busy server in a multiserver queue shown in Figure 3.2 models a call that has successfully reserved resources at this link (these resources may be held for the duration of a call or may be released shortly if a call is blocked downstream). Since the link capacity is limited, the maximum number of such busy servers in Figure 3.2 is limited and is denoted by C.

The steady state blocking probability of a link can be obtained by solving for the steady state distribution of the number of calls that are holding resources (i.e., are resident in the multiserver queues) on the link. For computing the steady state distribution of the number of calls (which will eventually either be accepted or blocked downstream) on the link, the original call blocking model shown in Figure 3.2 can be shown to be equivalent to the $M/G/C/C$ queue shown in Figure 3.3. The mean service time, $1/\mu$, for this $M/G/C/C$ queue is given by

$$\frac{1}{\mu} = \sum_{i=1}^{n} \frac{\gamma_i^s}{\Gamma} \cdot \delta_i^s + \sum_{i=1}^{n} \frac{\gamma_i^f}{\Gamma} \cdot \delta_i^f.$$

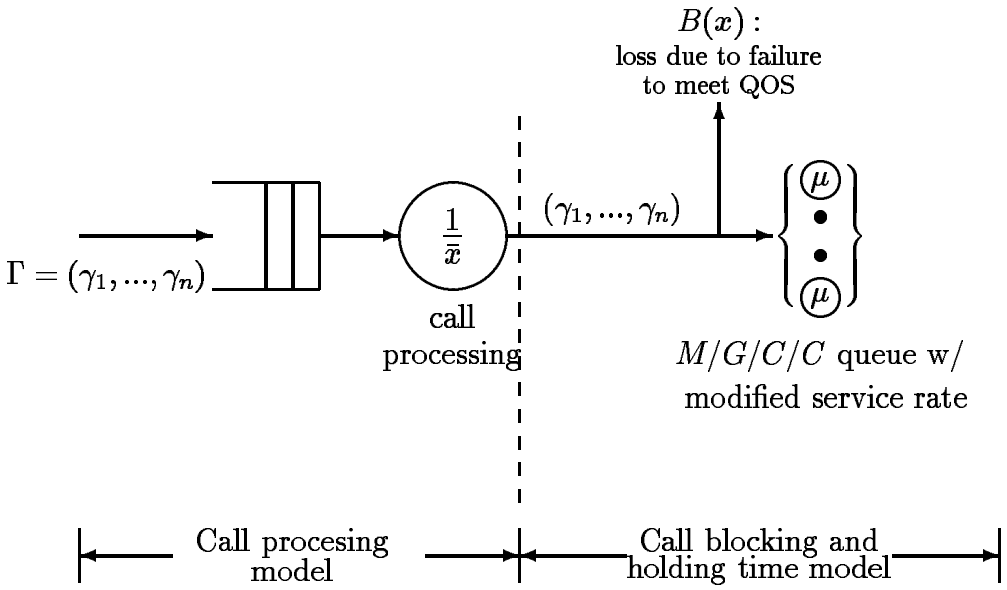


Figure 3.3 Equivalent queueing model for processing delay and call holding times

The arrival rate, $\tilde{\lambda}$, for this queue is state dependent and is given by

$$\tilde{\lambda}(i) = \Gamma \cdot (1 - B(i)) \quad i = 0, ..., C-1.$$

Given the arrival rate and service rate, we can compute the steady state distribution of the number of calls currently allocated bandwidth at this link, $\Pi = (\pi_0, \pi_1, ..., \pi_C)$ where π_x is the steady state probability that there are x calls holding resources at this link and is given by [41]

$$\pi_x = \frac{\prod_{i=0}^{x-1}(\tilde{\lambda}(i)/\mu)/x!}{\sum_{j=0}^{C}\prod_{i=0}^{j-1}(\tilde{\lambda}(i)/\mu)/j!}.$$

Once Π is known, the probability that an arriving call is blocked at this link due to admission control is given by

$$\mathcal{L} = \sum_{i=0}^{C} \pi_i \cdot B(i) \quad i = 0, ..., C.$$

3.3.3 Offered Call Requests, Call Setup Delay, and Call Blocking Probability in Homogeneous Networks

The computation of the link- and node-offered call arrival rate is complicated and depends heavily on the routing tree used. For ease of explanation, we show how the link- and node-offered call arrival rates are computed in a homogeneous network.

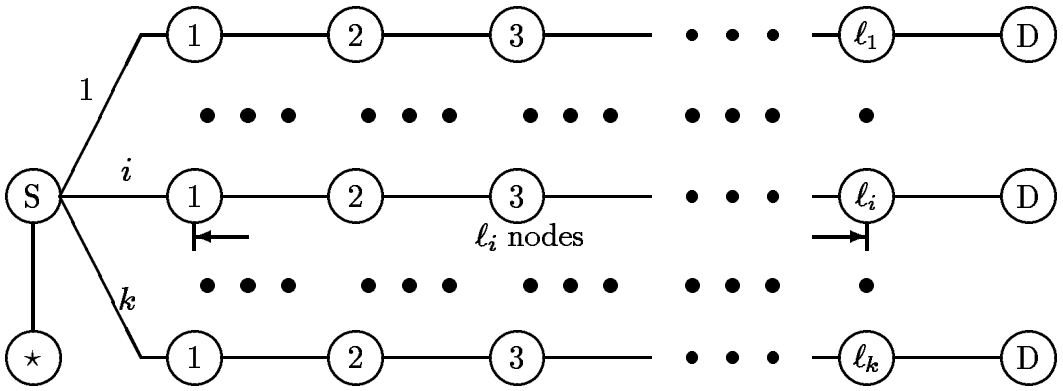


Figure 3.4 Augmented Route Tree for OOC Control Rules (homogeneous case).

3.3.3.1 Sequential Routing Rules

In the following three sections (Section 3.3.3.1.1, 3.3.3.1.2, and 3.3.3.1.3) we show how to compute the node- and link-offered call arrival rate, call blocking probability, and average call setup delay for the three sequential routing rules.

3.3.3.1.1 OOC Routing Rule

As shown in Figure 3.4, we assume that there are k possible paths between each source-destination pair, each with $\ell_i, i = 1, ..., k$, intermediate nodes. Let us begin by calculating certain quantities that will be required in the computation of the link- and node-offered call request rate, end-to-end call blocking probability, and mean end-to-end call setup delay.

First we define P^i_{fail} as the probability that a call fails on path i given that an attempt is made to set the call up on that path. Under the independence assumption, we have

$$\begin{aligned} P^i_{fail} &= \sum_{j=0}^{\ell_i} \mathcal{L}(1-\mathcal{L})^j, \\ &= 1-(1-\mathcal{L})^{\ell_i+1}. \end{aligned} \tag{3.1}$$

Define $\bar{\mathcal{N}}_i$ as the expected number of nodes visited along path i by a call setup message given that it fails on the selected path. $\bar{\mathcal{N}}_i$ is given by

$$\begin{aligned}\bar{\mathcal{N}}_i &= \sum_{j=1}^{\ell_i}(j+1)\frac{(1-\mathcal{L})^j\mathcal{L}}{P^i_{fail}}, \\ &= \frac{1-[(\ell_i+1)\mathcal{L}+1](1-\mathcal{L})^{\ell_i+1}}{\mathcal{L}P^i_{fail}} - \frac{\mathcal{L}}{P^i_{fail}}. \end{aligned} \tag{3.2}$$

Let $\bar{N}_{i,j}$ denote the expected number of nodes a call set up message visits after the jth node on path i given that it has successfully reserved resources at the jth node and fails at some node farther down path i. $\bar{N}_{i,j}$ is given by

$$\begin{aligned}\bar{N}_{i,j} &= \sum_{n=1}^{\ell_i-j} n\frac{(1-\mathcal{L})^{n-1}\mathcal{L}}{\sum_{m=1}^{\ell_i-j}(1-\mathcal{L})^{m-1}\mathcal{L}}, \\ &= \frac{1-[(\ell_i-j)\mathcal{L}+1](1-\mathcal{L})^{\ell_i-j}}{\mathcal{L}[1-(1-\mathcal{L})^{\ell_i-j}]}. \end{aligned} \tag{3.3}$$

Link-offered call requests

Let us now focus on a single *link* in the network. As shown in Figure 3.2, the incoming traffic to a link is divided into several classes. Under OOC control, the incoming traffic to a link is divided into $\sum_{i=1}^{k}(\ell_i+1)$ classes which will be referred to as (i,j), $i=1,...,k$, $j=0,...,\ell_i$, representing the jth link on the ith path between all source/destination pairs. The arrival rate of the (i,j)th class of traffic, which is referred to as $\gamma_{i,j}$, is the total call setup traffic offered to a link by call requests which reach this link as the jth link of path i for all source/destination pairs in the network. $\gamma_{i,0}$ can be viewed as the rate of exogenous call requests entering the network at a node incident to this link, and being offered to this link as the first link on the ith path from this source node. Because of the homogeneity assumption and the fact that each node has M statistically identical outgoing links, we have the following equations for all of the links.

$$\gamma_{1,0} = \frac{\lambda}{M}, \tag{3.4}$$

$$\gamma_{i,0} = \frac{\lambda}{M}\prod_{j=1}^{i-1}P^j_{fail}, \quad i=2,...,k, \tag{3.5}$$

$$\gamma_{i,j} = \gamma_{i,j-1}(1-\mathcal{L}), \quad j=1,...,\ell_i, \ \ i=1,...,k. \tag{3.6}$$

Recall that each class of offered call requests to link (i,j) can be further divided into two types: those that are eventually successful on this path and those that fail. The

arrival rate of each type of offered call requests, $\gamma^s_{i,j}$ and $\gamma^f_{i,j}$ respectively, and the mean call holding time, $\delta^s_{i,j}$ and $\delta^f_{i,j}$, are computed as follows,

$$\begin{aligned}
\gamma^s_{i,j} &= \gamma_{i,j}(1-\mathcal{L})^{\ell_i - j}, \quad j = 0, ..., \ell_i, \;\; i = 1, ..., k, && (3.7)\\
\gamma^f_{i,j} &= \gamma_{i,j} - \gamma^s_{i,j}, \\
&= \gamma_{i,j}(1-(1-\mathcal{L})^{\ell_i - j}), \quad j = 0, ..., \ell_i, \;\; i = 1, ..., k, && (3.8)\\
\delta^s_{i,j} &= (\ell_i - j) * (\bar{T} + D_p) + D_p + \bar{\tau}, && (3.9)\\
\delta^f_{i,j} &= \bar{N}_{i,j}(\bar{T} + D_p). && (3.10)
\end{aligned}$$

Recall that $\bar{T}$ is the mean processing delay (queueing plus service) at each node (which is still unknown at this point, since the overall call arrival rates are unknown), D_p is the propagation delay and $\bar{\tau}$ is the mean call holding time. Relations (3.7)-(3.10) can be used to compute the blocking probability on the link during an iteration of **Algorithm A** discussed in Section 3.3.2.

Node-offered call requests

Let us now focus on a single *node* in isolation. First, we assume that when a call request is blocked on the source node's jth outgoing link, the call request is immediately tried on the $(j+1)$st link. The rate of call requests which reach this node as the source node is referred to as G_0, and is given by:

$$G_0 = \lambda + M \sum_{i=1}^{k-1} \gamma_{i,0}(P^i_{fail} - \mathcal{L}). \qquad (3.11)$$

Note that G_0 includes *"first time"* exogenous call requests as well as call setup attempts which have been previously blocked. The node-offered call request rate, G, is then given by

$$G = G_0 + M \sum_{i=1}^{k} \sum_{j=1}^{\ell_i} \gamma_{i,j}. \qquad (3.12)$$

Call setup delay and call blocking probability

Since a call is lost if it is blocked on all paths, the probability that a call request is rejected is given by

$$P_{rej} = \prod_{i=1}^{k} P_{fail}^{i}. \tag{3.13}$$

The average call set-up delay, $\bar{T}_{setup}$, is then computed as follows:

$$\bar{T}_{setup} = \sum_{i=1}^{k} \left[\sum_{j=1}^{i-1} \bar{\mathcal{N}}_j + \ell_i + 1 \right] [\bar{T} + D_p] \frac{(1 - P_{fail}^{i}) \prod_{j=1}^{i-1} P_{fail}^{j}}{1 - P_{rej}}. \tag{3.14}$$

The term $\sum[\bar{\mathcal{N}}_j + \ell_i + 1]$ in equation (3.14) is the expected number of nodes a call setup message visits given that the call is successfully set up on the ith path. $\bar{T} + D_p$ is the expected processing and propagation delay. The final (fractional) term is the probability that the call is successfully set up on path i given that the call is not blocked.

3.3.3.1.2 SOC Routing Rule

The routing tree shown in Figure 3.5 is shared by the SOC routing rule and the crankback routing rule. The *loss nodes* in the tree are omitted. Each node in the routing tree is given a unique label according to the following rules. The node with label $(i_1, ..., i_{\ell-1}, i_\ell)$ is the i_ℓ-th child of node $(i_1, ..., i_{\ell-1})$ and has $k_{i_1,...,i_\ell}$ children labeled $(i_1, ..., i_\ell, 1), ..., (i_1, ..., i_\ell, k_{i_1,...,i_\ell})$. The source node is labeled (0) while its children are labeled $(1), (2), ..., (k_0)$ respectively. (Note that due to the homogeneity assumption, all O-D pairs have routing trees of the same form and each node will be node $(i_1, ..., i_\ell)$ in the routing tree of some O-D pair, $\forall(i_1, ..., i_\ell)$.) The set of all nodes that are directly connected to the destination node by a link is denoted by $\mathcal{D}$. A node $n \in \mathcal{D}$ always tries the direct link first.

The following notation, which is a straightforward generalization of the notation used for OOC, is used in the analyses of both SOC and Crankback.

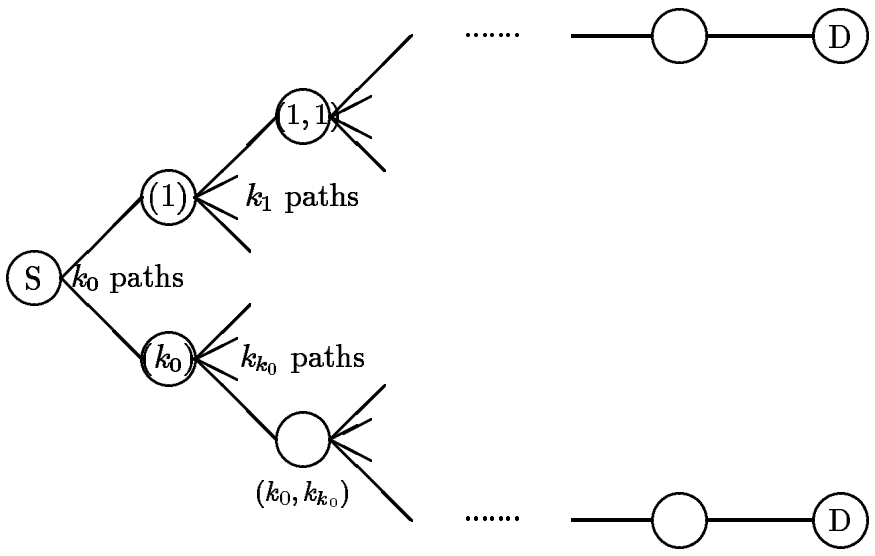


Figure 3.5 Routing tree for SOC and Crankback control rules (homogeneous case)

- $\bar{P}_{i_1,...,i_\ell}$: the probability that a call request is blocked at node $(i_1, ..., i_\ell)$ or later given that it has reached node $(i_1, ..., i_\ell)$.

- $\bar{N}_{i_1,...,i_\ell}$: the expected number of nodes visited at and after node $(i_1, ..., i_\ell)$ given that the call request is blocked at node $(i_1, ..., i_\ell)$ or later.

- $\bar{\bar{N}}_{i_1,...,i_\ell}$: the expected number of nodes visited at and after node $(i_1, ..., i_\ell)$ given that the call request is successfully set up through node $(i_1, ..., i_\ell)$ and subsequent nodes.

- $G_{i_1,...,i_\ell}$: the offered call request rate to the *node* under consideration by call requests which reach this node as the $(i_1, ..., i_\ell)$ node in the routing tree.

- $\gamma_{(i_1,...,i_\ell),j}$: the offered call request rate to the *link* under consideration by call requests which reach this link as the jth outgoing link of node $(i_1, ..., i_\ell)$ in the routing tree.

- $\gamma^s_{(i_1,...,i_\ell),j}$: the part of $\gamma_{(i_1,...,i_\ell),j}$ which will eventually be successfully set up. Recall that the mean resource holding time for successfully setup calls is $\delta^s_{(i_1,...,i_\ell),j}$.

- $\gamma^f_{(i_1,...,i_\ell),j}$: the part of $\gamma_{(i_1,...,i_\ell),j}$ which will be later blocked at some node. As before, the mean resource holding time for calls blocked downstream is $\delta^f_{(i_1,...,i_\ell),j}$.

Let us now calculate certain quantities that will be needed in the computation of both the link-offered call request rate, call setup delay and call blocking probability. The quantities $\bar{P}_{i_1,...,i_\ell}$, $\bar{N}_{i_1,...,i_\ell}$ and $\bar{\mathcal{N}}_{i_1,...,i_\ell}$ satisfy the following recursions. Note that the computation can be done by scanning the routing tree only once. We first compute the probability that a call request is blocked at node $(i_1,...,i_\ell)$ or later.

$$\bar{P}_{i_1,...,i_\ell} = [\sum_{j=1}^{k_{i_1,...,i_\ell}} \mathcal{L}^{j-1}(1-\mathcal{L})\bar{P}_{i_1,...,i_\ell,j}] + \mathcal{L}^{k_{i_1,...,i_\ell}} \tag{3.15}$$

with $\bar{P}_{i_1,...,i_\ell,1} = 0, \ \forall (i_1,...,i_\ell) \in \mathcal{D}$. (Recall that $(i_1,...,i_\ell,1)$ is the destination node.) The term $\mathcal{L}^{j-1}$ in equation (3.15) is the probability that a call setup message is blocked at this node on the first $j-1$ outgoing links. The term $(1-\mathcal{L})\bar{P}_{i_1,...,i_\ell,j}$ is the probability that a call setup message successfully reserves bandwidth on the jth outgoing link but is blocked downstream. The final term $\mathcal{L}^{k_{i_1,...,i_\ell}}$ is the probability that a call setup message is blocked at this node on all outgoing links (i.e., that none of the outgoing links to the destination can provide the requested QOS).

For $\bar{N}_{i_1,...,i_\ell}$, we have

$$\bar{N}_{i_1,...,i_\ell} = \left(\sum_{j=1}^{k_{i_1,...,i_\ell}} \frac{\mathcal{L}^{j-1}(1-\mathcal{L})\bar{P}_{i_1,...,i_\ell,j}}{\bar{P}_{i_1,...,i_\ell}} \bar{N}_{i_1,...,i_\ell,j} \right) + 1 \tag{3.16}$$

with $\bar{N}_{i_1,...,i_\ell,1} = 0, \ \forall (i_1,...,i_\ell) \in \mathcal{D}$. The first (fractional) term in equation (3.16) is the probability that a call setup message successfully reserves bandwidth on the jth outgoing link but fails downstream given that it is eventually blocked. The term "1" accounts for the visit to the current node. Note that, as we assumed in the OOC rule, a call request blocked on the jth outgoing link is immediately tried on the $(j+1)$st link.

Similarly, the expected number of nodes visited at and after node $(i_1,...,i_\ell)$ given that the call request is *successfully* set up through node $(i_1,...,i_\ell)$ and subsequent nodes is given by

$$\bar{\mathcal{N}}_{i_1,...,i_\ell} = \left(\sum_{j=1}^{k_{i_1,...,i_\ell}} \frac{\mathcal{L}^{j-1}(1-\mathcal{L})(1-\bar{P}_{i_1,...,i_\ell,j})}{1-\bar{P}_{i_1,...,i_\ell}} \bar{\mathcal{N}}_{i_1,...,i_\ell,j} \right) + 1 \tag{3.17}$$

with $\bar{\mathcal{N}}_{i_1,...,i_\ell,1} = 0, \ \forall (i_1,...,i_\ell) \in \mathcal{D}$. The first (fractional) term in equation (3.17) is the probability that a call setup message is successfully set up through the jth child of node $(i_1,...,i_\ell)$ given that it is successfully set up through node $(i_1,...,i_\ell)$.

Link-offered call requests

The quantities $\bar{P}_{i_1,...,i_\ell}$, $\bar{N}_{i_1,...,i_\ell}$ and $\bar{\mathcal{N}}_{i_1,...,i_\ell}$ are nodal performance measures. Now let us consider link-level measures and focus on a single link in the network and compute the offered call request rate at this link. The following equations are used to compute these rates and mean resources holding times of each class of calls,

$$\gamma_{(0),j} = \frac{\lambda}{M}\mathcal{L}^{j-1}, \quad j = 1,...,k_0, \tag{3.18}$$

$$\gamma_{(i),j} = \gamma_{(0),i}(1-\mathcal{L}), \mathcal{L}^{j-1} \quad i = 1,...,k_0, \ \ j = 1,...,k_i, \tag{3.19}$$

$$\gamma_{(i_1,...,i_\ell),j} = \gamma_{(i_1,...,i_{\ell-1}),i_\ell}(1-\mathcal{L})\mathcal{L}^{j-1}, \quad j = 1,...,k_{i_1,...,i_\ell}, \tag{3.20}$$

$$\gamma^s_{(i_1,...,i_\ell),j} = \gamma_{(i_1,...,i_\ell),j}(1-\bar{P}_{i_1,...,i_\ell,j}), \tag{3.21}$$

$$\gamma^f_{(i_1,...,i_\ell),j} = \gamma_{(i_1,...,i_\ell),j}\bar{P}_{i_1,...,i_\ell,j}, \tag{3.22}$$

$$\delta^s_{(i_1,...,i_\ell),j} = \bar{\mathcal{N}}_{i_1,...,i_\ell,j}(\bar{T}+D_p) + D_p + \bar{\tau}, \tag{3.23}$$

$$\delta^f_{(i_1,...,i_\ell),j} = \bar{N}_{i_1,...,i_\ell,j}(\bar{T}+D_p). \tag{3.24}$$

The term $\frac{\lambda}{M}$ in equation (3.18) is the portion of exogenous call requests offered to the link by call requests which reach this link as the first outgoing link of the source node. (The division by M results from the homogeneity assumption and the fact that each node has M statistically identical outgoing links.) The final term, $\mathcal{L}^{j-1}$, in equation (3.18) is the probability that a call request is blocked on the first $j-1$ links. (Recall that $\gamma_{(0),j}$ is the traffic offered to the link by call requests which reach this link as the jth outgoing link of the source node.) Similarly, the term $\gamma_{(0),i}(1-\mathcal{L})$ in equation (3.19) is the offered call requests which reach this link as the first outgoing link of node (i). Therefore, the right side of equation (3.19) gives the rate at which call requests reach this link as the jth outgoing link of node (i). The term $\gamma_{(i_1,...,i_\ell),j}(1-\bar{P}_{i_1,...,i_\ell,j})$ in equation (3.21) is the portion of the calls offered to the link by call requests which reach this link as the jth link of node $(i_1,...,i_\ell)$ and which are eventually *successfully* set up if not blocked on this link. (The reader is referred to Figure 3.2.)

Node-offered call requests

Let us now focus on a single node in isolation. In order to calculate the node-offered call request rate, let us focus on the M incoming link flows. Recall that the rate at which call requests reach this node as the $(i_0, ..., i_\ell)$ node for all source/destination pairs in the network is referred to as $G_{i_0,...,i_\ell}$. Because of the homogeneity assumption, $G_{i_0,...,i_\ell}$ can be computed as follows,

$$G_0 = \lambda, \tag{3.25}$$

$$G_j = M\ \gamma_{(0),j}(1-\mathcal{L}), \quad j = 1, ..., k_0, \tag{3.26}$$

$$G_{i_1,...,i_\ell} = M\ \gamma_{(i_1,...,i_{\ell-1}),i_\ell}(1-\mathcal{L}). \tag{3.27}$$

The term $\gamma_{(i_1,...,i_{\ell-1}),i_\ell}(1-\mathcal{L})$ in equation (3.27) is the call request rate arriving at the node by call requests which reach this node as the $(i_0, ..., i_\ell)$ node in routing tree from *one* incoming link, and is multiplied by M because each node has M statistically identical incoming links.

The node-offered call request rate, G, is then given by

$$G = \sum G_{i_1,...,i_\ell}.$$

The summation is for all nodes in the routing trees of all source/destination pairs in the network except the destination node.

Call setup delay and call blocking probability

Recalling that the source node has the label "0", the end-to-end call blocking probability is given by

$$P_{rej} = \bar{P}_0 \tag{3.28}$$

and the average call set-up delay is given by

$$\bar{T}_{setup} = \bar{N}_0(\bar{T} + D_p) \tag{3.29}$$

3.3.3.1.3 Crankback Routing Rule

The notation used in the analysis of the crankback routing rule is the same as that used in the analysis of the SOC routing rule. Again, we begin by presenting expressions for the quantities $\bar{P}_{i_1,...,i_\ell}, \bar{N}_{i_1,...,i_\ell}$ and $\bar{\mathcal{N}}_{i_1,...,i_\ell}$. The analysis is similar to that of SOC and hence we present the equations with little discussion,

$$\bar{P}_{i_1,...,i_\ell} = \prod_{j=1}^{k_{i_1,...,i_\ell}} [\mathcal{L} + (1-\mathcal{L})\bar{P}_{i_1,...,i_\ell,j}] \tag{3.30}$$

with $\bar{P}_{i_1,...,i_\ell,1} = 0, \ \forall (i_1,...,i_\ell) \in \mathcal{D}$,

$$\bar{N}_{i_1,...,i_\ell} = \left(\sum_{j=1}^{k_{i_1,...,i_\ell}} \frac{(1-\mathcal{L})\bar{P}_{i_1,...,i_\ell,j}}{\mathcal{L} + (1-\mathcal{L})\bar{P}_{i_1,...,i_\ell,j}} (\bar{N}_{i_1,...,i_\ell,j} + 1) \right) + \frac{\mathcal{L}}{\mathcal{L} + (1-\mathcal{L})\bar{P}_{i_1,...,i_\ell,k_{i_1,...,i_\ell}}} \tag{3.31}$$

with $\bar{N}_{i_1,...,i_\ell,1} = 0, \ \forall (i_1,...,i_\ell) \in \mathcal{D}$. The first term corresponds to the fact that if the call is blocked at node $(i_1,...,i_\ell)$, it must be blocked on each outgoing link or at some "downstream" node of this link. Also, recall that if the call blocks on the jth outgoing link, it is immediately tried on the $(j+1)$st link,

$$\begin{aligned} \bar{\mathcal{N}}_{i_1,...,i_\ell} = & \sum_{j=1}^{k_{i_1,...,i_\ell}} \frac{[\prod_{m=1}^{j-1}(\mathcal{L} + (1-\mathcal{L})\bar{P}_{i_1,...,i_\ell,m})](1-\mathcal{L})(1-\bar{P}_{i_1,...,i_\ell,j})}{1-\bar{P}_{i_1,...,i_\ell}} \\ & [(\sum_{m=1}^{j-1} \frac{(1-\mathcal{L})\bar{P}_{i_1,...,i_\ell,m}}{\mathcal{L} + (1-\mathcal{L})\bar{P}_{i_1,...,i_\ell,m}} (\bar{N}_{i_1,...,i_\ell,m} + 1)) + \bar{\mathcal{N}}_{i_1,...,i_\ell,j} + 1] \end{aligned} \tag{3.32}$$

with $\bar{\mathcal{N}}_{i_1,...,i_\ell,1} = 0, \ \forall (i_1,...,i_\ell) \in \mathcal{D}$.

Link-offered call requests

The link-offered call request rate is computed as follows:

$$\begin{aligned} \gamma_{(0),1} &= \frac{\lambda}{M}, \\ \gamma_{(0),j} &= \frac{\lambda}{M} \prod_{m=1}^{j-1} [\mathcal{L} + (1-\mathcal{L})\bar{P}_m], \quad j = 2,...,k_0, \end{aligned}$$

$$
\begin{aligned}
\gamma_{(i_1,\ldots,i_\ell),1} &= \gamma_{(i_1,\ldots,i_{\ell-1}),i_\ell}(1-\mathcal{L}),\\
\gamma_{(i_1,\ldots,i_\ell),j} &= \gamma_{(i_1,\ldots,i_\ell),1}\prod_{j=1}^{i-1}[\mathcal{L}+(1-\mathcal{L})\bar{P}_{i_1,\ldots,i_\ell,j}], \quad j=1,\ldots,k_{i_1,\ldots,i_\ell},\\
\gamma^s_{(i_1,\ldots,i_\ell),j} &= \gamma_{(i_1,\ldots,i_\ell),j}(1-\bar{P}_{i_1,\ldots,i_\ell,j}),\\
\gamma^f_{(i_1,\ldots,i_\ell),j} &= \gamma_{(i_1,\ldots,i_\ell),j}\bar{P}_{i_1,\ldots,i_\ell,j},\\
\delta^s_{(i_1,\ldots,i_\ell),j} &= \bar{\mathcal{N}}_{i_1,\ldots,i_\ell,i}(\bar{T}+D_p)+D_p+\bar{\tau},\\
\delta^f_{(i_1,\ldots,i_\ell),j} &= \bar{N}_{i_1,\ldots,i_\ell,i}(\bar{T}+D_p).
\end{aligned}
$$

Node-offered call requests

The node-offered call request rate is computed as $G = \sum G_{i_1,\ldots,i_\ell}$. where the summation is for all nodes in the routing trees of all source/destination pairs in the network except the destination node and

$$
G_{i_1,\ldots,i_\ell} = M\left(\gamma_{(i_1,\ldots,i_\ell),1}+\sum_{j=1}^{k_{i_1,\ldots,i_\ell}-1}\gamma_{(i_1,\ldots,i_\ell),j}(1-\mathcal{L})\bar{P}_{i_1,\ldots,i_\ell,j}\right).
$$

Call setup delay and call blocking probability

Equations (3.28) and (3.29), used in SOC, also applies here.

3.3.3.2 Controlled-Flooding Routing Rules

The analytical models for the two controlled-flooding routing rules are much more complicated and are presented in Appendix A.

3.4 Numerical Study on NSFNET

In this section we report the results of a study of the previously described policies on the NSFNET T3 backbone network. The NSFNET T3 backbone network as of July, 1992 is shown in Figure 3.6. We assume there are two T3 links, one for each direction, between each directly connected Core Nodal Switching Subsystem (CNSS) pair [1]. Thus, the network topology we study consists of 12 nodes, 30 T3-links and

[1]There is only one T3 link between each directly connected CNSS pair and the traffic of each direction is transmitted at 22.5Mbps.

Table 3.1 Total number of packets in and out of each Core Nodal Switching Subsystem (CNSS) in March 1992.

CNSS	Packets	CNSS	Packets
Hartford	1,477,934,832	New York	18,086,389
Washington	526,211,414	Greensboro	275,258,717
Cleveland	1,136,546,512	Chicago	12,710,280
St. Louis	194,983,478	Houston	533,028,743
Denver	5,540,142	Seattle	4,948,168
San Francisco	732,524,840	Los Angeles	263,799,005

132 source/destination pairs. Each node is assumed to have the same processing capacity. We assume the exogenous call arrival rate for each source/destination pair is proportional to the traffic load at each node reported in NSFNET [60]. Specifically, Table 3.1 shows the total number of packets in and out of each node, referred to as Core Nodal Switching Subsystem (CNSS) in NSF's report, March 1992. According to this table, the call arrival rate to each source/destination pair is constructed in the following manner. Let A_i be the number of packets in and out of node i in Table 3.1. For a given exogenous call arrival rate to the network, λ, the call arrival rate to node i, λ_i, is proportional to the number of packets in and out of that node, i.e. $\lambda_i = \lambda A_i / \sum_k A_k$. This traffic is divided between all potential destinations in proportion to the number of packets in and out of node j, i.e., $\lambda_{i,j} = \lambda_i A_j / \sum_{k \neq i} A_k$. In other words,

$$\lambda_{i,j} = \lambda \frac{A_i}{\sum_k A_k} \frac{A_j}{\sum_{k \neq i} A_k}.$$

We first validate our analytical models by simulation results. We then compare the performance of the five routing schemes discussed in previous sections and study the effects of call processing delay and the admission control function on the call setup delay.

3.4.1 Analytical Results Versus Simulation Results

The performance results comparing analysis and simulation are based on the parameter values below. In subsequent sections, we will vary these values and examine their effect.

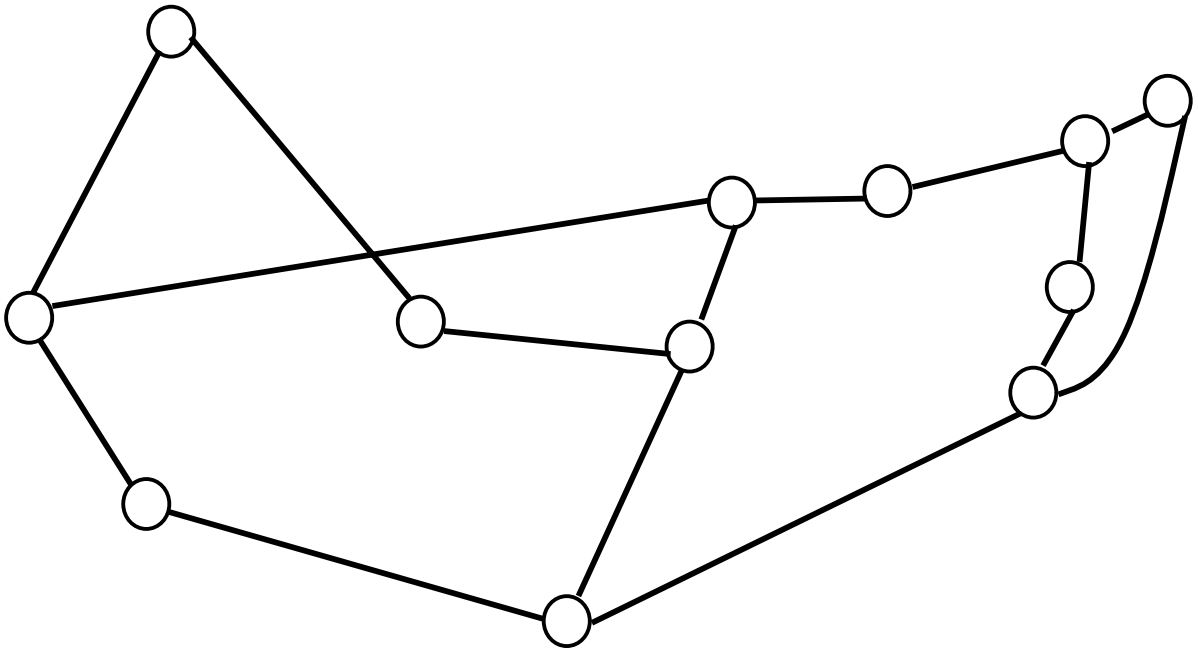


Figure 3.6 NSFNET topology

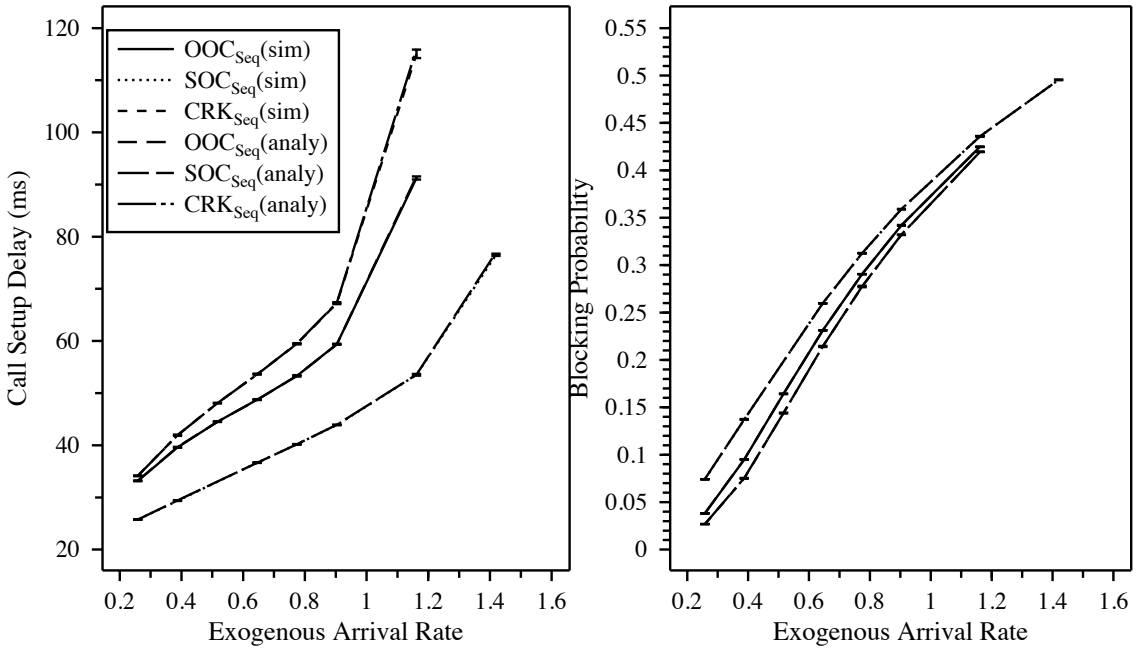
- The propagation speed is assumed to be 125 miles per millisecond, i.e., two third of the speed of light.
- The mean call holding time, $\bar{\tau}$, is set to 10 seconds.
- The mean service time for a call request is set to 2 milliseconds.
- The maximum number of connections a link can accommodate at a time, C, is set to 1406. (With T3 links, this is equivalent to assuming that a connection requires on average 32 KBits per second.)
- The admission control function is set to $B(x) = (\frac{x}{C})^2$.

Figure 3.7(a) and (c) compare the analytic results with the simulation results for the average call setup delay between Hartford and San Francisco. Figure 3.7(b) and (d) compare the analytic results with the simulation results for overall network blocking probability. Each simulation point is observed over 10 independent runs. Each run is run for 3000 units of call holding time. For each run, the initial 10%

is discarded. The vertical lines about each point indicate the 95 percent confidence interval. We note that the analytic and simulation results agree very closely under various traffic loads for all routing mechanisms.

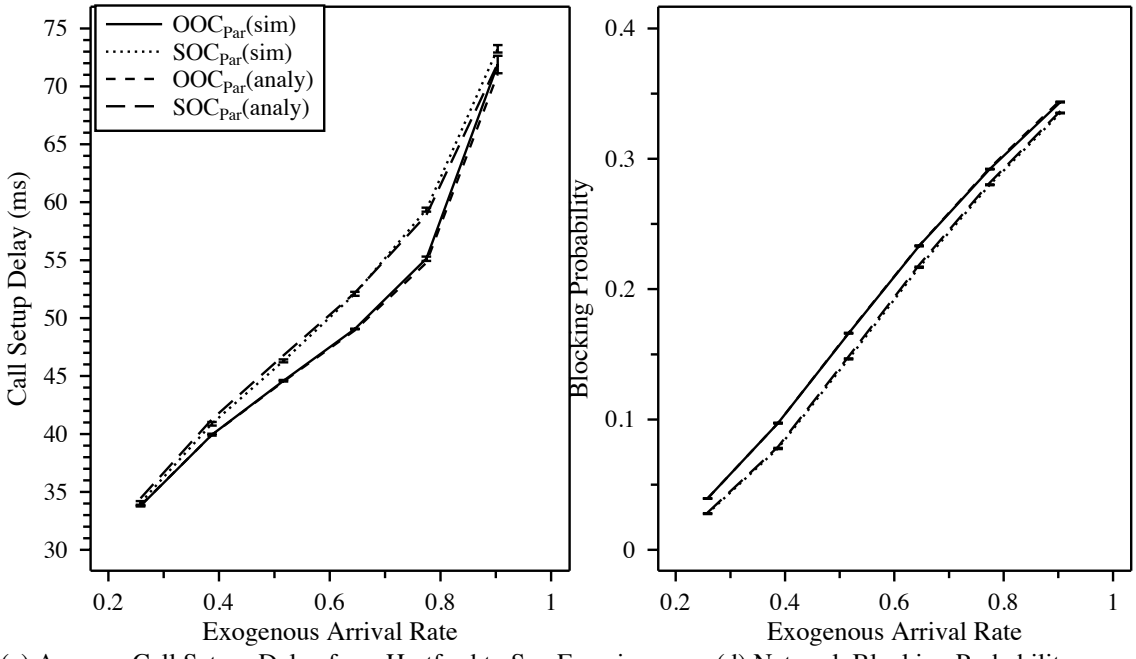
3.4.2 Comparison of Different Routing Algorithms

The trade-offs between call setup delays and call blocking probabilities for different routing algorithms are given in Figure 3.8. Consider the three sequential routing algorithms. From Figure 3.8, we can see the trade-off between the call blocking probability and the call setup delay. For a given exogenous call arrival rate, the crankback algorithm yields the lowest blocking probability but the highest average call setup delay. On the other hand, the SOC algorithm yields the highest blocking probability but lowest average call setup delay. However, for a given network throughput of accepted calls, as we will observe in Figure 3.9, the SOC algorithm always yields the smallest average call setup delay and highest maximum achievable throughput. (Note that the maximum achievable throughput is the asymptotic point at which the average call setup time goes to infinity. When the average call setup time approaches infinity, the connections that are in the process of being setup will hold the resources that have already being reserved for an infinite time and thus the blocking probability will approach unity.) This observation is slightly different from what we observed in an earlier study [50]. In [50], we studied a 8-node hypercube network and observed that the SOC algorithm yielded a slightly lower maximum achievable throughput than the crankback algorithm. The main reason for the difference is that when the network topology is very sparse (as in the NSFnet case), the trade-off between the call blocking probability and the call setup delay becomes significant. In order to reduce the call blocking probability, we would like to have more alternate paths in the routing trees. But because the network topology is very sparse, increasing the number of alternate paths will unavoidably include paths with longer propagation delays and more intermediate nodes which, consequently increases the average call setup delay. The SOC algorithm thus performs better in this network because calls are offered more chances to be routed through "short" paths than under the other two routing algorithms.



(a) Average Call Set-up Delay from Hartford to San Francisco Sequential rules

(b) Network Blocking Probability Sequential rules



(c) Average Call Set-up Delay from Hartford to San Francisco Parallel Rules

(d) Network Blocking Probability Parallel Rules

Figure 3.7 Simulation results vs analysis results

In comparing the results of the parallel (flooding) algorithms to the sequential algorithms, we observe that for a given exogenous call arrival rate, the sequential OOC algorithm and the parallel OOC algorithm yield almost the same blocking probability. The parallel SOC/crankback algorithm yields a slightly higher blocking probability than the sequential crankback algorithm for a given exogenous call arrival rate because it reserves more resources during call setup. Since the parallel algorithms require much more processing, the processing elements quickly saturate and, thus, yield much lower achievable maximum throughput. For the given parameter values, we also observe that parallel schemes yield lower average call setup delay only for small exogenous arrival rates. This is because at small exogenous arrival rates, processing delays are relative small at each node, thus the parallel algorithms can take the advantage of attempting call setups in parallel over all possible paths.

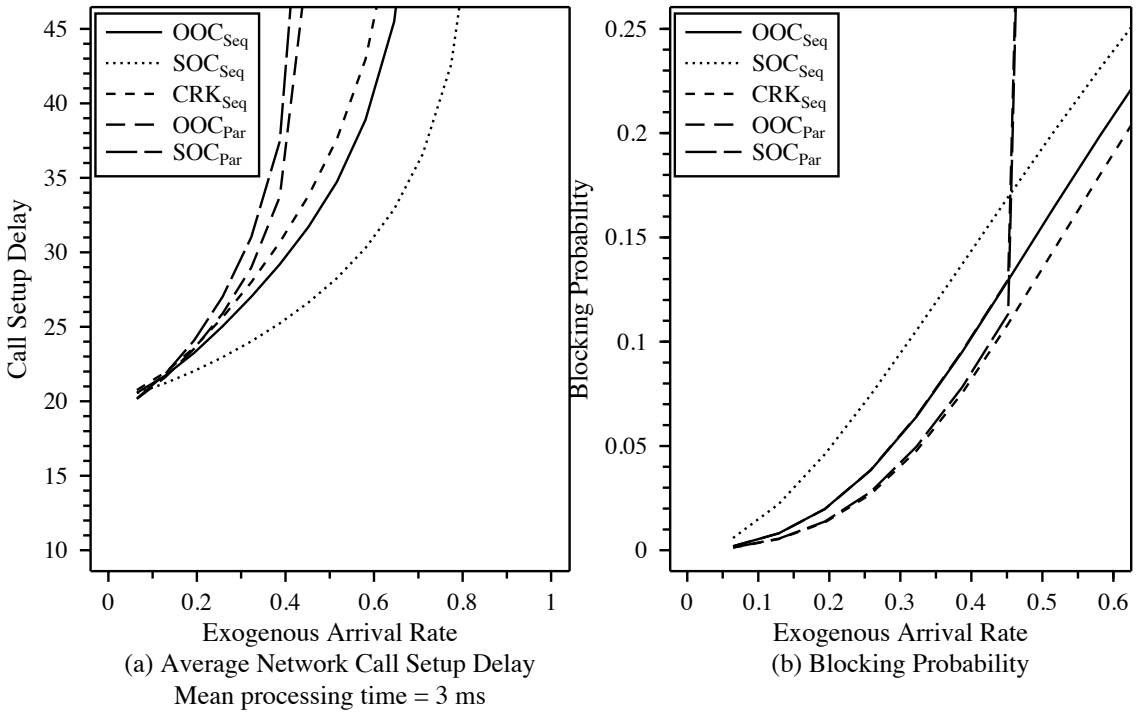


Figure 3.8 Comparison of different routing schemes

3.4.3 The Effects of Call Processing (Service) Time

The effect that the mean control packet processing (service) time has on the call setup delay and accepted call throughput is shown in Figure 3.9. By comparing Figure 3.9(a) and Figure 3.9(b), we observe that increasing the processing requirements for a call (i.e., from 1 millisecond to 3 milliseconds) affects the performance of these five routing algorithms significantly. We know that parallel routing algorithms generate more call request messages than sequential algorithms. Therefore when processing capacity is very limited, parallel routing algorithms will saturate the processing elements very quickly. Thus, sequential algorithms can offer much higher throughputs. Certainly, at very low traffic loads, we always expect to see that parallel algorithms can yield lower mean call set up delays than sequential algorithms. This is also confirmed by the analytic results. On the other hand, when the processing capacity is abundant and the communication bandwidth is the bottleneck, we observe that parallel algorithms yield not only lower mean call setup delays but also higher throughputs. The reason why sequential OOC and crankback algorithms yield lower maximum achievable throughputs is due to overflow call requests from previously attempted paths.

3.4.4 The Effects of Admission Control Function

The preceding results have assumed $B(x) = (\frac{x}{C})^2$. In Figure 3.9 we study the effects of different forms of admission control by varying $B(x)$. Three forms of the admission control function are studied: a convex function ($B(x) = (\frac{x}{C})^2$), a linear function ($B(x) = \frac{x}{C}$), and a concave function ($B(x) = \sqrt{\frac{x}{C}}$).

By comparing the graphs in Figure 3.9, we observe the following interesting behavior. First, call setup time is very sensitive to the processing time requirements when the admission control function is convex. In other words, the processing capacity can easily become the performance bottleneck if the admission control function is a convex function. As a consequence, the parallel algorithms, which require much more processing capacity, perform poorly in Figure 3.9(b) because processing elements are the bottleneck of the network. On the other hand, in Figure 3.9(d), the processing

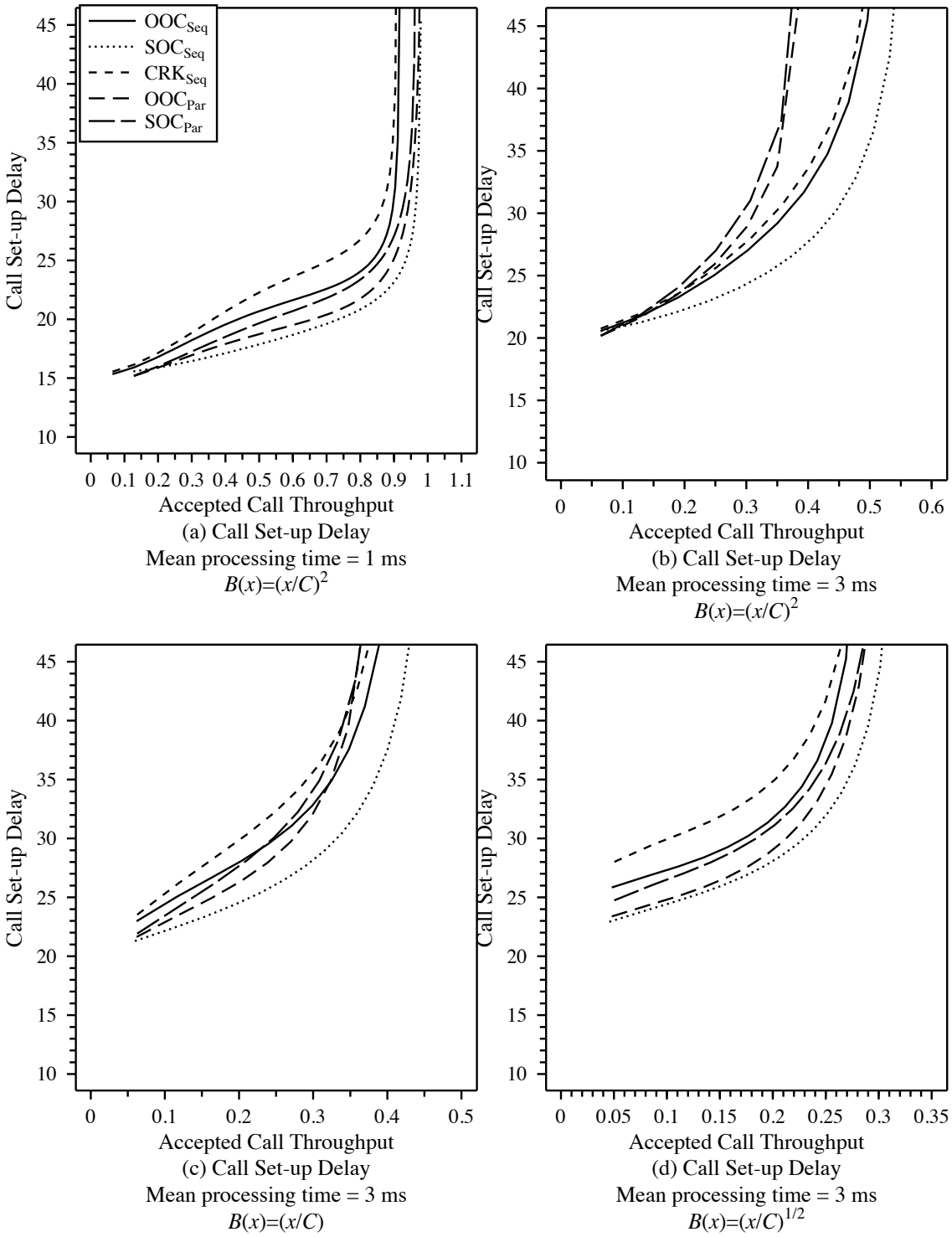


Figure 3.9 The effects of processing delay and admission control function.

capacity is abundant and the parallel algorithms yield both lower call setup delay and higher achievable maximum throughput.

Second, with a convex admission control function, all routing algorithms yield higher throughputs than with a concave or linear admission control function. This is because, for a given number of connections on a link, a concave admission control function and a linear admission control function block more calls than a convex admission control function does.

Third, the performance of the crankback routing algorithm becomes significantly worse than the other two algorithms when the admission control function changes from convex to concave. Again, this is because there are more alternate paths in the crankback algorithm than in the OOC algorithm. Thus, a call is given more chances to try longer paths in the crankback algorithm. As a result, the call blocking probability under the crankback routing algorithm also becomes significantly lower than under the other two algorithms.

3.5 Discussion

3.5.1 Modeling Resources Held by Blocked Calls

One of the differences between our work and previous works in the literature is that we explicitly model the resources held by blocked calls (the reader is referred to Figure 3.2). Whether it is important to model resources held by blocked calls depends on the ratio between the sum of call processing (service) time and propagation delay and the call holding time. When this ratio is larger than 0.01, we observe that ignoring the resources held by blocked calls introduces noticeable error in computing network performance measures such as blocking probability and call setup delay.

If the ratio of the sum of the call processing time and propagation delay and the call holding time is significantly small, we can ignore the modeling of resources held by blocked calls in our analysis. In this case, the computation of γ_i^f, δ_i^f, $\bar{N}_i$ and the first part of δ_i^s, which corresponds to the downstream processing delay and propagation delay, can be ignored in our analysis. Furthermore, since the call blocking model in Figure 3.2 is now independent of the call processing delay, we can also ignore the computation of node-offered traffic and nodal call processing delay in the

iterative algorithm (Algorithm A). After we obtain the link blocking probabilities from Algorithm A, we then can compute the processing delay at each node, average call setup delay and call blocking probability for each O-D pair in the same manner as in section 3.3.

3.5.2 The Effect of Propagation Delay

The effect of propagation delay on call setup delay and accepted call throughput has been studied in [50]. The influence of propagation delay is twofold. First, given the processing delays at the nodes of a path, increasing the propagation delay of this path will also increase the mean call setup delay by the same amount. This has a more significant effect on the sequential OOC and crankback algorithms because more time is required to crankback blocked call requests to the source node (under the sequential OOC algorithm) or upstream nodes (under the sequential crankback algorithm). Second, an increase of the propagation delay also means that the blocked calls hold their resources longer. This can have significant effect on the call blocking probability if the amount of blocked calls and their resource holding times are not negligible as compared to the amount of successful calls and normal call holding times. This influence is especially important for flooding algorithms because they generate more call requests and, consequentially, more blocked calls. However, these two influences have little effect on the call processing delay or the network blocking probability as long as the propagation delays are relatively small as compared to the call holding time. On the other hand, in the analysis of flooding schemes, modeling propagation delays is very important in determining the probability that a successfully received call request follows a particular path.

3.6 Summary

In this chapter, we examined the influence of network parameters such as routing algorithms, propagation delays, admission control functions, and processing capacities on the mean call setup time and call blocking probability. The influence of routing on the mean call setup time and call processing capacity was examined under five routing algorithms: three well-known algorithms in the circuit-switching literature and two

controlled flooding versions of these algorithms. The influence of call admission control was modeled and examined with three different forms of "call admission control function". The NSF T3 network was used in our examples as the underlying network topology. We developed analytical models for evaluating the average call setup time and call blocking probability for these five algorithms and validated our analysis by simulations. The results of our study indicate that for a given exogenous call arrival rate, there is a trade-off between call setup time and call blocking probability. The crankback routing algorithm yields the lowest call blocking probability but also the highest call setup delay. On the other hand, the SOC routing algorithm yields the highest call blocking probability but also the lowest call setup delay. As expected, flooding algorithms yield smaller call setup delays only when the call processing capacity is abundant. We also find that the form of the admission control function has a significant influence on the call setup time and the processing loads at processing elements.

The contribution of this work is to quantitatively study these important factors on call setup time and call blocking probability. In order to examine the influence of call processing delay and propagation delay, resources (includes processing resources and transmission resources) held by blocked calls during the call setup time were explicitly modeled; this has been ignored in all past research. We observe that it is important to model resources held by blocked calls only when the call processing (service) time or the propagation delay is not significantly small as compared to call holding time. This might be true for some applications in high-speed networks such as file transfer, or, if we do the call setup at the burst level for bursty sources.

In high-speed networks, multicast routing becomes very important because of the support of new applications such as multimedia, multiparty conferencing service. One way of setting up a multicast connection is to set up multiple point-to-point connections, each connects the source node to a destination node. Another way to set up a multicast connection is to build a tree-shaped route which connects the source node to all destionation nodes. No matter which approach is adopted, multicast routing requires much more processing capacity and yields a higher call setup delay than point-to-point routing. In our future work, we plan to extend our analytical

models to study the interaction of call processing delays and multicast routing in high-speed networks.

CHAPTER 4

MDP ROUTING IN HIGH-SPEED NETWORKS

The great similarity between routing in circuit-switched networks and high-speed networks has inspired us to examine and apply adaptive circuit-switched routing algorithms to high-speed networks. Due to advances in switching technologies, adaptive routing in circuit-switched networks has been of considerable interest since the early 80's [68]. We can classify current adaptive routing algorithms into two categories: Least-Loaded Path-based (LLP-based) and Markov Decision Process-based (MDP-based). The MDP approach has been shown to be more general and efficient than the LLP approach [68]. Thus the main focus of this chapter is to design and evaluate adaptive routing algorithms for future high-speed networks based on the MDP approach.

Two problems can be identified when we design MDP routing algorithms for future high-speed networks. First, the required computational complexity becomes extremely high when the network supports more than one class of traffic. For example, in [31], Dziong and Mason extend their previous work on traditional circuit-switched networks to multirate circuit-switched networks handling multiple classes of calls, each with its own bandwidth requirement. Although a statistical link independence assumption is made to reduce the complexity of the MDP model, the computational complexity still grows quickly as the number of classes of traffic increases. Thus we first focus on developing an approximation scheme toward MDP routing for multi-class networks which maintains low computational complexity while still providing quite accurate routing information. A rough approximation scheme is also proposed by Dziong *et al.* [30]. Our approximation scheme provides more accurate routing information than the scheme from [30] while requiring no additional computation. We also show that routing algorithms based on our approximate scheme yield competitive

performance as compared to the routing algorithms proposed in [31, 32] which requires considerably more computation.

The second problem with MDP routing in high-speed networks is that the link independence assumption, which is made in order to decouple link states and hence reduce the computational complexity, may not be valid in high-speed networks. Thus, it is questionable whether routing algorithms based on the link independence assumption can still yield good performance. In the second part of this chapter, we investigate the effect of the link independence assumption on the performance of MDP routing algorithms by analyzing a simple network. Our results show that even when the link independence assumption is invalid, routing algorithms based on this assumption can still yield near optimal performance.

The remainder of this chapter is structured as follows: in section 4.1, we design and evaluate computationally feasible MDP routing algorithms for high-speed networks. Our investigation of the effect of the link independence assumption on the performance of MDP routing algorithms is presented in section 4.2. Finally, section 4.3 summarizes this chapter and presents suggestions for future work.

4.1 Computationally Feasible MDP Routing in High-Speed Networks

In this section, we first summarize the formulation of the routing problem for multi-class networks as a Markov decision process. We then show how the link independence assumption can be used to decompose the resulting Markov decision process. In order to reduce the size of the state space of the Markov decision process, a simplified model for a link with multiple classes of traffic is introduced and, based on this simplified link model, a set of simple expressions for estimating the state-dependent expected revenue loss for adding a call, the so-called "*link shadow price*", is derived. Three MDP routing algorithms based on approximate link shadow prices are then proposed and evaluated. For performance comparison purposes, we also design and evaluate an LLP-based routing algorithm for high-speed networks. The performance comparison of these four routing algorithms is presented at the end of this section.

4.1.1 The MDP Approach Towards Routing

In this section, we first define our network model. We then formulate the routing problem in multi-class networks as a continuous-time Markov decision process [31].

4.1.1.1 Network Model

Consider a network consisting of a set of nodes, N, a set of links (trunk groups), L, and a set of all possible Origin-Destination (O-D) pairs, W. The network handles K classes of traffic labeled $k = 1, ..., K$. Assume that class k calls require b_k units of bandwidth and that they arrive at O-D pair w according to a Poisson process with rate λ_k^w. (Without loss of generality, we assume $1 = b_1 \leq b_2 \leq ... \leq b_{K-1} \leq b_K$.) Let $\lambda_k = \sum_{w \in W} \lambda_k^w$ be the total exogenous arrival rate of class k calls to the network. The call holding time for a class k call is assumed to be exponentially distributed with mean $1/\mu_k$ where $1/\mu_1 = 1$. Such a network, as described above, is referred to as a multirate network [23].

A multirate network can be used to describe both ISDN's and future B-ISDN's. First, when it is used to describe an ISDN network with Synchronous Transfer Mode (STM), a unit of bandwidth is equivalent to a time slot (or channel). Second, when it is used to describe a future B-ISDN with Asynchronous Transfer Mode (ATM), the bandwidth requirement of each class of calls is equivalent to the *effective bandwidth* of the call [33, 37, 42, 53]. For ease in explanation, we will refer to the unit of bandwidth as a "circuit", as in traditional circuit-switched networks, throughout this chapter.

When a call arrives to the network, it will either be carried on a path or lost, depending on the routing policy being used. We assume that each class k call of O-D pair w carried on the network produces r_k^w units of revenue. From another point of view, the network loses r_k^w units of revenue for each class k call of O-D pair w that is rejected. We assume that there is no additional cost associated with any of the actions of accepting a call, rejecting a call, or completing a call. The performance metric in which we are interested is the *fractional reward loss* [31, 32], which is defined as the ratio of the loss revenue to the expected offered revenue. More formally,

$$fractional\ reward\ loss = \frac{\sum_{w \in W} \sum_{k=1}^{K} r_k^w \lambda_k^w B_k^w}{\sum_{w \in W} \sum_{k=1}^{K} r_k^w \lambda_k^w}$$

where B_k^w is the blocking probability of class k traffic of O-D pair w.

We are interested in state-dependent routing policies, i.e., those policies that use network state information to make routing decisions. When a call arrives to the network, the routing policy assigns it to the most "efficient" path (based on some computation using current state information). Otherwise the call is rejected. The objective of a routing scheme is to minimize the fractional reward loss.

4.1.1.2 Problem Formulation

In this section, the multi-class routing problem is formulated as a Markov decision process. We assume that for each class k traffic of each O-D pair w, $w \in W$, there is a pre-defined set of possible paths, $\Phi_w^k = \{\phi_1^{w,k}, ..., \phi_{n_w^k}^{w,k}\}$. The state of the network is described by a $K \times J$ matrix $\mathbf{x} = [x_{k,j}]$ where $x_{k,j}$ denotes the number of class k calls carried on path j, and J is the total number of distinct paths in the network. Let $\mathbf{X}$ be the set of all feasible states, i.e.,

$$\mathbf{X} = \{\mathbf{x} | \sum_k \sum_{j \in \Psi_\ell} b_k x_{k,j} \leq C_\ell, \quad \forall \ell \in L\},$$

where C_ℓ is the link capacity of link ℓ and Ψ_ℓ is the set of paths which contain link ℓ.

When the network is in state $\mathbf{x} \in \mathbf{X}$ and a class k call arrives to O-D pair w, one of the following actions can be taken: reject the call or route it on path $j \in \Phi_w^k$ (without violating any capacity constraints). Let $A(\mathbf{x}, w, k)$ be the set of all possible actions when the network is in state $\mathbf{x}$ and a class k call of O-D pair w arrives. Thus,

$$A(\mathbf{x}, w, k) = \{0\} \cup \{j | \mathbf{x} + \delta_k^j \in \mathbf{X}, \ \forall j \in \Phi_w^k\}$$

where δ_k^j is a $K \times J$ matrix with 1 in the position of row k, column j and zeros in all other positions. Here $a = 0$, $a \in A(\mathbf{x}, w, k)$, corresponds to rejecting the new call, and $a = j$ corresponds to routing the call on path $j \in \Phi_w^k$.

For a given action $a \in A(\mathbf{x}, w, k)$, the *infinitesimal transition rate* from state $\mathbf{x}$ to state $\mathbf{y}$, $\mathbf{x}, \mathbf{y} \in \mathbf{X}, \mathbf{x} \neq \mathbf{y}$, is given by

$$Q_{\mathbf{x},\mathbf{y}}(a) = \begin{cases} \lambda_k^w & \mathbf{y} = \mathbf{x} + \Delta_k^w(\mathbf{x}, a) \\ x_{k,j}\mu_k & \mathbf{y} = \mathbf{x} - \delta_k^j \\ 0 & \text{otherwise.} \end{cases}$$

where $\Delta_k^w(\mathbf{x}, a)$ is defined to be the following $K \times J$ matrix:

$$\Delta_k^w(\mathbf{x}, a) = \begin{cases} \delta_k^j & \text{if } a = j > 0, \\ \mathbf{0} & \text{otherwise.} \end{cases}$$

where $\mathbf{0}$ is a $K \times J$ matrix of zeros.

We are interested in the class of non-randomized, stationary policies that are allowed to use state information, i.e., policies which take deterministic actions, that depend on the state but not the history of the system. Let Π be the set of all non-randomized, stationary policies. Define $\pi : \mathbf{X} \times W \times \{1, ..., K\} \rightarrow \{0, 1, ..., J\}$ where $\pi(\mathbf{x}, w, k)$ is the action taken by policy π for an incoming class k call of O-D pair w when the network is in state $\mathbf{x}$. If $\pi(\mathbf{x}, w, k) = 0$, the incoming call is rejected. Otherwise, it is routed to path j if $\pi(\mathbf{x}, w, k) = j$.

Let $q(\mathbf{x}, w, k, a)$, $a \in A(\mathbf{x}, w, k)$, be the revenue loss (cost) rate when the process is in state $\mathbf{x}$ and action a has been chosen for an incoming class k call of O-D pair w. Recall that the cost of rejecting a class k call of O-D pair w is r_k^w. Thus $q(\mathbf{x}, w, k, a)$ is given by:

$$q(\mathbf{x}, w, k, a) = \begin{cases} r_k^w \lambda_k^w & a = 0, \\ 0 & \text{otherwise.} \end{cases}$$

After defining the state transition matrix and revenue loss rate matrix, we are now able to define the expected revenue loss for a policy π in the interval of time $(0, t)$ with $\mathbf{x}$ as its initial state, $V_{\mathbf{x}}(t, \pi)$. If policy π is completely ergodic (i.e., the process contains only one recurrent chain), then $V_{\mathbf{x}}(t, \pi)$ is given by [46, 68]:

$$V_{\mathbf{x}}(t, \pi) = g(\pi)t + v(\mathbf{x}, \pi)$$

where $g(\pi)$ is called the loss rate of policy π [68] (also, the "gain" of the policy in [46]) and $v(\mathbf{x}, \pi)$ is the relative value of state $\mathbf{x}$ under policy π.

It can be shown that all stationary routing policies are completely ergodic policies. Thus there always exists an optimal policy in the class of stationary policies [46]. An

optimal policy, π^*, is the policy such that $V_{\mathbf{x}}(t,\pi^*) = \min_{\pi}\{V_{\mathbf{x}}(t,\pi)\}$ for all $\mathbf{x}$. That is, an optimal policy is the policy that minimizes the expected loss revenue in any time t with any possible initial state. In the following, we will show how an optimal policy can be obtained based on Markov decision theory.

An optimal policy can be found by a policy iteration procedure based on the following iteration cycle:

- **Value Determination Step:** for a given policy during the ith iteration, $\pi^{(i)}$, solve the following set of equations

$$\begin{aligned} g(\pi^{(i)}) &= \sum_k \sum_w q(\mathbf{x}, w, k, \pi^{(i)}(\mathbf{x}, w, k)) \\ &\quad + \sum_k \sum_w \lambda_k^w [v(\mathbf{x} + \Delta_k^w(\mathbf{x}, \pi^{(i)}(\mathbf{x}, w, k)), \pi^{(i)}) - v(\mathbf{x}, \pi^{(i)})] \\ &\quad + \sum_k \sum_j x_{k,j}\mu_k [v(\mathbf{x} - \delta_k^j, \pi^{(i)}) - v(\mathbf{x}, \pi^{(i)})], \quad \forall \mathbf{x} \in \mathbf{X}, \end{aligned} \tag{4.1}$$

  for all relative values $v(\mathbf{x}, \pi^{(i)})$ and policy loss rate $g(\pi^{(i)})$ by setting one of the relative values to zero.

- **Policy Improvement Step:** find the policy, which will be used for the next iteration, that minimizes the right hand side of equation (4.1) for each state $\mathbf{x}$ using the relative values obtained from the previous step. That is,

$$\pi^{(i+1)}(\mathbf{x}, w, k) = \arg \min_{a \in A(\mathbf{x},w,k)} \left\{ \begin{array}{l} \sum_k \sum_w \{q(\mathbf{x}, a) + \lambda_k^w [v(\mathbf{x}', \pi^{(i)}) - v(\mathbf{x}, \pi^{(i)})]\} \\ + \sum_k \sum_j x_{k,j}\mu_k [v(\mathbf{x} - \delta_k^j, \pi^{(i)}) - v(\mathbf{x}, \pi^{(i)})] \end{array} \right\}, \quad \forall \mathbf{x} \in \mathbf{X},$$

  where $\mathbf{x}' = \mathbf{x} + \Delta_k^w(\mathbf{x}, a)$. The resulting policy is called an improved policy [46].

To get more insight into the policy improvement step in the policy iteration cycle, consider the action taken by the improved policy when a class k call of O-D pair w arrives and the system is in state $\mathbf{x}$. Recall that Φ_w^k is the set of possible paths for

O-D pair w for a class k call. The possible actions that can be taken are: either reject the call or route it on path j, $j \in \Phi_w^k$. If

$$r_k^w \leq v(\mathbf{x} + \delta_k^j, \pi^{(i)}) - v(\mathbf{x}, \pi^{(i)}), \quad \forall j \in \Phi_w^k,$$

then the improved policy will reject the call when in state $\mathbf{x}$, i.e., $\pi^{(i+1)}(\mathbf{x}, w, k) = 0$. Similarly, if

$$\min_{j \in \Phi_w^k} (v(\mathbf{x} + \delta_k^j, \pi^{(i)}) - v(\mathbf{x}, \pi^{(i)})) \leq r_k^w,$$

and

$$j' = \arg \min_{j \in \Phi_w^k} (v(\mathbf{x} + \delta_k^j, \pi^{(i)}) - v(\mathbf{x}, \pi^{(i)})),$$

then the improved policy will always route the call to path j' when in state $\mathbf{x}$, i.e., $\pi^{(i+1)}(\mathbf{x}, w, k) = j'$. In the above, $v(\mathbf{x} + \delta_k^j, \pi^{(i)}) - v(\mathbf{x}, \pi^{(i)})$ can be viewed as the *path cost* of adding a class k call on path j given that the system is in state $\mathbf{x}$. An improved policy should always route a call on the path that has the minimum path cost and this cost must be less than the revenue that will be received by carrying that call. Otherwise, the call should be blocked.

4.1.2 Decomposition of the MDP

Clearly, solving the set of equations (4.1) is computationally infeasible for any realistic network due to the huge state space of the Markov decision process. Thus, numerous studies, e.g., [28, 68], have proposed a link independence assumption and a consequent decomposition of the path cost into a set of separable link costs, referred to as a path cost separability assumption. In the following, we first introduce these two assumptions. We then show the decomposed Markov decision process resulting from these two assumptions.

- **The link independence assumption:**

  This assumption assumes that ([28, 31, 56, 68]):

  1. calls from any class k arrive to any link ℓ according to independent Poisson processes with rate λ_k^ℓ.

2. a call carried on an n-link path behaves like n independent calls, i.e., the link holding times for the call on each of the n single links are statistically independent.

- **The path cost separability assumption:**

  This assumption is first made in [68]. The idea is to assume that the cost functions associated with adding a call of class k on path j to the network, $v(\mathbf{x}+\delta_k^j,\pi)-v(\mathbf{x},\pi)$, are *separable*, i.e.,

$$v(\mathbf{x}+\delta_k^j,\pi)-v(\mathbf{x},\pi) \;=\; \sum_{\ell\in\phi_j}[v_\ell(\mathbf{y}_\ell+\mathbf{e}_k,\pi)-v_\ell(\mathbf{y}_\ell,\pi)] \quad \forall\mathbf{x} \qquad (4.2)$$

  where $v_\ell(\mathbf{y}_\ell,\pi)$, which is to be defined later, is called the relative value of link ℓ, $\mathbf{e}_k$ is a vector of size K with a 1 in position k and zeros in all other positions and $\mathbf{y}_\ell$ is the state of link ℓ on path j corresponding to the network state $\mathbf{x}$. That is, given the network state $\mathbf{x}$, the behavior of each link $\ell\in L$ of the network can be described by a vector $\mathbf{y}_\ell=(y_{\ell,1},...,y_{\ell,K})$ where $y_{\ell,k}$ denotes the number of class k calls carried on link ℓ. To simplify equation (4.2), let

$$p_k^\ell(\mathbf{y}_\ell)=v_\ell(\mathbf{y}_\ell+\mathbf{e}_k,\pi)-v_\ell(\mathbf{y}_\ell,\pi).$$

  Equation (4.2) can then be rewritten as

$$v(\mathbf{x}+\delta_k^j,\pi)-v(\mathbf{x},\pi)=\sum_{\ell\in\phi_j}p_k^\ell(\mathbf{y}_\ell).$$

  Here $p_k^\ell(\mathbf{y}_\ell)$ is referred to as the state-dependent link shadow price [31], and can be viewed as the cost of adding a class k call on link ℓ when the link is in state $\mathbf{y}_\ell$.

The link independence assumption and the path cost separability assumption are satisfied if direct routing is used and the network is fully connected, i.e., each O-D pair is connected by a direct link. In direct routing, calls are only allowed to be routed via a direct link. Under the assumption that the arrival process of each O-D pair is statistically independent, each link is, therefore, also statistically independent. Now suppose we want to route a call on a path with more than one link, while still

restricting all other calls to be direct routed. The path cost for adding this call is thus simply the sum of individual link costs.

With these two assumptions, the Markov decision process can be decomposed as follows. Given the separability assumption, the path cost required in the policy improvement step of the policy iteration procedure can be obtained by solving for the state-dependent link shadow prices at each individual link on the path. By invoking the link independence assumption, we obtain the state-dependent individual link shadow prices by modeling each link as a Markov process.

Unfortunately the decision of whether to route a call on a path with multiple links may depend on the states of all links on the path for an arbitrary policy π. Thus, for a given policy π, we cannot form a Markov process for this policy on the basis of a single link state because the state transitions depend on the states of other links. A simple approach is to execute the policy iteration only once and assume that the initial policy is a direct routing policy regardless of the routing policy used. With this assumption, we obtain the link shadow prices by modeling each link as a Markov process with a cost as described below.

Let π_d be a direct routing policy. When the link is in state $\mathbf{y}_\ell$ and a class k call arrives, policy π_d either accepts or rejects it. Define $\pi_d^\ell : \mathbf{Y}_\ell \times \{1, ..., K\} \rightarrow$ {accept,reject} where $\mathbf{Y}_\ell$ is the set of all feasible states of link ℓ, i.e.,

$$\mathbf{Y}_\ell = \{\mathbf{y}_\ell | \sum_k b_k y_{\ell,k} \leq C_\ell\},$$

and $\pi_d^\ell(\mathbf{y}_\ell, k)$ is the action taken by policy π_d for an incoming class k call when the link is in state $\mathbf{y}_\ell$. Since policy π_d rejects a link only if there are not enough free circuits available, $\pi_d^\ell(\mathbf{y}_\ell, k)$ is given by:

$$\pi_d^\ell(\mathbf{y}_\ell, k) = \begin{cases} \text{accept} & \text{if } \mathbf{y}_\ell + \mathbf{e}_k \in \mathbf{Y}_\ell, \\ \text{reject} & \text{otherwise.} \end{cases}$$

Assume the link arrival rates, λ_k^ℓ, are given. The link revenue loss rate, $q_\ell(\mathbf{y}_\ell, k, a)$, $a \in$ {accept,reject}, when the link is in state $\mathbf{y}_\ell$ and action a has been chosen for an incoming class k call is given by:

$$q_\ell(\mathbf{y}_\ell, k, a) = \begin{cases} r_k^\ell \lambda_k^\ell & a = \text{reject}, \\ 0 & \text{otherwise}, \end{cases}$$

where r_k^ℓ is the expected revenue loss for rejecting a class-k call on this link (which is referred to as *link loss*). The link loss will be defined differently for the different

routing schemes examined in subsequent sections; we thus defer the formal definition of the link loss to the next section.

The relative values, $v_\ell(\mathbf{y}_\ell, \pi_d)$, of link ℓ under policy π_d are obtained by solving the set of equations in the value determination step:

$$\begin{aligned} g_\ell(\pi_d) &= \sum_k q_\ell(\mathbf{y}_\ell, k, \pi_d^\ell(\mathbf{y}_\ell, k)) + \sum_k \lambda_k^\ell [v_\ell(\mathbf{y}_\ell + \mathbf{e}_k(\mathbf{y}_\ell, \pi_d), \pi_d) - v_\ell(\mathbf{y}_\ell, \pi_d)] \\ &\quad + \sum_k y_{\ell,k}\mu_k [v_\ell(\mathbf{y}_\ell - \mathbf{e}_k, \pi_d) - v_\ell(\mathbf{y}_\ell, \pi_d)], \quad \forall \mathbf{y}_\ell \end{aligned} \tag{4.3}$$

where $\mathbf{e}_k(\mathbf{y}_\ell, \pi_d) = \mathbf{e}_k$ if policy π_d accepts a class k arrival given that the link is in state $\mathbf{y}_\ell$. Otherwise, $\mathbf{e}_k(\mathbf{y}_\ell, \pi)$ is a zero vector.

Thus link shadow prices can be obtained by solving the set of equations (4.3) once the traffic load offered to each link is known. In [68], the values of the link offered traffic correspond to the traffic assignments of the optimal non-alternate routing policy. It may be more desirable to estimate the actual traffic load during network operation rather than rely on such a *priori* traffic estimation. Thus later studies, such as [28, 31, 56], all adopted the approach in which the traffic loads of individual links are measured on-line and state-dependent link shadow prices are updated periodically.

4.1.3 Link Model Simplification

The goal of simplifying the link model is to derive a set of simple expressions for computing the state-dependent *link shadow prices*. When we formulate a Markov process for a single link, the state space can easily contain tens of thousands of states. For example, the state space for a link with 200 circuits and two classes of traffic that each requires 1 or 2 circuits at a time, respectively, contains 10201 states! Thus, it is not practical to obtain the set of state-dependent link shadow prices by solving the set of linear equations in the value determination step, as in [31]. Our approach will thus be to develop an approximation to the link state dynamics that yields a birth-death process. This will allow us, as in [68], to derive a set of simple expressions for computing link shadow prices. In this section, we will first describe the simplified link model and then show how a Markov decision process can be formed based on this model.

4.1.3.1 A Simplified Link Model

Wilkinson [82] showed that the cumulative distribution function (CDF) of the link occupancy of a link with non-Poisson arrivals can be very accurately approximated by a negative binomial distribution when the link blocking probability is higher than 0.001. Recently, Chung and Ross [23] have shown that such an approximation can also be applied to multirate loss networks, and that for the purpose of computing blocking probabilities, the approximate link steady state distribution (a negative binomial distribution) can be approximated well by the steady state distribution of a birth-death process. Our approach is to form a Markov decision process based on this birth-death process. In the following, we summarize how a link with multiple classes of traffic can be approximated as a birth-death process using this technique, [23]. A discussion of how this approximation can be used to derive a set of simple expressions for computing the state-dependent link shadow prices is then presented in the next section.

Let us describe the state of a link by the number of busy circuits, i. Given the link independence assumption of the previous section, the CDF of the link occupancy of each network link will be approximated by the following birth-death process. Let $\rho_k^\ell = \lambda_k^\ell/\mu_k$ be the offered load for class-k calls to this link. Then the birth-death process has state space 0,...,C where C is the capacity of the link. When in state i, the process has a death rate of i and a birth rate of $\xi^2/\sigma^2 + i(1 - \xi/\sigma^2)$, where $\xi = \sum_{k=1}^K b_k\rho_k$ and $\sigma^2 = \sum_{k=1}^K b_k^2\rho_k$. The numerical results in [23] show that the blocking probability of the link can be accurately approximated by the steady state distributions of this birth-death process under various traffic conditions, especially under heavy traffic conditions.

4.1.3.2 Markov Decision Process with the Simplified Link Model

In this section, we derive a set of simple expressions for computing the "approximate" state-dependent link shadow prices by forming a Markov process with cost using the simplified (birth-death) link model. Let us focus on a single link, ℓ. Let $\eta_k = r_k^\ell\lambda_k^\ell$. For convenience, we will omit the subscript (or superscript) ℓ in our notation. (e.g., the expected revenue loss rate is denoted by g, instead of g_ℓ.)

Assume the initial policy is the direct routing policy discussed at the end of the previous section. Recall that the CDF of the link occupancy can be approximated by a birth-death process with birth rate $\bar{\lambda}_i = \xi^2/\sigma^2 + i(1 - \xi/\sigma^2)$. The set of equations in the value determination step (i.e., equation (4.3)) is given by:

$$\begin{aligned} g &= 0 + \bar{\lambda}_0(v(1) - v(0)), \\ g &= 0 + \bar{\lambda}_i(v(i+1) - v(i)) - i(v(i) - v(i-1)), \quad 0 < i \leq C-1, \\ g &= \textstyle\sum_{j=1}^{K} \eta_j - C(v(C) - v(C-1)). \end{aligned}$$

The state-dependent link shadow prices, $p_k(i)$, are then computed as

$$p_k(i) = (v(i + b_k) - v(i))/\mu_k$$

where $v(i) - v(i-1)$ is obtained by solving the above $C+1$ equations.

Because of the special structure of the set of linear equations, the values of $v(i) - v(i-1), 1 \leq i \leq C$, can be computed by the following set of simple expressions.

$$v(C) - v(C-1) = \frac{\sum_{j=1}^{K} \eta_j}{\bar{\lambda}_{C-1}} \frac{E(\bar{\lambda}, C)}{E(\bar{\lambda}, C-1)}, \tag{4.4}$$

$$v(i) - v(i-1) = \frac{g}{\bar{\lambda}_{i-1} E(\bar{\lambda}, i-1)}, \quad 1 \leq i < C, \tag{4.5}$$

where

$$E(\bar{\lambda}, i) = \frac{\frac{1}{i!} \prod_{j=0}^{i-1} \bar{\lambda}_j}{\sum_{n=0}^{i} \frac{1}{n!} \prod_{j=0}^{n-1} \bar{\lambda}_j}, \tag{4.6}$$

and

$$g = \sum_{j=1}^{K} \eta_j - C(v(C) - v(C-1)). \tag{4.7}$$

In order to compute the link shadow prices, we need on-line measurements of the link offered loads, $\{\lambda_k^\ell\}$ [31, 56]. We will discuss in detail how they are estimated at the time that we describe our policies in the next section.

4.1.3.3 Compensation Parameters

From our numerical experiments, as shown in section 4.1.5, we have found that the approximate link shadow prices are very accurate when b/C is sufficiently small

(where $b = max_k(b_k)$). However, when b/C is not small enough, we need to introduce some class-dependent compensation parameters, α_k, and modify equations (4.4), (4.5) as follows,

$$\begin{aligned}
v(C) - v(C-1) &= \frac{\sum_{j=1}^{K} \eta_j}{\bar{\lambda}_{C-1}} \frac{E(\bar{\lambda}, C)}{E(\bar{\lambda}, C-1)} - \frac{\sum_{j=2}^{K} \alpha_j \eta_j}{\bar{\lambda}_{C-1}} \frac{E(\bar{\lambda}, C)}{E(\bar{\lambda}, C-1)} (1 + E(\bar{\lambda}, C-1)), \\
v(i) - v(i-1) &= \frac{g}{\bar{\lambda}_{i-1} E(\bar{\lambda}, i-1)} - \frac{\sum_{j=k}^{K} \alpha_j \eta j}{\bar{\lambda}_{i-1}}, \\
& \qquad C - b_k + 2 \leq i < C - b_{k-1} + 2 \quad 2 \leq k \leq K, \\
v(i) - v(i-1) &= \frac{g}{\bar{\lambda}_{i-1} E(\bar{\lambda}, i-1)}, \quad 1 \leq i < C - b_K + 2,
\end{aligned}$$

where $E(\bar{\lambda}, i)$ and g are given in equations (4.6) and (4.7).

We have observed that when b/C is sufficiently small, α_k can be set to 0 for all k. However, there are several reasons for setting $\alpha_k > 0$ for $k > 1$ (α_1 is always 0). First, when b/C is not small enough, the negative binomial distribution does not match the CDF of the link occupancy very well. Second, when mixing two or more classes of traffic, the approximate model cannot capture the fact that in the exact model there will be no class k arrivals when the link occupancy is less than b_k. A complete discussion of how values of the α_k's are chosen is deferred to section 4.1.5.

4.1.4 Routing Policies

Four routing policies will be studied in this chapter. The first three policies are adaptive state-dependent routing algorithms based on Markov decision theory and use the approximate state-dependent link shadow prices obtained in the previous section. They differ in how link rewards and path costs are assigned. These three routing policies share the following algorithm:

- **Initial step:**

  Use direct, non-alternate routing as the initial policy. Assign link-loads $\{\lambda_k^\ell\}$ (where λ_k^ℓ is the offered load of class-k call on link ℓ) accordingly.

- **Loop**

  1. At regular intervals of length Δ_t, use link-measurements (which may depend on the policies) to re-estimate the link-offered loads, λ_k^ℓ.

2. Use the new estimated link-offered loads, λ_k^ℓ, from the previous step to compute new values of the link shadow prices. The routing policy then routes any new subsequent call according to the new set of link shadow prices. The method by which link shadow prices are used to select the routing path, or reject the call, is policy dependent.

Clearly, the value of Δ_t can affect the performance of routing policies. In our numerical examples, Δ_t is set to twenty time units.

4.1.4.1 The Priority ASDR Policy

Our ASDR scheme is a modified adaptive version of the SDR scheme proposed in [56]. The following is a detailed description of priority ASDR.

Call classification: In priority ASDR, calls are classified according to their bandwidth requirements. (For simplicity, we assume all calls with the same bandwidth requirements have the same mean holding time. This assumption can be easily relaxed.) That is, calls are classified into K classes with a class k call requiring b_k circuits at a time and $1 = b_1 < b_2 < ... < b_{K-1} < b_K$. Note that as in the original ASDR, we do not distinguish between direct calls and alternate calls (calls that are carried on multi-link paths).

Link loss: Since calls are all treated as direct calls, the link loss is set to the expected revenue loss of rejecting a call. That is, $r_k^\ell = r_k$.

Link-loads $\{\lambda_k^\ell\}$: For each link, ℓ, the following statistics are measured for each class of calls, $1 \leq k \leq K$.

- carried load $\tilde{\lambda}_k^\ell$: average number of calls carried per unit of time. The hatted notation, $\tilde{}$, will be used to indicate the carried, as opposed to offered, load.
- blocking probability B_k^ℓ: the fraction of time that the link is in "blocking" states. Blocking states for class k are those states with less than b_k free circuits.

The offered load of class k call on link ℓ, λ_k^ℓ, is then estimated as

$$\lambda_k^\ell = \frac{\tilde{\lambda}_k^\ell}{1 - B_k^\ell}.$$

Recall that the set of offered loads, $\{\lambda_k^\ell\}$, is used in computing the link shadow prices, as described in section 4.1.3.2.

Path cost: The cost of adding a class k call on a path with links $j_1, ..., j_m$, in the respective states $i_1, ..., i_m$, is given by [68]:

$$path\_cost \;=\; \sum_{n=1}^{m} p_k^{j_n}(i_n). \tag{4.8}$$

Path selection: The priority ASDR policy adopts a modified path selection method which assigns higher priority to paths with shorter length. In other words, paths with shorter path length are tried before paths with longer path length, where path length is defined as the number of links on the path. For paths with the same path length, only the path with the minimum path cost is tried. A call is routed to a path if the path cost is less than the call reward. If path costs of all candidate paths equal or exceed the call reward, the call is rejected.

4.1.4.2 The Priority MDPD Policy and the MDPD' Policy

The priority MDPD and MDPD' routing policies were introduced and studied in [32]. In this chapter, we study simplified versions of these two policies, in which (1) the state-dependent link shadow prices are approximated by the expressions obtained in the previous section, and (2) certain on-line link-measurements are simplified. A detailed description follows.

Call classification: Both policies classify calls in the same manner as priority ASDR.

Link loss: When a call is carried on a multi-link path, it has been suggested in [28] that the reward of the call be divided into link rewards, one for each of the links on the path carrying the call. This idea can also be applied to the case of dividing the revenue loss of rejecting a multi-link call into link losses. Four division rules

are studied in [31] and have been shown to exhibit comparable performances. However, our numerical results show that this, in general, is not true. In our study, we implement two division rules. Under rule 1, which is referred to as D1, the call reward (loss) is evenly divided among the links on the path. Under rule 2, which is referred to as D2, the call reward (loss) is distributed among the links on the path in proportion to the average link shadow prices. That is, the link reward loss of link ℓ for rejecting a class k call of O-D pair w on path j is given by

$$r^{\ell}_{k,\phi_j} = r^{w}_{k} \frac{\bar{p}^{\ell}_{k}}{\sum_{n \in \phi_j} \bar{p}^{n}_{k}}.$$

Because of the decomposition of the call revenue loss, calls carried on a link with the same bandwidth requirement and holding time may have different link reward losses. When calls with the same bandwidth requirement and holding time but different losses are aggregated into one class (as will be done in our algorithms, since calls are classified only by their bandwidth requirement), a modified link loss for this new class of call is computed as follows.

Let us focus on a single link ℓ. Assume Γ is the set of traffic classes that are to be aggregated. For a given policy, let r^{ℓ}_{k} be the link loss of a class k call and $\lambda^{\ell}_{k,\phi_j}$ be the measured offered load which will be specified later, $k \in \Gamma$. The modified link loss for the aggregated class of traffic, r^{ℓ}_{Γ}, is given by [31]:

$$r^{\ell}_{\Gamma} = \frac{\sum_{(k,\phi_j) \in \Gamma} r^{\ell}_{k,\phi_j} \lambda^{\ell}_{k,\phi_j}}{\sum_{i \in \Gamma} \lambda^{\ell}_{k,\phi_j}}. \tag{4.9}$$

Then r^{ℓ}_{Γ} is used as the expected revenue loss for rejecting a call belonging to this aggregated class in our model.

Link loads: Both policies estimate link-offered loads using the same procedure as priority ASDR. However, in order to be able to decompose the multi-link call loss into individual link losses, MDPD policies need to measure, for each class of traffic on each link, the carried load of all paths that traverse this link. That is, for each link $\ell \in L$, both policies measure the carried load of each class of calls

on each path $\phi_j \in \Psi_\ell$, $\tilde{\lambda}^\ell_{k,\phi_j}$. The offered load of class k on path ϕ_j observed by link ℓ, λ^ℓ_{k,ϕ_j}, is then estimated by

$$\lambda^\ell_{k,\phi_j} = \frac{\tilde{\lambda}^\ell_{k,\phi_j}}{1 - B^\ell_k},$$

where B^ℓ_k is the measured blocking probability of class k calls on link ℓ. The estimated offered loads, λ^ℓ_{k,ϕ_j}, are then used in equation (4.9) for computing the link loss.

Path cost: The method of computing path costs in priority MDPD is the same as in ASDR. Notice that *path net-gain* defined in [31] is equivalent to $(r_k - path\_cost)$.

Consider now the MDPD' policy. It was demonstrated in [32] that priority should not be always given to direct links. Instead, [32] proposes a method of modifying the path net-gain to improve the performance of a routing scheme without giving priority to direct calls. The idea is to increase the probability of choosing paths with a higher derivative of the average network reward with respect to the rate of calls carried on the path. The path cost is thus modified as follows [32]:

$$path\_cost' = \sum_{n=1}^{m} [(1-\beta) p_k^{j_n}(i_n) + \beta \bar{p}_k^{j_n}],$$

where $\bar{p}_k^{j_n}$ is the average link shadow price measured on-line from the previous policy iteration cycle. Unfortunately, there is no systematic methodology for choosing the optimal value of β. From numerous of the tested cases, the authors of [32] have found that the optimal value of β falls in the interval $[0.4, 0.7]$ and the performance of the MDPD' routing policy is quite insensitive to the value of β given that $\beta \in [0.4, 0.7]$. In our numerical examples, we have found that the MDPD' policy yields very good performance by setting $\beta = 0.3$. (Note that the optimal value of β in our study is different from [32] because of some different approaches (e.g., link-offered load measures, the use of the simplified link model, ..., etc.).)

Path selection: Priority MDPD uses the same method of path selection as priority ASDR. MDPD' uses a similar algorithm except that no priority is given to direct links.

4.1.4.3 The LLP policy

The last policy we study is based on the least-loaded path (LLP) routing algorithm [40]. LLP has been shown to be very efficient [6, 16], particularly when coupled with an appropriate trunk reservation policy [4]. Under LLP with a trunk reservation policy, a call is first offered to the direct link. If it is blocked, then the path with the maximum number of path circuits is tried, where the path circuit of a path is defined as the minimum number of free circuits over the path's links. The number of free circuits of a link, in turn, is defined as $C_s - x_s - t_s$ where C_s is link capacity, x_s is the number of busy circuits, and t_s is the trunk reservation level.

We propose a new LLP policy for multi-class networks with general topology. For each source-destination pair, we define an ordered set of candidate paths in which paths are ranked in increasing order of path length for each class of traffic. The path length of a path is defined as the number of links on the path. If more than one path has the same length, priority is given to the path with the largest number of path circuits. As in the original LLP, the path circuit of a path is defined as the minimum number of free circuits over the path's links. The number of free circuits of a link of class k traffic is defined by:

$$free\_circuits_s = \begin{cases} C_s - x_s & \text{if the path is one of the shortest paths} \\ & \text{of this O-D pair,} \\ C_s - x_s - t_s^k & \text{otherwise,} \end{cases}$$

where t_s^k is the trunk reservation level for class k traffic.

The problem of choosing the trunk reservation level on each link for each class of traffic in order to yield optimal performance is beyond the scope of this study. We have, however, tried to tune the the trunk reservation levels on each link to yield the best performance in our simulation results.

4.1.5 Numerical Results

In our numerical results, we first study the accuracy of our approximate link shadow prices obtained using the simplified link model. We then study the performance of routing algorithms based on our approximate link shadow prices by comparing their performance with the performance of routing algorithms based on exact link

shadow prices presented in [32]. Finally, we study the application of our four routing algorithms on the NSF T3 backbone network.

4.1.5.1 Comparison of the Approximate Link Shadow Prices with Exact Results

We first examine the accuracy of our approximate link shadow prices for a link with one class of traffic which requires more than one circuit at a time. The results are shown in Figure 4.1. The exact values are obtained by solving the system in which the bandwidth requirement of each call is normalized to unity (i.e., $C' = 20, b' = 1$ in (a) and $C' = 60, b' = 1$ in (b)). From Figure 4.1, we can see that the approximate state-dependent link shadow prices match the exact values very well when b/C is sufficiently small (e.g., 0.05).

Next, we study a link with more than one class of traffic. We first examine the case where all classes of traffic have the same mean call holding time. By setting $\alpha_2 = 0.1$ and $\alpha_3 = 0.15$, as we can see in Figure 4.2, the approximate state-dependent link shadow prices still match the exact values very well. Notice that a state in our simplified model corresponds to a set of states in the exact model. In Figure 4.2, the minimum and the maximum of the state-dependent link shadow prices of the corresponding set of states in the exact model are plotted.

The need for class-dependent compensation parameters, α_k's, in computing link shadow prices can be seen in Figure 4.2(a) and (c). (Note that the approximate link shadow prices obtained without α_k's are also plotted for the purpose of comparison.) Let us consider the case of Figure 4.2(a). As we can see, the cost of adding a class 1 call is more expensive when the link state is 95 (i.e., $C - b_2$) than when the link state is 96. Intuitively, this is because when the link state is $C - b_2$, the cost for accepting a class 1 call includes both blocking future class 1 calls as well as class 2 calls. However, the cost for accepting a class 1 call when the link state is $C - b_2 + 1$ includes only blocking future class 1 calls because a class 2 call will be blocked regardless of whether a class 1 call is accepted at this state. The simplified link model cannot capture such a phenomenon because all classes of calls have been modeled as requiring a single circuit. The parameters α_k's attempt to compensate for this deficiency. Note that this phenomenon occurs only in link state i, $C - b_2 \leq i \leq C$. Equivalently, the use

of these parameters also only affects the shadow prices for those states. From our experiments, we have found that the choices of α_i's are very sensitive to the number of classes present in the network and are less sensitive to the mean or variance of the traffic. We also found that the approximated link shadow prices match the exact values very well under various configurations by setting $\alpha_2 = 0.1$ in the case of $K = 2$ and $\alpha_2 = 0.1, \alpha_3 = 0.15$ in the case of $K = 3$.

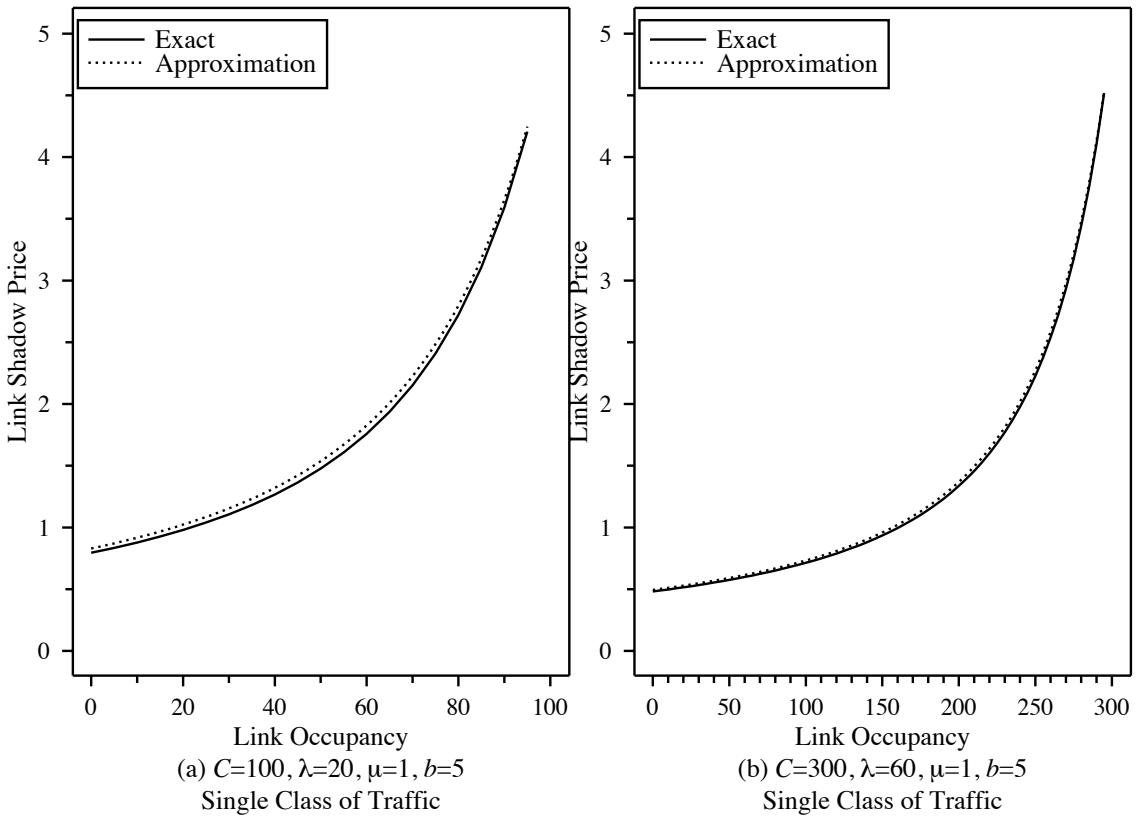


Figure 4.1 Applying Markov decision process to the simplified link model: single class of traffic.

In applying Markov decision theory to the approximate model, we have implicitly made the assumption that the exact state-dependent link shadow prices of states in the exact model that result in the same overall link occupancy (even though the numbers of calls from different classes are different) should be roughly the same. This

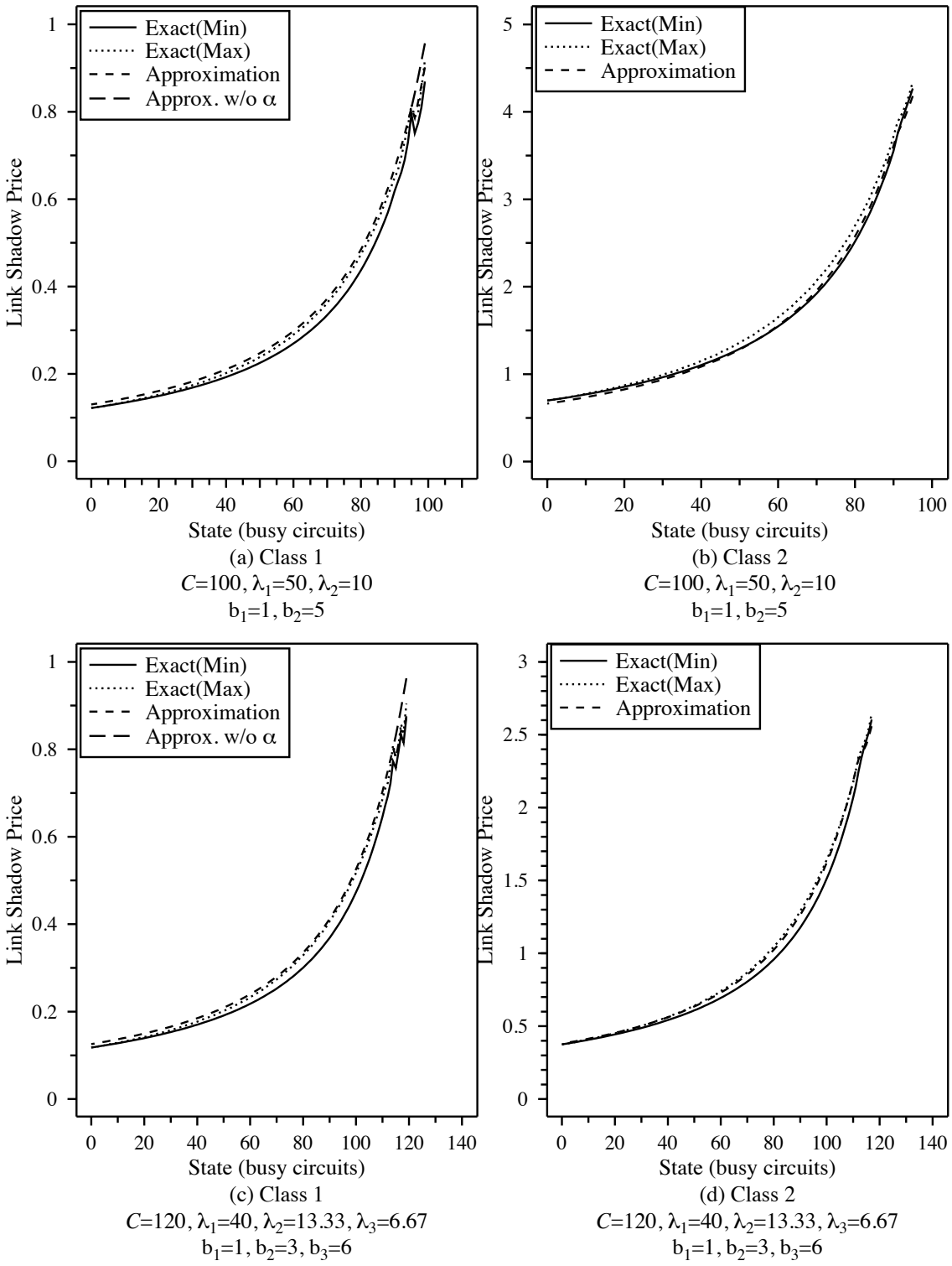


Figure 4.2 Applying Markov decision process to the simplified link model: multiple classes of traffic.

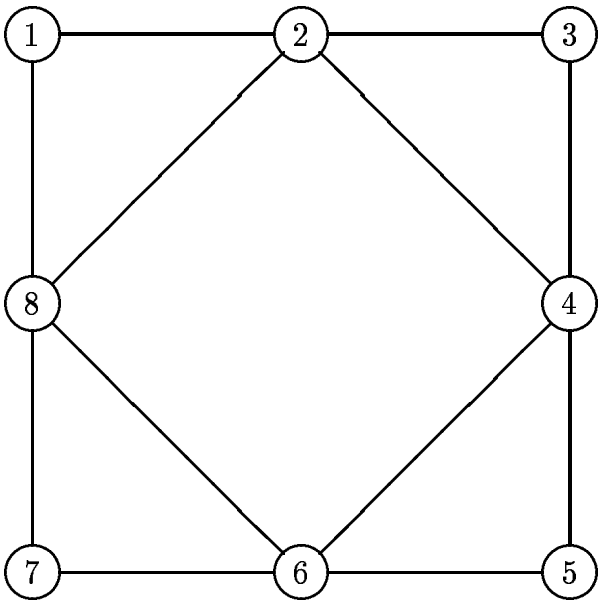
is validated in our numerical results for the case where all classes of calls have the same mean call holding time and the ratio of b/C is small enough.

When we extend the approximation model to the general case in which each class of traffic has a different mean call holding time, we observe that this implicit assumption becomes questionable. That is, for a given state in the simplified model, the link shadow prices of the corresponding states in the exact model can be quite different. This is because, when the mean holding time of a wide-band call is much longer than that of a narrow-band call, the expected revenue loss (i.e., link shadow price) of a state with a wide band call will be much higher than the loss of a state with equivalent number of narrow band calls. However, we are still interested in investigating how a routing policy will perform even when shadow price information is inaccurate.

4.1.5.2 Performance of Routing Algorithms Using Approximate Link Shadow Prices

In [32], Dziong and Mason study a ill-dimensioned network, W8N (Table 4.1), in order to demonstrate that priority MDPD and LLP can perform poorly while the modified MDPD scheme, MDPD', still yields excellent performance. (The traffic loads in Table 4.1 are set such that the MDPD' policy yields a fractional reward loss of approximately 1%, which is referred to as the nominal traffic load, and a fractional reward loss of approximately 5%, which is referred to as the overloaded condition.) In this section, we study the performance of our priority MDPD, LLP and modified MDPD schemes on the same network with the same traffic loads. (Note that for priority MDPD and modified MDPD policies, we have studied two reward decomposition rules, D1 and D2. These routing policies are denoted as MDPD(D1), MDPD'(D1), MDPD(D2), and MDPD'(D2) respectively.) The purpose of this study is to demonstrate how performance degrades when less accurate link shadow prices are used. Table 4.2 compares the performance of our routing policies with Dziong and Mason's results. In Table 4.2, each point estimate is obtained from 10 to 20 independent runs, depending on when the width of the 95% confidence interval becomes sufficiently small, i.e., the width of the confidence interval is less than 10% of the value of the point estimate under the nominal traffic load and 5% the value of

Table 4.1 W8N topology and traffic configurations: link cap. $= 120$, $b_2 = 12$, $\mu_2 = 0.1$.



set of O-D pairs	nominal		overload	
	λ_1	λ_2	λ_1	λ_2
A	4.2	0.021	5.1	0.0255
B	56.0	0.224	68.0	0.2720

set	O-D pairs
A	(1,2), (2,3), (3,4), (4,5), (5,6), (6,7), (7,8), (1,8)
B	(2,4), (4,6), (6,8), (2,8), (2,6), (4,8)

Table 4.2 Comparison of fractional reward losses (%) of policies studied in this dissertation with policies by Dziong, *et al.*

Traffic load	Policies studied		
	ASDR	MDPD(D1)	MDPD'(D1)
nominal	19.91 ± 0.38	18.37 ± 0.28	0.95 ± 0.07
overload	25.82 ± 0.13	25.78 ± 0.24	5.40 ± 0.17
Traffic load	Policies studied		
	LLP	MDPD(D2)	MDPD'(D2)
nominal	10.13 ± 0.08	18.54 ± 0.23	0.89 ± 0.09
overload	12.98 ± 0.15	26.06 ± 0.36	5.21 ± 0.24
Traffic load	Policies by Dziong, *et al*		
	LLP	MDPD(D2)	MDPD'(D2)
nominal	12.81 ± 0.20	25.81 ± 0.37	0.70 ± 0.14
overload	18.43 ± 0.23	33.87 ± 0.25	4.92 ± 0.29

the point estimate under the overloaded condition. Each run is run for 5000 units of call holding time. For each run, the initial 10% of the samples were discarded. 95% confidence intervals for simulation results are indicated in the table.

Our results agree with those in [32]; priority MDP and LLP perform poorly whereas MDPD' performs well. It is very interesting to observe that our priority MDPD yields even better performance than the one presented in [32]. On the other hand, we do see some performance degradation when we compare the two modified MDPD schemes. Fortunately, such degradation is not very significant. Considering the decrease in complexity, we thus believe that this makes our approach towards computing approximate link shadow prices quite useful. Furthermore, as the link capacity increases, the approximated link shadow prices will become more accurate and we would expect the degradation to be reduced.

A rough approximation scheme is also proposed in [30]. The advantage of our approximation scheme is that our scheme provides more accurate link shadow prices than this other scheme while requiring no additional computation. As a result, routing algorithms using our approximation scheme are, in general, able to yield better performance. For example, we simulated the MDPD'(D2) policy using Dziong *et. al.*'s approximation scheme on the W8N network with the same traffic loads shown in Table 4.1. Our results show that the MDPD'(D2) policy yields a fractional reward

Table 4.3 Total number of packets in and out of each Core Nodal Switching Subsystem (CNSS) in November 1992.

CNSS	Packets	CNSS	Packets
Hartford	3,073,648,060	New York	763,223,725
Washington	3,065,093,082	Greensboro	1,196,883,668
Cleveland	2,592,664,056	Chicago	1,095,185,554
St. Louis	1,620,282,595	Houston	892,957,594
Denver	1,381,641,510	Seattle	1,464,635,724
San Franciso	3,009,969,812	Los Angeles	812,279,913

loss of 1.41% with a 95% confidence interval of $(1.33\%, 1.49\%)$ under the nominal traffic condition and a fractional reward loss of 5.93% with a 95% confidence interval of $(5.68\%, 6.18\%)$ in the overloaded traffic condition. By comparing these results with the results shown in Table 4.2, we observe that the MDPD'(D2) policy yields much better performance using our approximation scheme.

4.1.5.3 Routing in NSF T3 Backbone Network

In this section, we study the performance of the four routing policies on a more realistic network, the NSFNET T3 backbone network. The NSFNET T3 backbone network as of July, 1992 is shown in Figure 3.6. The assumptions made in chapter 3 are also adopted here. In addition, we assume two classes of traffic are presented to the network. The first class of traffic requires 64Kbps bandwidth with unit mean call holding time (i.e., $\mu_1 = 1$) [1]. The second class of traffic requires 768Kbps bandwidth with mean call holding time, μ_2, as a simulation parameter [2]. By normalizing the bandwidth requirements to class 1 traffic, we have $b_1 = 1, b_2 = 12$ and the link capacity $C = 703$.

Case study 1:

In this numerical example, the exogenous call arrival rate for each source/destination pair is set in proportion to the traffic statistics reported for NSFNET in November

[1] Consider this class of calls as voice calls using the G.711 coding protocol.

[2] Consider the second class of traffic as video sources using the H.261 protocol with $p = 12$ [11].

1992 [60]. Specifically, Table 4.3 shows the total number of packets in and out of each CNSS. We compute the call arrival rate to each source/destination pair from this table in the following manner. Let A_i be the number of packets in and out of CNSS i in Table 4.3. We assume that the call arrival rate to CNSS i is proportional to A_i and the fraction destined to CNSS j is proportional to A_j. Hence, the call arrival rate of traffic from CNSS i to CNSS j is

$$\lambda_{i,j} = \lambda \frac{A_i}{\sum_k A_k} \frac{A_j}{\sum_{k \neq i} A_i},$$

where λ is the exogenous call arrival rate to the network. We also assume that 75% of the traffic is class 1 traffic and that the mean call holding time of class 2 calls is 0.1 (i.e., $\mu_2 = 0.1$).

We study two ways of assigning call rewards. The first reward-assignment rule assumes that each class k call carried on the network brings $r_k^w = b_k/\mu_k$ units of revenue to the network, regardless of how far the source node is away from the destination node. In the second reward-assignment rule, we assume the call reward depends not only on the bandwidth requirement but also on the distance between the source and destination node. That is, the revenue brought by a class k call of O-D pair w is given by

$$r_k^w = d_w \frac{b_k}{\mu_k},$$

where d_w is the number of links of the shortest path between O-D pair w.

We first examine the performance of different routing policies under the first call reward policy. The traffic load has been tuned, as described above, to yield a fractional reward loss of approximately 1%, which is referred to as the nominal traffic load, and a fractional reward loss of approximately 5%, which is referred to as the overloaded condition. The simulation results of the six routing policies previously studied are shown in Table 4.4 under the nominal traffic load and the overloaded conditions. (For the rest of our simulation results presented in this section, each simulation point is observed over 10 independent runs. Each run is run for 3000 units of mean call holding time of class 1 calls. For each run, the initial 10% of the samples were discarded. 95% confidence intervals for simulation results are indicated in the table.) From Table 4.4, we observe that all routing algorithms yield competitive performance

under both nominal and overloaded traffic loads. The main reason why MDP policies do not perform significantly better than LLP is that, among the 30 links, 6 links are heavily loaded while the remaining links are lightly loaded. Links that are overloaded are those between Houston and Greensboro, Chicago and Cleveland, and Cleveland and New York. Since at least one of these overloaded links is needed to connect a call from west to east (or from east to west), there is no alternate path that is superior. Furthermore, since call rewards are independent of the distance of the O-D pairs, no calls should be purposely blocked to yield better network performance.

We next study the performance of these routing policies when the second call reward policy is used. As we can see from Table 4.5, MDP policies perform much better than LLP. The main reason for this is that MDP policies will try to route as many "long-distance" calls as possible because they bring more revenue while LLP is blind to this property. We also observe that MDPD(D2) and MDPD'(D2) perform much better than other MDP policies. Finally, ASDR provides comparable performance to LLP and is much worse than the MDP policies. We conclude that when call reward is dependent not only on the bandwidth requirement but also on other parameters such as the distance between the endpoints of the O-D pair, properly dividing the reward loss of a rejected multi-link call becomes very important. However, the better performance is achieved at an additional computational cost.

In [32, 49], an ill-dimensioned network has been used to show that giving priority to shorter paths will degrade the routing performance. However, in this study, we observe almost no performance degradation by giving priority to shorter paths.

Case study 2:

In this case study, the proportional ratio of call arrival rate of each O-D pair is set according to Table 4.6. These relative traffic loads have been chosen to produce balanced link traffic loads. We also assume that 95% of the traffic is class 1 traffic and that both classes have the same mean call holding time.

The performance of six routing policies under this traffic configuration is shown in Tables 4.7 and 4.8 using the first and second call reward policies respectively. From these two tables, we observe that MDP policies perform significantly better than

Table 4.4 Comparison of fractional reward loss (%) of the six routing policies: case study 1, call reward assignment rule 1.

Policy	Nominal	Overload
LLP	0.92 ± 0.07	5.27 ± 0.12
ASDR	0.94 ± 0.05	5.10 ± 0.07
MDPD(D1)	0.85 ± 0.07	5.09 ± 0.12
MDPD'(D1)	0.87 ± 0.05	5.08 ± 0.10
MDPD(D2)	0.85 ± 0.05	5.15 ± 0.10
MDPD'(D2)	0.92 ± 0.10	5.04 ± 0.12

Table 4.5 Comparison of fractional reward loss (%) of the six routing policies: case study 1, call reward assignment rule 2.

Policy	Nominal	Overload
LLP	1.17 ± 0.09	6.28 ± 0.12
ASDR	1.10 ± 0.04	6.17 ± 0.14
MDPD(D1)	1.02 ± 0.07	5.91 ± 0.07
MDPD'(D1)	1.09 ± 0.11	5.93 ± 0.12
MDPD(D2)	0.82 ± 0.07	4.31 ± 0.10
MDPD'(D2)	0.81 ± 0.05	4.33 ± 0.10

LLP in most cases. This is because the traffic load at each link is more balanced, and MDP policies balance the link traffic load better than LLP by using link shadow prices. We also observe that MDPD(D2) and MDPD'(D2) do not perform better than other MDP policies. This suggests that in a well-dimensioned network, the reward decomposition rule for rejecting a multi-link call is not as important as in an ill-dimensioned network. Thus we conclude that the ASDR policy is competitive with other MDP policies while requiring the least computational complexity.

In this case study, we still observe no performance degradation for routing algorithms that give priority to paths with shorter length.

In section 4.1.3.3, we have introduced class-dependent compensation parameters, $\alpha'_k s$, to improve the accuracy of the estimated link shadow prices when b/C is not small enough. When b/C is sufficiently small, e.g., in our NSFNET example, it is not necessary to have these compensation parameters. We have conducted numerous

Table 4.6 Traffic configuration for case study 2.

Source	Destination	traffic load ratio $\lambda_{i,j}/\lambda_{baseline}$
Hartford	New York	12.5
Hartford	Atlanta	9.5
New York	Hartford	9.5
New York	Washington	12.5
New York	Cleveland	4.5
Washington	New York	9.5
Washington	Atlanta	12.5
Atlanta	Hartford	12.5
Atlanta	Washington	9.5
Atlanta	Houston	0.5
Cleveland	New York	4.5
Cleveland	Chicago	1.5
Chicago	Cleveland	1.5
Chicago	St. Louis	8.5
Chicago	San Francisco	6.5
St. Louis	Chicago	8.5
St. Louis	Houston	4.5
St. Louis	Denver	7.5
Houston	Atlanta	0.5
Houston	St. Louis	4.5
Houston	Los Angeles	9.5
Denver	St. Louis	7.5
Denver	Seattle	12.5
Seattle	Denver	12.5
Seattle	San Francisco	9.5
San Francisco	Chicago	6.5
San Francisco	Seattle	9.5
San Francisco	Los Angeles	10.5
Los Angeles	Houston	9.5
Los Angeles	San Francisco	10.5
Other O-D pairs ($\lambda_{baseline}$)		1.0

Table 4.7 Comparison of fractional reward loss (%) of the six routing policies: case study 2, call reward assignment rule 1.

Policy	Nominal	Overload
LLP	0.92 ± 0.03	4.45 ± 0.03
ASDR	0.78 ± 0.03	3.33 ± 0.03
MDPD(D1)	0.70 ± 0.03	3.18 ± 0.04
MDPD'(D1)	0.66 ± 0.02	3.13 ± 0.02
MDPD(D2)	0.70 ± 0.01	3.17 ± 0.03
MDPD'(D2)	0.68 ± 0.01	3.13 ± 0.04

Table 4.8 Comparison of fractional reward loss (%) of the six routing policies: case study 2, call reward assignment rule 2.

Policy	Nominal	Overload
LLP	1.07 ± 0.02	5.18 ± 0.04
ASDR	0.80 ± 0.01	4.89 ± 0.05
MDPD(D1)	0.81 ± 0.03	4.94 ± 0.04
MDPD'(D1)	0.75 ± 0.02	4.89 ± 0.02
MDPD(D2)	0.85 ± 0.01	4.80 ± 0.04
MDPD'(D2)	0.73 ± 0.03	4.88 ± 0.06

simulation studies and find that in our NSFNET example, the performance of our MDP routing policies is very insensitive to whether compensation parameters are used or not in computing link shadow prices.

4.2 The Effect of the Link Independence Assumption on the Performance of MDP Routing Algorithms

When the routing problem is formulated as a Markov decision process, the statistical link independence assumption is needed to reduce the enormous complexity of the exact model. In a real network, there exist certain positive correlations among network links that are ignored under the link independence assumption. This may be especially true in future high-speed networks in which network topologies will be very sparse. Thus the performance of the routing algorithms based on the link independence assumption becomes questionable. In this section, we study

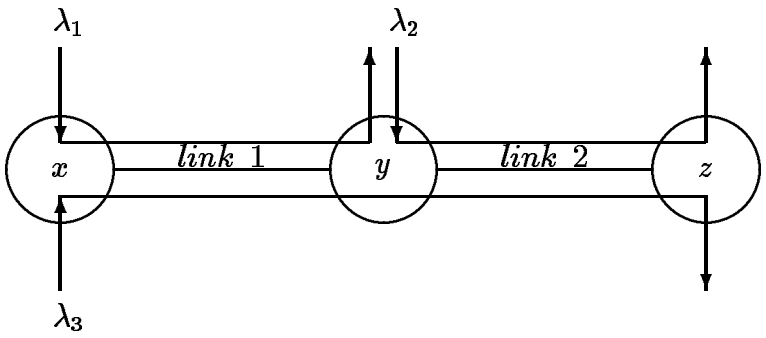


Figure 4.3 A simple network for investigating the effect of link independence assumption

the influence of the link independence assumption on the performance of routing algorithms.

4.2.1 Network Model

We consider a simple network, shown in Figure 4.3, consisting of 3 nodes, 2 links, and 3 O-D pairs. Let us consider the case where only one class of calls is offered to each O-D pair. Let b_i, $1/\mu_i$, and λ_i be the bandwidth requirement, mean holding time, and arrival rate of the class of calls offered to O-D pair i respectively. We assume that the arrival process of each O-D pair is a Poisson process and that the call holding times of each O-D pair are exponentially distributed random variables. We also assume that each call of O-D pair i carried on the network brings $r_i = b_i/\mu_i$ units of revenue to the network. Let C_j be the link capacity of link j. The purpose of this network model is to allow us to control the correlation of the two links by adjusting the arrival rate of the class of calls offered to O-D pair 3. At one extreme, in which $\lambda_3 = 0$, the occupancy distributions of link 1 and link 2 are statistically independent. In the other extreme case, in which $\lambda_1 = \lambda_2 = 0$, link 1 and link 2 have exactly the same link occupancy distribution.

Three routing policies are studied in this section. The first policy is the *optimal* policy which is obtained by formulating the routing problem of the whole network as a Markov decision process as discussed in section 4.1.1.2. The second policy is

the priority MDPD policy with reward decomposition rule D1 introduced in section 4.1.4.2. The third policy is the priority ASDR policy presented in section 4.1.4.1. Note that since each O-D pair has only one path, the routing problem is not to find the most "efficient" path on which to carry an incoming call, but to decide whether an incoming call should be carried. The difference between the optimal policy and the other two policies is that a link independence assumption is made in the derivation of the priority MDPD policy and the priority ASDR policy. That is, in these two policies, each link is treated independently and the arrival process of O-D pair 3 at each link is assumed to be a Poisson process. In fact, by assuming $r_i = b_i/\mu_i$, all policies will always accept calls from O-D pair 1 and O-D pair 2 if there are free circuits available. Thus they only differ in the decision of whether an incoming call of O-D pair 3 should be purposely blocked even if there are free circuits available on both links. For the optimal policy, the decision is made based on a three dimensional state space (i.e., the number of calls of each O-D pair). For the other two policies, the decision is made only based on a two dimensional state space (i.e., the number of busy circuits on each link).

This study has two goals. First, by comparing the performance of the optimal policy and the priority MDPD policy, we are able to investigate the degradation of the performance of routing algorithms due to the use of the link independence assumption. Second, by comparing the priority MDPD policy and the priority ASDR policy, we are able to see if it is important to decompose the reward of a multi-link call into individual link rewards when the link independence assumption is not valid.

Recall that we are interested in the fractional reward loss which, for an arbitrary policy π, is

$$fractional\ reward\ loss(\pi) = \frac{g(\pi)}{\sum_{i=1}^{3} \lambda_i r_i}, \tag{4.10}$$

where $g(\pi)$ is obtained by solving the set of equations in (4.1). It is straightforward to obtain the reward loss for the optimal policy. For the other two policies, we can obtain their performance by simulation or by analysis. In order to obtain their performance by analysis, we need to estimate the rate of the arrival process of O-D pair 3 at

each link. In our study, they are estimated by solving for the *Erlang fix point* of the following set of equations:

$$\lambda_3^1 = \lambda_3(1 - B_2) \tag{4.11}$$

$$\lambda_3^2 = \lambda_3(1 - B_1) \tag{4.12}$$

$$B_i = E(\lambda_i, b_i, \mu_i, \lambda_3^i, b_3, \mu_3, C_i) \quad i = 1, 2 \tag{4.13}$$

where λ_3^i is the offered traffic rate of O-D pair 3 to link i, B_i is the blocking probability for the calls of O-D pair 3 at link i, and $E()$ is a modified Erlang's B formula. For $b_1 = b_2 = b_3$ and $\mu_1 = \mu_2 = \mu_3$, $E()$ is exactly the Erlang B formula. Otherwise, we need to solve the steady state distribution for a two dimensional CTMC. Given the estimated offered traffic of O-D pair 3 at each link, we are able to derive the priority MDPD policy and the priority ASDR policy analytically. The reward losses of these two policies can then be obtained by equation (4.10).

4.2.2 Numerical Results

Let us first consider a homogeneous, symmetric case where each class of calls behaves exactly the same and each link has the same traffic load. As we observe in Table 4.9, the performance of the priority MDPD differs little from the optimal policy. However, the priority ASDR performs much worse when the correlation of the occupancy distributions of the two links is very high. This suggests that the decomposition of the reward of a multi-link call into individual link rewards is very important when link independence is not valid.

Since we have achieved our second goal, we can discard the priority ASDR policy and focus on investigating the effect of the link independence assumption on the performance of the priority MDPD policy. Let us consider the case where the traffic loads on the two links are different. In our parameter settings, we purposely set the capacity of link 1 so that link 1 is the network bottleneck. The performance of the two routing policies is shown in Table 4.10. (Note that we have chosen the network parameters such that we expect the difference in the performances of the optimal policy and MDPD to be most significant.) We find that even when the traffic load

Table 4.9 Comparison of fractional reward loss of optimal, MDPD, and ASDR policies, $b_1 = b_2 = b_3 = 1, C_1 = C_2 = 50$.

λ_1	λ_2	λ_3	Optimal	MDPD	ASDR
45	45	5	9.809%	9.815%	9.847%
40	40	10	9.637%	9.731%	9.955%
30	30	20	9.892%	9.997%	11.071%
25	25	25	10.134%	10.261%	11.656%
20	20	30	10.430%	10.461%	12.603%
10	10	40	11.000%	10.996%	14.537%

Table 4.10 Comparison of fractional reward loss of optimal and MDPD policies, $b_1 = b_2 = b_3 = 1, Cap = 50$.

λ_1	λ_2	λ_3	Optimal	MDPD
30	20	20	7.738%	7.741%
30	5	20	9.525%	9.525%

of the network is nonsymmetric, the priority MDPD policy still yields near optimal performance.

Let us now vary the link capacity of link 2 while maintaining the traffic load of the two links relatively the same. As shown in Table 4.11, there is no significant difference between the performance of the priority MDPD policy and the performance of the optimal policy.

Now let us look at the cases in which calls are not homogeneous. We first investigate the case where the calls offered to O-D pair 3 require more than one unit of bandwidth. Approximate link shadow prices are obtained as described in section 3. As shown in Table 4.12, even when the link shadow prices are approximate

Table 4.11 Comparison of fractional reward loss of optimal and MDPD policies, $b_1 = b_2 = b_3 = 1$.

C_1	C_2	λ_1	λ_2	λ_3	Optimal	MDPD
50	40	30	20	20	10.504%	10.634%
50	25	30	5	20	11.559%	11.619%

Table 4.12 Comparison of fractional reward loss of optimal and MDPD policies, $b_1 = b_2 = 1, b_3 = 3, C_1 = C_2 = 60$.

λ_1	λ_2	λ_3	Optimal	MDPD
50	50	2	6.379%	6.383%
40	40	5	6.691%	6.720%
30	30	10	11.728%	11.870%
30	30	8	7.254%	7.279%
20	20	11	7.988%	7.994%

and a link independence assumption is made, the priority MDPD policy still yields near optimal performance.

As indicated in section 4.1.5.1, the approximate link shadow prices obtained from the simplified link model are inaccurate when each class of call has a different mean call holding time. Thus we consider the case where the class of call offered to O-D pair 3 has a longer mean holding time than the other two classes of calls. We set the network parameters as follows: $b_1 = b_2 = 1, b_3 = 3, \mu_3 = 0.5, C = 60$. The fractional reward losses we obtained for the optimal policy and the MDPD policy are 7.280% and 7.340% respectively. We observe that the priority MDPD policy still performs reasonably well.

4.2.3 Conclusion

From our experiments, we have observed that routing algorithms based on an invalid link independence assumption can still yield near optimal performance. However, we do observe that the performance of a routing algorithm that does not decompose the reward of a multi-link call into individual link rewards degrades significantly.

4.3 Summary

In the first part of this chapter, we have described a simplified link model which can be used to derive a set of simple expressions to estimate state-dependent link shadow prices. We also have shown that the approximate link shadow prices are very accurate in some regimes of interest. Even in the case where the approximate link

shadow prices are not very accurate, we have shown in [49] that the degradation of the performance of routing policies due to these inaccuracies is not very significant.

A numerical study was performed using the NSFNET T3 backbone network. In one case where the traffic was set according to NSFNET traffic statistics, we observed little difference in the performance of MDP routing algorithms and the LLP algorithm when the call rewards are independent of the distance between the endpoints of O-D pairs. However, these MDP algorithms do perform better when call rewards depend on the distance of O-D pairs. In the second case we studied, in which traffic was set so that the link offered traffic was more balanced, MDP routing algorithms always yield better performance than LLP. We also observed from this study that, in a general network topology such as NSFNET, giving priority to shorter paths does not degrade the performance.

In the second part of this chapter, we investigated the effect of the link independence assumption on the performance of routing algorithms. The link independence assumption is a fundamental assumption that is needed in order to obtain feasible Markov decision-based routing algorithms. Although the link independence assumption may not be valid in general networks, we have found that the effect of an invalid link independence assumption on the performance of MDP-based routing algorithms is not very significant given that the rejection cost of a multi-link call is properly distributed into individual links on the path.

From our study, we have found that among MDP routing policies, an ASDR-like policy requires the least computational complexity and on-line measurements and yields competitive performance when the network traffic load on each link is well balanced and the link independence assumption is valid. However, when one of these two conditions is not true, the performance of ASDR policy degrades quickly and MDPD-like policies are preferred.

CHAPTER 5

MDP ROUTING IN VIRTUAL PATH NETWORKS

5.1 Introduction

The CCITT specifications on future Broadband ISDN propose a Virtual Path (VP) concept as a component of traffic control and resource management in B-ISDN [18, 20]. The VP concept was first introduced by Addie *et al.* [1], in parallel with Kanada *et al.* [51]. One of its goals is to help reduce the control cost in future high-speed networks by grouping connections sharing common paths through the network into a single unit. A VP can be viewed as a single logical direct link between a source node and a destination node which consists of a set of several physical links which together form the path. Therefore, a network with VP's can be viewed as a virtual network with VP's as links.

The use of VP's for traffic control has several advantages. For example, VP's can be used to simplify call admission control, implement a form of priority control by segregating traffic types requiring different QOS, efficiently distribute messages for the operation of traffic control schemes, facilitate traffic aggregation, simplify network architecture, and enhance network services [20, 72]. Most of these advantages have also resulted in a reduction of processing complexity at the switches (nodes) within the "interior" of the network.

The use of VP's also plays an important role in network resource management. By reserving resources on VP's, such as buffer space and bandwidth, processing requirements for call establishment are substantially decreased. This is because a new virtual circuit (VC) [1] can be established by executing the admission control function at the source node of a VP and without performing any call processing at transit

[1]Throughout this chapter, the following three terminologies, "call", "VC", and "connection", are interchangeable.

nodes. However, reserving resources on VP's also reduces the statistical multiplexing gains of the network, resulting in an increased network blocking probability. Thus, there exists a trade-off between reduced control processing requirements and reduced resource utilizations. Since nodal control processing is very likely to be a significant cost in future high-speed networks, we believe that the total network costs can be reduced by reserving resources on VP's [13, 14, 66, 72]. Throughout this chapter, we will use the term "VP capacity", found in previous research [13, 14, 47, 66, 67, 72], to refer to the resources reserved on VP's. However, readers should keep in mind that "VP capacity" includes not only transmission bandwidth but also buffer space.

Most previous research in VP networks has focused on studying the trade-off between control cost, i.e., the amount of processing required at switches, and transmission cost, i.e., the amount of bandwidth required at physical links, for a given network topology and traffic pattern. For example, in [13], Burgin presents a cost-benefit analysis of reserving capacity on VP's in a homogeneous network. In this study, three VP capacity reservation schemes are studied. In the first scheme, no capacity is reserved on VP's. In the second scheme, dedicated capacity is reserved on all VP's and these reservations are updated periodically. In the last scheme, dedicated capacity is reserved on a VP only if the anticipated traffic demand on the VP exceeds some threshold. Control costs, which include the processing required for call establishment and VP capacity management, and transmission costs for each scheme are estimated and compared. The author concludes that since transmission costs will be reduced significantly, the scheme that reserves resources on all VP's is the best technique for future BISDN networks. In [47], Hui *et al.* also present a study of the reduction of bandwidth efficiency resulting from reserving capacity on VP's. In this study, only the first two schemes of [13] are studied on a 5-node network with single class of traffic. Their results show that reserving VP capacity for each VP in such a simple network requires roughly 50% bandwidth more than the case where no VP capacity is reserved in order to provide the same source-destination blocking probabilitiy.

The study of the trade-off between control cost and transmission cost by Ohta *et al.* [66, 67] is rather different from [13] and [47]. Instead of reserving capacity on VP's for fixed length periods, VP capacities are dynamically increased/decreased on

demand in units of "blocks". Specifically, when a call arrives at a source node of a VP and finds insufficient VP capacity, the VP controller will try to reserve one more block of capacity to accommodate this call. If this reservation is unsuccessful, the call is blocked. (A block of capacity is predefined by the network manager and is sufficient to accommodate several calls.) On the other hand, when the VP reserved capacity is one block more than needed after a call is terminated, a block of capacity is released. By assuming that every call is homogeneous and that the traffic offered to each VP is identical, the authors study the effects of the block size on the trade-off between control cost (which only accounts for the processing required for VP capacity management), and transmission cost. Since the bandwidth on each link is assumed fixed, the transmission cost is expressed in terms of bandwidth utilization efficiency. The effect of statistical multiplexing is not considered in allocating bandwidth for VP's.

Our research focus differs significantly from these previous works. Although we also study two VP capacity reservation schemes, our focus is on how to improve bandwidth efficiency through adaptive routing when capacity is reserved on all VP's. A related previous work is [43], in which LLP-based routing algorithms were developed to reduce the network call blocking probability for homogeneous VP networks. Our research differs from this work in the following respects. First, instead of using an LLP approach, we develop MDP-based routing algorithms for VP networks. The reason for using an MDP approach is that we have shown the MDP approach to be more general and efficient than the LLP approach in other contexts (see chapter 4). Second, the study in [43] is limited to the case in which each VP only supports one QOS requirement. In our study, this constraint is relaxed. Finally, besides studying a VP capacity reservation strategy which reserves dedicated resources on VP's, we also propose a statistical VP capacity reservation strategy in which no dedicated resources are reserved on VP's. The advantage of the statistical reservation strategy is that better statistical multiplexing gains of the network can be achieved. However, we will see that this advantage is offset somewhat by the need to establish a more stringent QOS requirement at physical links.

The remainder of this chapter is organized as follows: in section 5.2, a VP network

is described in more detail. For ease of explanation, a specific traffic source model, QOS criterion, and call admission scheme that will be used in our numerical examples are described at the end of the section. The two VP capacity reservation strategies are described in detail in section 5.3. Implementing the statistical reservation strategy requires establishing multiple classes of QOS on a physical link. This introduces problems for which we propose several solutions, including one referred to as the ADAPTIVE scheme, at the end of section 5.3. In section 5.4, the routing problem in VP networks is discussed and four routing policies based on the MDP approach are proposed. The performances of the four routing policies are evaluated and compared to the performances of two base-line policies in section 5.5. Performance metrics of interest to us include network blocking probability, the largest blocking probability among all source-destination pairs, call admission cost, and resource management cost. The first two metrics are considered to be part of the transmission cost, while the last two metrics contribute to the control cost. The first base-line policy, called the DIRECT policy, reserves dedicated resources on VP's and offers all calls to direct VP's only. The second base-line policy, called the NOVP policy, does not reserve capacity for VP's. Instead, calls are setup in the traditional hop-by-hop fashion. Finally, section 5.6 summaries this chapter and discusses how the performance of routing policies will be affected under different QOS criteria and call admission schemes.

5.2 Network Model

5.2.1 A Virtual Path Network

The CCITT specifications on ATM [17] specify a cell-based, connection-oriented, packet-switched transport mode within the network. In ATM networks, when a connection (virtual circuit) is established, a route with a guaranteed QOS is assigned to this connection. Cells of the virtual circuit are then transferred along the assigned route and expect to receive the guaranteed QOS.

In order to facilitate traffic control, simplify node structure, reduce nodal processing and control of routing, and ease resource management, the Virtual Path (VP) concept has been proposed [18, 19, 20, 72]. A VP is broadly defined as a labeled path (a set of adjacent links) which may carry more than one virtual circuit. Virtual circuits

which belong to the same VP have the same endpoints. The purpose of introducing VP's into ATM networks is to facilitate the management of multiple virtual circuits by treating them as a group. That is, all virtual circuits in a VP can be transported, processed, and managed in a similar manner. For example, since all virtual circuits of a virtual path share the same routing table, establishment of a virtual circuit as well as maintenance of routing tables can be simplified.

A logical VP network can be defined by viewing switches as nodes and VP's as links. Consider the physical network shown in Figure 5.1(a), which consists of 5 switches (nodes) and 12 unidirectional physical links. If a VP is assigned to each source-destination pair, as shown in Figure 5.1(b), a 5-node fully connected VP network can be formed which consists of 20 unidirectional *logical links* (VP's). By reserving resources on VP's, connection establishment on a VP is very similar to setting up a call on a direct link in traditional circuit-switched networks. Thus, as we will show later, routing algorithms in circuit-switched networks can be easily applied to VP networks.

One rationale for using VP's for traffic control is to separate traffic in order to prevent statistical multiplexing among different types of traffic [20]. Different types of traffic, e.g., voice and data, usually have very different traffic characteristics and QOS requirements. The question of how to multiplex more than one type of traffic while providing different QOS requirements at a switch is a very complicated, open problem. However, the problem can be much simplified if different types of traffic are separated by assigning a VP with dedicated resources to each type of traffic. As a consequence, more than one VP may be established between the same source-destination pair, with each carrying different types of traffic. Throughout this chapter, we assume that VP's are employed to separate different types of traffic and focus on a VP network which only carries a single type of traffic. We also assume that connections in a VP network require a homogeneous end-to-end QOS guarantee. Such a VP network is referred to as a homogeneous VP network in [44]. However, unlike [44], we will assume that virtual circuits in a VP connection may have different QOS requirements, as described in [20]. That is, a virtual path may provide more than one class of QOS in order to meet the different QOS requirements of different virtual circuits. The rationale for

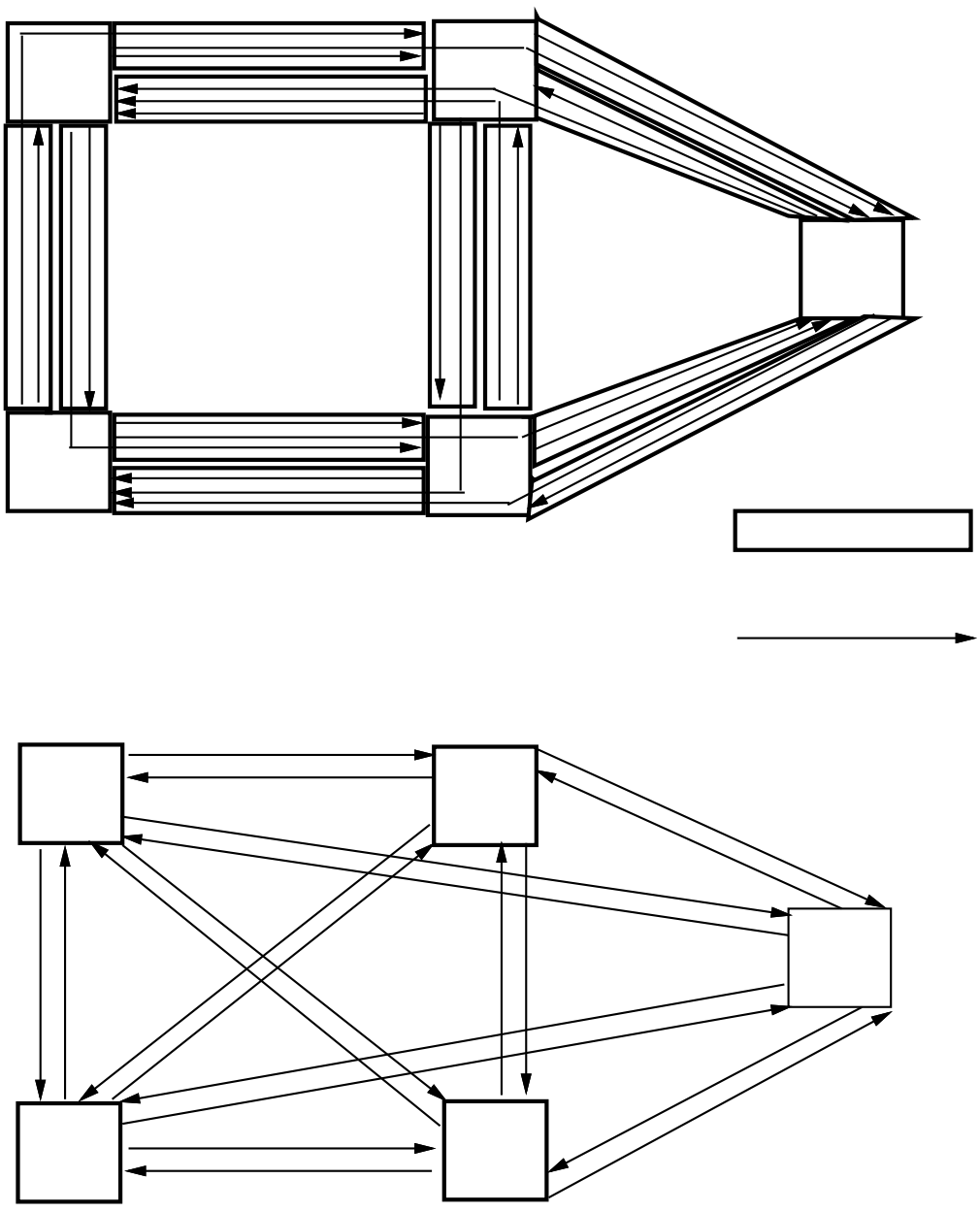


Figure 5.1 Example: A 5-node network and its corresponding VP network.

relaxing this assumption is, as we will see in section 5.4, that when a VC is routed on an alternate path, i.e., a path consists of two VP's, a more stringent QOS is required at each VP on this alternate path (since the path consists of two concatenated VP's) in order to meet the specified end-to-end QOS requirement.

As in previous chapters, throughout this chapter, we assume that VC's arrive at each source-destination pair according to a Poisson process and the call holding times are assumed to be exponentially distributed random variables with mean of unity.

5.2.2 Traffic Model, QOS Criterion, and Call Admission Scheme

So far, we have not explicitly defined a traffic model, specific QOS criterion, or call admission scheme, primarily because they are orthogonal to the routing problem. However, in order to explain VP capacity reservation schemes and routing policies, we now describe a traffic model, QOS criterion, and call admission scheme which will be used in our numerical examples. We emphasize, however, that the design of our routing policies is independent of the underlying traffic model, QOS criterion, and call admission policy.

The traffic model that we use throughout our numerical examples is a two-state fluid-flow model [5, 42]. This model describes a source which is either in an "on" state, transmitting at its peak rate, or in an "off" state, transmitting at zero bit rate. The durations of the "on" and the "off" states are assumed to be exponentially distributed random variables. When this model is used to describe a packet voice source [26, 45, 64, 77], the source will stay in the "on" state for an average of 352 msec, transmitting at 32 Kbps, or stay in the "off" state for an average of 650 msec, transmitting at zero bit rate. Based on this two-state fluid-flow model, a traffic source can be described by three parameters: its peak rate, r (bps); utilization, ρ, i.e., the fraction of time the source is at "on" state; and the mean "on" period, b (seconds) [42]. For example, the voice source can be described by the triple $(32000, 0.3513, 0.352)$.

Different types of traffic may require different QOS guarantees. For example, voice sources may require small cell delays and cell jitters while text sources may require a small or zero cell loss rate. Throughout our examples, we assume that the QOS criterion is specified by $Q(D, \epsilon)$, which means that no more than a fraction, ϵ,

of cells of a virtual circuit should experience more than D seconds of queueing delay [2]. Note that, unlike [43], the value of D does not include the propagation delay. We also assume that the buffer space allocated for each link (or VP) only holds up to D seconds of cells and arriving cells are discarded if the buffer is full. A lost cell is considered as a cell with infinite queueing delay. Therefore, this QOS criterion takes into account not only bounds on the delay jitter but also on the cell loss rate.

Recently, a number of call admission schemes have been proposed for high-speed networks (e.g. [33, 37, 42, 53]). We use the admission control scheme proposed by Guérin *et al.* [42], which is based on the fluid-flow approximation model of [5]. Specifically, consider a number of homogeneous sources described by (r, ρ, b) with a QOS requirement $Q(D, \epsilon)$, which are multiplexed onto a link of capacity C bps. Based on fluid-flow approximation techniques [33, 42], the maximum number of sources that can be multiplexed onto the link without violating the QOS constraint is given by

$$N = \lfloor \frac{C}{\hat{c}} \rfloor \tag{5.1}$$

where $\hat{c}$ is called the equivalent capacity of a single source and is given by [42]

$$\hat{c} = \frac{\alpha b(1-\rho)r - x + \sqrt{[\alpha b(1-\rho)r - x]^2 + 4x\alpha b\rho(1-\rho)r}}{2\alpha b(1-\rho)} \tag{5.2}$$

where $\alpha = \ln(1/\epsilon)$ and $x = D \times C$.

5.3 Resource Management

Both VP's and QOS requirements play an important role in network resource management in B-ISDN. In this section, we first describe two VP capacity reservation strategies. Implementing the second reservation strategy requires establishing multiple classes of QOS on a physical link. Thus, at the end of this section, we also discuss possible solutions for providing multiple classes of QOS on a physical link.

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[2] Although our traffic model does not have the concept of "cells" or "packets", we assume that the data generated during the "on" period of a source are packed into integer number of "cells" and the cell loss probability is very close to the buffer overflow probability (see also [64]).

5.3.1 VP Capacity Reservation

Having noted that call setup processing can be eliminated from the transit nodes when resources are reserved on VP's, it is necessary to examine different strategies for reserving resources on VP's. In the following, we present two VP capacity reservation strategies: a *deterministic* strategy and a *statistical* strategy. The basic distinction between these two strategies is whether separate capacity is reserved for the VP's. The deterministic strategy reserves dedicated resources, such as buffer space and bandwidth, for VP's while the statistical strategy allows statistical cell multiplexing among the cells from different VP's on a physical link.

The deterministic strategy, advocated by [13, 43, 47], reserves separate link capacity for each VP passing through the link. This approach treats a VP as if it were a physical link. A certain amount of buffer space at the source node and bandwidth at each physical link on this VP are *dedicated* to this VP. When a physical link is shared by several VP's, the cells of different VP's are transmitted in a TDM-like manner such that the instantaneous bit rate transmitted on a VP is limited to its reserved bandwidth. (For implementation details, see [78].) Consequently, the sum of the reserved VP bandwidths on a link is not permitted to exceed the total capacity of this link. Thus no multiplexing is performed among VP's at the link-level. However, statistical cell multiplexing is still performed among virtual circuits within a VP. Therefore, it is possible that the offered peak rate to a VP exceeds its allocated bandwidth over a short period of time. In such a case, cells that cannot be transmitted are buffered at the reserved buffer space at the source node of the VP. Since the instantaneous transmission rate of a VP is limited to its reserved bandwidth, the buffer space required and the queueing delays incurred at transit nodes/links of the VP are very small, and can be assumed to be negligible [13, 43, 47]. Therefore, the maximum number of virtual circuits that can be supported by a VP without violating a given QOS guarantee can be computed locally at the source node based on the knowledge about the the size of the reserved buffer space and the amount of the reserved bandwidth.

The statistical strategy, inspired by [73, 74], allows statistical multiplexing of cells from different VP's on a physical link. In this approach, instead of explicitly reserving

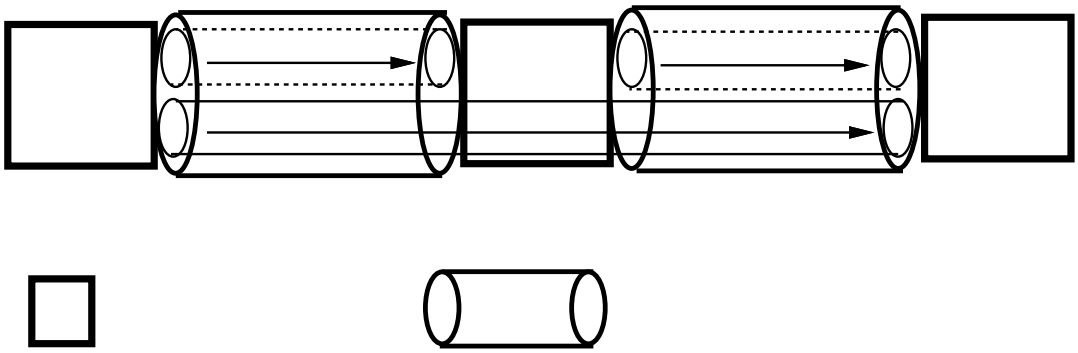

Figure 5.2 Example: A simple VP network

buffer space and link bandwidth for VP's, each VP is allocated a *dedicated* number of virtual circuits, each with a guaranteed QOS, such that call setup processing can still be done locally at the source node as long as the existing number of virtual circuits within the VP is less than this number. The actual reservation is done by pretending as if each VP has accepted as many as the dedicated number of virtual circuits into the network.

To help explain the statistical strategy, let us consider the simple network shown in Figure 5.2. Assume each link is a T1 link with capacity of 1.544 Mbps. Also assume that homogeneous packet voice sources, described by a two-state fluid-flow model with parameters $(32000, 0.3513, 0.352)$, are presented to the network. Let the QOS requested by virtual circuits within both VP 1 and VP 2 be $Q(0.016, 5 \times 10^{-7})$ and the QOS requested by virtual circuits of VP 3 be $Q(0.032, 10^{-6})$. We assume that cells of different virtual circuits are treated equally. Since cells of all virtual circuits are multiplexed (on both physical links), cells of VP 3 will experience queueing delays at both links. (Note that this situation does not happen in the deterministic strategy.) However, since the QOS requested by VC's of VP 3 is less stringent than that of VP 1 and 2 in this example, the QOS of VC's carried on VP 3 is still guaranteed, provided the QOS established on each link is $Q(0.016, 5 \times 10^{-7})$. By invoking equation (5.1), the maximum number of virtual circuits that can be accepted on each link without

violating the QOS requirement $Q(0.016, 5\times10^{-7})$ is 55. Let N_1, N_2, N_3 be the number of virtual circuits to be reserved on VP 1, 2 and 3 respectively. Then such reservation request is *feasible*, i.e., the request can be granted, if the following two inequalities are satisfied:

$$N_1 + N_3 \leq 55,$$

$$N_2 + N_3 \leq 55.$$

The advantage of the statistical approach is that it provides better statistical cell multiplexing gains than the deterministic strategy. However, this advantage is offset by the fact that a more stringent QOS guarantee needs to be provided by each link, as we observed in the previous example in which the QOS provided at each physical link is more stringent than the end-to-end QOS requirement of VP 3. In general, consider a VP which consists of n links and wants to reserve some N virtual circuits, each having an end-to-end QOS requirement $Q(D, \epsilon)$. In the process of granting such request, two problems can be identified. First, the traffic characteristics of a virtual circuit may change as the traffic travels along the path; these changes must be characterized. Second, we must determine how to assign the end-to-end QOS requirement to each link such that the overall end-to-end QOS can be satisfied. Instead of solving these two complicated problems, our approach is to assume that the traffic characteristics of a virtual circuit are unaffected as the traffic travels along the path and to assign an equal amount of the end-to-end QOS of a virtual circuit to each link on the path, the so called equal allocation (EQ) policy [64, 65, 69, 74]. As a result, a request reserving N virtual circuits with an end-to-end QOS requirement $Q(D, \epsilon)$ on a VP can be decomposed into requests, each reserving N virtual circuits on each individual link, each with a QOS requirement $Q(D/n, \epsilon/n)$. Thus when a physical link is shared by more than one VP, the QOS requirement from different VP's may be different under this strategy even though we assume that all connections in a homogeneous VP network require the same end-to-end QOS guarantee. This issue of how to establish multiple classes of QOS service in a single link is deferred to the next section.

When capacity is reserved on VP's, it may be desirable to dynamically adjust the allocation to improve link bandwidth utilization as well as to adapt to dynamic changes in network traffic flows. The authors of [13, 47] suggest that the reallocation

of VP capacity should be done periodically on a much longer time scale than the interarrival time of successive calls. On the other hand, Sato *et al.* [66, 67] argue that this should be done in a more dynamic way, i.e., the reserved capacity on a VP should be dynamically increased/decreased on demand in fixed units of capacity, called blocks. As stated in the introduction to this chapter, the choice of the time interval between two VP capacity reallocations in the former case and the block size in the latter case determine the trade-off between control cost and transmission cost. The determination of which approach is the best is beyond the scope of this thesis, and we adopt the first approach. Furthermore, we assume that the time interval between two VP capacity reallocations is significantly larger than the call setup time. Under this assumption, routing of virtual circuits can be performed as if the topology and the capacity of the VP network were fixed [43].

The question of how to optimally assign capacity to VP's for a given network topology is also beyond the scope of this thesis. Determining the optimal assignment of VP capacity is very similar to, but much more complex than, the dimensioning problem in circuit-switched networks. Furthermore, one needs to address issues such as fairness, adaptability, (i.e., dynamically reallocation), and diversity, (i.e., handling diverse traffic sources with different QOS requirements), subject to the constraints imposed by the network topology such as connectivity and link capacity. In this study, we assume that the capacity reserved on VP's is given [43].

5.3.2 Multiple Classes of QOS

The implementation of the statistical VP capacity reservation strategy requires a physical link to accommodate multiple service classes with different QOS requirements. In order to achieve high bandwidth utilization, a sophisticated cell transfer control scheme can be developed. For example, a priority scheme which classifies and serves cells according to their QOS requirements was proposed in [29] to provide two classes of QOS on a physical link. However, it is expensive to implement such a complicated cell control scheme at switches and the resulting QOS is difficult to predict. In the following, we focus on non-priority schemes which are relatively easy to implement.

A simple approach is to uniformly provide the most stringent QOS to all cells [43, 74]. Since all cells are treated equally, no special cell control scheme is required. However, link utilization will be decreased because an unnecessarily stringent QOS is provided to *all* cells, regardless of their actual QOS requirements. We will refer to this approach as the UNIFORM scheme.

A complementary approach to the UNIFORM scheme is to separate cells according to their QOS requirements, similar to the approach adopted by the deterministic VP capacity reservation strategy. The main idea is to allocate dedicated buffer space and bandwidth to each class of QOS. Cells with the same QOS requirement are treated equally and are served by their dedicated bandwidth based on a first-come-first-serve discipline. The advantage of this approach is that cells will receive their actual QOS requirements. However, this advantage is offset by the reduction in the statistical cell multiplexing gains because of the partition of the resources. We will refer to this approach as the SEPARATE scheme.

To implement the SEPARATE scheme, we need a mechanism to determine the amount of buffer space and bandwidth to dedicate to each class of QOS. Let us consider an example where two classes of QOS are to be established on a link of capacity C. Let $Q(D_1, \epsilon_1)$ and $Q(D_2, \epsilon_2)$ be the two classes of QOS to be established. Let N_1 and N_2 be the maximum number of virtual circuits of these two classes that can be simultaneously supported without violating the QOS requirements. Assume that $Q(D_1, \epsilon_1)$ is more stringent than $Q(D_2, \epsilon_2)$. Our approach to determine the amount of buffer space and bandwidth to dedicate to each class of QOS is as follows. For a given N_1, we first determine the size of buffer space, B_1, and the amount of bandwidth, C_1, required by the class 1 traffic. Then we evaluate N_2 by equation (5.1) using $\alpha = \ln(1/\epsilon_2)$ and $x = D_2 \cdot (C - C_1)$. To determine the size of buffer space, B_1, and the amount of bandwidth, C_1, required to serve the class 1 traffic at the requested QOS, the following iterative algorithm is used where C_1^i and B_1^i are the bandwidth and buffer space, respectively, to be allocated to class 1 traffic obtained from the ith iteration. Recall that we have homogeneous traffic sources described by (r, ρ, b).

- **Initial step:** Allocate bandwidth to class 1 traffic according to its peak rate, i.e., $C_1^1 = N_1 \times r$. The buffer space required is then determined by $B_1^1 = D_1 \cdot C_1^1$.

- **Iteration:** Let $\hat{c}^i$ be the equivalent capacity of class 1 traffic obtained from the ith iteration. Repeat the following two steps until $(C_1^{i-1} - C_1^i) < \varepsilon$, where ε is the stopping criterion. (In our examples, ε is set to 1 bps.)

  1. Use $x = B_1^{i-1}$ to compute $\hat{c}^i$ by equation (5.2).
  2. Re-calculate C_1^i and B_1^i by

$$C_1^i = \hat{c}^i N_1,$$

$$B_1^i = D_1 \cdot C_1^i$$

Extending this iterative algorithm to a more general case with arbitrary classes of QOS is very simple, and, thus, is omitted.

Whether the SEPARATE scheme is better than the UNIFORM scheme depends on the link capacity, the QOS requirements, the number of QOS classes, and the number of virtual circuits in each class of QOS. For example, consider a simple example where homogeneous packet voice sources, described by a two-state fluid-flow model with parameters $(32000, 0.3513, 0.352)$, are presented to a link and two classes of QOS are to be established. In Figure 5.3, we compare the maximum number of class 1 and class 2 virtual circuits that can be supported on the link under these two schemes with different link parameter settings. A point (N_1, N_2) on the curve of a cell control scheme in Figure 5.3 can be interpreted as follows: in order to support N_1 class 1 virtual circuits, N_2 is the maximum number of class 2 virtual circuits that can be supported on the link without violating the QOS requirement under this cell control scheme. We observe that when the number of virtual circuits of the first class, N_1, is small, the SEPARATE scheme can support more class 2 VC's than the UNIFORM scheme. On the other hand, when N_1 is large, we see the opposite result. The rationale for such an observation is that, as stated above, when N_1 is small, the capacity required to support N_1 class 1 VC's is also small which implies the reduction in the statistical multiplexing gains due to the separation of resources is also small. Thus, by providing a less stringent QOS to the class 2 VC's, the SEPARATE scheme can support more class 2 VC's. However, as N_1 increases, the reduction in the statistical multiplexing gains due to the separation of resources also increases. Hence, at some point, the UNIFORM scheme becomes better than

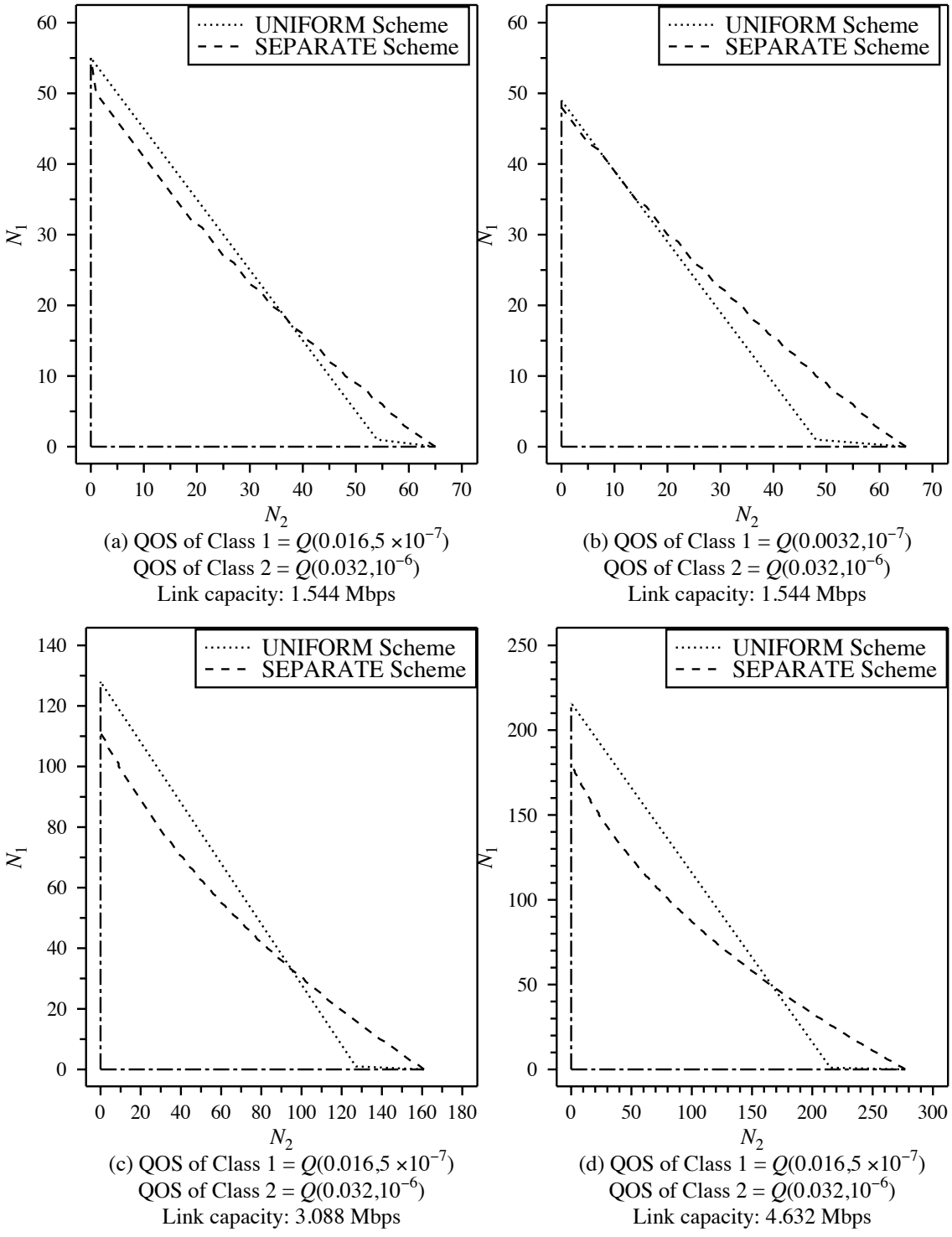

Figure 5.3 Comparison of the maximum number of VC's that can be supported under different cell control schemes.

the SEPARATE scheme. From Figure 5.3, we also observe that, for a given N_1, the difference between the maximum number of class 2 virtual circuits that can be supported under these two scheme increases as the link capacity or the difference between the two QOS requirements increases.

We adopt a hybrid scheme, referred to as the ADAPTIVE scheme, which is a combination of the UNIFORM scheme and the SEPARATE scheme. Consider the previous case where two classes of QOS are to be established. For a given N_1, the ADAPTIVE scheme will adopt the UNIFORM scheme if it supports more class 2 virtual circuits than the SEPARATE scheme. Otherwise, the ADAPTIVE scheme will adopt the SEPARATE scheme. Figure 5.4 shows the maximum number of class 1 and class 2 virtual circuits that can be supported under the ADAPTIVE scheme with a T1 link and the two classes of QOS to established are $Q(0.016, 5 \times 10^{-7})$ and $Q(0.032, 10^{-6})$ (same parameters as in Figure 5.3(a)).

5.4 VP Routing

When resources are reserved on VP's, routing in VP networks is very similar to routing in traditional circuit-switched networks. Each virtual path can be viewed as a logical link with allocated capacity. Thus, as shown in Figure 5.1, the logical view of a VP network can be fully connected even though the physical network is very sparse. Furthermore, when distinct VP's are used for separating different types of traffic, routing in homogeneous VP networks is also very similar to routing in circuit-switched networks. Therefore, in this section, we study how routing algorithms in circuit-switched networks can be applied to VP networks. Since we have shown that MDP routing is more efficient than LLP routing in chapter 4, we will focus on MDP routing in VP networks.

5.4.1 MDP Routing in Circuit-Switched Networks

The formulation of the routing problem in traditional circuit-switched networks as a Markov decision process is very similar to the formulation presented in section 4.1 except that only one class of traffic is present in the network. The link independence assumption is still required to make the MDP tractable [68]. However, since the

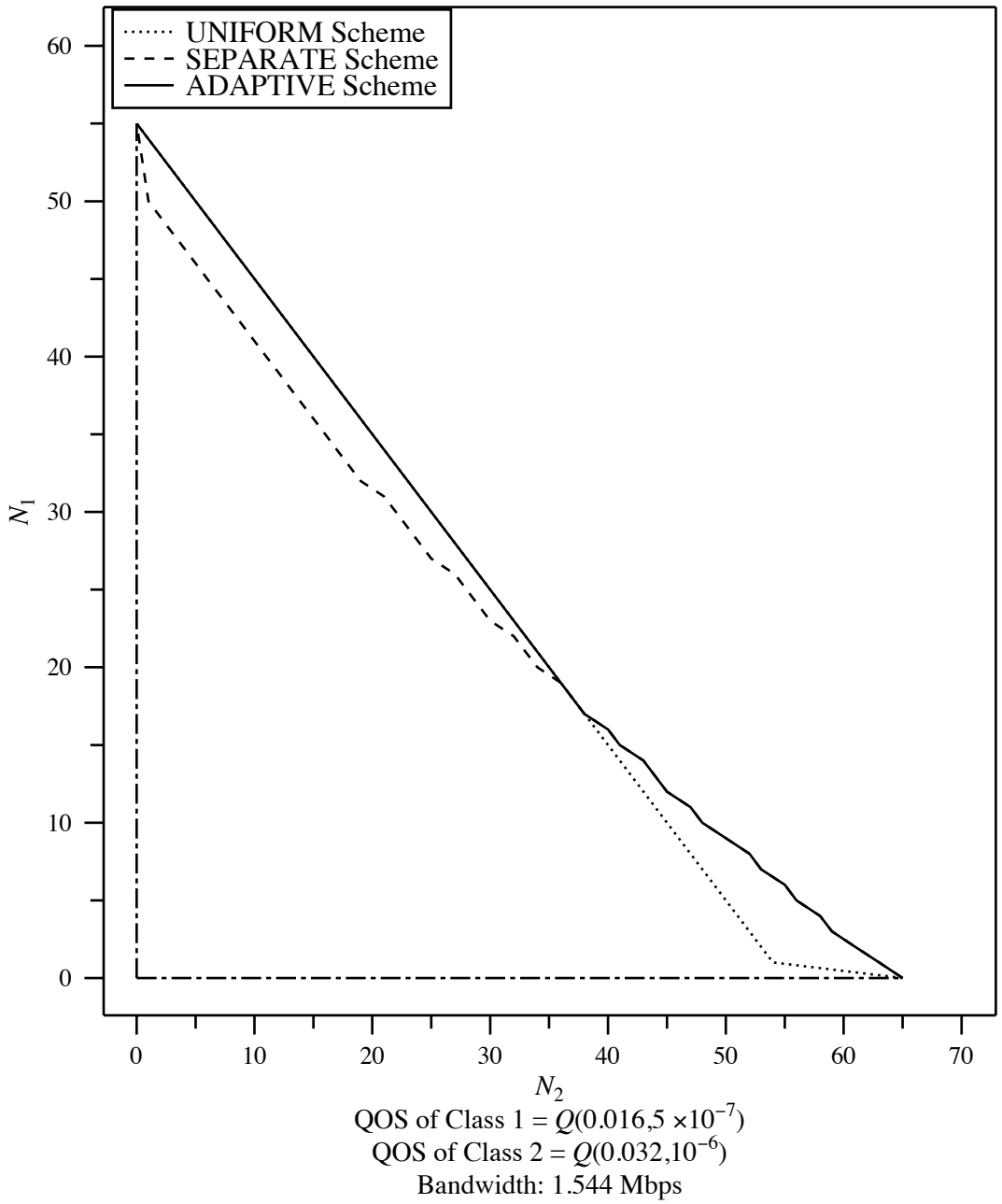

Figure 5.4 Maximum number of VC's that can be supported under the ADAPTIVE scheme.

network handles only one type of traffic, the link shadow prices can be computed efficiently without the use of the Pascal approximation technique [23, 49]. As a result of simplifying equations (4.4), (4.5) for a single class of traffic, the state-dependent link shadow price at link state i, i.e., a link with i busy circuits, is given by

$$p^{\ell}(i) = \frac{r^{\ell} E(\lambda^{\ell}, C_{\ell})}{E(\lambda^{\ell}, i)}, \tag{5.3}$$

where $E()$ is the Erlang's B formula, λ^{ℓ} is the arrival rate, C_{ℓ} is the link capacity, and r^{ℓ} is the link loss which is defined by routing policy (see section 4.1.4).

In evaluating link shadow prices, the decomposition rules for distributing the revenue loss of rejecting a multi-link call into link losses, r^{ℓ}, can still be used. Routing policies based on the computed link shadow prices, e.g., SDR [68], are very similar to the three MDP-based routing policies studied in section 4.1.4.

5.4.2 MDP Routing in VP Networks

Dynamic alternate routing schemes, such as MDP-based routing, are used to reduce the network's blocking probability in circuit-switched networks. Similarly, dynamic alternate VP routing schemes should improve a VP-based network's blocking probability. To perform MDP routing in a VP network, we first define a set of possible paths for each source-destination pair. A candidate path may consist of one or more VP's. Since a VP network is very likely to be densely connected, we restrict all candidate paths to consist of at most two VP's. From a previous study on fully connected networks [49], we have found that priority MDP routing policies, which give priority to direct links, yield better performance than non-priority MDP routing policies. Thus, we consider MDP routing schemes in VP networks which always offer a call to the direct VP first. If a call is blocked on the direct VP, it is then offered to the alternate path with the minimum *path cost* (see section 4.1.4). A call is rejected if the minimum path cost of all alternate paths is greater than the call reward. Before we develop MDP routing algorithms for VP networks, however, several issues need to be addressed.

When a call is to be routed on an alternate path consisting of two VP's, we again encounter the same problems facing the statistical strategy for reserving capacity on

VP's, i.e., how to characterize the traffic characteristics of a VC as it travels along the alternate path and how to assign the end-to-end QOS requirement to each link of the alternate path. As in the previous section, we assume that the traffic characteristics of a virtual circuit are unaffected as the traffic travels through a VP. The problem of dividing the end-to-end QOS into local QOS is subtler because it depends on the underlying VP capacity reservation strategy. If a deterministic reservation strategy is used, the equal allocation policy, i.e., assigning an equal amount of the end-to-end QOS of the virtual circuit to each VP on the alternate path, is also adopted. As a consequence, two classes of QOS, one for direct calls and one for alternate calls, need to be established on a VP which contains alternate routed calls. (Recall that a homogeneous VP network handles only one type of traffic with homogeneous end-to-end QOS requirements and alternate paths are limited to at most two VP's.) The ADAPTIVE scheme proposed in the previous section is applicable for providing two QOS classes on a VP with dedicated capacity.

On the other hand, if a statistical VP capacity reservation strategy is adopted, the equal allocation policy becomes problematic because it is very difficult to provide two classes of QOS on a VP under this strategy. As previously discussed, when multiple classes of QOS are established on a physical link, a more stringent QOS can be provided to VP's. That is, some VP's may receive better QOS than requested. Therefore, under the statistical reservation strategy, our approach for providing alternate routing is to restrict the paths in the set of possible paths such that an alternate path is considered as "possible" only if the sum of the QOS of the two VP's on this path satisfies the end-to-end QOS requirement.

5.4.3 MDP Routing Policies

In this section, we develop four MDP routing policies under the two different VP capacity reservation strategies. In general, the following algorithm is shared by these four routing policies.

- **Initial step:**

  Use direct VP routing as the initial policy. Assign link-offered loads, λ^{ℓ}, accordingly. (λ^{ℓ} is the total offered link load on link ℓ.)

- **Loop**

  1. At regular time intervals of length Δ_t, use link-measurements to re-estimate the link-offered loads, λ^ℓ.
  2. Use the new estimated link-offered loads, λ^ℓ, from the previous step to compute new values of the link shadow prices. The computation of link shadow prices and the method by which link shadow prices are used to select the routing path, or reject the call, will be policy dependent, as discussed below.

The four MDP routing policies discussed below are quite similar. We first describe the priority MDPD policy under the deterministic reservation strategy in detail. Other policies are described only in the manner in which they differ from the priority MDPD policy. In MDP routing, the most important step is to compute the link shadow prices. In order to compute the link shadow prices, a MDP routing policy first estimates link-offered loads and link losses. After the link shadow prices are computed, the cost for a path can then be calculated. The resulting path cost is then used for path selection. Thus, in the following description, we describe how link losses and link-offered loads are estimated, link shadow prices and path cost are computed, and the "best" path is selected under different routing policies.

5.4.3.1 Routing Policies under Deterministic Reservation Strategy

The priority MDPD policy (MDPD_D):

This policy is similar to the priority MDPD policy specified in section 4.1.4.2. A detailed description follows.

Link loss: As with the MDPD policy in section 4.1.4.2, the revenue loss for rejecting a call on an alternate path is divided into link losses. (Recall that a "link" in a VP network is actually a VP.) In order to simplify the computation of link loss, we will only study a division rule which divides the revenue loss for rejecting an alternate call evenly among the links (VP's) on the alternate path. Since all alternate paths consist of two VP's, the link loss for rejecting an alternate call is

independent of the alternate path. Therefore, the link loss, r^ℓ, under this policy is computed as follows:

$$r^\ell = \frac{r\lambda_d^\ell + 0.5r\lambda_a^\ell}{\lambda_d^\ell + \lambda_a^\ell} \tag{5.4}$$

where r is the call reward, λ_d^ℓ is the estimated offered load on the direct path, and λ_a^ℓ is the estimated offered load on all alternate paths. The reason why the cost of rejecting an alternate call is $0.5r$ in equation (5.4) is that all alternate paths consist of two links. Thus the loss of rejecting an alternate call at each link is given only half of r (due to the even division rule). Estimates for the link loads λ_d^ℓ and λ_a^ℓ are given next.

Link loads: MDPD_D estimates link-offered loads using the same procedure as the priority MDPD specified in section 4.1.4.2. However, instead of measuring the carried load of all paths that traverse this link, MDPD_D only needs to measure the carried load on the direct path and the total carried load on all alternate paths (see equation (5.4)). Specifically, the measured carried load on the direct path, $\tilde{\lambda}_d^\ell$, is the average number of direct calls carried per unit of mean call holding time. The measured carried load on alternate paths, $\tilde{\lambda}_a^\ell$, is the average number of alternate calls carried per unit of time. The offered link loads are then given by

$$\lambda_d^\ell = \frac{\tilde{\lambda}_d^\ell}{1 - B^\ell},$$

$$\lambda_a^\ell = \frac{\tilde{\lambda}_a^\ell}{1 - B^\ell},$$

where B^ℓ is the measured link blocking probability. The total offered link load on link ℓ, λ^ℓ, is the sum of offered loads on the direct and alternate paths, i.e.,

$$\lambda^\ell = \lambda_d^\ell + \lambda_a^\ell.$$

Link shadow prices: Computing the link shadow prices in VP networks when the deterministic reservation strategy is adopted is much more complicated than in traditional circuit-switched networks. Thus approximation techniques are required. The complication arises from the fact that when a VC is routed on an

alternate path, it requires a more stringent QOS at each VP on this alternate path, since QOS is an end-to-end measure and multiple VP's now constitute the end-to-end path. Thus, unlike in traditional circuit-switched networks, the link shadow prices in VP networks for adding a direct call and an alternate call are not equivalent for a given link state.

Although more accurate link shadow prices can be obtained by modeling each link as a two-dimensional MDP in which the link state keeps track of both the number of direct calls and alternate calls, the computational cost is too high. Our approach is to develop a simple approximation technique such that the link shadow prices for both direct calls and alternate calls can be estimated based on simple expressions such as equation (5.3). The main idea is to make a VP look like a link in a circuit-switched networks with a fixed number of "circuits". In a single rate circuit-switched network, both direct calls and alternate calls require one circuit. Thus the link shadow prices for adding a direct call and an alternate call are the same. However, in a VP network, alternate calls may require more capacity than direct calls due to a more stringent QOS requirement. Our approach is to measure the capacity required by an alternate call in units of the capacity required by a direct call. By doing so, a VP can then be viewed as a link with a fixed number of "circuits". A direct call always require one circuit, while an alternate call may require more than one circuit due to its stricter QOS requirement. Given that a VP has a fixed number of circuits, equation (5.3) can be applied to compute the link shadow prices for both direct calls and alternate calls. In the following, we first show how to measure the capacity required by an alternate call in units of the capacity required by a direct call. We then show how link shadow prices for both direct calls and alternate calls are estimated.

When an alternate call is added to a VP, the maximum number of direct calls that can be added to this VP may decrease by more than one. For example, consider the link studied in Figure 5.4. Let class 1 calls be alternate calls and class 2 calls be direct calls. (Recall that the QOS of class 1 is more stringent than that of class 2.) If there are no alternate calls, the link can support up to 65 direct VC's

without violating the QOS requirement. However, when one alternate call is added to the link, only 63 direct VC's can be supported (under the ADAPTIVE scheme). Thus, our approach is to define the capacity required by an alternate call as the difference between the number of the maximum number of direct VC's that can be supported on the VP before and after adding the alternate call. We refer to this capacity as the "virtual capacity" required by the alternate call because this capacity is measured in number of direct calls, or equivalently in units of the capacity required by a direct call, not in the amount of the actual capacity required to carry the alternate call. Clearly, the virtual capacity required by an alternate call depends on the number of existing alternate calls (see Figure 5.4).

In order to compute the virtual capacity required by an alternate call, we introduce some additional notation and functions. Let the link state be described by (i, j) where i is the number of existing direct VC's and j is the number of existing alternate VC's. Assume link ℓ can support up to N^ℓ direct VC's without violating the QOS requirement. On the other hand, when only alternate VC's are carried on the link, assume that N_a^ℓ is the maximum number of alternate VC's that can be supported. Since alternate calls require a more stringent QOS requirement, N_a^ℓ is no greater than N^ℓ, i.e., $N_a^\ell \leq N^\ell$. Let $\mathcal{N}^\ell(j)$ be the maximum number of direct VC's that can be supported on link ℓ when there exist j alternate virtual circuits on the link. Therefore, the quantity $\mathcal{N}^\ell(j) - \mathcal{N}^\ell(j+1)$ is the virtual capacity required to carry an alternate call when there already exists j alternate VC's on the link.

Since the capacity required to carry an alternate call is now measured in units of the capacity required by a direct call, we can map the original two dimensional state space into a one dimensional state space. Once we have a one dimensional state space, we will be able to use equation (5.3) to compute link shadow prices. (The idea is similar to the use of the Pascal approximation technique in section 4.1.3 where all classes of calls have been modeled as class 1 calls.) Define $\mathcal{V}^\ell$: $\{0, ..., N^\ell\} \times \{0, ..., N_a^\ell\} \rightarrow \{0, ..., N^\ell\}$ where $\mathcal{V}^\ell(i, j)$ is called the transformed

link state of link ℓ when there are i direct VC's and j alternate VC's on link ℓ. That is, $\mathcal{V}^\ell(i,j)$ maps a two dimensional state into a one dimensional state. The value of $\mathcal{V}^\ell(i,j)$ is defined by

$$\mathcal{V}^\ell(i,j) = \begin{cases} i + \mathcal{N}^\ell(0) - \mathcal{N}^\ell(j), & \text{if } i \leq \mathcal{N}^\ell(j) \\ \text{undefined}, & \text{otherwise}, \end{cases}$$

where $\mathcal{N}^\ell(0) - \mathcal{N}^\ell(j)$ is the capacity, in units of the capacity required by a direct call, required to support j alternate VC's. Thus $\mathcal{V}^\ell(i,j)$ gives us the total capacity, in units of the capacity required by a direct call, required to support i direct VC's and j alternate VC's. Note that, according to the definition, $\mathcal{V}^\ell(i,j) = N^\ell$ if and only if $i = \mathcal{N}^\ell(j)$. This implies that if link ℓ is at state (i,j) and $\mathcal{V}^\ell(i,j) < N^\ell$, then link ℓ can accept at least one more direct call.

The state dependent link shadow price, $p_d^\ell(i,j)$, for adding a direct call on link ℓ when there exists i direct VC's and j alternate VC's on the link is given by

$$p_d^\ell(i,j) = \frac{r^\ell E(\lambda^\ell, N^\ell)}{E(\lambda^\ell, \mathcal{V}^\ell(i,j))}. \tag{5.5}$$

Note that this exactly parallels equation (5.3).

The computation of the approximate link shadow prices for alternate calls needs to take into account the fact that adding an alternate call may decrease the maximum number of direct calls that can be supported on the link by more than one due to a more stringent QOS requirement. By the use of virtual capacity, the state-dependent link shadow prices for adding an alternate call, $p_a^\ell(i,j)$, are estimated by

$$p_a^\ell(i,j) = \sum_{v=\mathcal{V}^\ell(i,j)}^{\mathcal{V}^\ell(i,j+1)-1} \frac{r^\ell E(\lambda^\ell, N^\ell)}{E(\lambda^\ell, v)}. \tag{5.6}$$

The summation in equation (5.6) corresponds to the virtual capacity required to add an alternate call at link state (i,j).

Path cost: The cost of adding a VC on an alternate path with links ℓ_1, ℓ_2, in respective link states $(i_1,j_1),(i_2,j_2)$, is given by:

$$path\_cost = p_a^{\ell_1}(i_1,j_1) + p_a^{\ell_2}(i_2,j_2)$$

Path selection: Under the MDPD_D policy, a call is always offered to the direct path first. If it is blocked, i.e., $\mathcal{V}^{\ell}(i,j) = N^{\ell}$, the call is then offered to the alternate path with the minimum path cost. A call is rejected if the minimum path cost equals or exceeds the call reward.

The priority MDPD policy with dynamic VP capacity sharing (MDPD_DS):

In this policy, we introduce a new concept, *VP capacity borrowing*, to further reduce the blocking probability of the VP network. The idea is to allow one VP to borrow capacity from another VP if they share the same source node and the set of links in the former VP is a subset of that of the latter. For example, consider VP 1 and VP 3 in Figure 5.2. When a VC offered to VP 1 is blocked, instead of offering the VC to an alternate path which might consist of VP 3 and some other VP, the VC could be carried on VP 1 if VP 1 could "borrow" some capacity from VP 3 to carry this VC. The "borrowed" capacity would be "returned" as soon as one of existing VC's on VP 1 terminates.

Borrowing and returning capacity from one VP to another is performed only at the common source node, and is done so in the same manner as VP capacity is reserved periodically under the deterministic VP capacity reservation strategy. Note that no processing needs to be done at transit nodes. This is a very important feature because, with this new operation, call setup processing can still be done at the source node without involving transit nodes.

To ensure that the QOS of VC's on other VP's are unaffected and that processing at transit nodes can be eliminated, a VP can borrow capacity from another VP only if both VP's share the same source node. For example, in Figure 5.2, VP 1 can borrow capacity from VP 3 without affecting the QOS of VC's on VP 2 and all existing VC's on VP 3. The reason is that the increased traffic on VP 1 does not propagate to the second physical link and the amount of traffic on VP 3 which travels from the first link to the second link is decreased, not increased. On the other hand, VP 2 cannot borrow capacity from VP 3 without requiring processing at the source node of VP 3 since this source node (of VP 3) would need to be informed of the change at the

second link in order to restrict the maximum number of VC's that can be accepted on VP 3. Therefore, in order to eliminate the additional processing at other nodes, VP 2 is not allowed to borrow capacity from VP 3.

The MDPD_DS policy operates very similar to MDPD_D, except for some minor changes regarding path selection and link-load measurements. These changes are described below.

Link loads: The advantage of temporarily borrowing capacity from another VP to carry a VC is that this VC is still carried on a single VP and at the original QOS requirement. This differs from the MDPD_D policy, in which an alternate path with two VP's and a resulting more stringent per-link QOS would be required. Therefore, the VC can be viewed as a direct VC which is carried on the VP loaning the capacity. For example, in Figure 5.2, if VP 1 borrows some capacity from VP 3 to carry a VC, this VC can be counted as if it were a direct VC on VP 3. The methods used in MDPD_D for measuring the link carried loads for direct VC's and alternate VC's are slightly changed to reflect this property.

Path selection: As in MDPD_D, the MDPD_DS policy always offers a VC to the direct VP first. If the VC is blocked, MDPD_DS then tries to "borrow" some capacity from other VP's in order to temporarily assign this capacity to the direct VP. If the borrowing is successful, the VC is carried on the direct VP using this borrowed capacity. Otherwise, the VC is offered to the set of alternate paths in the same manner as MDPD_D.

Trunk reservation: It is expected that a decrease in the blocking probability on the VP which borrows capacity is offset by an increase in the blocking probability on the VP which loans the capacity. This may cause some fairness problems. For example, it may be necessary to guarantee that the blocking probability on the VP which loans its capacity under the MDPD_DS policy to be no larger than the blocking probability it would received if no alternate routing is employed. If this is the case, a trunk reservation scheme could be employed to insure the blocking probability on the VP which loans its capacity does not exceed some maximum value. Specifically, a VP will loan its capacity to other VP's only if its

residual capacity, i.e., the additional number of direct VC's that it can support without violating the QOS requirements of existing VC's, is larger than a certain threshold, R.

5.4.3.2 Routing Policies under Statistical Reservation Strategy

A major difference between the routing policies under the statistical reservation strategy and the routing policies under the deterministic reservation strategy is the computation of the link shadow prices. As stated at the end of section 5.4.2, only one class of QOS is established on each VP under the statistical reservation strategy. Therefore, the computation of link shadow prices for the routing policies under the statistical reservation strategy is exactly the same as in traditional circuit-switched networks. That is, equation (5.3) can be applied directly to compute the link shadow prices. Furthermore, the costs of adding a direct call and an alternate call are the same. With the exception of this difference, routing policies under the statistical reservation strategy are very similar to the routing policies under the deterministic reservation strategy.

The priority MDPD policy (MDPD_S):

The MDPD_S operates slightly differently from MDPD_D in the selection of the set of possible paths for each source-destination pair. As discussed in section 5.4.2, an alternate path is considered as "possible" under the statistical reservation scheme only if the sum of the QOS of the two VP's on this path satisfies the end-to-end QOS requirement.

The priority MDPD policy with dynamic VP capacity sharing (MDPD_SS):

Borrowing and returning capacity from one VP to another VP is much easier to implement under the statistical reservation scheme because all cells are statistically multiplexed on physical links and the traffic characteristics of a VC are assumed to be unaffected as the VC travels through the network. The QOS received by a VC

which is routed on the borrowed capacity is always guaranteed to be more stringent than the original QOS requirement because the VC only travels over a subset of the set of links of the borrowed VP. Thus, the only operation required at the source node is to adjust the maximum number of VC's that can be accepted on these two VP's.

The MDPD_SS policy has the same restriction on selecting the set of possible paths as MDPD_S.

5.5 Numerical Examples

In this section, we study the performance of the four routing algorithms under two VP capacity reservation strategies on a simple 5-node network through simulation. We first describe the performance metric in which we are interested, followed by a detailed description of the network configurations under the two VP capacity reservation strategies. Finally, the performances of the four routing policies are compared with those of two base-line policies.

5.5.1 Performance Metric

When evaluating a routing algorithm, the performance metric in which we are interested includes four components: the overall VC blocking probability of the network, the largest VC blocking probability of a source-destination pair among all source-destination pairs, the call admission cost, and the resource management cost. The first two metrics are considered as transmission costs while the last two metrics contribute to control costs. We assume the network topology remains fixed during the evaluation of a routing algorithm. Thus, the transmission cost of the network, which is measured by traffic-carrying capacity in [13], bandwidth cost in [47], and transmission efficiency in [66, 67], is measured by the network blocking probability in our study. However, as stated in the previous section, when VP's are allowed to dynamically borrow capacity from other VP's, it is also desirable to have a minimum blocking probability guarantee for all source-destination pairs. We will refer to this guarantee as the guarantee of the largest blocking probability among all O-D pairs.

The control cost is decomposed into a call admission cost and a resource management cost. The call admission cost is defined as the average number of times the

call admission function is invoked in setting up a VC. For example, to set up a VC on a direct path, the call admission function is invoked only once, i.e., at the source node. On the other hand, to set up a VC on an alternate path with two "links" (VP's), the call admission function must be invoked twice, once at the source node and once at the transit node (i.e., the node connecting the two VP which make up the alternate path). The resource management cost is similarly defined as the average number of times the resource management function is invoked in setting up a VC. For example, setting up a VC on a direct path does not require invoking a resource management function. Setting up a VC on the borrowed capacity requires invoking the resource management function at the source node. Due to the requirement of a more stringent QOS at each link, setting up a VC on an alternate path invokes the resource management function both at the source node and the transit node.

Other measures of VP control cost have been introduced in the literature (e.g., the cost for periodically reserving VP capacity [13]). We do not consider such costs in our simulation since we assume that this is done on a much slower time scale than the VC setup time.

5.5.2 Network Parameters

We study routing policies in a 5-node network illustrated in Figure 5.1(a). We assume links a, b, e and f all have a bandwidth of 4.632 Mbps, each with an input buffer space of 148.224 Kbits (i.e., 32 msec worth of data). The remaining links are assumed to have bandwidth 3.088 Mbps, each with input buffer space 98.816 Kbits. The VP configurations are shown in Figure 5.1(b). We assume that homogeneous voice sources, described by a two-state fluid flow model with parameters $(32000, 0.3513, 0.352)$, with QOS $Q(0.032, 10^{-6})$ are offered to the network. Recall that VC's are assumed to be generated for each source-destination pair according to a Poisson process and the VC holding times are assumed to be exponentially distributed random variables with mean of 1.

VP capacity is assigned as follows. Under the deterministic VP capacity reservation strategy, each VP is assigned a dedicated bandwidth of 1.544 Mbps and a buffer space of 49.408 Kbits. The ADAPTIVE scheme is used to establish two classes of

QOS on a VP, one for direct VC's and the other one for alternate VC's. The resulting admissible region is shown in Figure 5.4.

Under the statistical VP capacity reservation strategy, the bandwidth and buffer space of each physical link are shared among all VP's. The number of VC's reserved for each VP is set such that it is as close as possible to the number of direct VC's that can be supported on the VP when the deterministic VP capacity reservation strategy is used. The ADAPTIVE scheme is also used for establishing multiple QOS classes on a physical link. However, the large number of VC's requiring a more stringent QOS have resulted in uniformly providing the most stringent QOS to all VC's on each physical link. The final assignment of the number of VC's reserved on each VP is to reserve 88 VC's on VP 0, 4, 10, and 14 and 64 VC's on each of the remaining VP's. We observe a very interesting fact from this assignment. Due to the need to support a more stringent QOS on each physical link, the number of VC's that can be reserved on some VP's is reduced while it is increased for some other VP's. This is because, on one hand, allowing multiplexing among all VP's sharing a physical link increases the statistical multiplexing gain. On the other hand, this advantage is offset by establishing a more stringent QOS requirement on each link. We have found that, in this example, when a physical link is shared by two VP's, the increase in statistical multiplexing gain is less than the loss incurred in providing a more stringent QOS requirement. For example, link c is shared by VP 1 and 2. The maximum number of VC's that can be supported without violating a QOS guarantee of $Q(0.016, 5 \times 10^{-7})$ is 128 which is then evenly allocated to each VP. However, both VP's are able to accommodate 65 VC's without violating this QOS guarantee under the deterministic reservation strategy. On the other hand, when a physical link is shared by more than two VP's, we see the opposite effect. For example, link a is shared by VP 0, 3 and 9. The maximum number of VC's that can be supported is 216, which is much larger than $65 \times 3 = 195$. However, since VP 3 and 9 also consist of other links which are shared by only two VP's, the only possible allocation is to reserve 88 VC's on VP 0 and 64 VC's on both VP 3 and 9.

5.5.3 Numerical Results

The performance of the four routing policies are evaluated through simulations. To evaluate the trade-off between transmission and control costs as a result of MDP routing and reserving VP capacity, we also compare the performance of the four routing policies to two base-line routing policies: the DIRECT policy and the NOVP policy. In the DIRECT policy, a deterministic reservation strategy is adopted and all VC's are offered to direct VP's only. A VC blocked on the direct path is rejected. The NOVP policy does not reserve capacity on VP's and alternate routing is exercised based on the LLP approach. For a fair comparison, paths in the set of possible paths for each O-D pair are limited to those paths which consist of at most two links. The ADAPTIVE scheme is also adopted for establishing multiple classes of QOS on a physical link.

The performance of the four MDP routing algorithms and two base-line algorithms are compared in Figure 5.5. The following observations can be made.

1. MDP routing algorithms are able to provide significantly lower network blocking probability with a slightly increased control cost as compared to the DIRECT policy.

2. The lowest network blocking probability is achieved by not reserving capacity on VP's. However, this advantage is offset by the significant increase in control cost as shown in Figure 5.5(c) and Figure 5.5(d).

3. Routing policies that allow one VP to borrow capacity from another VP yield significantly lower network blocking probability than policies that do not.

4. Routing policies under the statistical VP capacity reservation strategy provide, in general, lower network blocking probability and resource management cost than policies with the deterministic strategy. (In fact, routing policies with the statistical strategy do not invoke any resource management function at all.) One exception occurs when the traffic load is very light (e.g., 50 in this example). In this case, routing policies under the deterministic strategy yield lower network

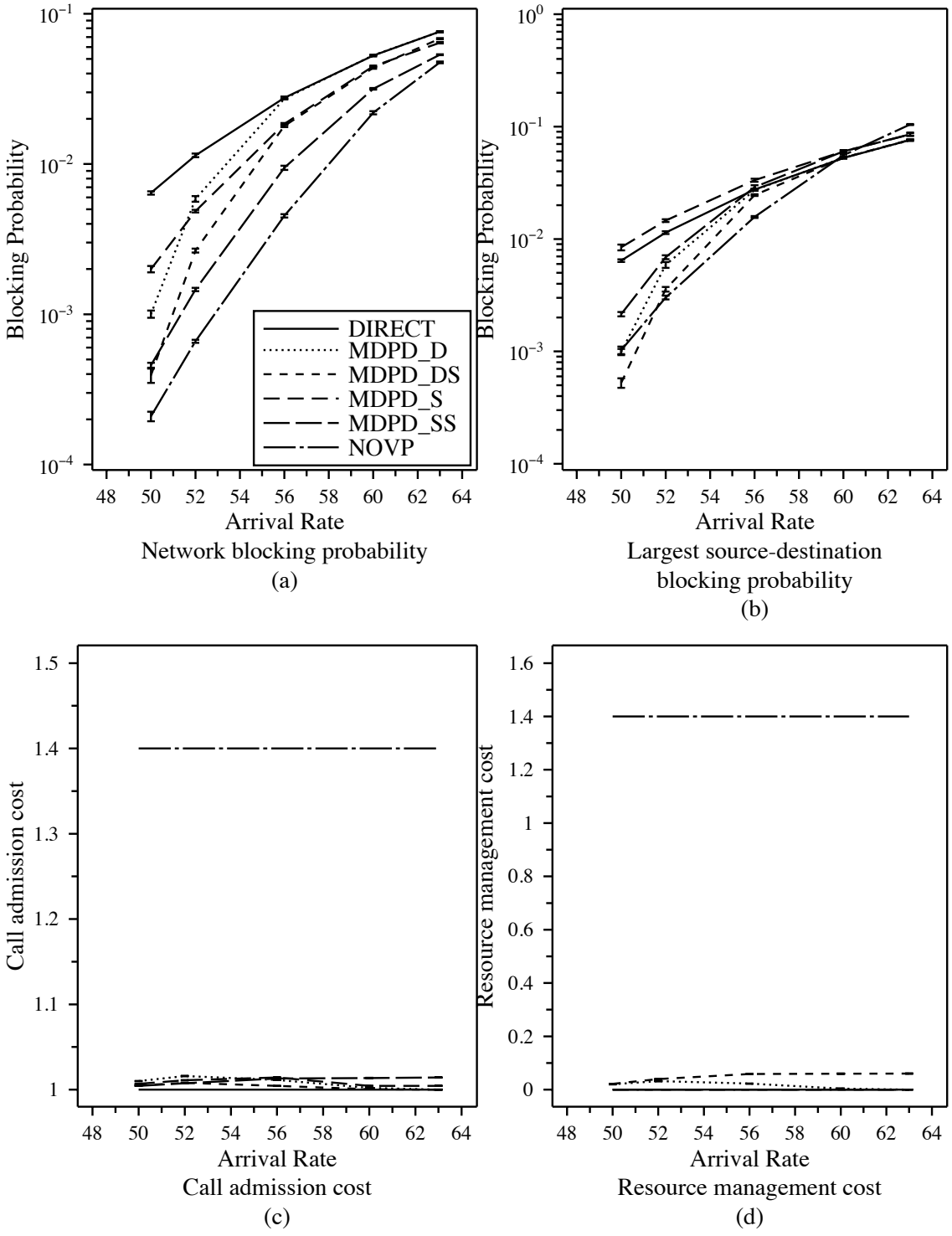

Figure 5.5 Performance of different routing policies

blocking probability because the sizes of the possible path sets are larger under these policies.

5. Although routing policies under the statistical reservation strategy yield lower network blocking probabilities, the largest blocking probability among all O-D pairs cannot be guaranteed to be less than the largest blocking probability guarantee provided by the DIRECT policy. In our simulation, a trunk reservation technique, similar to the technique used in LLP routing, has been adopted in these policies to improve the value of the guarantee of the largest blocking probability among all O-D pairs. That is, an alternate VC will be carried on a VP only if the residual capacity on the VP, i.e., the number of additional direct VC's that can be added, is larger than some threshold, R. However, due to the decrease in the number of VC's reserved and the size of the possible alternate path set for some VP's, it is impossible, under these two policies, to always guarantee the largest blocking probability among all O-D pairs to be no larger than the guarantee provided by the DIRECT policy. We consider this to be a major drawback of these two policies. On the other hand, providing a guarantee of the largest blocking probability among all O-D pairs at a slightly larger value is still easy to achieve because a dedicated number of VC's has been reserved on each VP.

6. The largest blocking probability among all O-D pairs achieved under the NOVP policy is also larger than the that of the DIRECT policy. Although it is possible to implement some reservation scheme, similar to the trunk reservation technique, to reduce the largest blocking probability among all O-D pairs under the NOVP policy, the development of such scheme is not trivial and is beyond the scope of this study. Indeed, the ease of providing a guarantee on the maximum blocking probability that will be seen by any source-destination pair is one of the advantages for reserving capacity on VP's.

5.6 Summary and Discussion

In this chapter, MDP routing in VP networks with reserved VP capacity was studied. We first examined two VP capacity reservation strategies. In the deterministic strategy we studied, dedicated resources were reserved on VP's. In the statistical strategy, instead of reserving dedicated resources, a dedicated number of VC's, each with a guaranteed QOS, was allocated to each VP. Although the second strategy was able to achieve better statistical multiplexing effect, physical links were required to provide a more stringent QOS. We also proposed a ADAPTIVE scheme for more efficiently establishing multiple classes of QOS on a physical link or VP.

Four MDP routing policies were designed and evaluated in this study. The performance metrics studied include network blocking probability, the largest blocking probability among all O-D pairs, call admission cost, and resource management cost. The performances of the four MDP routing policies were compared to the performances of two base-line routing policies. Our numerical results showed that MDP routing algorithms were able to provide significantly lower network blocking probability with a slightly increased control cost. We also observed that routing policies with the statistical reservation strategy yield, in general, a lower network blocking probability and resource management cost than policies with the deterministic strategy. However, the largest blocking probability among all O-D pairs under these two policies cannot be guaranteed to be no larger than the guarantee of the largest blocking probability among all O-D pairs provided by the DIRECT policy. Finally, the NOVP policy, which does not reserve capacity on VP's, provided the lowest network blocking probability. However, this was offset by the increase in control cost and difficulty in providing a guaranteed blocking probability for each O-D pair.

The performance of the four MDP routing algorithms is affected by both the QOS criterion and the underlying call admission scheme. In this study, we have limited our study to a single QOS criterion, which takes both cell delay and cell loss rate into account. If the buffer space at each physical link is fixed and only cell loss probability is considered in the QOS requirement, the four MDP routing policies will be able to yield lower network blocking probabilities because without the time constraint, more dedicated buffer space will be able to be allocated to each VP. Similarly, if a call

admission scheme which more precisely models the effect of statistical multiplexing (e.g., the call admission function based on a stationary approximation technique in [42]) is adopted, the routing policies with the statistical reservation strategy will be able to take more advantage of the increase of the statistical multiplexing gain of the network and, thus, will perform better. Furthermore, the drawback of being unable to provide a guarantee of the largest blocking probability among all O-D pairs which is no larger than the guarantee provided by the DIRECT policy will be mitigated because more VC's can be allocated to VP's, a result of achieving better statistical multiplexing effect.

Although the performance of routing algorithms may change when different QOS criteria are required or different call admission schemes are adopted, we emphasize that the choice of QOS criterion and the design of the call admission scheme are orthogonal issues to the routing problem in VP networks. The four MDP routing algorithms we developed in this chapter should remain unaffected by changes in the QOS criteria or in the call admission schemes used.

CHAPTER 6

SUMMARY AND FUTURE WORK

6.1 Summary of the Dissertation

In this dissertation, we have examined the routing problem in future high-speed networks, taking into account the new network characteristics introduced by new technologies and new applications. We have examined the influence of these new network characteristics on the routing problem and designed and evaluated adaptive routing algorithms to account for these new features.

In the first part of this thesis, we examined the influence of network parameters such as routing algorithms, propagation delays, aspects of the admission control procedure, and processing capacities on the mean call setup time and call blocking probability. The influence of routing on the mean call setup time and call processing capacity was examined under five routing algorithms: three well-known algorithms in the circuit-switching literature and two controlled flooding versions of these algorithms. The influence of call admission control was modeled and examined with three different forms of a "call admission control function". We developed analytical models for evaluating the average call setup time and call blocking probability for these five algorithms and validated our analysis by simulations. The results of our study showed that the average call setup delay and call throughput are very sensitive to call processing service time, admission control function, and routing algorithm, but are relatively insensitive to the propagation delay.

The contribution of this work is to quantitatively study these important factors on call setup time and call blocking probability. In order to examine the influence of call processing delay and propagation delay, resources (including processing resources and transmission resources) held by blocked calls during the call setup time were explicitly modeled; this has been ignored in all past research. We observe that it is important to model resources held by blocked calls only when the call processing

(service) time or the propagation delay is not significantly small as compared to call holding time. This might be true for some applications in high-speed networks such as some forms of file transfer, or, if we do the call setup at the burst level, for bursty sources.

The main focus of the remainder of the thesis was on designing and evaluating adaptive routing algorithms for future high-speed networks based on Markov decision theory. We first focused on developing computationally feasible MDP routing algorithms for high-speed networks without the use of VP's. Since the required computational complexity of MDP routing for multi-class networks becomes extremely high, we developed an approximation scheme toward MDP routing for multi-class networks. The algorithm maintains low computational complexity while still providing quite accurate routing information. We showed that routing algorithms based on our approximate scheme yield competitive performance as compared to the routing algorithms which require considerably more computation for obtaining more accurate routing information. We also demonstrated that MDP routing is more general and efficient than the LLP approach.

We have also shed light upon interesting properties of MDP routing. First, we found that the manner in which the revenue loss of a multi-link call is distributed into individual link losses affects the performance of MDP routing algorithms significantly. Second, we showed, through a simple network model, that the performance of MDP-based routing algorithms is insensitive to the validity of the link independence assumption.

Since the VP concept has been proposed to simplify traffic control and resource management in future high-speed networks, in the last part of this thesis we focused on designing and evaluating MDP routing algorithms for VP networks. We first examined two VP capacity reservation strategies and showed that routing policies are able to yield significantly lower blocking probability when the statistical reservation strategy is adopted. We designed and evaluated four MDP routing algorithms for homogeneous VP networks and showed that the network blocking probability can be significantly reduced by MDP routing. We also have proposed a VP capacity sharing

concept and showed that routing algorithms which account for this concept are able to further reduce the network blocking probability.

The major contribution of this dissertation was to design and evaluate routing algorithms in the context of high-speed networks taking into account new technologies (such as the VP concept) and new application-driven requirements (such as diverse applications with divergent QOS requirements). Based on Markov decision theory, we have demonstrated a possible way for defining the cost of a path and developed an efficient way for computing such a cost. Based on the estimated path cost, we have proposed several computationally feasible MDP routing algorithms for future high-speed networks. In addition, our simulation results on NSFNET have demonstrated the feasibility of using these algorithms in general high-speed networks.

6.2 Future Work

The work reported in this dissertation can be extended in a number of directions. Among several interesting directions, we suggest four problems:

Multicast routing: One of the most popular applications in high-speed networks is multimedia, multiparty conferencing services. Multicast routing is required for providing such services. Most researchers have focused on designing near optimal, polynomial-time multicast routing algorithms and assumed that the cost of each link is given. However, without knowing the exact meaning of the "link cost", it is difficult to claim the performance of a heuristic algorithm is good or bad. We believe that our approximate link shadow prices can be used to close this gap. Thus we plan to design multicast routing algorithms based on approximate link shadow prices. As in point-to-point routing, a multicast call should not be always granted even if the network has enough resources to carry the call at the request time. A new criterion, similar to call reward in point-to-point routing, must be devised to determine whether it is worthwhile to carry a multicast call for a given network traffic load. It is also very important to be able to evaluate the performance of a multicast routing algorithm more precisely. In past research, the performance of a heuristic multicast routing

algorithm is evaluated by comparing the difference between the costs of multicast paths found by the heuristic algorithm and the costs of multicast paths found by the optimal algorithm. Without defining the link cost, we feel that the results of such performance evaluation are uninterpretable. We believe that the *fractional reward loss* is a better performance metric for evaluating the performance of multicast routing algorithms.

The multicast routing problem is even more pronounced in VP networks where the problem is coupled with the physical network configuration. For example, the set of physical links of one VP may be a subset of that of another VP. In such a case, a multicast connection of the former VP may have a "free ride" on the same multicast connection of the latter VP. On the other hand, the densely connected topology of VP networks may simplify the design of multicast routing algorithms. For example, we have learned from point-to-point routing that an alternate path should be limited to consist of at most two links. Therefore, a polynomial-time multicast routing algorithm can be obtained by applying Dijkstra's shortest path algorithm and limiting the search depth to be two.

State Aggregation: Although in our link simplified model, we have reduced the link state space significantly by defining the link state as the number of busy circuits on the link, the processing cost for switches to keep such link state information may still be too expensive in future high-speed networks. Thus a state aggregation technique, such as that proposed in RTNR [8], may be necessary for state-dependent routing in high-speed networks. State aggregation is not very difficult to implement in RTNR because of its use of the LLP approach. The question of how to apply the state aggregation technique to MDP routing is much more complex and demands further research. Mitra *et al.* have presented an analytical model for evaluating the performance of LLP routing algorithms under the use of a state aggregation technique (the so-called ALBA routing) in [62]. It would also be interesting to see if a similar analytical model could be developed for MDP routing with state aggregation.

Implementing MDP Routing Algorithms: In designing our MDP routing algorithms, we have assumed that when a call arrives at a source node, the instantaneous global network state (specifically, the link shadow prices of all links), is available at the source node. This is an unrealistic assumption because switches in real networks only exchange state information periodically, e.g., once every 5 minutes. Thus, just as Bellcore's DR5 is a practical approximation of the SDR scheme [22], we plan to develop practical versions of our MDP routing algorithms and evaluate their performance.

Dynamic VP Capacity Allocation: One approach toward reducing network blocking probability when capacity is reserved on VPs is to have a dynamic VP capacity reservation algorithm which can adapt to dynamic changes in network traffic flows. In our future research, we are interested in not only designing such algorithms but also integrating these capacity allocation algorithms into routing algorithms.

APPENDIX A

ANALYTICAL MODELS FOR CONTROLLED-FLOODING ALGORITHMS IN HOMOGENEOUS NETWORKS

In this appendix, we present the computation for the link- and node-offered call arrival rate, average call setup delay, and call blocking probability for controlled-flooding versions of OOC routing and SOC/Crankback routing in homogeneous networks.

A.1 OOC Routing Rule

One recent work which also studies the performance of a flooding algorithm is [76]. In [76], the author presents an analysis for the performance of the MKS circuit-switching communication system designed by PKI. The call setup procedure in this system makes use of a flooding scheme to find a free path between any two subscribers (nodes). The quantity of interest in [76] is the probability that the *first* call setup request message received at the destination node has followed a given path. As discussed in [76], an exact way of evaluating this probability is very hard. However, by assuming the waiting times along different paths are statistically independent, we are able to compute the quantity we need. Our analysis differs from [76] by the fact that we have a predefined routing tree for each O-D pair. This simplifies our analysis and enables us to obtain close form expressions.

In the analyses of flooding algorithms, a critical quantity that must be computed is the probability that a call request received by the destination node is a duplicated request. In the following analysis, we thus focus on the computation of this probability. Note that the notation and assumptions used in Chapter 3.3 are also adopted here. To be consistent with the sequential routing algorithms, we also assume that the destination node discards the duplicated request in zero time.

Let $P_{dup}(i)$ be the probability that a call request received through path i is duplicated. To compute $P_{dup}(i)$, we need to introduce some more notation. Let

T_i be the total waiting time (processing delay + round trip propagation delay) along path i given that the call request is successfully received through path i. We assume that processing delay (queueing + service) is exponentially distributed with mean $1/\mu$. The sum of processing delay through ℓ_i nodes is then given as an Erlang-ℓ_i random variable. The sum of propagation delays is a deterministic random variable with value $\ell_i D_p$. The CDF and pdf of T_i can be expressed as:

$$F_{T_i}(t) = 1 - e^{-\mu(t-\ell_i D_p)}\left(\sum_{i=0}^{\ell_i-1} \frac{(\mu(t-\ell_i D_p))^i}{i!}\right), \quad t \geq \ell_i D_p,$$

$$f_{T_i}(t) = \frac{\mu(\mu(t-\ell_i D_p))^{\ell_i-1} e^{-\mu(t-\ell_i D_p)}}{(\ell_i-1)!}, \quad t \geq \ell_i D_p.$$

To compute $P_{dup}(i)$, let us consider the situation where call requests are successfully set-up on paths $i_1, ..., i_m$ and i, the one under consideration. Let S be a set of paths such that $S = \{i_1, ..., i_m\}$. Let the random variable T_j be the total waiting time on path j and the random variable T^S_{min} be the time for the destination node to receive the first call request from one of the paths in S. If $\{T_j\}_{j \in S}$ is a sequence of independent random variables, T^S_{min} is given by the following equation.

$$F_{T^S_{min}}(t) = 1 - \prod_{j \in S}[1 - F_{T_j}(t)].$$

$P_{dup}(i)$ is then given by

$$P_{dup}(i) = \sum_{\substack{i \notin S \\ S \subseteq \mathcal{S}}} P'_S P(T^S_{min} < T_i)$$

where $\mathcal{S}$ is the set of all possible sets of paths between the same O-D pair of path i and P'_S is the probability that the call request are successfully set-up only on paths in S given that the call request is successfully set-up on path i.

$$P'_S = \prod_{j \in S}(1 - P_{fail}(j)) \prod_{j \notin S \bigcup \{i\}} P_{fail}(j).$$

Link-offered call requests

The offered call request rate of each class of calls at the link under consideration is computed as follows. The reader is referred to Section 3.3.3 for the definition of notations used.

$$\gamma_{i,0} = \frac{\lambda}{M}, \quad i = 1, ..., k,$$

$$\begin{aligned}\gamma_{i,j} &= \gamma_{i,j-1}(1-\mathcal{L}), \\ &= \frac{\lambda}{M}(1-\mathcal{L})^j, \quad j=1,...,\ell_i.\end{aligned}$$

Each class of calls can then be divided into three types. The arrival rate of the first type of call, which will eventually be accepted if not blocked on the link under investigation, is referred to as $\gamma^s_{i,j}$; the arrival rate of the second type of calls, which will be blocked downstream if not blocked on this link, is referred to as $\gamma^f_{i,j}$; the third type of calls, which has arrival rate $\gamma^{dup}_{i,j}$, will eventually be rejected not because it is blocked on some downstream link but because it is a duplicated request for the same call. The arrival rates and call holding times for these types of traffic are given by:

$$\begin{aligned}\gamma^s_{i,j} &= \gamma_{i,j}(1-\mathcal{L})^{\ell_i-j}(1-P_{dup}(i)), \quad i=1,...,k \;\; j=0,1,...,\ell_i, \\ \gamma^f_{i,j} &= \gamma_{i,j}(1-(1-\mathcal{L})^{\ell_i-j}), \quad i=1,...,k \;\; j=0,1,...,\ell_i, \\ \gamma^{dup}_{i,j} &= \gamma_{i,j}(1-\mathcal{L})^{\ell_i-j}P_{dup}(i), \quad i=1,...,k \;\; j=0,1,...,\ell_i, \\ \delta^s_{i,j} &= (\ell_i-j)*(\bar{T}+D_p)+D_p+\bar{\tau}, \\ \delta^f_{i,j} &= \bar{N}_{i,j}(\bar{T}+D_p), \\ \delta^{dup}_{i,j} &= (\ell_i-j)*(\bar{T}+D_p)+D_p.\end{aligned}$$

Recall that $\bar{N}_{i,j}$ is the expected number of nodes a call setup message visits after the jth node on path i given that it has successfully reserved resources at the jth node and fails at some node further down path i. $\bar{N}_{i,j}$ is given by equation (3.3).

Node-offered call requests

The aggregated rate of offered calls to a node is computed as follows,

$$G = \lambda + M\sum_{i=1}^{k}\sum_{j=1}^{\ell_i}\gamma_{i,j}.$$

Call setup delay and call blocking probability

Recall that

$$P_{fail}(i) = 1-(1-\mathcal{L})^{\ell_i+1}$$

and the end-to-end call blocking probability is

$$P_{rej} = \prod_{i=1}^{k} P_{fail}(i).$$

To compute the average call setup time, let T^S_{min} be the call setup time given that call requests are successfully set up on paths in $S \subseteq \mathcal{S}$. $\bar{T}^S_{min}$ can be obtained by

computing the Laplace transform of $f_{T^S_{min}}(t)$. The average call set-up delay can then be computed as follows.

$$T_{setup} = \sum_{S \subseteq \mathcal{S}} P_S \bar{T}^S_{min}$$

where P_S is the probability that the call request are successfully set up only on paths in S given that at least one call request is successfully set up,

$$P_S = \frac{\prod_{j \in S}(1 - P_{fail}(j)) \prod_{j \notin S} P_{fail}(j)}{1 - P_{rej}}.$$

A.2 SOC/Crankback Routing Rule

The fact that a call request received by the destination node can be a duplicate request makes the analysis much more complicated. We first introduce some additional notations:

- $\mathcal{T}_{i_1,...,i_\ell}$: Let $\mathcal{T}_{i_1,...,i_\ell}$ be the delay of the first call request sent by node $(i_1, ..., i_\ell)$ to the destination given that at least one call request originated from node $(i_1, ..., i_\ell)$ is received successfully by the destination node. $\mathcal{T}_{i_1,...,i_\ell}$ is approximated by following:

$$\mathcal{T}_{i_1,...,i_\ell} = \hat{T}_{i_1,...,i_\ell} + \mathcal{D}_p + T \tag{A.1}$$

  where random variable T is the processing delay required at node $(i_1, ..., i_\ell)$ and random variable $\mathcal{D}_p$ is the propagation delay. $\hat{T}_{i_1,...,i_\ell}$ is the time from a call request being successfully forwarded to node $(i_1, ..., i_\ell)$'s children in the routing tree until the call request is received by the destination node. Thus $\hat{T}_{i_1,...,i_\ell} = 0$ if $(i_1, ..., i_\ell)$ is label for the destination node. Otherwise, let m be a subset of the children of node $(i_1, ..., i_\ell)$. The CDF of $\hat{T}_{i_1,...,i_\ell}$ is then given by:

$$F_{\hat{T}_{i_1,...,i_\ell}}(t) = \sum_{m \subseteq \mathcal{M}_{(i_1,...,i_\ell)}} P_m F_{T^m_{min}}(t). \tag{A.2}$$

  where $\mathcal{M}_{(i_1,...,i_\ell)}$ is the set of all possible subsets of the children of node $(i_1, ..., i_\ell)$. The term P_m in equation (A.2) is the probability that the call requests sent to the nodes in m are eventually received by the destination node.

$$P_m = \frac{\prod_{(i_1,...,i_\ell,j)\in m}(1-\mathcal{L})(1-\bar{P}_{i_1,...,i_\ell,j})\prod_{(i_1,...,i_\ell,j')\notin m}(\mathcal{L}+(1-\mathcal{L})\bar{P}_{i_1,...,i_\ell,j'})}{1-\bar{P}_{i_1,...,i_\ell}}. \tag{A.3}$$

The term $\bar{P}_{i_1,...,i_\ell}$ in equation (A.3) is the probability that a call request is blocked at node $(i_1,...,i_\ell)$ or later. The computation of this probability is the same as in the sequential crankback routing rule.

The random variable T^m_{min} in equation (A.2) is the time from the call request is successfully forwarded by node $(i_1,...,i_\ell)$ until the call request first received by the destination node given that only call requests forwarded by nodes in m are successfully received. Thus T^m_{min} is the minimum of $\mathcal{T}_{i_1,...,i_\ell,j}$'s, for all $(i_1,...,i_\ell,j)\in m$; i.e.,

$$T^m_{min} = \min_{(i_1,...,i_\ell,j)\in m} \mathcal{T}_{i_1,...,i_\ell,j}. \tag{A.4}$$

The distribution of $\min(\mathcal{T}_{i_1,...,i_\ell,j})$ can be computed as in the case of parallel OOC routing rule.

- $BT_{i_1,...,i_\ell}$: the delay until node $(i_1,...,i_\ell)$ determines that all of the call requests that it forwarded are rejected given that they are rejected. It is defined as,

$$BT_{i_1,...,i_\ell} = \max_{1\le j\le k_{i_1,...,i_\ell}} (BT_{i_1,...,i_\ell,j}) + T + \mathcal{D}_p. \tag{A.5}$$

  $BT_{i_1,...,i_\ell} = 0$ if $(i_1,...,i_\ell)$ is label for the destination node.

- $P_{dup}((i_1,...,i_\ell),j)$: Let $P_{dup}((i_1,...,i_\ell),j)$ be the probability that a call request sent by node $(i_1,...,i_\ell)$ on its jth outgoing link is a duplicated request given that the call request is eventually successfully received by the destination node. Let us denote the jth outgoing link of node $(i_1,...,i_\ell)$ by ℓ. For ease of explanation, let us introduce some additional notations.

  - Let $\mathcal{S}$ be the set of all possible paths between the O-D pair under consideration. Note that different paths may share common nodes and links.
  - Let S_ℓ be the set of paths which contain link ℓ. That is, $\forall p\in S_\ell, \ell\in p$.

- Denote the complement of set S by $\bar{S}$. That is, $S \bigcup \bar{S} = \mathcal{S}$.
- Let $A(\ell)$ be the event that at least one call setup message is successfully received through a path in S_ℓ (i.e., at least one successfully received message has successfully reserved some resource on link ℓ.)
- Let $A^{\ell}_{S_1,S_2}$ be the event that call setup messages through paths in $S_1 \bigcup S_2$ are successfully received and messages through paths in $\bar{S}_1 \bigcap \bar{S}_2$ are blocked, where $S_1 \subseteq S_\ell$ and $S_2 \subseteq \bar{S}_\ell$.
- Let $Z(\ell)$ be the set of nodes between and including the source node and node $(i_1, ..., i_\ell)$. That is, $Z(\ell) = \{(0), (i_1), (i_1, i_2), ..., (i_1, ..., i_\ell)\}$.
- Let $\psi(S, a)$ be a subset of S such that $\forall p \in \psi(S, a), a \in p$.
- Let $\chi(a, S)$, where $\forall p \in S, a \in p$, be the time between when node a sends messages and the destination node receives the first message given that messages sent through paths in S are all successfully received.

For a given event $A^{\ell}_{S_1,S_2}$, a successfully received message through link ℓ is a duplicated request if at least one message sent through a path in S_2 is received first by the destination node. Thus, $P_{dup}(\ell | A^{\ell}_{S_1,S_2})$ is given by

$$P_{dup}(\ell | A^{\ell}_{S_1,S_2}) = 1 - P(\bigwedge_{a \in Z(\ell)} \chi(a, S_1) < \chi(a, \psi(S_2, a))). \quad \text{(A.6)}$$

Therefore, $P_{dup}(\ell)$ is given by

$$P_{dup}(\ell) = \sum_{\substack{S_1 \subseteq S_\ell \\ S_1 \neq \emptyset}} \sum_{S_2 \subseteq \bar{S}_\ell} P(A^{\ell}_{S_1,S_2} | A(\ell)) P_{dup}(\ell | A^{\ell}_{S_1,S_2}) \quad \text{(A.7)}$$

where $P(A^{\ell}_{S_1,S_2} | A(\ell))$ is the conditional probability for event $A^{\ell}_{S_1,S_2}$ given event $A(\ell)$. Clearly, the computation is combinatorial and is very cumbersome. As noted in [76], approximation techniques are needed for general large networks. However, for the sparse network such as NSFNET we study in our numerical example, the computation is still tractable.

- $\bar{P}_{i_1,...,i_\ell}$: the probability that a call request is blocked at node $(i_1, ..., i_\ell)$ or later given that it has reached node $(i_1, ..., i_\ell)$. This probability is computed as we did in analysis of the crankback routing rule.

Link-offered call requests

The equations used to compute the rate of link-offered calls are presented without further discussion.

$$\begin{aligned}
\gamma_{(0),j} &= \frac{\lambda}{M}, \quad j = 1, ..., k_0, \\
\gamma_{(i),j} &= \gamma_{(0),i}(1-\mathcal{L}), \quad i = 1, ..., k_0, \ \ j = 1, ..., k_i, \\
\gamma_{(i_1,...,i_\ell),j} &= \gamma_{(i_1,...,i_{\ell-1}),i_\ell}(1-\mathcal{L}), \quad j = 1, ..., k_{i_1,...,i_\ell}, \\
\gamma^s_{(i_1,...,i_\ell),j} &= \gamma_{(i_1,...,i_\ell),j}(1-\bar{P}_{i_1,...,i_\ell,j})(1-P_{dup}((i_1,...,i_\ell),j)), \\
\gamma^f_{(i_1,...,i_\ell),j} &= \gamma_{(i_1,...,i_\ell),j}\bar{P}_{i_1,...,i_\ell,j}, \\
\gamma^{dup}_{(i_1,...,i_\ell),j} &= \gamma_{(i_1,...,i_\ell),j}(1-\bar{P}_{i_1,...,i_\ell,j})P_{dup}((i_1,...,i_\ell),j), \\
\delta^s_{(i_1,...,i_\ell),j} &= \bar{T}_{i_1,...,i_\ell,j} + D_p + \bar{\tau}, \\
\delta^f_{(i_1,...,i_\ell),j} &= \bar{BT}_{i_1,...,i_\ell,j} + D_p, \\
\delta^{dup}_{(i_1,...,i_\ell),j} &= \bar{T}_{i_1,...,i_\ell,j} + D_p.
\end{aligned} \tag{A.8}$$

Node-offered call requests

Recall that $G_{i_1,...,i_\ell}$ is the rate at which call requests reach this node as the $(i_1, ..., i_\ell)$ node for all source/destination pairs in the network. $G_{i_1,...,i_\ell}$'s can be easily computed by

$$\begin{aligned}
G_0 &= \lambda, \\
G_i &= \lambda(1-\mathcal{L}), \quad i = 1, ..., k_0, \\
G_{i_1,...,i_\ell} &= G_{i_1,...,i_{\ell-1}}(1-\mathcal{L}), \\
&= \lambda(1-\mathcal{L})^\ell.
\end{aligned}$$

The node-offered call request rate, G, is then given by

$$G = \sum_{\forall(i_1,...,i_\ell)} G_{i_1,...,i_\ell}.$$

Call setup delay and call blocking probability

Recall that the source node has the label "0". Thus the end-to-end call blocking probability is given by

$$P_{rej} = \bar{P}_0,$$

and the average call set-up delay is given by

$$\bar{T}_{setup} = \bar{T}_0.$$

APPENDIX B

ANALYTICAL MODELS FOR HETEROGENEOUS NETWORKS

With the same analytical technique, the analytical model can be extended to heterogeneous networks. In this Appendix, we show how the homogeneous assumption can be relaxed.

Without the homogeneity assumption, each O-D pair has its own routing tree, each node may have a different processing delay, each link may have a different blocking probability and propagation delay. Thus we need to introduce some new notation:

- W: the set of all O-D pairs, $W = \{w\}$.
- R_w: the routing tree of O-D pair w.
- λ_w: the external call arrival rate for O-D pair w.
- $\bar{T}_x$: the average processing delay through node x.
- $\mathcal{L}_{(i,j)}$: the steady state blocking probability of link (i,j).
- $D_{(i,j)}$: the propagation of link (i,j).

The network performance is obtained by examining every O-D pair. In the following analysis, we focus on an arbitrary O-D pair w.

B.1 Sequential Rules

B.1.1 OOC Routing Rule

Assume that there are k_w paths in the routing tree R_w. R_w is an ordered set of paths, that is,

$$R_w = (R_w^1, R_w^2, ..., R_w^{k_w}),$$

$$R_w^i = (a_{w,i,1}, ..., a_{w,i,\ell_w^i+1}),$$

where R_w^i is the ith path in routing tree R_w with $\ell_w^i - 1$ intermediate nodes, $a_{w,i,1}$ is the source node and a_{w,i,ℓ_w^i+1} is the destination node $(\forall i)$.

The following notation is used for the analysis of OOC routing rule:

- Ψ_x: the set of paths which consists of node x,

$$\Psi_x = \{R_w^i | x \in R_w^i\}.$$

- $\Psi_{(x,y)}$: the set of paths which consists of link (x, y),

$$\Psi_{(x,y)} = \{R_w^i | (x, y) \in L(R_w^i)\}.$$

- $L(r)$: a function yields the set of links on route r. For example, Let $r = (a_1, a_2, ..., a_\ell)$. $L(r)$, the set of links in route r, is

$$L(r) = \{(a_1, a_2), (a_2, a_3), ..., (a_{\ell-1}, a_\ell)\}.$$

- $Sub(r, i, j)$: a sub-vector of r defined by

$$Sub(r, i, j) = (a_i, a_{i+1}, ..., a_j), \ 1 \leq i \leq j \leq \ell.$$

- $P_{fail}^{w,i}$: the probability that a call request is blocked on the ith path of R_w.

- $\mathcal{T}_{R_w^{i,j}}$: the delay from the time a call request is sent by the jth node of path R_w^i until it is successfully received by the destination node given that the call request will be eventually successfully received.

- $BT_{R_w^{i,j}}$: the delay from the time a call request is sent by the jth node of path R_w^i until it is blocked at some node downstream given that the call request will be blocked some node later on.

The computation is very similar to homogeneous case, we, thus, present the equations without further discussion.

$$P_{fail}^{w,i} = \sum_{j=1}^{\ell_w^i} \mathcal{L}_{L(Sub(R_w^i,j,j+1))} [\prod_{(a,b) \in L(Sub(R_w^i,1,j))} (1 - \mathcal{L}_{(a,b)})]$$

$$\bar{T}_{R_w^{i,j}} = \sum_{a \in Sub(R_w^i, j+1, \ell_w^i)} \bar{T}_a + \sum_{(a,b) \in L(Sub(R_w^i, j, \ell_w^i + 1))} D_{(a,b)}$$

$$\bar{BT}_{R_w^{i,j}} = \sum_{n=j+1}^{\ell_w^i} [(\sum_{a \in Sub(R_w^i, j+1, n)} \bar{T}_a + \sum_{(a,b) \in L(Sub(R_w^i, j, n))} D_{(a,b)}) P_{R_w^{i,j,n}}]$$

where $P_{R_w^{i,j,n}}$ is the probability that it is blocked at nth node given that it is blocked at some node after jth node on path R_w^i.

$$P_{R_w^{i,j,n}} = \frac{[\prod_{(a,b) \in L(Sub(R_w^i, j+1, n))}(1 - \mathcal{L}_{(a,b)})]\mathcal{L}_{L(Sub(R_w^i, n, n+1))}}{\sum_{m=j+1}^{\ell_w^i}[\prod_{(a,b) \in L(Sub(R_w^i, j+1, m))}(1 - \mathcal{L}_{(a,b)})]\mathcal{L}_{L(Sub(R_w^i, m, m+1))}}$$

Node-offered call requests

Consider the node x which is reached as the jth node of path i in route tree R_w. Let g_{x,R_w^i} be the call requests offered to node x from path R_w^i. g_{x,R_w^i} is given by:

$$g_{x,R_w^i} = \begin{cases} \lambda_w[\prod_{m=1}^{i-2} P_{fail}^{w,m}](P_{fail}^{w,i-1} - \mathcal{L}_{L(Sub(R_w^{i-1}, 1, 2))}) & x \text{ is the source node of } R_w, \\ \lambda_w[\prod_{m=1}^{i-1} P_{fail}^{w,m}][\prod_{(a,b) \in L(Sub(R_w^i, 1, j))}(1 - \mathcal{L}_{(a,b)})] & \text{otherwise.} \end{cases}$$

The aggregated traffic offered to node x, G_x, is then given by

$$G_x = \sum_{R_w^i \in \Psi_x} g_{x,R_w^i}.$$

Link-offered call requests

Now consider the link (x,y) which is reached as the jth link of path R_w^i. Let $\gamma_{(x,y),R_w^i}$ be the traffic offered to this link from path R_w^i.

$$\gamma_{(x,y),R_w^i} = \lambda_w[\prod_{m=1}^{i-1} P_{fail}^{w,m}][\prod_{(a,b) \in L(Sub(R_w^i, 1, j))}(1 - \mathcal{L}_{(a,b)})]$$

As in homogeneous, the link-offered traffic is further classified into two traffic streams:

- $\gamma_{(x,y),R_w^i}^s$: the traffic which will be successfully set up,

$$\gamma_{(x,y),R_w^i}^s = \lambda_w[\prod_{m=1}^{i-1} P_{fail}^{w,m}](1 - P_{fail}^{w,i}).$$

The mean call holding time, $\delta_{(x,y),R_w^i}^s$, is given by

$$\delta_{(x,y),R_w^i}^s = \bar{\tau} + \bar{T}_{R_w^{i,j}}.$$

- $\gamma^{f}_{(x,y),R^i_w}$: the traffic which will be blocked at some hop down the path,

$$\gamma^{f}_{(x,y),R^i_w} = \gamma_{(x,y),R^i_w} - \gamma^{s}_{(x,y),R^i_w}.$$

The mean call holding time, δ^{f}_{xy,R^i_w}, is given by

$$\delta^{f}_{(x,y),R^i_w} = \bar{BT}_{R^{i,j}_w}.$$

Call setup delay and call blocking probability

The end-to-end call blocking probability, P_{rej} is given by

$$P^{w}_{rej} = \prod_{i=1}^{k_w} P^{w,i}_{fail}$$

Let $\bar{T}_{R^i_w}$ be the expected time for the source node of w to receive a blocking message from path i given that it is blocked at path i.

$$\bar{T}_{R^i_w} = \sum_{j=2}^{\ell^i_w} \frac{\mathcal{L}_{L(Sub(R^i_w,j,j+1))}[\prod_{(a,b)\in L(sub(R^i_w,1,j))}(1-\mathcal{L}_{(a,b)})]}{P^{w,i}_{fail}} \left[\sum_{a\in Sub(R^i_w,1,j)} \bar{T}_a + \sum_{(a,b)\in L(Sub(R^i_w,1,j))} D_{(a,b)}\right]$$

Finally, the average call set up delay for O-D pair w, $\bar{T}^{w}_{setup}$, is given by

$$\bar{T}^{w}_{setup} = \sum_{i=1}^{k_w}[\sum_{j=1}^{i-1} \bar{T}_{R^j_w} + \sum_{a\in RD^i_w} \bar{T}_a + \sum_{(a,b)\in RD^i_w} D_{(a,b)}]\frac{(1-P^{w,i}_{fail})\prod_{j=1}^{i-1} P^{w,j}_{fail}}{1-P^{w}_{rej}}$$

B.1.2 SOC Routing Rule

In the following analysis, the routing tree of each O-D pair is labeled as in the homogeneous case. The set of all nodes that are directly connected to the destination node of O-D pair w by a link is now denoted by $\mathcal{D}_w$. We also denote the set of O-D pairs which has node x in its routing tree by Ψ_x:

$$\Psi_x = \{w | x \in R_w\}.$$

And the set of O-D pairs which has link (x,y) in its routing tree is denoted by $\Psi_{(x,y)}$:

$$\Psi_{(x,y)} = \{w | (x,y) \in L(R_w)\}.$$

where $L(R_w)$ is the set of links in routing tree R_w.

Now we define some notation that will be used in the analysis of SOC as well as Crankback routing rules.

- $\bar{P}^{w}_{i_1,...,i_\ell}$: the probability that a call request is blocked at node $(i_1,...,i_\ell)$ or later in tree R_w.

- $\bar{BT}^{w}_{i_1,...,i_\ell}$: the expected time for node $(i_1,...,i_\ell)$ to know that a call request is blocked given that the call request is blocked at node $(i_1,...,i_\ell)$ or later.

- $\bar{T}^{w}_{i_1,...,i_\ell}$: the expected time for node $(i_1,...,i_\ell)$ to know that a call request is successfully set up given the call request is successfully set up through node $(i_1,...,i_\ell)$.

- $g_{(i_1,...,i_\ell),w}$: traffic offered to node $(i_1,...,i_\ell)$ in routing tree R_w.

- $g^{j}_{(i_1,...,i_\ell),w}$: the call request traffic offered from node $(i_1,...,i_\ell)$ to its jth child in routing tree R_w.

- G_x: the aggregated traffic offered to node x.

- $\gamma_{(x,y),w}$: the call request offered traffic on link (x,y) corresponding to O-D pair w.

- $\gamma^{s}_{(x,y),w}$: the portion of $\gamma_{(x,y),w}$ that will be successfully set up.

- $\gamma^{f}_{(x,y),w}$: the portion of $\gamma_{(x,y),w}$ that will be blocked.

We also define Ω^{w}_{x} as the set of outgoing links of node x in routing tree R_w. Following recursive equations are used to compute $\bar{P}^{w}_{i_1,...,i_\ell}$, $\bar{BT}^{w}_{i_1,...,i_\ell}$ and $\bar{T}^{w}_{i_1,...,i_\ell}$.

$$\bar{P}^{w}_{i_1,...,i_\ell} = \left(\sum_{j=1}^{k_{i_1,...,i_\ell}} \Big(\prod_{(a,b)\in Sub(\Omega^{w}_{(i_1,...,i_\ell)},1,j-1)} \mathcal{L}_{(a,b)} \Big)(1-\mathcal{L}_{((i_1,...,i_\ell),(i_1,...,i_\ell,j))})\bar{P}^{w}_{i_1,...,i_\ell,j}] \right) + \prod_{(a,b)\in\Omega^{w}_{i_1,...,i_\ell}} \mathcal{L}_{(a,b)}$$

with $\bar{P}^{w}_{i_1,...,i_\ell,1} = 0, \ \forall (i_1,...,i_\ell) \in \mathcal{D}_w$.

$$\bar{BT}^{w}_{i_1,...,i_\ell} = \sum_{j=1}^{k_{i_1,...,i_\ell}} \frac{\prod_{(a,b)\in Sub(\Omega^{w}_{(i_1,...,i_\ell)},1,j-1)} \mathcal{L}_{(a,b)}(1-\mathcal{L}_{((i_1,...,i_\ell),(i_1,...,i_\ell,j))})\bar{P}^{w}_{i_1,...,i_\ell,j}}{\bar{P}^{w}_{i_1,...,i_\ell}} (\bar{BT}^{w}_{i_1,...,i_\ell,j} + D_{((i_1,...,i_\ell),(i_1,...,i_\ell,j))}) + \bar{T}_{(i_1,...,i_\ell)}$$

with $\bar{BT}^{w}_{i_1,...,i_\ell,1} = 0, \ \forall (i_1,...,i_\ell) \in \mathcal{D}_w$.

$$\bar{T}^{w}_{i_1,...,i_\ell} = \sum_{j=1}^{k_{i_1,...,i_\ell}} \frac{\prod_{(a,b)\in Sub(\Omega^{w}_{(i_1,...,i_\ell)},1,j-1)} \mathcal{L}_{(a,b)}(1-\mathcal{L}_{((i_1,...,i_\ell),(i_1,...,i_\ell,j))})(1-\bar{P}^{w}_{i_1,...,i_\ell,j})}{1-\bar{P}^{w}_{i_1,...,i_\ell}} (\bar{T}^{w}_{i_1,...,i_\ell,j} + D_{((i_1,...,i_\ell),(i_1,...,i_\ell,j))}) + \bar{T}_{(i_1,...,i_\ell)}$$

with $\bar{T}^{w}_{i_1,...,i_\ell,1} = 0, \ \forall (i_1,...,i_\ell) \in \mathcal{D}_w$.

Node-offered call requests

The call request offered to each node and link in routing tree R_w is given by

$$\begin{aligned} g_{0,w} &= \lambda_w, \\ g^{j}_{0,w} &= g_{0,w} \prod_{(a,b)\in Sub(\Omega^{w}_{(0)},1,j-1)} \mathcal{L}_{(a,b)} \quad j = 1,...,k_0, \\ g_{(i_1,...,i_\ell),w} &= g^{i_\ell}_{(i_1,...,i_{\ell-1}),w}(1-\mathcal{L}_{((i_1,...,i_{\ell-1}),(i_1,...,i_\ell))}), \\ g^{j}_{(i_1,...,i_\ell),w} &= g_{(i_1,...,i_\ell),w} \prod_{(a,b)\in Sub(\Omega^{w}_{(i_1,...,i_\ell)},1,j-1)} \mathcal{L}_{(a,b)}. \end{aligned}$$

The aggregated traffic offered to node x is given by

$$G_x = \sum_{w\in\Psi_x} \sum_{(i_1,...,i_\ell)\equiv x} g_{(i_1,...,i_\ell),w},$$

where "$(i_1,...,i_\ell) \equiv x$" denotes node x has id $(i_1,...,i_\ell)$ in tree R_w.

Link-offered call requests

The call requests offered to link $((i_1,...,i_\ell),(i_1,...,i_\ell,j))$ is given by

$$\gamma_{((i_1,...,i_\ell),(i_1,...,i_\ell,j)),w} = g^{j}_{(i_1,...,i_\ell),w}$$

As before, this traffic is further classified into:

- $\gamma^{s}_{((i_1,...,i_\ell),(i_1,...,i_\ell,j)),w}$: the type of traffic that will be successfully set up,

$$\gamma^{s}_{((i_1,...,i_\ell),(i_1,...,i_\ell,j)),w} = \gamma_{((i_1,...,i_\ell),(i_1,...,i_\ell,j)),w}(1 - \bar{P}^{w}_{i_1,...,i_\ell,j}).$$

The mean call holding time for this type of traffic is given by

$$\delta^{s}_{((i_1,...,i_\ell),(i_1,...,i_\ell,j)),w} = \bar{T}_{i_1,...,i_\ell,j} + D_{((i_1,...,i_\ell),(i_1,...,i_\ell,j))}.$$

- $\gamma^{f}_{((i_1,...,i_\ell),(i_1,...,i_\ell,j)),w}$: the type of traffic that will be blocked,

$$\gamma^{f}_{((i_1,...,i_\ell),(i_1,...,i_\ell,j)),w} = \gamma_{((i_1,...,i_\ell),(i_1,...,i_\ell,j)),w}\bar{P}^{w}_{i_1,...,i_\ell,j}.$$

The mean call holding time for this type of traffic is given by

$$\delta_{((i_1,...,i_\ell),(i_1,...,i_\ell,j)),w} = \bar{BT}^{w}_{i_1,...,i_\ell,j} + D_{((i_1,...,i_\ell),(i_1,...,i_\ell,j))}.$$

Call setup delay and call blocking probability

The end-to-end call blocking probability for O-D pair w, P^{w}_{rej}, is

$$P^{w}_{rej} = \bar{P}^{w}_{0}.$$

and the average call set up delay for O-D pair w, $\bar{T}^{w}_{setup}$, is

$$\bar{T}^{w}_{setup} = \bar{T}^{w}_{0}.$$

B.1.3 Crankback Routing Rule

Similar to SOC routing rule, we compute $\bar{P}^{w}_{i_1,...,i_\ell}$, $\bar{BT}^{w}_{i_1,...,i_\ell}$, and $\bar{T}^{w}_{i_1,...,i_\ell}$ by following recursive equations. Recall that $(i_1, ..., i_\ell, 1)$ is the destination node if $(i_1, ..., i_\ell) \in \mathcal{D}_w$.

$$\bar{P}^{w}_{i_1,...,i_\ell} = \prod_{j=1}^{k_{i_1,...,i_\ell}} [\mathcal{L}_{((i_1,...,i_\ell),(i_1,...,i_\ell,j))} + (1 - \mathcal{L}_{((i_1,...,i_\ell),(i_1,...,i_\ell,j))})\bar{P}^{w}_{i_1,...,i_\ell,j}]$$

with $\bar{P}^{w}_{i_1,...,i_\ell,1} = 0, \ \forall (i_1, ..., i_\ell) \in \mathcal{D}_w$.

$$\bar{BT}^w_{i_1,...,i_\ell} = \sum_{j=1}^{k_{i_1,...,i_\ell}} \frac{(1-\mathcal{L}_{((i_1,...,i_\ell),(i_1,...,i_\ell,j))})\bar{P}^w_{i_1,...,i_\ell,j}}{\mathcal{L}_{((i_1,...,i_\ell),(i_1,...,i_\ell,j))}+(1-\mathcal{L}_{((i_1,...,i_\ell),(i_1,...,i_\ell,j))})\bar{P}^w_{i_1,...,i_\ell,j}} (\bar{BT}^w_{i_1,...,i_\ell,j}+\bar{T}_{i_1,...,i_\ell}+D_{((i_1,...,i_\ell),(i_1,...,i_\ell,j))}) + \frac{\mathcal{L}_{((i_1,...,i_\ell),(i_1,...,i_\ell,k_{i_1,...,i_\ell}))}}{\mathcal{L}_{((i_1,...,i_\ell),(i_1,...,i_\ell,k_{i_1,...,i_\ell}))}+(1-\mathcal{L}_{((i_1,...,i_\ell),(i_1,...,i_\ell,k_{i_1,...,i_\ell}))})\bar{P}^w_{i_1,...,i_\ell,k_{i_1,...,i_\ell}}}\bar{T}_{i_1,...,i_\ell}$$

with $\bar{BT}^w_{i_1,...,i_\ell,1}=0, \ \forall(i_1,...,i_\ell)\in\mathcal{D}_w$.

$$\bar{T}^w_{i_1,...,i_\ell} = \sum_{j=1}^{k_{i_1,...,i_\ell}} \frac{\prod_{m=1}^{j-1}[\mathcal{L}_{((i_1,...,i_\ell),(i_1,...,i_\ell,m))}+(1-\mathcal{L}_{((i_1,...,i_\ell),(i_1,...,i_\ell,m))})\bar{P}^w_{i_1,...,i_\ell,m}]}{1-\bar{P}^w_{i_1,...,i_\ell}} (1-\mathcal{L}_{((i_1,...,i_\ell),(i_1,...,i_\ell,j))})(1-\bar{P}^w_{i_1,...,i_\ell,j}) [(\sum_{m=1}^{j-1}\frac{(1-\mathcal{L}_{((i_1,...,i_\ell),(i_1,...,i_\ell,m))})\bar{P}^w_{i_1,...,i_\ell,m}}{\mathcal{L}_{((i_1,...,i_\ell),(i_1,...,i_\ell,m))}+(1-\mathcal{L}_{((i_1,...,i_\ell),(i_1,...,i_\ell,m))})\bar{P}^w_{i_1,...,i_\ell,m}}(\bar{BT}^w_{i_1,...,i_\ell,m}+\bar{T}_{i_1,...,i_\ell}))+\bar{T}^w_{i_1,...,i_\ell,j}+\bar{T}_{i_1,...,i_\ell}]$$

with $\bar{T}^w_{i_1,...,i_\ell,1}=0, \ \forall(i_1,...,i_\ell)\in\mathcal{D}_w$.

Node-offered call requests

The traffic offered to each link in routing tree R_w is given by

$$\begin{aligned}
g^1_{0,w} &= \lambda_w, \\
g^i_{0,w} &= \lambda_w \prod_{j=1}^{i-1}[\mathcal{L}_{((0),(j))}+(1-\mathcal{L}_{((0),(j))})\bar{P}^w_j] \quad i=2,...,k_0, \\
g^1_{(i_1,...,i_\ell),w} &= g^{i_\ell}_{(i_1,...,i_{\ell-1}),w}[1-\mathcal{L}_{((i_1,...,i_{\ell-1}),(i_1,...,i_\ell))}], \\
g^i_{(i_1,...,i_\ell),w} &= g^1_{(i_1,...,i_\ell),w}\prod_{j=1}^{i-1}[\mathcal{L}_{((i_1,...,i_\ell),(i_1,...,i_\ell,j))}+(1-\mathcal{L}_{((i_1,...,i_\ell),(i_1,...,i_\ell,j))})\bar{P}^w_{i_1,...,i_\ell,j}] \\
&\qquad i=1,...,k_{i_1,...,i_\ell}.
\end{aligned}$$

The traffic offered to node $(i_1,...,i_\ell)$ in routing tree R_w is then given by

$$g_{(i_1,...,i_\ell),w} = g^1_{(i_1,...,i_\ell),w}\sum_{j=1}^{k_{i_1,...,i_\ell}-1} g^j_{(i_1,...,i_\ell),w}(1-\mathcal{L}_{((i_1,...,i_\ell),(i_1,...,i_\ell,j))})\bar{P}^w_{i_1,...,i_\ell,j}$$

The aggregated offered traffic to node x can then be computed by

$$G_x = \sum_{w\in\Psi_x}\sum_{(i_1,...,i_\ell)\equiv x} g_{(i_1,...,i_\ell),w}.$$

Link-offered call requests

The call requests offered to link $(i_1, ..., i_\ell), (i_1, ..., i_\ell, j)$ is

$$\gamma_{(i_1,...,i_\ell)(i_1,...,i_\ell,j),w} = g^j_{(i_1,...,i_\ell),w}.$$

This traffic is further classified into:

- $\gamma^s_{((i_1,...,i_\ell),(i_1,...,i_\ell,j)),w}$: the type of traffic that will be successfully set up,

$$\gamma^s_{((i_1,...,i_\ell),(i_1,...,i_\ell,j)),w} = \gamma_{((i_1,...,i_\ell),(i_1,...,i_\ell,j)),w}(1 - \bar{P}^w_{i_1,...,i_\ell,j}).$$

The mean call holding time for this type of traffic is

$$\delta^s_{((i_1,...,i_\ell),(i_1,...,i_\ell,j)),w} = \bar{T}^w_{i_1,...,i_\ell,j} + D_{((i_1,...,i_\ell),(i_1,...,i_\ell,j))}.$$

- $\gamma^f_{((i_1,...,i_\ell),(i_1,...,i_\ell,j)),w}$: the type of traffic that will be blocked downstream,

$$\gamma^f_{((i_1,...,i_\ell),(i_1,...,i_\ell,j)),w} = \gamma_{((i_1,...,i_\ell),(i_1,...,i_\ell,j)),w}\bar{P}^w_{i_1,...,i_\ell,j}$$

The mean call holding time for this type of traffic is

$$\delta^f_{((i_1,...,i_\ell),(i_1,...,i_\ell,j)),w} = \bar{BT}^w_{i_1,...,i_\ell,j} + D_{((i_1,...,i_\ell),(i_1,...,i_\ell,j))}.$$

Call setup delay and call blocking probability

Equations used in SOC rule also applies here.

B.2 Controlled-Flooding Rules

Extending the analytical methodology to controlled-flooding algorithms for heterogeneous networks is quite straightforward. However, the computation itself is rather complicated. As stated in Appendix A, the most difficult computation in the analysis of controlled-flooding algorithms is the the computation of the probability that a call request received through a particular path is a duplicated request and the average call set up time when more than one call requests have been received. In this section, we derive close form expressions for computing these two quantities for OOC routing rule with the limitation that each routing tree is limited to have at most three paths.

The same technique can be applied to SOC/crankback rule. However, the derivation of close form expressions depends heavily on the structure of the routing tree. We are, thus, not able to show the derivation systematically. Therefore, we choose to omit the derivation of these two quantities for SOC/crankback routing rule in this dissertation.

B.2.1 OOC Routing Rule

In the following, we show how to compute the probability that a call request received through a particular path is a duplicated request and the average call set up time when more than one call requests have been received for controlled-flooding OOC algorithm in heterogeneous networks. The notation used in Appendix A and Appendix B.1 is also adopted here.

B.2.1.1 Computation of $P_{dup}(k)$

Let us first consider the computation of the probability that a call request received through path k is duplicated, $P_{dup}(k)$. Let T_k be the total waiting time (processing delay + round trip propagation delay) along path k given that the call request is successfully received through path k. We assume that there are ℓ_k intermediate nodes on path k and the processing delay (queueing delay + service time) at each node is exponentially distributed with mean $1/\mu_i$, $1 \leq i \leq \ell_k$. Assume that $\mu_i \neq \mu_j, \ \forall i, j$ on path k. Then the CDF and PDF of T_k is given by [79]

$$\begin{aligned} F_{T_k}(t) &= 1 - \sum_{i=1}^{\ell_k} \frac{\prod_{j \neq i} \mu_j}{\prod_{j \neq i}(\mu_j - \mu_i)} e^{-\mu_i (t - Dp_k)}, \\ f_{T_k}(t) &= \sum_{i=1}^{\ell_k} \frac{\prod_{j=1}^{n} \mu_j}{\prod_{j \neq i}(\mu_j - \mu_i)} e^{-\mu_i (t - Dp_k)}, \end{aligned}$$

where Dp_k is the round trip propagation delay for the source node to receive the acknowledgment of the call request through path k.

To compute $P_{dup}(k)$, we need to know how many call requests have been successfully received. It is very difficult to derive a general expression for any number of call requests that are successfully received. Here, we derive the cases of two and three successfully received call requests. Let us consider the case where two call

requests have been successfully received through path a and b respectively. For easy explanation, we introduce a different notation for average processing delay at each node. Let $1/\mu_{a_i}, 1 \leq a_i \leq \ell_a$, be the average processing delays for those nodes on path a. Similarly, Let $1/\mu_{b_j}, 1 \leq b_j \leq \ell_b$, be the average processing delays for those nodes on path b. Let Dp_a and Dp_b be the propagation delay of path a and b respectively. Assume $Dp_a \leq Dp_b$. Then the probability that the call request received through path b is a duplicated request (i.e., it is received by the destination node later than the call request through path a) given that the call requests sent through both paths are all successfully received is given by:

$$P_{dup}(b) = 1 - \sum_{i=1}^{\ell_a} \sum_{j=1}^{\ell_b} \frac{\mu_{b_j} A_i B_j}{\mu_{a_i} + \mu_{b_j}} e^{\mu_{a_i}(Dp_a - Dp_b)},$$

where

$$\begin{aligned} A_i &= \frac{\prod_{j \neq i} \mu_{a_j}}{\prod_{j \neq i}(\mu_{a_j} - \mu_{a_i})} \quad i = 1, ..., \ell_a, \\ B_i &= \frac{\prod_{j \neq i} \mu_{b_j}}{\prod_{j \neq i}(\mu_{b_j} - \mu_{b_i})} \quad i = 1, ..., \ell_b. \end{aligned}$$

The probability that the call request through path a is duplicated can easily derived by:

$$P_{dup}(a) = 1 - P_{dup}(b).$$

Now consider the case where three call requests have been successfully received through path a, b and c. Adapt similar notation as above and assume that $Dp_a \leq Dp_b \leq Dp_c$. Then $P_{dup}(a), P_{dup}(b)$ and $P_{dup}(c)$ can be computed as following:

$$\begin{aligned} P_{dup}(a) = & \sum_{i=1}^{\ell_a} A_i e^{\mu_{a_i}(Dp_a - Dp_b)} - \sum_{i=1}^{\ell_a} \sum_{j=1}^{\ell_b} \frac{\mu_{a_i} A_i B_j}{\mu_{a_i} + \mu_{b_j}} e^{\mu_{a_i}(Dp_a - Dp_b)} \\ & + \sum_{i=1}^{\ell_a} \sum_{j=1}^{\ell_b} \frac{\mu_{a_i} A_i B_j}{\mu_{a_i} + \mu_{b_j}} e^{\mu_{a_i}(Dp_a - Dp_c) + \mu_{b_j}(Dp_b - Dp_c)} \\ & - \sum_{i=1}^{\ell_a} \sum_{j=1}^{\ell_b} \sum_{k=1}^{\ell_c} \frac{\mu_{a_i} A_i B_j C_k}{\mu_{a_i} + \mu_{b_j} + \mu_{c_k}} e^{\mu_{a_i}(Dp_a - Dp_c) + \mu_{b_j}(Dp_b - Dp_c)}, \end{aligned} \tag{B.1}$$

$$
\begin{aligned}
P_{dup}(b) &= 1 - \sum_{i=1}^{\ell_a}\sum_{j=1}^{\ell_b} \frac{\mu_{b_j} A_i B_j}{\mu_{a_i}+\mu_{b_j}} e^{\mu_{a_i}(Dp_a - Dp_b)} \\
&\quad + \sum_{i=1}^{\ell_a}\sum_{j=1}^{\ell_b} \frac{\mu_{b_j} A_i B_j}{\mu_{a_i}+\mu_{b_j}} e^{\mu_{a_i}(Dp_a - Dp_c)+\mu_{b_j}(Dp_b - Dp_c)} \\
&\quad - \sum_{i=1}^{\ell_a}\sum_{j=1}^{\ell_b}\sum_{k=1}^{\ell_c} \frac{\mu_{b_j} A_i B_j C_k}{\mu_{a_i}+\mu_{b_j}+\mu_{c_k}} e^{\mu_{a_i}(Dp_a - Dp_c)+\mu_{b_j}(Dp_b - Dp_c)}, \\
P_{dup}(c) &= 1 - \sum_{i=1}^{\ell_a}\sum_{j=1}^{\ell_b}\sum_{k=1}^{\ell_c} \frac{\mu_{c_k} A_i B_j C_k}{\mu_{a_i}+\mu_{b_j}+\mu_{c_k}} e^{\mu_{a_i}(Dp_a - Dp_c)+\mu_{b_j}(Dp_b - Dp_c)}.
\end{aligned}
$$

B.2.1.2 Computation of $\bar{T}_{setup}$

When more than one call request are successfully received, we need to compute the time for the first call request received. Again, let us only consider the cases where there are two or three call requests that have been successfully received. Let us first consider the case where two call requests through path a and b have been successfully received. Let T_a and T_b be the total waiting time along path a and b. Let the random variable $T_{min}^{a,b}$ be the time for the source node to receive the acknowledgment of the first call request. Then $T_{min}^{a,b}$ is the minimum of T_a and T_b. The CDF and PDF of $T_{min}^{a,b}$ is given by:

$$
\begin{aligned}
F_{T_{min}^{a,b}}(t) &= \begin{cases} 1 - \sum_{i=1}^{\ell_a} A_i e^{-\mu_{a_i}(t - Dp_a)} & Dp_b \geq t \geq Dp_a, \\ 1 - \sum_{i=1}^{\ell_a}\sum_{j=1}^{\ell_b} A_i B_j e^{\mu_{a_i} Dp_a + \mu_{b_j} Dp_b} e^{-(\mu_{a_i}+\mu_{b_j})t} & t \geq Dp_b, \end{cases} \\
f_{T_{min}^{a,b}}(t) &= \begin{cases} \sum_{i=1}^{\ell_a} \mu_{a_i} A_i e^{-\mu_{a_i}(t - Dp_a)} & Dp_b \geq t \geq Dp_a, \\ \sum_{i=1}^{\ell_a}\sum_{j=1}^{\ell_b} (\mu_{a_i}+\mu_{b_j}) A_i B_j e^{\mu_{a_i} Dp_a + \mu_{b_j} Dp_b} e^{-(\mu_{a_i}+\mu_{b_j})t} & t \geq Dp_b. \end{cases}
\end{aligned}
$$

The expectation of $T_{min}^{a,b}$ is then given by:

$$
\bar{T}_{min}^{a,b} = \sum_{i=1}^{\ell_a} A_i \left(Dp_a + \frac{1 - e^{\mu_{a_i}(Dp_a - Dp_b)}}{\mu_{a_i}}\right) + \sum_{i=1}^{\ell_a}\sum_{j=1}^{\ell_b} \frac{A_i B_j}{\mu_{a_i}+\mu_{b_j}} e^{\mu_{a_i}(Dp_a - Dp_b)}.
$$

Similarly, for the case where three call requests have been successfully received, we can derive the $\bar{T}_{min}^{a,b,c}$ as follows:

$$\begin{aligned}\bar{T}_{min}^{a,b,c} &= \sum_{i=1}^{\ell_a} A_i(Dp_a + \frac{1 - e^{\mu_{a_i}(Dp_a - Dp_b)}}{\mu_{a_i}}) + \\ &\sum_{i=1}^{\ell_a}\sum_{j=1}^{\ell_b} \frac{A_i B_j}{\mu_{a_i} + \mu_{b_j}}(e^{\mu_{a_i}(Dp_a - Dp_b)} - e^{\mu_{a_i}(Dp_a - Dp_c) + \mu_{b_j}(Dp_b - Dp_c)}) + \\ &\sum_{i=1}^{\ell_a}\sum_{j=1}^{\ell_b}\sum_{k=1}^{\ell_c} \frac{A_i B_j C_k}{\mu_{a_i} + \mu_{b_j} + \mu_{c_k}} e^{\mu_{a_i}(Dp_a - Dp_c) + \mu_{b_j}(Dp_b - Dp_c)}.\end{aligned}$$

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