Signalling and Scheduling for Efficient Bulk Data Transfer in Circuit-switched Networks

by

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Supervisor: Prof. D.G. Kourie

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Summary

The number and size of files transferred over the Internet is sufficiently significant to warrant the investigation of faster and more efficient mechanisms for their transfer. This research begins by comparing the performance of circuit-switching and TCP/IP (which is most commonly used in the Internet) when bulk data is transferred. Experimental and analytical analysis suggests that circuit-switching is indeed better suited for bulk data transfer than TCP/IP.

Circuit-switching suffers from the drawback that connections at rates higher than DS0 (64Kbps) can only be obtained in provisioned mode. For circuit-switching to be usable for bulk data transfer, it should support on-demand circuits at much higher data rates, for example OC1 (51.84Mbps) to OC768 (40 Gbps). A signalling protocol has been designed to be implemented in hardware (thus providing high throughput). It enables the setup of high-bandwidth on-demand circuits in Time Division Multiplexing (TDM) networks. An accompanying routing protocol to be implemented in software is also designed. Further, design decisions are considered for a transport protocol that enables the reliable transfer of bulk data over a circuit-switched connection.

When transferring bulk data over a connection-oriented network that implements preventative congestion control mechanisms, the connection has a deterministic holding time that can be computed from the file size, data transfer rate and propagation delays.

With the knowledge of connection duration it is possible to move away from the current blocking mode of operation (where a call is blocked when a switch does not have sufficient resources to handle it) and to implement a queueing mode of operation where connections are scheduled. By scheduling connections it is possible to reply to a connection request with a later start time if the required resources are not available at the time of the request.

Two scheduling schemes have been designed. Simulations show that at 70% loading, these schemes offer start time delays that are up to 85% smaller and channel utilization of up to 37% larger than a simple queueing mode of operation where call holding times are ignored when connections are set up.

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Opsomming

Die hoeveelheid en grootte van lêers wat tans oor die Internet versend word is noemenswaardig. Om hierdie rede is 'n ondersoek geloods op soek na meer doeltreffende meganismes vir die versending van massa data. Hierdie navorsing begin deur die werkverrigting van *circuit-switching* te vergelyk met díe van TCP/IP (wat tans die mees algemene protokol is vir data versending oor die Internet). Eksperimentele en analitiese ondersoeke suggereer dat *circuit-switching* wel meer geskik is vir massa data versending as TCP/IP.

In *circuit-switched* netwerke is konneksies van slegs DS0(64Kbps) op aanvraag beskikbaar. All konneksies hoër as DS0 moet voorlopig bespreek word en word gewoonlik opgestel met die hulp van menslike interaksie. Vir *circuit-switching* om 'n noemenswaardige invloed te hê op massa data versending is daar 'n vereiste dat die netwerke konneksies teen baie hoër snelhede op aanvraag opgestel kan word. Byvoorbeeld, konneksies van OC1(51.84Mbps) tot OC768(40Gbps) gaan 'n baie groter invloed hê op die spoed van massa data versending as DS0 konneksies. 'n Protokol om konneksies op te stel (*signalling protocol*) is ontwerp om in apparatuur ge-implementeer te word. Deur die protokol in apparatuur te implementeer word die hoeveelheid konneksies wat per eenheid tyd opgestel kan word meer as wanneer 'n protokol gebruik word wat in programmatuur implementeer is. Hierdie protokol maak dit moontlik om konneksies vanaf OC1 tot OC192 op aanvraag in 'n TDM netwerk op te stel. 'n Saamgaande roeteringsprotokol (*routing protocol*), geteiken vir programmatuur implementasie, is ook ontwerp. Verder is opsies oorweeg vir 'n transport protokol wat betroubare versending van massa data oor 'n *circuit-switched* konneksie sal waarborg.

Wanneer massa data versend word oor 'n konneksie-geöriënteerde netwerk wat voorkomende meganismes vir kongestiebeheer implementeer, het die konneksie 'n deterministiese tydsduur wat bereken kan word vanaf die hoeveelheid data, die spoed van die konneksie en die totale vertragings tussen die twee nodes wat data uitruil.

Met die kennis van die tydsduur van die konneksie is dit moontlik om weg te beweeg van die huidige hantering van konneksies waar 'n konneksie geblok word indien daar op enige stadium van opstelling nie genoeg hulpbronne is om die konneksie te kan hanteer nie. Dit is nou moontlik om konneksies te skeduleer. Deur konneksies te skeduleer is dit moontlik om 'n versoek vir 'n konneksie te beantwoord met 'n later begintyd indien daar nie genoeg hulpbronne beskikbaar is wanneer die versoek arriveer nie.

Twee skeduleringskemas is ontwerp. Simulasies van die skemas wys dat teen 70% lading, die skemas vertraging in begintyd van konneksies verlaag met tot 85% en die benutting van kanale verhoog met tot 37% wanneer die skemas vergelyk word met 'n eenvoudige skema waar tydsduur ignoreer word tydens konneksie opstelling.

Acknowledgements

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Acronyms

ATM	Asynchronous Transfer Mode
CAC	Connection Admission Control
CL	Connectionless
CO	Connection oriented
ICMP	Internet Control Message Protocol
IETF	Internet Engineering Task Force
IP	Internet Protocol
OSPF	Open Shortest Path First
PDH	Plesiochronous Digital Hierarchy
PSTN	Public Switched Telephone Network
QoS	Quality of Service
RIP	Routing Information Protocol
SCTP	Stream Control Transmission Protocol
SDH	Synchronous Digital Hierarchy
SONET	Synchronous Optical Network
TCP	Transmission Control Protocol
TDM	Time Division Multiplexing
TOS	Type of Service

TTL Time To Live

UDP User Datagram Protocol

WDM Wavelength Division Multiplexing

Chapter 1

Introduction

Based on a classification of the different networking and switching modes, three types of networking techniques are in existence today. A network can be packet-switched or circuit-switched and a packet-switched network can either be connectionless (CL) or connection-oriented (CO). Circuit-switched networks are, by their very nature, connection-oriented. In such networks, the switching action is based on the "position" of the arriving bits. "Position" is determined by the incoming interface number and time slot in a Time Division Multiplexing (TDM) switch, or wavelength in a Wavelength Division Multiplexing (WDM) switch. In CO packet-switched networks, packets that arrive on the connection after it has been set up carry the labels corresponding to the connection in their headers. When packets arrive at a switch in a CO packet-switched network, the mapping tables are consulted, labels are modified to corresponding values for the outgoing link (if needed), and packets are forwarded to the next node. No resources are reserved along the path between hosts communicating over a CL packet-switched network before data exchange is initiated between them. Each packet header contains the complete address of the destination and the packet is routed through the CL network based upon this information. Example networks from the above categories are presented in Table 1.1.

Networking mode / Switching mode	Connection-oriented	Connectionless
Packet-switching	ATM,RSVP/IP,MPLS	IP
Circuit-switching	Telephony TDM network, SONET/SDH,WDM	

Table 1.1: Classification of networking techniques

Sending end \ Consuming end	Live	Stored
Live	Interactive/Live streaming	Recording
Stored	Stored streaming	File transfers

Table 1.2: Classification of data transfers

Data transfers can be classified according to Table 1.2. The three categories where either the source or destination is "live" can be classified as "real-time" where the data is either sent live by the source or consumed live by the destination or both. Examples of these transfers include telephone conversations, telnet sessions, video and audio streaming (using compression techniques). These data transfers typically generate bursty traffic, have delay and jitter constraints, and endure for long sessions. For these applications the use of CO packet-switched networks are ideal because bandwidth is not wasted between bursts of data, which will occur when circuit-switched connections are used.

The remaining category involves file transfers. The data transferred is stored both at the sending and receiving end. Bulk-data transfers from applications such as web accesses, file transfers and electronic mail, could be "small" or "large." If the time to transfer a file is much larger than call setup delay, the file is classified as "large." Otherwise it is considered small. Small transfers are best handled by a CL network to avoid the overhead of call setup delay. Schwartz [5] (page 511), McDonald [6] (page 195) and Miyahara et al [7] shows that large bulk-data transfers are better handled in a circuit-switched mode than in a CL packet-switched mode. This is because the per-packet header and acknowledgment overhead in CL networks is greater than call setup overhead in circuit-switched networks. There is no intrinsic burstiness since a large file consisting of bits can be simply transferred across a network at a constant rate. This results in better performance (lower delays) when a large file is transferred over a circuit-switched network. Nevertheless, in practice, CL IP (TCP/IP) networks are mostly used for bulk data transfers. Using a CO packet-switched network for bulk transfer will suffer similar delays because the connection setup overhead together with the packet header and acknowledgement overhead will contribute to larger delays when transferring bulk data using a CO packet-switched network than transferring the same data using a circuit-switched network.

Two types of circuit-switched networks are in use today. Time Division Multiplexing (TDM) is the simplest circuit-switched network where the switching action is performed based on the incoming interface number and time slot. Wavelength Division Multiplexing (WDM) switches base their switching decision on the incoming interface number and wavelength of the arriving bits. Currently, in both types of circuit-switched networks, on-demand (referred to as "switched" mode of operation as opposed to "provisioned" mode) circuits can only be obtained at the DS0 (64 Kbps) rate, and the only application that uses switched circuits is telephony. All higher rate circuits are used in provisioned (hardwired) mode where circuits are established a priori across the network and are changed infrequently.

To support large bulk-data transfers from applications such as web accesses, file transfers and electronic mail, a network will have to accommodate high-bandwidth (higher than DS0) on-demand circuits between any two nodes of the network. Considering that the high-speed circuit-switched network, Synchronous Optical Networks/Synchronous Digital Hierarchy (SONET/SDH), is able to provide circuits with bandwidth from OC1(51.84Mbps) up to OC768 (40 Gbps), there is a need for a mechanism to set up on-demand circuits for the benefit of increasing performance experienced by file transfer applications. Transferring a large file over even the least-bandwidth SONET circuit (51.84Mbps) will improve delays considerably when compared to the commonly used TCP protocol over the connectionless IP network.

In a CO network, connection setup and release procedures, along with the associated message exchanges, constitute the signalling protocol. Moving away from the telephony application, a new signalling protocol is required for the circuit-switched networks to support these high-bandwidth on-demand circuits.

Signalling protocols are typically implemented in software. This is due to the complexity of signalling messages, and the amount of state information that needs to be kept by all switches participating in connection setup. The fact that signalling protocols are implemented in software points to a shortcoming in circuit-switched networks. Switch fabrics are continually improved to handle data at faster speeds, but that is only after a connection has been set up. Implementing signalling protocols in hardware could significantly increase switches' call handling capabilities. The expected increase in call handling capacities of switches will dramatically affect the role circuit-switched networks play in future data transfer applications. This dissertation presents a novel signalling protocol intended for hardware implementation.

In support of the new signalling protocol, an accompanying routing protocol (to be implemented in software) is designed. The routing protocol provides enough information to the signalling protocol for efficient connection setup along the shortest path between the source and destination.

An experimental analysis of circuit-switching, done as part of this research, suggests that an improvement in network utilization can be expected based on an important characteristic of file transfer applications. The characteristic of interest is that if a file is transferred over a circuit-switched network (that implements preventative congestion control as opposed to reactive congestion control in TCP/IP networks), the holding time of the required connection can be deduced from the file size. In connection-oriented networks, connections are admitted without knowledge of the call holding time. If resources are not available, the connection request is simply denied and connection release procedures are initiated by the signalling protocol. This dissertation explores the option of queueing calls in connection-oriented networks instead of blocking them when network resources are unavailable. A simple call queueing algorithm is to hold up call setup messages at each switch along an end-to-end path until resources become available. This approach is used in the analysis described in Miyahara et al's paper [7]. This scheme suffers from poor network utilization and long call queueing delays. However, if calls have known holding times, it is possible to design call scheduling algorithms that result in reduced call queueing delays and improved network utilization. Algorithms for such call scheduling are proposed and the quantitative benefits of algorithms that exploit knowledge of call holding times are demonstrated.

The author was a member of the CATT (Center for Advanced Technology in Telecommunications) research group at Polytechnic University, Brooklyn, New York, and is responsible for the design of the signalling protocol, the supporting routing protocol as well as the design and simulation of the scheduling algorithms. The scheduling algorithms presented in Chapter 5 has a US patent pending, reference number IDS 122874. Other CATT team members are currently exploring hardware implementation of the signalling protocol presented in Chapter 3, and busy designing a transport protocol to be used over a circuit for bulk data transfers.

Chapter 2 explores the performance gains that can be expected when a circuit-switched network is used for bulk data transfers. To provide a better understanding of the performance gains an experiment was designed to compare data transfer characteristics of

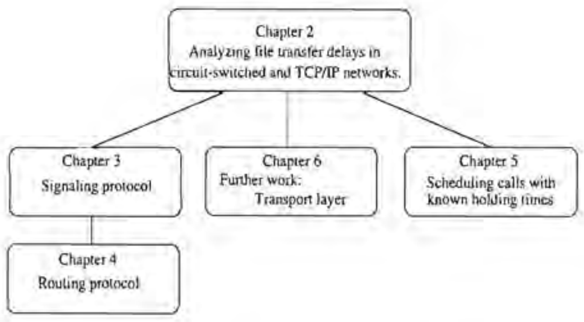

Chapter 2
Analyzing file transfer delays in circuit-switched and TCP/IP networks.
Chapter 3
Signaling protocol
Chapter 6
Further work:
Transport layer
Chapter 5
Scheduling calls with known holding times
Chapter 4
Routing protocol

Figure 1.1: Chapter layout

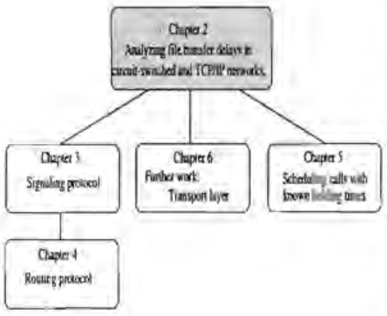
TCP/IP and circuit-switched networks. Chapter 3 introduces the signalling protocol for use in a connection-oriented circuit-switched network, in this case a TDM network, and Chapter 4 provides a solution for the supporting routing protocol. Chapter 5 describes some solutions that can be used to improve connection admission by making use of the holding time of the connection to schedule the start of a connection for a later time if the requested resources are not available at the time of connection request. Chapter 6 notes future work, which is the transport protocol that is required to enable the reliable transfer of data over a circuit-switched network.

Excluding the introduction and conclusion, the layout of the chapters can be presented with Figure 1.1. The conclusions drawn in Chapter 2 showed the need for a signalling protocol for circuit-switched networks (described in Chapter 3) and a transport protocol that enables the reliable transfer of data between a source and destination connected with a circuit-switched connection (discussed in Chapter 6). Chapter 5 investigates the queueing mechanism used in the comparison presented in Chapter 2. The signalling protocol requires a routing protocol to be able to set up a connection along the shortest path between a source and destination, this protocol is described in Chapter 4.

Chapter 2

Analyzing file transfer delays in circuit-switched and TCP/IP networks

2.1 Introduction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In theory circuit-switched networks have been shown to have lower response times for large data transfers than packet-switched networks [5] (page 511), [6] (page 195) and [7]. Since transferring a large stored file from one computer to another has no intrinsic burstiness associated with it, the total per-packet header and acknowledgement overhead in packet-switched networks is greater than the call setup overhead in circuit-switched networks.

To be able to transfer files over a circuit switched network, it is necessary that the network provide high-bandwidth on-demand circuits. We need high-bandwidth to be able to transfer large amounts of data quickly, and on-demand circuits are needed because these transfers could occur between any two nodes connected to the network. Currently in circuit-switched networks, connections of T1(1.5Mbps) and higher rates are set up in provisioned mode. This lack of flexibility might be the single biggest obstacle standing between circuit switching and its success in data networks. Indeed, currently we primarily use packet switching techniques like IP and ATM for the majority of network traffic, including large file transfers.

This chapter attempts to associate real world numbers with the comparisons in [5] and [7], starting by moving from general packet switching techniques to TCP/IP. TCP/IP is currently the packet switching technique of choice when large files are transferred over the Internet. Comparing TCP/IP to circuit-switching under laboratory conditions is intended to help us understand the performance we currently experience and to anticipate the performance we should expect if we decided to make use of circuit-switching for all large file transfers. Note that in this discussion, a large file is considered to be a file for which the total transfer time of the per-packet overhead and the acknowledgement overhead is greater than the call setup overhead in circuit-switched networks.

Due to the unavailability of a circuit-switched network that provides high-bandwidth on-demand circuits, we used UDP together with a simple network configuration to simulate circuit-switching. Section 2.2 describes the equations we designed for the comparison, Section 2.3 introduces the reader to the testing environment and the goals of the specific tests, Section 2.4 describes the results of the tests and Section 2.5 is the conclusion.

2.2 Comparing TCP/IP to circuit-switched networks

When transferring a file, delay can be categorized as follows:

- propagation delay
- emission delay
- queueing delay
- processing delay

In a large network with long round trip times, propagation delay will have a significant influence on the transfer time when TCP/IP is used. This is because of the presence of ACK messages. Before the maximum window size is used by the client, the propagation delay incurred by the ACKs will be significant.

Emission delays (time taken to put the bits on the wire) are also more significant when TCP/IP is used to transfer a file. When transferring data using TCP/IP, emission delays are present at the sending host and all the routers along the path. In the case of circuit switching, emission delays are only present at the sending host (see Figure 2.1). However, on closer examination, under optimal conditions pipelining could occur as will be explained below.

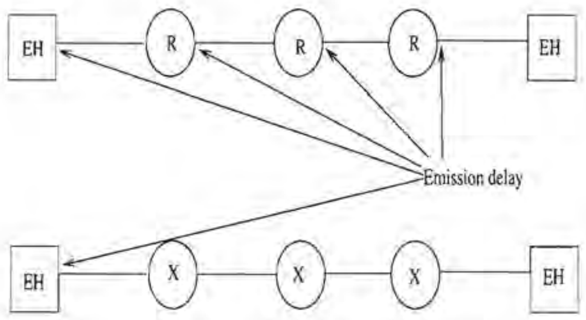


Figure 2.1: Emission delays, TCP/IP vs. circuit switching

The queueing and processing delays (experienced at routers when incoming packets are queued and processed) are also only present when sending the file via TCP/IP, not when the file is sent using circuit switching. Once a circuit has been set up between the source and destination hosts, the data is switched by each intermediate switch without it experiencing queueing and processing delays.

To better understand the total time taken to transfer a large file using TCP/IP vs. a fast circuit, we first propose mathematical models of the time delays in each respective case. Sections 2.2.1 and 2.2.2 present the mathematical models for circuit-switching and TCP/IP respectively.

2.2.1 Circuit-switched networks

$$T_{ckt} = T_{setup} + kT_{wait} + T_{transfer}(f) \tag{2.1}$$

$$T_{setup} = kT_{proc} + T_{r.t.prop} + 2(k+1)\left(\frac{s + ovhd}{rate}\right) \tag{2.2}$$

$$T_{transfer}(f) = \frac{T_{r.t.prop}}{2} + \frac{f + ovhd}{rate} \tag{2.3}$$

In the above equations, all the times (T) are the average times, s is the length of the signalling message, f is the file size, $T_{r.t.prop}$ denotes the round-trip propagation delay and k indicates the number of switches. T_{wait} is the time to wait for the requested resources to be freed (assuming a queueing mode of operation is used, as opposed to a call blocking mode). T_{proc} indicates the total processing delay that includes route determination, Connection Admission Control (CAC) and switch programming, incurred at a switch to set up a connection.

The overhead (*ovhd*) included in Equation (2.2) and (2.3) is the overhead introduced by the data link layer (for example SONET). If the transport layer implements a FEC (Forward Error Correction) scheme (supplemented with NAKs), the overhead will include more than just the headers added by the lower protocol layers.

Equation (2.1) is the total time needed to transfer a file using a circuit. Together with the processing delay (T_{proc}) experienced by a connection request captured in the term T_{setup}, the signalling message requesting the connection is also placed in a queue at each switch to wait for the required resources to be released. Only when the connection request reaches the head of the queue, and its requested resources are available, will the request be passed on to the next switch. Hence the term kT_{wait}.

Equation (2.2) includes the processing delay at switches, propagation delay, and emission delay (which is the length of signalling messages divided by the data rate used for signalling). It is assumed that a circuit is set up using two messages: typically a SETUP is sent in the forward direction from the source to the destination; if all the switches along the path and the destination are able to handle the connection, a SETUP SUCCESS message is sent in the reverse direction from the destination to the source. After processing each of these messages at each switch, a signalling message has to be re-introduced to the circuit-switched network. This has the consequence that the emission delay is computed for each switch (each switch places the signalling message "on the wire" to the next switch) as well as for the end hosts responsible for sending the message. Thus signalling messages have to be placed on $(k+1)$ links during connection setup in both the forward and reverse directions. Hence the term $2\,(k+1)$.

During file transfer (see Equation (2.3)), there is no queueing delay since these are circuit switches and only emission and propagation delays are incurred. The emission delay is also only incurred at the sending host. This is because after a circuit has been set up, the host initially places the bits on the wire after which it is switched by each intermediate node without experiencing queueing, processing or emission delays. It is assumed that the file transfer is initiated from the sending host, and the connection is closed by the sending host after the error free file transmission has completed. The consequence of this assumption is that the file transfer does not include a round trip delay, but only delay for transfer in one direction.

Time to release a circuit incurs the same three delays as to set up a circuit. When the sending host closes the connection, it sends a message with its request to its ingress switch. After receiving a reply to this message, the connection is considered closed by the sending host. The total time for release is not directly included in the file transfer delay model given here, but the effect of release messages on queueing delays of setup messages should be considered.

2.2.2 TCP/IP

The following equations are obtained if it is assumed that there is no loss of packets and hence that the congestion window grows steadily with the sender sending 1 segment, and then 2, and then 4 and so on, until the whole file is sent. One ACK is assumed for each "group" of segments. It is also assumed that *ssthresh* is not reached and that TCP continues operating in slow start. In these equations k indicates the number of routers, m indicates the maximum segment size (in bytes) in TCP/IP and e-e indicates end-to-end.

$$T_{TCP} = T_{e-esetup} + T_{transfer}(f) + T_{e-eclose} \tag{2.4}$$

Equation (2.4) is the total time needed to transfer a file using TCP.

$$T_{e-esetup} = T_{e-eproc} + T_{r.t.prop.} + 3(k+1)\left(\frac{s + ovhd}{rate}\right) \tag{2.5}$$

Equation (2.5) represents the three-way handshake that is needed to set up a TCP connection. It includes the processing delay at routers (which includes queueing delay and service time), propagation delay, and emission delay. The term $T_{e-eproc}$ captures the total queueing and processing delay experienced by all the routers along the path to the destination. The emission delay is the byte size of signalling messages (s) divided by the data rate. In the case of TCP the signalling messages are all 40 bytes. The emission delay should also take into account the overhead added by the data link layer. For example, if Ethernet is used to transfer the messages, the overhead is 58 bytes.

$$T_{e-eclose} = T_{e-eproc} + T_{r.t.prop.} + 2(k+1)\left(\frac{c + ovhd}{rate}\right) \tag{2.6}$$

Equation (2.6) represents the closing of a TCP connection. Although there are more than 2 messages involved in closing a connection, this equation should typically only correspond to the last 2 messages. This is because only the last two messages do not contain data or acknowledgements corresponding to received data. The time delays caused by other segments responsible for closing the connection are included in the file transfer time computation. The time taken to close a TCP connection incurs the same three delay components as were discussed in setting up a connection.

$$T_{transfer}(f) = NT_{r.t.prop} + T_{emission} + T_{total-svc} + T_{ack} \tag{2.7}$$

Equation (2.7) represents the total transfer time of a file of size f.

$$N = \left\lceil log_2\left(\frac{f}{m}+1\right)\right\rceil \tag{2.8}$$

Equation (2.8) represents the number of groups of segments (N) needed to transfer a file of size f in segments that are m bytes in length. In the special case where the file size is such that each segment in each of the N groups is fully utilized, then: $f = \left(1+2+4+8+16+...+2^{N-1}\right)m = \left(2^N-1\right)m$. That is, $N = log_2\left(\frac{f}{m}+1\right)$. In the general case the relationship $2^{N-1} < \frac{f}{m}+1 \leq 2^N$ holds, justifying the ceiling function in Equation (2.8).

The first term in Equation 2.7 results from our assumption that one ACK is sent for each group of segments. Thus, a round trip propagation delay is incurred for each of the N groups of segments.

$$T_{emission} = \sum_{n=0}^{N-2}(k+2^n)\left(\frac{m+ovhd}{rate}\right) + \left(k+1+\left\lceil\frac{f}{m}\right\rceil - 2^{N-1}\right)\left(\frac{m+ovhd}{rate}\right) \tag{2.9}$$

Equation (2.9) is obtained if we assume the emission delay on each link is pipelined. Per-link emission delay is the time to transmit a segment at the data rate *rate*. We first consider the emission delay experienced by groups in which each segment is fully utilized, that is, group x contains 2^{x-1} segments. For $n = 0,\dots,(N-2)$, consider the $(n+1)st$ group. Each such group contains 2^n segments whose emission delay must be explained. Given that the emission delay experienced by one segment when it is placed on a link is $\frac{m+ovhd}{rate}$, the first segment of the group will experience $(k+1)\left(\frac{m+ovhd}{rate}\right)$ emission delay. If there are k switches on the path, there are $k+1$ links. The remaining 2^n-1 segments will each experience $\frac{m+ovhd}{rate}$ delay due to pipelining. Hence the term $(k+2^n)\left(\frac{m+ovhd}{rate}\right)$ for emission delays.

The second term in equation (2.9) handles the emission delay experienced by the last group of segments that may not contain a number of segments that is an integer power of 2. The first segment of group N experiences $(k+1)\left(\frac{m+ovhd}{rate}\right)$ emission delay, the remaining number of segments, which is $\left\lceil\frac{f}{m}\right\rceil - 2^{N-1}$, each experience $\frac{m+ovhd}{rate}$ emission delay.

$$T_{total-svc} = \sum_{n=0}^{N-2} (k + 2^n - 1) T_{dp} + \left(k + \left\lceil \frac{f}{m} \right\rceil - 2^{N-1}\right) T_{dp} \tag{2.10}$$

Equation (2.10) is obtained if we assume the processing delay is pipelined at each router. In this equation the per-packet processing delay at routers, which includes queueing delays, is T_{dp}. Again, we first consider the emission delay experienced by groups in which each segment is fully utilized. For $n = 0, \ldots, (N-2)$, consider the $(n+1)st$ group. The first segment of each group takes kT_{dp} to reach the far end. After it has reached the destination, at the expiration of each T_{dp} one of the remaining $2^n - 1$ segments completes being processed. Hence the term $(k + 2^n - 1) T_{dp}$ represents the processing delays of group $(n+1)$. The first term is the sum of these delays for $n = 0, 1, \ldots, N-2$.

The second term of equation (2.10) corresponds to the processing delay experienced by the last group of segments that may not contain a number of segments that is an integer power of 2. The first segment of group N experiences kT_{dp} processing delay on its way to the destination. The remaining $\left\lceil \frac{f}{m} \right\rceil - 2^{N-1}$ segments each experience T_{dp} processing delay.

$$T_{ack} = N\frac{40 + ovhd}{rate} + NT_{queue} + NT_{svc} \tag{2.11}$$

The total delay experienced by ACK messages is given by Equation (2.11). For each group of segments, one ACK message is sent. This equation includes the emission delay, queueing delay and service time experienced by each ACK message traversing the network. The propagation delay experienced by ACKs are already included in the first term of Equation (2.7).

2.3 Laboratory experiment

This section describes an experiment run in a laboratory, measuring file transmit delays. Since the nodes were closely located in the laboratory, the influence of propagation delay on the overall transmission time was negligible. Instead, the emission and queueing delays were kept in mind when the network configuration was made.

Tests were conducted with a Sun SPARC4 workstation, an Intel PIII500 workstation and two Cisco 2500 routers, all connected with 10 Mbps Ethernet.

In all experiments the term "server" refers to the machine that sends a large file to the "client" that requested the file.

The network configurations presented in Figure 2.2 and Figure 2.3 were used.

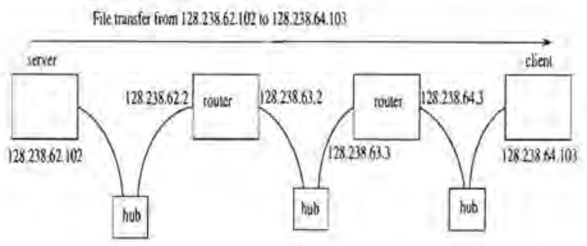


Figure 2.2: Network configuration used when transferring file with TCP/IP

The configuration used in the TCP/IP tests resembles a typical network. In this way the typical emission and low-load queueing delays were reproduced. Also, no extra load was introduced to this network. The only other traffic on the network was some negligible routing updates. This enables the measurement of the best case transfer time using TCP/IP.

UDP does not have the slow start and connection admission feature of TCP and consequently simply sends data at a constant rate. This enables us to emulate file transfer over a circuit (without the connection setup phase) by transferring a file using UDP. The network configuration used for the UDP test shows the "dedicated channel" between the sending and receiving host. Thus to emulate the absence of queueing delays in a circuit-switched network, we used a direct connection between the client and server. In our first experiment we realized the need for flow control with UDP. For this reason the client was the PIII500MHz machine receiving a file from the slower Sun SPARC4 - to prevent the client's buffers to overflow.

To measure the time taken to transfer a file with UDP, the client sent one ACK message back to the server to indicate that the file was received correctly. In all the tests, the times were measured at the server.

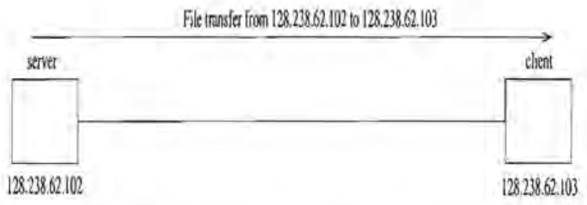


Figure 2.3: Network configuration used when transferring file with UDP

```
13:44:38.354435 128.238.62.102.32788 > 128.238.64.103.6234: S
        1569190891:1569190891(0) win 8760 <mss 1460> (DF)
13:44:38.354482 128.238.64.103.6234 > 128.238.62.102.32788: S
        169214293:169214293(0) ack 1569190892 win 32120
        <mss 1460> (DF)
13:44:38.356011 128.238.62.102.32788 > 128.238.64.103.6234: .
        ack 1 win 8760 (DF)
```

Figure 2.4: Establishing a TCP connection

2.4 Numerical results

In this analysis, the contribution of propagation delay is ignored in all experimental cases. Considering the network configuration used in these tests, with a typical distance of 7 meters between hosts, the propagation delay would be $\frac{7}{2.3*10^8} = 0.0304ns$, which is negligible in these tests.

Tests done with the help of the *traceroute* command found on most networked machines revealed that the average processing time for one packet in the end hosts is 0.4ms (this is the total processing time for both hosts). The average per-packet processing delay at each router (T_{dp}), which includes queueing delays, is 0.6ms.

2.4.1 TCP/IP

End-to-end call setup using TCP/IP is handled by a three-way handshake. Figure 2.4 shows message exchanges of a successful connection establishment between hosts 128.238.62.102 and 128.238.64.103 in *tcpdump* [1] format.

$$T_{e-esetup} = 3 \times k \times T_{dp} + 3 \times (k+1) \times \left(\frac{s + ovhd}{rate}\right) \tag{2.12}$$

By ignoring propagation delay from Equation (2.5), it can be changed to Equation (2.12). The processing and queueing delays experienced by the three connection-establishment segments is given by $(3 \times k \times T_{dp})$. The second term denotes the emission delay experienced by the three segments when they pass through a network with k routers ($k+1$ links). Our tests were run over an Ethernet network, introducing an overhead of 18 bytes. The message size in this case is 40 bytes, which is 20 bytes for the IP header and

```
13:44:51.427515 128.238.62.102.32788 > 128.238.64.103.6234: F
                9730049:9731181(1132) ack 1 win 8760 (DF)
13:44:51.427536 128.238.64.103.6234 > 128.238.62.102.32788: .
                ack 9731182 win 30987 (DF)
13:44:51.427775 128.238.64.103.6234 > 128.238.62.102.32788: F
                1:1(0) ack 9731182 win 32120 (DF)
13:44:51.429244 128.238.62.102.32788 > 128.238.64.103.6234: .
                ack 2 win 8760 (DF)
```

Figure 2.5: Closing a TCP connection

20 bytes for the TCP header.

After a file has been transferred with TCP/IP, the connection is closed gracefully. The server sends a FIN segment containing the last segment of data. This segment is ACKed by the client after which it sends a FIN segment of its own. On receipt of this FIN segment the server closes the connection, the client closes the connection after it receives the ACK for the FIN segment from the server. An example of this exchange of messages is shown in Figure 2.5.

The first two segments of the closing sequence will be handled by Equation (2.7). So, this equation is only applicable to the last two segments.

The propagation delay will be ignored, so the equation applied to our tests will be Equation (2.13).

$$T_{e-eclose} = 2 \times k \times T_{dp} + 2 \times (k+1) \times \left(\frac{c + ovhd}{rate}\right) \tag{2.13}$$

In Equation (2.13) the processing and queueing delay experienced by the last two segments is given by $(2 \times k \times T_{dp})$, and the second term denotes the emission delay experienced by the two segments when they pass through a network with k routers.

The TCP/IP implementation used for the tests implemented the ACK strategy of "ACK every other segment." This is different from our assumption used in Section 2.2, which stated that an ACK is generated for every group of segments. In none of our tests was delay introduced at the server due to a late acknowledgement. For this reason, the term T_{ack} shall be ignored in our analysis. This change to Equation (2.7), together with the removal of the term containing propagation delay produces Equation (2.14) in which the emission and processing delays are represented. Equations (2.9) and (2.10) respectively model these delays.

$$T_{transfer}(f) = T_{total-svc} + T_{emission} \tag{2.14}$$

2.4.2 UDP as an emulator of the circuit switched mode

There is no connection setup phase in UDP, so the total transfer time can be given by Equation (2.15) which is similar to Equation (2.3) (the equation for the time taken to transfer a file of size f over a circuit switched network). The reason why the propagation delay is not halved to indicate the transfer time is because, in the UDP case, an ACK message is sent from the client to the server to indicate successful receipt of the data. The propagation delay will still be ignored due to its insignificant size, so the equations can be thought of as exactly the same in the laboratory environment. The second term of Equation (2.16) shows the ACK that is sent from the client to the server to indicate successful receipt of the file.

$$T_{transfer} = T_{r.t.prop} + \frac{f + ovhd}{rate} \tag{2.15}$$

$$f = filesize + 512 \tag{2.16}$$

2.4.3 The experimental results

Figure 2.6 shows the results for different file sizes f given in Table 2.1, ranging from 13 to 25661582 bytes, and the corresponding experiments done to compare the predicted values against real world tests. In the TCP tests, the number of routers (k) is 2, the message size (m) is 1024 bytes and the overhead ($ovhd$) is 58 bytes (per message of size m). In the UDP tests, the overhead ($ovhd$) is 46 bytes per segment.

In the graph (Figure 2.6) the predicted values (using our mathematical models) for TCP/IP are also plotted. To predict the file transfer delay using TCP/IP, we attempted to provide bounds for the experimental results by computing the file transfer delay under the assumptions of ideal pipelining ("optimistic") and no pipelining ("pessimistic"). Both the optimistic and the pessimistic graphs were plotted using Equations (2.12), (2.13) and (2.14). The difference between the graphs is in how the emission and processing delays (the terms $T_{total-svc}$ and $T_{emission}$ from Equation (2.14)) are computed. In the optimistic graph the emission and processing delays are based on ideal pipelining and expressed by

File sizes (bytes)
13
1205
4820
5898
14460
24796
62039
332008
544541
1091306
3309132
5654721
6120664
9731180
12226702
15918652
25661582

Table 2.1: Values used for file sizes (f) in comparison of circuit-switching and TCP/IP

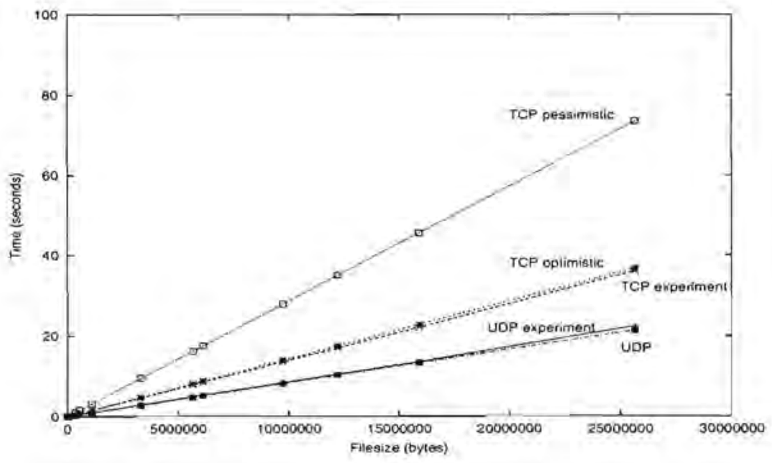


Figure 2.6: Comparison of transport protocols TCP and UDP

Equations (2.9) and (2.10) respectively. The pessimistic graph represents the emission and processing delays based on no pipelining as expressed by Equations (2.17) and (2.18).

$$T_{emission} = \left\lceil \frac{f}{m} \right\rceil \times \left(\frac{m + ovhd}{rate} \right) \times k \tag{2.17}$$

$$T_{total-svc} = \left\lceil \frac{f}{m} \right\rceil \times T_{dp} \times k \tag{2.18}$$

To see if UDP is a good emulator of circuit-switching Equation 2.15 was also plotted in Figure 2.6 (line "UDP").

A closer look at the TCP results suggests that the pipelining does occur. The "optimistic" graph presents a much better estimate of TCP performance than the "pessimistic" graph. Even if the TCP connection passes through k routers the dominant value is the emission delay on one link and the service time at one router, as anticipated in Section 2.2. The graph also indicates that UDP is a good emulator of circuit-switching.

The difference between the delays experienced with the two transport protocols is mostly due to the service time experienced by segments while passing through the network. For example, using Equation 2.14 and assuming ideal pipelining, the transfer of a file with a size of 12226702 bytes results in a total transfer time of 18.015 seconds of which 7.341 seconds is the service time (computed using Equation (2.10)). Using equation (2.15), the transfer of the same file will only take 10.221 seconds in the UDP network.

2.4.4 The analytical results

The laboratory did not allow much freedom to note the behaviour of the two transport protocols under operational conditions where large propagation delays might occur or a large number of routers might be in place (for analysis of emission delays in TCP/IP). Nevertheless, the experimental results from the previous section suggest that our equations provide a good estimate of the delay we can expect when a file is transferred using TCP/IP and a circuit-switched network. Thus, we continue to apply our equations with variables that indicate different network conditions and configurations. This allows us to compare the performance of TCP/IP to circuit switching more thoroughly.

The two network conditions of most interest are large propagation delays and large emission delays (large number of hops in a TCP/IP network). In a TCP/IP network, large propagation delays typically coincide with many hops. To divide the contributions

	Effect of propagation delay		Effect of emission delay	
Parameter	circuit-switching	TCP/IP	circuit-switching	TCP/IP
f	Table 2.1	Table 2.1	Table 2.1	Table 2.1
m	1024 bytes	1024 bytes	1024 bytes	1024 bytes
$ovhd$	4.4%	58 bytes	4.4%	58 bytes
$rate$	10Mbps	10Mbps	10Mbps	10Mbps
k	2	2	22	22
T_{proc}	100		100	
$T_{r.t.prop}$	554.5ms	554.5ms	50ms	50ms

Table 2.2: Parameters used in analytical comparison of circuit-switching and TCP

made to the total delay by the propagation and emission delays respectively, two scenarios are explored and the behaviour of TCP/IP and circuit-switching in each scenario is examined. Firstly, a network with large propagation delays, but few intermediate hops was investigated. Secondly, a network with a large number of hops, but a relative smaller propagation delay was used.

The baseline for these analytical investigations was obtained from the following experiments. Testing with *traceroute* and *ping* revealed that the average round trip time between a certain machine in the USA and a certain machine in Europe is 554.5 ms. Another *traceroute* test between a machine in the USA and a machine in South Africa revealed a hop count of 22.

The same round trip time and number of hops is assumed in the circuit switching and TCP/IP computations. It is assumed that circuit switching is done with SONET (thus introducing a 4.4% overhead). The same rate of 10Mbps is also assumed in the formulae for computing circuit switching and TCP/IP times. It is assumed that 100ms of queueing and processing (parameter T_{proc} from Equation (2.2)) is required at each circuit switch to be able to set up a connection.

In each scenario the analysis of TCP was conducted for two implementations. In the first implementation there is only one ACK sent for each group of segments (using Equation (2.7)). The second implementation considered was the "ACK every other segment" strategy commonly used in TCP implementations. When making use of this strategy, the destination of a file transfer sends an acknowledgement for every second data segment received (except the fist data segment when the group's size is one segment, in which case an acknowledgement is sent for one segment). This acknowledgement indicates the successful receipt of two data segments. Acknowledging every two data segments rather than every group of segments enables the source of the transfer to enlarge the group size while transmitting data, without waiting for an explicit acknowledgement that a group has been successfully received by the destination. Thus, once the group size is sufficiently large, the source will continue sending data without pausing to wait for an acknowledgement.

The "ACK every other segment" strategy is difficult to capture analytically. During the transfer of the first few groups, delays experienced by the ACK messages will not entirely be "absorbed" by the transfer of the actual data. A compromise was made to only include the last ACK message in the file transfer delay computation. That is Equations (2.7) and (2.11) were changed to Equations (2.19) and (2.20) respectively. This is just an estimation of the transfer time until a more appropriate equation can be found that captures the behaviour of the "ACK every other segment" strategy.

$$T_{transfer}(f) = T_{r.t.prop} + T_{emission} + T_{total-svc} + T_{ack} \tag{2.19}$$

$$T_{ack} = \frac{40 + ovhd}{rate} + T_{queue} + T_{svc} \tag{2.20}$$

Table 2.2 provides a summary of the parameters used in the analytical comparison.

Isolating effect of propagation delay

The propagation delay component can be isolated for the first scenario by combining the round trip time from the baseline with a small number of routers (hops). In this scenario the round trip time was chosen to be 554.5 ms and the value for k, the number of routers/switches was chosen to be 2.

Using Equation (2.3) for circuit switching, Equations (2.7), (2.9), (2.10) and (2.11) for TCP/IP with an "ACK every group of segment" strategy, Equations (2.9), (2.10), (2.19) and (2.20) for TCP/IP with an "ACK every other segment" strategy with the variables described above results in the graph presented in Figure 2.7. The interesting part shall be seen when we zoom into the graph where smaller files are concerned, which is shown in Figure 2.8. Looking at Figure 2.8 it is clear that TCP is much better suited for the transfer of small files, whereas circuit switching proves to be the network technique to use for the transfer of large files. It can also be seen that using the technique of "ACK every

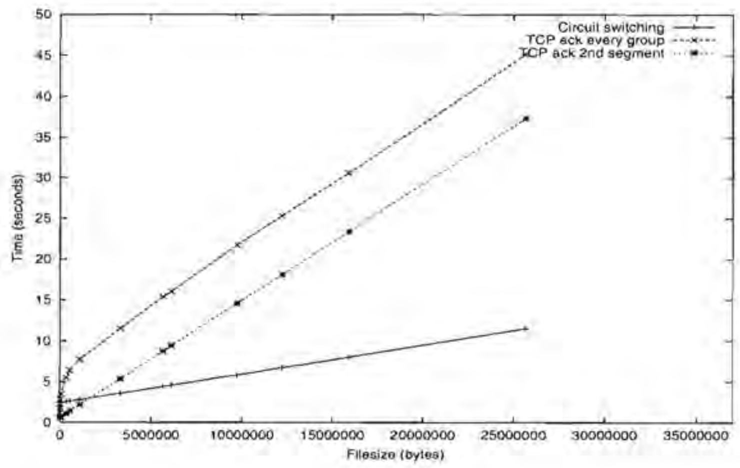


Figure 2.7: Effect of propagation delay on TCP and circuit switching

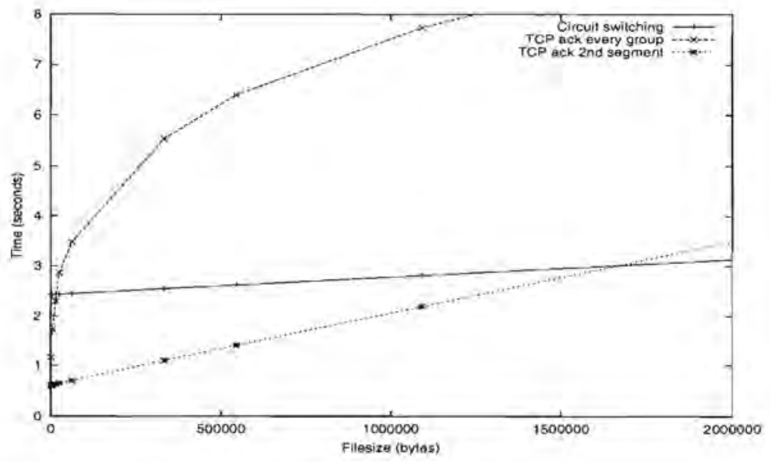


Figure 2.8: Effect of propagation delay on TCP and circuit switching, smaller files

other segment" seems to be more efficient than the "ACK every group" technique when a network with large propagation delays is used.

Isolating effect of emission delay

The emission delay component can be isolated for the second scenario by combining the number of routers (hops) from the baseline with a small propagation delay. The value 50ms was chosen for the propagation delay and the number of hops was chosen to be 22.

The same equations that were used in the examination of the propagation delay contribution are used with the new variables, and the results of this is shown in Figure 2.9. Again, a section of the graph is shown in Figure 2.10. The point at which the emission delay when using TCP/IP becomes more significant than the call setup delay in circuit

115784 85x
b15236614

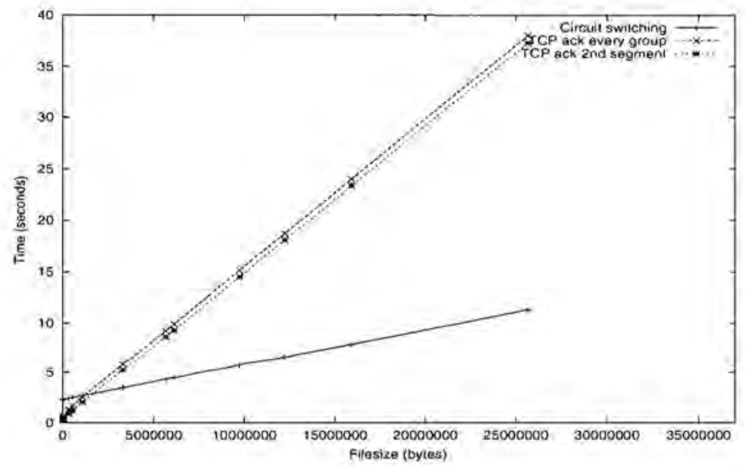


Figure 2.9: Effect of emission delay on TCP and circuit switching

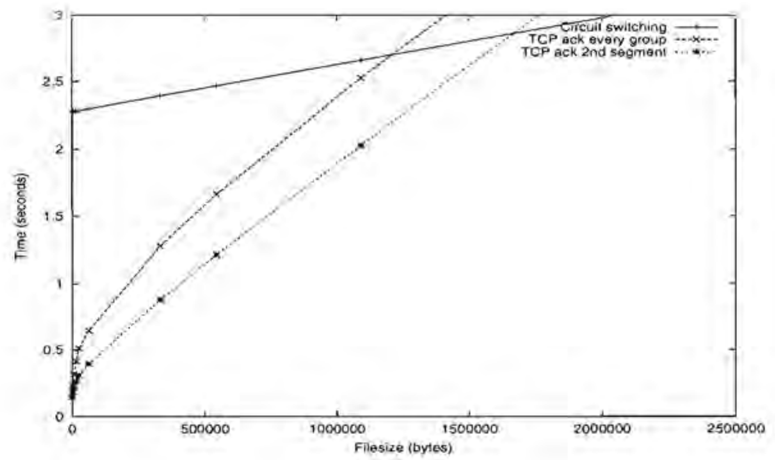


Figure 2.10: Effect of emission delay on TCP and circuit switching, smaller files

switching can be seen clearly in Figure 2.10. Sending any file larger than the file size at that point (just over 1GB) with circuit switching will result in smaller delays than using TCP/IP.

2.5 Conclusion

Queueing delays has changed over the years. In particular with the introduction of differentiated services it became necessary that all packet classification actions be performed at "wire speed" [8]. Thus, the processing of incoming packets has become fast enough to keep up with the rate of incoming packets. Queueing delay is instead incurred at the output port where the link insertion time and the number of packets destined for that link determine the queueing delay. An analysis of the contribution of queueing delay to the total delay is absent in the foregoing investigation. However such an analysis is unlikely to affect the broad conclusions to be drawn from the investigation. This is because the impact of queueing delay in circuit-switched networks is only present during connection setup. During data transfer, all data is switched without experiencing queueing delay. On the other hand, queueing delay will always contribute to the delay experienced by data transfers over CL packet-switched networks.

Thus the evidence we have seen generally suggests that circuit switching is better suited for large file transfers than TCP. Using TCP, emission, propagation and queueing delay all contribute significantly to the total delay when a large file is transferred . Propagation delay and call setup time does contribute to the total delay experienced when a large file is transferred using circuit switching. The result of interest is the fact that there is always a certain file size, that serves as the dividing marker between TCP and circuit switching. Transferring any file larger than this size using circuit switching will result in transfer speeds faster than if TCP was used.

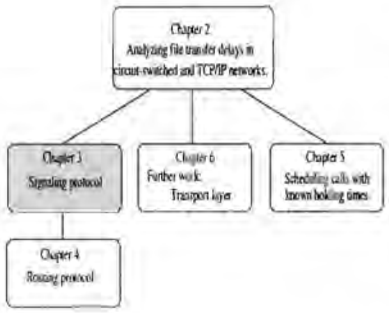
The need for transferring large files increases every day. Workstations are currently equipped with more storage space than would have been imagined a few years ago. There has also been an increase in the number of large file transfers over the internet. The increase is caused, for example, by the increase in occurrence of multimedia content available for free and by the availability of large software packages for download over the internet. Continuing this trend, the need to use circuit-switching will come to the fore more forcefully. However, as pointed out at the start of this chapter, circuit switching as a strategy can only succeed in an environment that provides high-bandwidth on-demand circuits.

Chapter 3 will propose a potential way of achieving this, based on appropriate original protocols.

Chapter 3

Signalling protocol

3.1 Introduction


Chapter 2
Analyzing file transfer delays in circuit-switched and TCP/IP networks.
Chapter 3
Signaling protocol
Chapter 6
Further work:
Transport layer
Chapter 5
Scheduling calls with known holding times
Chapter 4
Routing protocol

Two types of circuit switched networks are in use today. Time Division Multiplexing (TDM) is the simplest where the switching action is performed based on the incoming interface number and time slot. Wavelength Division Multiplexing (WDM) switches base their switching decision on the incoming interface number and wavelength of the arriving bits. This work focuses on a TDM based circuit switched network.

Existing TDM circuit switched networks support the Plesiochronous Digital Hierarchy (PDH) up to the T3 (45 Mbps) rate and the Synchronous Optical Network (SONET)/Synchronous Digital Hierarchy (SDH) for higher rates up to OC768 (40 Gbps). Currently, on-demand (referred to as "switched" mode of operation as opposed to "provisioned" mode) circuits can only be obtained at the DS0 (64 Kbps) rate, and the only application that uses switched circuits is telephony. All higher rate circuits are used in provisioned mode. From Chapter 2 we note that for large file transfers, circuit switched networks that operate at higher data rates than the DS0 rate could be efficient to use. Also, since any end host can download files from any server, a switched mode of operation is necessary. These requirements introduce us to the need for a signalling protocol for high speed TDM networks that will enable them to offer the required switched mode of operation. This chapter discusses the design decisions made for such a signalling protocol and presents the protocol itself. Sections 3.2 to 3.4 describe the decisions made during the design of the signalling protocol and an overview of the protocol itself. Sections 3.5 to 3.8 explain the protocol specification, and Section 3.9 introduces some future improvements to the signalling protocol.

3.2 Implementing the signalling protocol in hardware

One of the early decisions we made before designing the signalling protocol was to implement the protocol in hardware. Typically signalling protocols are implemented in software due to the complexity of the protocols and the need to keep the implementation flexible for evolution reasons. On the other hand, hardware implementations would yield a performance improvement so significant that the role and use of high speed circuit switched networks would change dramatically. To obtain such a performance gain while retaining some flexibility we recommend the use of reconfigurable hardware such as Field Programmable Gate Arrays (FPGAs).

Hardware implementations of networking protocols are on the rise. It has been quite common to implement physical-layer and data-link layer protocols in hardware. For example, Ethernet chips have been available for a long time for use in network interface cards. An Ethernet chip in a network interface card plugged into a host (workstation or pc) simply performs a 6 byte (MAC) address matching function. Chips are also available to perform Ethernet switching, where frames are switched on their destination MAC addresses.

Moving up the protocol stack, network-layer protocols are now being implemented in hardware. First, chipsets were created for ATM header processing as well as ATM switching. ATM is the network-layer protocol of a packet-switched Connection-Oriented (CO) network. In such networks, since a connection is first set up, the identifiers on which the switching is performed are simpler than in a packet-switched connectionless network-layer protocol, where switching is performed on destination addresses as in IP. More recently, network-layer protocols for even connectionless networks, such as IP, are being implemented in hardware. The throughput of IP routers (now called IP "switches") is greatly improved with hardware implementations. Also, the latency incurred per packet, which includes queueing and processing delays at each switch, is reduced. Thus there are many advantages to hardware implementations of networking protocols. Loss of flexibility as protocols are upgraded is cited as the main drawback of hardware implementations. However, now with reconfigurable hardware, this drawback should no longer be a serious concern. It is also feasible to use hardware only for the basic operations of the protocol and relegate more complex and infrequent operations (for example, processing of options fields in the IP header) to software.

All of the protocols discussed above are used to carry user data, and are hence referred to as "user-plane" protocols. Our interest is in applying this same technique of improving performance through hardware implementations to "control-plane" protocols. Signalling protocols are control-plane protocols used to set up and release connections. These protocols are only needed in CO networks.

Current implementations of signalling protocols for both circuit-switched and packet-switched CO networks are done in software. Call setup signalling messages are queued at processors associated with switches and handled typically on a first-come-first-serve basis. Queueing delays are incurred under moderate and heavy load conditions. These delays add up across the switches dominating propagation delays and causing end-to-end call setup delays to be significant. Also, the processors handling signalling messages often become bottlenecks [2]-[4]. Hardware implementations of signalling protocols will greatly help improve both call setup delays and call handling capacities. By implementing the signalling protocol in hardware the throughput will increase significantly, thus a CO network will be able to admit more connections in a time period.

There are many challenges to implementing signalling protocols in hardware that have not been encountered in hardware implementations of user-plane protocols. The main difference is that in handling signalling protocols, the hardware needs to maintain state information of connections, whereas all the user-plane protocols implemented in hardware are stateless. Ethernet frames, ATM cells and IP datagrams are simply forwarded without any state being maintained for each packet or "flow".

The work items to provide a practical environment for such a signalling protocol include:

1. specification of a signalling protocol
2. implementation of the signalling protocol in hardware (using FPGAs)
3. design and analysis of how these signalling protocol chips can be used in a switch controller, and
4. design and analysis of end-to-end applications that can use high bandwidth on-demand circuits.

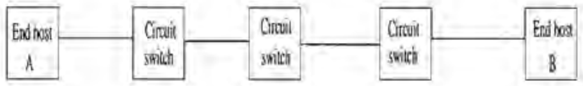


Figure 3.1: Ideal network in which signalling protocol will be used

This chapter only discusses the first work item, the specification of the signalling protocol. Work has already begun on prototypes of the hardware implementing a scaled down version of the signalling protocol and future work in this project shall deal with the other pending work items.

3.3 Mode of transport for signalling messages

The end goal of the signalling protocol is to be able to set up circuit switched connections on-demand. Before the specification of the protocol can be dealt with, one first has to consider the delivery of the signalling messages. Ideally the signalling messages should traverse the same type of network in which the connection is being set up. For example, in Figure 3.1 when end host A requests a connection to end host B, it should send its first signalling message requesting a connection (SETUP) to its ingress switch. In a circuit switched network, this means that there has to be a connection between end host A and its ingress switch. Also, for the signalling messages to traverse the network to the destination host (end host B), there has to be a dedicated signalling channel between all the switches and end hosts. To reserve such a channel in a TDM network for signalling results in a waste of valuable bandwidth. For example, if the circuit switch is a SONET/SDH switch with a cross connect rate of OC1 (51.84Mbps), the smallest channel available is OC1 which, when dedicated to signalling, would be very wasteful.

The proposed solution is to use another network as depicted in Figure 3.2. We propose carrying signalling messages on IP as out-of-band messages. Thus, when end host A wants to set up a connection to end host B, it sends a SETUP message to its ingress switch via its closest IP router. This has the consequence that routes and end hosts are identified by their IP addresses in signalling messages. Using IP to transport the signalling messages has two very important disadvantages: reliability and security. These issues will be considered under the improvements to the signalling protocol discussed in Section 3.9.

Using IP to send the signalling messages has a side benefit of reducing the size of the SETUP message by eliminating the need to include the source address in the SETUP

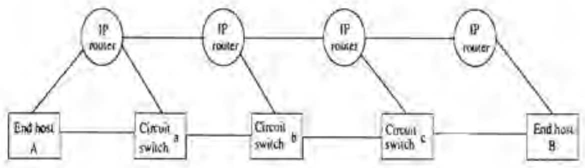


Figure 3.2: Typical network in which signalling protocol will be used

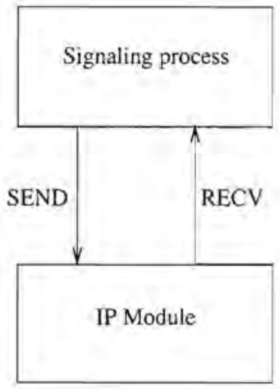


Figure 3.3: Primitives used between signalling process and IP module

message. IP provides the functionality of specifying source and destination addresses of the signalling messages in the IP header, the source address is needed at the receiving host to be able to reply to the connection request (SETUP SUCCESS). For example, when a connection is being set up between host A and B, the SETUP message sent from switch a to switch b in Figure 3.2 carries the source and destination addresses of end hosts A and B as parameters of the message, but carries the addresses of switch a and b in the IP header source and destination address fields. Each transit node can store the IP address of the previous node to use for the reverse SETUP SUCCESS message without it ever being included in the SETUP message.

The interface between the signalling process and the IP layer is implementation dependent. The following is from the IP specification [10]. "The functional description of user interfaces to the IP is, at best, fictional, since every operating system will have different facilities." Hence, the interface between IP and the signalling process should be developed according to the implementation. An example upper layer interface, consisting of two calls, that satisfies the requirements for the user (in this case the signalling process) to IP module communication is depicted in Figure 3.3.

The parameters of the calls are as follows ("=>" means returns) [10], the addition of the *SpecDest* parameter to the RECV call is in accordance with [11], which updates [10]

Parameter	Description	Length(bits)
src	source address	32
dst	destination address	32
prot	protocol	8
TOS	type of service	8
TTL	time to live	8
BufPTR	buffer pointer	Implementation dependent
len	length of buffer	Implementation dependent
Id (optional)	Identifier	16
DF	Don't Fragment	1
opt	option data	Implementation dependent
result	response OK = datagram sent ok Error = in arguments or local network	Implementation dependent

Table 3.1: Description of parameters in SEND primitive [10]

in this respect:

```
SEND (src,dst,prot,TOS,TTL,BufPTR,len,Id,DF,opt => result)
```

The description of the parameters in the SEND primitive is given in Table 3.1.

```
RECV (BufPTR, prot, => result, src, dst, SpecDest, TOS, len, opt)
```

The description of the parameters in the RECV primitive is given in Table 3.2.

Due to the freedom allowed in these two calls' specification, both have been changed to reflect the requirements of the signalling process. The changes are limited to the omission of unnecessary parameters. The *Id* is already optional, and not required by the signalling process and no optional data needs to be passed between the signalling process and the IP module. The length of the signalling messages (which are the only messages/data passed

Parameter	Description	Length (bits)
BufPTR	buffer pointer	Implementation dependent
prot	protocol	8

result	response
	
	OK = datagram received ok
	Error = error in arguments
		Implementation dependent

len	length of buffer	Implementation dependent
src	source address	32
dst	destination address (may be broadcast or multicast address)	32
SpecDest	specific destination address	32
TOS	type of service	8
opt	option data	Implementation dependent

Table 3.2: Description of parameters in RECV primitive [10]

between the signalling process and the IP module) is either included in the signalling message itself, or it can be deduced from the message type. This eliminates the need for the IP module to include the length of the message when it is passed to the signalling process (the RECV primitive). The remaining parameters have the same description and length as before. The SEND call has been changed to:

```
SEND (src,dst,prot,TOS,TTL,BufPTR,len,DF => result)
```

The RECV call have been changed to the following:

```
RECV (BufPTR, prot, => result, src, dst, SpecDest)
```

Other implementation decisions are:

- The Type of Service (TOS) parameter indicates to the IP network the quality of service desired for the data transfer. It shall always be set to 0001 0100 which indicates a request for low delay and high reliability when a signalling message is transferred.

- The *len* parameter will be a 16 bit value that is an exact copy of the message length information element in the SETUP message. For all the other messages, it shall contain the length of the signalling message.

- The usage of two destination address parameters (*dst* and *SpecDest*) in the RECV call is required due to the IP facility for broadcasting and multicasting. The specific-destination address (*SpecDest*) is defined to be the destination address (*dst*) in the IP header unless the header contains a broadcast or multicast address, in which case the specific-destination is an IP address assigned to the physical interface on which the datagram arrived [11]. No signalling message should be destined for a multicast group or sent as a broadcast message over the network. For this reason, the signalling message shall immediately be discarded when the two destination parameters are not equal.

- Use 8 bits for the result parameter, and the value 200 shall indicate success.

- The protocol value of 253 will be used to indicate the signalling layer. This value is currently unassigned according to [21].

- When data is passed to the IP layer, the TTL parameter will always be set to 16. This number indicates the maximum acceptable number of routers between TDM switches.

- The DF parameter shall always be set to 1, which indicates that this message may not be fragmented by the IP network. This is because the maximum size for the SETUP message is 407 bytes and given that the IP specification, [10], dictates that all hosts must be prepared to accept datagrams of up to 576 bytes, the message will never need to be fragmented by the IP network (even if a maximum sized header is included in the datagram).

3.4 Signalling protocol description

This signalling protocol is intended to be used in a circuit-switched CO network where connections are used for large file transfers, which have been proven to be the ideal application for high speed circuit-switched CO networks. Also, in a quest for a simple design of this protocol, a homogeneous network (where all the switches operate at the same cross connect rate) is assumed to be the target network. Changes to the signalling protocol when it is used in a heterogenous network will be discussed under the improvements to the signalling protocol in Section 3.9. With the above implementation considerations, certain assumptions can be made in the design of this protocol. They are:

- Only unidirectional two-party connections are allowed. The reason for this is that the primary application is large file transfers. Connections would then primarily be needed to transfer a large file from a server to a client.

- The setup requests are generated by the sending end of the unidirectional connections.

- The bandwidth requested in a SETUP message is either equal to, or an integral multiple of the (homogeneous network) switch cross connect rate. The bandwidth can be requested with the specification of a range that indicates minimum and maximum bandwidth requirements. Using these two parameters, the network attempts to allocate the maximum requested bandwidth. If any node along the path of the connection cannot provide the requested bandwidth, the highest bandwidth less than the maximum requested and larger or equal to the minimum requested is reserved.

Destination node address	Data rate OC1			...	Data rate OC192		
	Next hop option 1	Next hop option 2	Next hop option 3		Next hop option 1	Next hop option 2	Next hop option 3

Table 3.3: Routing table

Again, the file transfer application permits this since a file can be transferred at any rate, unlike streamed video or audio where a specific data rate is required.

Connection setup consists of four steps:

- Route determination
- Connection admission control (CAC) and interface selection
- Time slot selection
- Switch fabric configuration

On completion of data transfer, corresponding release procedures are executed to free all resources reserved for the connection. These connection setup and release procedures, along with the associated message exchanges, constitute the signalling protocol.

There are two types of routing, hop-by-hop routing and source routing. In hop-by-hop routing each node determines the next-hop node to reach the destination. In source routing, the ingress switch determines the end-to-end route, which is then carried in the connection setup signalling message. Each intermediate node merely consults this parameter to determine the next-hop node to which to route the connection. In both cases, routes can either be computed dynamically when a connection setup message arrives or can be precomputed and stored in routing tables as shown in Table 3.3. Since we are targeting hardware implementation of this protocol, we select hop-by-hop routing and route precomputation. Due to the granularity of the rates available in a TDM network, the construction of the routing table is simplified in that next hop nodes need only be kept for each rate provided by the network.

This simplifies the *route determination action* performed at the node to a simple routing table lookup. It also simplifies the parsing of SETUP messages by not requiring the

Next hop node	Interface number	Total bandwidth	Available bandwidth	Cost

Table 3.4: Available bandwidth table including cost

hardware to process a source route parameter. The signalling protocol essentially trusts its associated routing protocol to ensure loop-free routing data. However, to prevent indefinite looping, we do provide for a Time-To-Live (TTL) field in the SETUP message requiring switches to decrement the TTL field at each hop, and to stop a SETUP request received with the TTL field equal to 1. Making use of route precomputation has the consequence that real-time parameters (for example, the current load in a node) are typically not taken into account. This means the next-hop node according to the routing table might not have resources available for a connection at the time when the connection request arrives. This introduces the need for more next hop options which is included in the routing table as shown in Table 3.3. Including more than one "next hop" allows a node to send a request for a connection to an alternative node in the case when a previously chosen "next hop" does not have enough resources available.

Connection admission control (CAC) in a TDM switch consists of determining whether there is sufficient available bandwidth for the connection on any of the interfaces to the next-hop node identified in the route-lookup step. The connection admission control action is performed by examining the available bandwidth column of a table such as the one shown in Table 3.4, or by using some other connection admission mechanisms as will be discussed in Chapter 5. In this table a record of available bandwidth is kept on a "per-interface" basis. Signalling protocol hardware reads and writes data into the "available bandwidth" column of this table as it admits and releases calls. The other columns are written by the node administrator during configuration, except the "cost" column that will be discussed in Chapter 4.

The incoming interface numbers are obtained from information in the SETUP message. As connection setup proceeds hop-by-hop, each switch places its outgoing interface number as a parameter in the SETUP message. Assume node I receives a SETUP message from its neighbour I-1 indicating an outgoing interface number *O*. The interface number *O* corresponds to an interface on node I-1. Node I needs to determine the number of its interface(s) that is connected to interface *O* on node I-1. This is done by consulting a

Neighbour node		Interface number
Address	Interface number	

Table 3.5: Connectivity table

connectivity table, as shown in Table 3.5.

For example, interface number 5 on node I is connected to interface number 2 in its neighbour node I-1 as shown in Figure 3.4. If the SETUP message received from node I-1 indicates that the connection has been routed on its interface number 2, node I needs to use 5 as the incoming interface number.

Time slot selection is closely coupled with the interface selection that occur during CAC. Once it has been determined that there is enough bandwidth available on an interface, time slot selection will determine which time slots on the chosen interface will be used for data transfer. The time slots on an interface that will be used for the actual connection can be determined by the node during processing of the SETUP message, or the next-hop node may choose the time slots and provide the current node with the information in the success reply (SETUP SUCCESS) message. Whichever mechanism is used, the next-hop and the current node has to agree upon the time slots used for the connection before data transmission commence. With the introduction of maximum bandwidth selection, a node cannot know during processing of the SETUP message the exact amount of bandwidth that can be allocated by all nodes along the path to the destination. The bandwidth for a connection will only be known during the processing of the SETUP SUCCESS message. For this reason it is necessary for the next-hop node to include the time slots that will be used for the connection in the SETUP SUCCESS message.

One more benefit is received when the next-hop node is responsible for time slot selection. When time slot selection and/or switch configuration is done by a node upon receipt of a SETUP message the processing required when a connection has to be released is increased significantly. Keeping in mind our goal of hardware implementation, the processing in any node has to be kept to a minimum, which is accomplished by including the time slot numbers in the SETUP SUCCESS message.

Switch fabric configuration consists of writing data tables that map incoming interface and time-slot numbers to outgoing interface and time-slot interfaces as shown in Table 3.6.

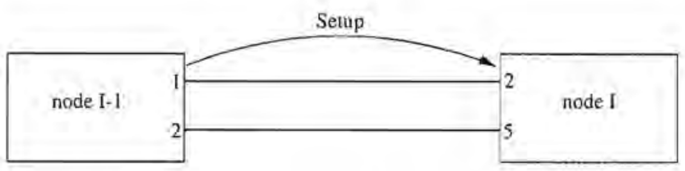


Figure 3.4: Connectivity information

Incoming channel identifier		Outgoing channel identifier	
Interface number	Time-slot number	Interface number	Time-slot number

Table 3.6: Switch mapping table

The connection setup actions, route determination and connection admission control/interface selection, occur sequentially as the setup procedure moves from node-to-node. Time slot selection is initiated upon receipt of the SETUP SUCCESS message, and switch fabric configuration is initiated either on a timer trigger, or with another message of which the SETUP SUCCESS message sent in the reverse direction is an example. When the SETUP SUCCESS message reaches the starting end host, it can send data on the newly established connection.

A connection could be released by the node that requested the connection (source), the node to which the connection was requested (destination) or any intermediate switch as a result of an error (for example, the physical connection breaks). At each node that forms part of the connection, the connection release procedure releases all resources reserved for the connection by editing the available bandwidth and switch mapping tables.

3.5 Signalling protocol specification

3.5.1 Messages

This section deals with the specification of the four messages of the signalling protocol, they are SETUP (Table 3.7), SETUP SUCCESS (Table 3.8), RELEASE (Table 3.9) and RELEASE CONFIRM (Table 3.10).

Information element	Length (bits)	Reference
Message type	4	Section 3.5.2.1
Destination address	32	Section 3.5.2.2
Source address	32	Section 3.5.2.3
Connection reference (previous)	12	Section 3.5.2.4
Time to live (TTL)	8	Section 3.5.2.5
Min bandwidth	8	Section 3.5.2.6
Max bandwidth	8	Section 3.5.2.6
Interface number	8	Section 3.5.2.7
Checksum	16	Section 3.5.2.8

Table 3.7: SETUP

Information element	Length (bits)	Reference
Message length	16	Section 3.5.2.9
Message type	4	Section 3.5.2.1
Connection reference (previous)	12	Section 3.5.2.4
Connection reference (own)	12	Section 3.5.2.4
Bandwidth	8	Section 3.5.2.6
Time slot number(s)	8-1536	Section 3.5.2.7
Checksum	16	Section 3.5.2.8

Table 3.8: SETUP SUCCESS

Information element	Length (bits)	Reference
Message type	4	Section 3.5.2.1
Connection reference (own)	12	Section 3.5.2.4
Cause	8	Section 3.5.2.10
Checksum	16	Section 3.5.2.8

Table 3.9: RELEASE

Information element	Length (bits)	Reference
Message type	4	Section 3.5.2.1
Connection reference (own)	12	Section 3.5.2.4
Cause	8	Section 3.5.2.10
Checksum	16	Section 3.5.2.8

Table 3.10: RELEASE CONFIRM

3.5.2 Description of Information elements

3.5.2.1 The *Message type* information element indicates the type of the received message. It can be any of four messages as depicted in Table 3.11. The protocol might be expanded to include more message types, which can be used for status enquiries, general notifications and updates. To support the scheduling scheme in Chapter 5 additional messages may be needed to trigger the programming of the switch fabric.

Bits	Message type
4 3 2 1	
0 0 0 1	SETUP
0 0 1 0	SETUP SUCCESS
0 0 1 1	RELEASE
0 1 0 0	RELEASE CONFIRM

Table 3.11: Message types

3.5.2.2 *Destination address* is the IP address of the end host to which a connection is being requested.

3.5.2.3 *Source address* is the IP address of the end host requesting the connection.

3.5.2.4 A *Connection reference* can be assigned locally or globally. When only one node is responsible for the assignment of connection references that will be used by all the nodes forming part of a connection (*globally assigned*), complexity is added to the signalling protocol. For this reason, each node will be responsible to select a number that is used by the signalling protocol processing engines to keep track of connections. This number has local significance, changes hop by hop, and remains fixed for the lifetime of a connection. The connection reference is assigned by the signalling protocol to identify the node forming part of the connection (as opposed to a link of a node). This is to enable the signalling protocol to be as light-weight as possible. By allowing the signalling protocol to select a connection reference from one range only (as opposed to a range associated with each outgoing link), its connection reference assignment is simplified. The SETUP message contains the *connection reference* being used by the node that sent the message. The RELEASE and RELEASE CONFIRM messages contain the *connection reference* being used by the node to which the message is sent. In reply to a SETUP message, the SETUP SUCCESS message to the previous node contains the *connection reference* of the node to which the message is being sent (which is the same as the one included in the received SETUP message), and the connection reference reserved for the connection by the node that sent the SETUP SUCCESS message. That is, the SETUP SUCCESS message sent from node i to node $i-1$ contains the connection references assigned by node i and node $i-1$.

3.5.2.5 The *Time to live (TTL)* information element is used to prevent indefinite looping. It is decremented at every node, and if it reaches the value 1 at any transit node (which is any node except the end nodes), the setup is deemed a failure.

3.5.2.6 The *Min bandwidth* and *Max bandwidth* information elements contain the minimum and maximum bandwidths requested for the connection in the SETUP message. The *Bandwidth* information element in the SETUP SUCCESS message contains the bandwidth agreed on for the connection. These information elements should contain the bandwidth in multiples of OC1 connections. Thus, when an OC48 connection is requested, the bandwidth (minimum and maximum) field shall contain "00110000". This allows for a request for a connection with bandwidth of up to OC192.

3.5.2.7 The *interface number* and *time-slot number(s)* information elements contain the values selected for the channels that will carry user data. Like connection references, these numbers also change at each hop. We allow for multiple time slot numbers in case the requested bandwidth is greater than the switch cross connect rate. Due to synchronization problems when one connection is handled by more than one interface, a connection is only able to be handled by one interface with the usage of one or more time slots.

3.5.2.8 A *checksum* information element allows for error checking on the received message. If the checksum fails, the message is immediately discarded by the signalling process.

3.5.2.9 The *Message length* information element is needed in the SETUP SUCCESS message because multiple time slots might be needed to accommodate the connection's requested bandwidth on the selected interface. The *Message length* information element indicates the number of 16 bit words of the message contents, that is, including the bytes used for the message length and the message type. If the message length cannot be expressed as a number of 16 bit words, padding may be used.

3.5.2.10 The *cause* information element indicates to the receiver of a RELEASE or RELEASE CONFIRM message the reason why the connection is being cleared. These values can follow the messages as defined in ITU-T Recommendation Q.2610. According to this recommendation, the information element has a variable length, but we shall only need the definitions of the different types of errors and their respective values.

3.5.3 Tables

The signalling protocol relies on the use of five tables, four of which have been discussed in Section 3.4, they are:

1. Routing table, Table 3.3;
2. Available Bandwidth table, Table 3.4;
3. Connectivity table, Table 3.5;
4. Switch mapping table, Table 3.6; and
5. State table, Table 3.12.

The state of every connection being established, released, or in use is maintained both at the end hosts and at the switches. The state diagram for the signalling protocol is as shown in Figure 3.5. In the *Setup received* state, two of the connection setup actions are performed. The last two connection setup actions, *time slot selection* and *switch programming*, is performed upon entering the *Established* state. In the *Release received* state, the release actions are performed.

Connection references			State	Bandwidth	Previous-node address	Next-node address	Outgoing channel identifiers	
Own	Prev node	Next node					Interface	Time slot

Table 3.12: State table

A state table such as the one shown in Table 3.12 is updated each time a signalling message is received. Since *connection references* are local, the connection references used by the previous and next nodes forming part of the connection are stored in the state table. The signalling protocol will always use the connection reference it assigned to the node (*own*) as an index to this table.

The *state* entry indicates the state of each connection with a number assigned to each of the states shown in Figure 3.5. The *bandwidth*, *previous-* and *next-node address* fields are maintained for each connection. The last column stores the *outgoing channel identifiers* of which there could be many per connection.

There are six states defined for this protocol. They are:

1. **Closed** No connection exits

2. **Setup received** This state exists in a succeeding node after it has received a SETUP message from the preceding node, but has not yet responded.

3. **Setup sent** This state exists in a preceding node after it has sent a SETUP message to the succeeding node, but has not yet received a response.

4. **Established** This state exists when a connection has been established.

5. **Release received** This state exists when a node has received a request from a preceding or succeeding node to release the connection, or when the higher layer (application) requests a connection release.

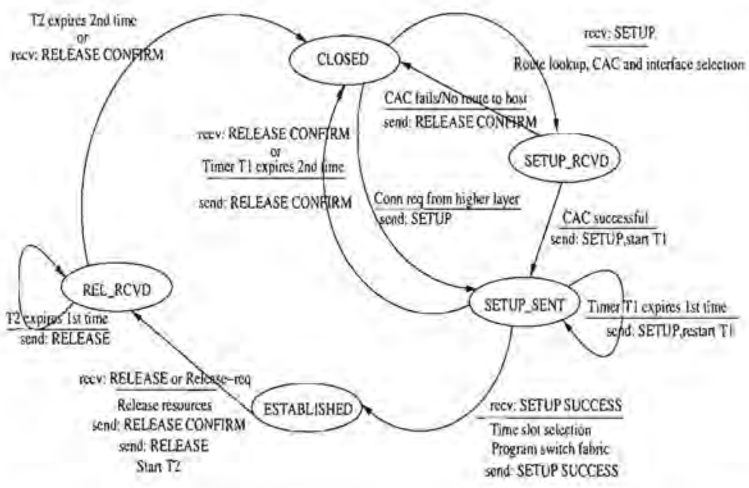


Figure 3.5: Signalling protocol state transition diagram

preceding node
succeeding node

Figure 3.6: preceding/succeeding

3.6 Setup and release procedures for homogeneous networks

The setup and release procedures and the corresponding message exchanges are now briefly explained for switches setting up a unidirectional connection.

In these explanations preceding node refers to a network node that comes immediately before another network node in the connection setup path. Succeeding node refers to a network node that comes immediately after another network node in the connection setup path, see Figure 3.6. For clarity, the use of the IP network to carry signalling messages is ignored in this description.

3.6.1 Connection request

On receipt of a SETUP message from a preceding node, a node shall enter the *Setup received* state and execute all connection admission procedures. If there are not enough resources available for the connection, connection clearing procedures will be initiated by sending the message RELEASE CONFIRM to the preceding node with the "cause" field set to *bandwidth unavailable*. The node shall also return to the *Closed* state. If enough resources are available the node shall perform interface selection and the connection establishment shall continue.

Connection establishment is initiated by the preceding node by sending a SETUP message to the succeeding node which is determined by a route table lookup. After sending the SETUP message, the preceding node enters state *Setup sent*. If, after a certain timeout, there has been no response to the SETUP message, the message may be retransmitted. After the second expiration, the preceding node enters state *Closed* and sends the message RELEASE CONFIRM to the node it received the SETUP message from with the "cause" field set to *timer expired*.

On receipt of the SETUP message, the succeeding node shall enter the *Setup received* state, and the above procedures are repeated at that node.

On receipt of a SETUP message, the node decrements the TTL. If the value results in 1, it shall enter the *Closed* state and send a RELEASE CONFIRM message to the node it received the SETUP message from. The "cause" field shall be set to *TTL expired*.

3.6.2 Connection establishment

Upon receiving a response from the application that the connection has been accepted, the destination node initiates the final phase of connection setup by choosing available time slot numbers on the incoming interface and sending the message SETUP SUCCESS to its preceding node. After this it enters the *Established* state. This message indicates to the preceding node that a connection has been established from its interface to the called party. On receipt of a SETUP SUCCESS message, a switch finalizes connection setup by performing the switch configuration step. The SETUP SUCCESS message propagates to the end host that requested the connection, which on receipt of the message, may start with data transmission.

Figure 3.7 shows an example of the message sequence when connection establishment is initiated by Host1, and is successful.

Figure 3.8 shows an example of the message sequence when connection establishment is initiated by Host1, but is rejected by Host2.

3.6.3 Clearing a connection

A call clearing request can be initiated by any of the two parties involved in data transmission. It is also possible for any node forming part of the connection to initiate call clearing due to a failure, administrative action or some other exception.

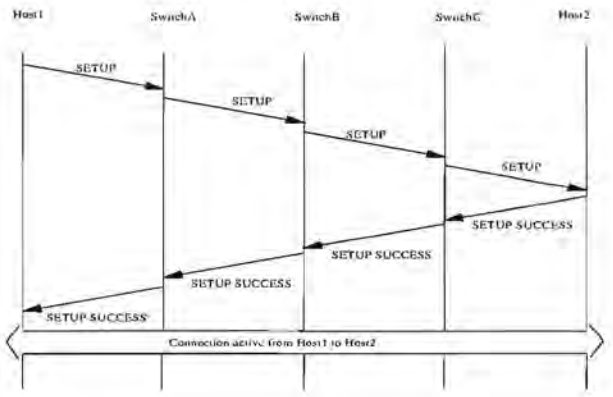


Figure 3.7: Successful setup of a connection

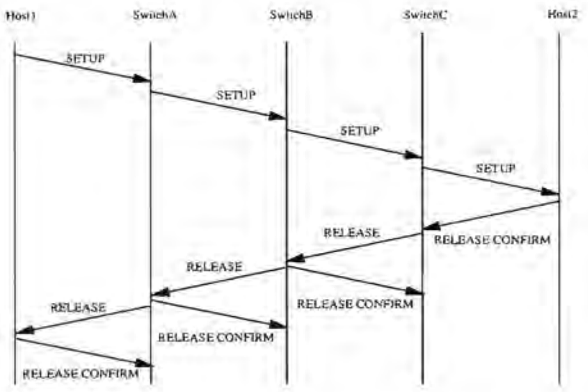


Figure 3.8: Unsuccessful setup of a connection

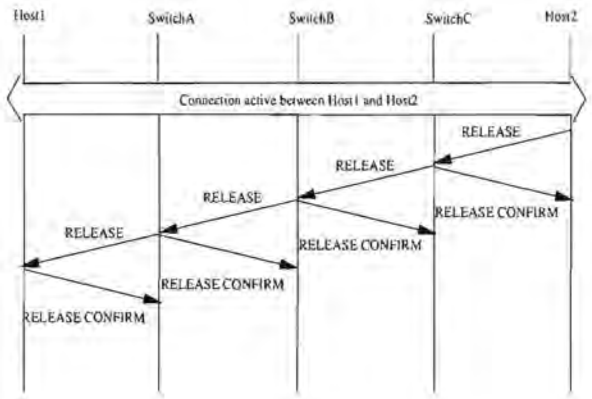


Figure 3.9: Normal connection release

Clearing the connection is initiated by the preceding node by sending a RELEASE message and initiate procedures for clearing the connection. These procedures are the notification of the application if appropriate, releasing resources held by the connection and clearing from all data tables the information of the connection. After this, it enters the *Release sent* state. In this state it expects the RELEASE CONFIRM message, after which it shall enter the *Closed* state.

Upon receipt of the RELEASE message, the succeeding node shall enter the *Release received* state and initiate the procedures for clearing the connection. After this, the succeeding node shall send a RELEASE CONFIRM message to the preceding node, send a RELEASE message to the next node in the connection and enter the *Closed* state.

Figure 3.9 shows an example of connection release initiated by the host that received the request for the connection.

Being a unidirectional connection, it is assumed that all connection releases will be initiated by the host receiving the request for a connection. This is because when a unidirectional connection is used to transfer a large file from a source to a destination, the destination is the only party that will know when data transmission has completed. Thus, if this end host receives a RELEASE message, there had to be an error in the network and the software should receive an interrupt indicating an error.

Figure 3.10 shows an example of message exchanges when an intermediate node requests the release of a connection.

During connection release it becomes apparent why the need exists to store three connection references in the state table. On receipt of a RELEASE message, a node has to determine which connection the message refers to. In the RELEASE message there are

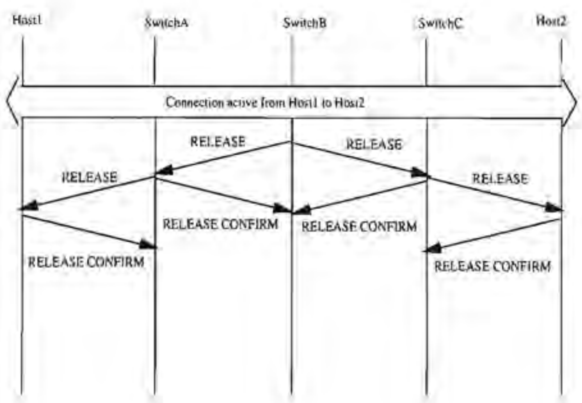


Figure 3.10: Connection release by an intermediate node

two indications, firstly there is the *connection identifier* in the message itself, secondly there is the *source address* which can be obtained from the IP header.

When a RELEASE message is received, the connection it refers to is found easily in the state table by using the connection reference included in the message as an index to the table. The source address (from the IP header) is then compared to the previous and next node entries in the state table to determine where the message came from. For example, if the message came from the previous node, a RELEASE message is sent to the "next" node containing the "Next node connection reference" and a RELEASE CONFIRM message is sent to the previous node containing the "Previous node connection reference."

3.7 Error conditions

The cause values and timers used by the signalling protocol are defined in this section. An 8 bit cause value is inserted in every RELEASE and RELEASE CONFIRM message to indicate the reason for failure. Depending on the cause value, the recipient of the RELEASE or RELEASE CONFIRM message may retry a connection request by altering the error causing parameters. For example, if a RELEASE CONFIRM message with a cause value of 1010 0010 (*Bandwidth unavailable*) is received in response to a SETUP message, the end host requesting the connection has the option of reducing the bandwidth requested and sending a new SETUP message.

This section also defines two timers that will be used in the signalling protocol.

3.7.1 Cause values

The cause values use the same format, and some values as specified in [13], although the definition and usage of the cause values may differ. This format for the cause values has also been used in [14] and [9].

In the RELEASE and RELEASE CONFIRM messages, the cause value field is 8 bits long, with the first (most significant) bit always set to 1. The cause value is divided into two fields, a class (bits 5 through 7) and a value within the class (bits 1 through 4). The cause classes are given in Table 3.13 and the cause values are given in Table 3.14.

Class	Nature of event
000	normal event
001	normal event
010	resource unavailable
101	invalid message
110	protocol error

Table 3.13: Cause classes

3.7.2 Timers

Whenever a signalling message fails a checksum test, it will be discarded immediately. There are no retransmission requests included in the current protocol. This, together with the usage of an unreliable network protocol (IP) to transfer the signalling messages prompts the requirement for some mechanism to handle erroneous or lost signalling messages. A solution is to make use of two timers in this signalling protocol, T1 and T2.

T1 is started whenever a SETUP message is sent. It is stopped when a SETUP SUCCESS or RELEASE CONFIRM message is received. On first expiry, the SETUP message is resent (and T1 is restarted). On second expiry, the connection is released with error *Nr. 11, Timer T1 expired* and the Closed state is entered.

T2 is started whenever a RELEASE message is sent. It is stopped when a RELEASE or a RELEASE CONFIRM message is received. On first expiry, the RELEASE message is resent (and T2 restarted). On second expiry, the connection is released with error *Nr. 12, Timer T2 expired* and the Closed state is entered.

The usage of timers becomes more apparent in the case when no reply is received for

Nr.	Class	Cause value	Definition
1	000	0001	Unallocated (unassigned) number
2	000	0011	No route to destination
3	001	0000	Normal/unspecified connection release
4	010	0010	Bandwidth unavailable
5	101	0001	Invalid connection reference value
6	110	0000	Mandatory information element is missing
7	110	0001	Message type non-existent or not implemented
8	110	0011	Information element/parameter non-existent or not implemented
9	110	0100	Invalid information element contents
10	110	0101	Message not compatible with call state
11	110	1000	Timer T1 expired
12	110	1001	Timer T2 expired
13	110	1111	Protocol error, unspecified
14	110	1010	TTL in SETUP expired

Table 3.14: Cause values

a RELEASE message. It is now possible for a connection to be released by a node, but if its communication with its neighbour is interrupted during the connection release, the neighbour will have resources reserved for the connection indefinitely. The addition of the T2 timer will only be able to solve this problem if we have a reliable network or a reliable transport protocol for the signalling messages. Section 3.9 shall attempt to solve this problem by introducing a reliable transport protocol.

3.7.3 Handling of error conditions

This section describes the actions to be taken when certain errors occur during connection setup.

- Whenever the integrity check of a message (using the included checksum) fails, the message shall be discarded.
- When a SETUP message is received with an illegal destination address, for example 127.0.0.1, a RELEASE CONFIRM message will be sent to the previous node with cause value *Nr. 1, "Unallocated (unassigned) number"*.
- If there is no entry in the routing table that matches the destination node, a RELEASE CONFIRM will be sent to the previous node with cause value *Nr. 2, "No route to destination"*.
- A RELEASE CONFIRM message with cause value *Nr. 4, "Bandwidth unavailable"* will be sent to the previous node if there is insufficient bandwidth available on the outgoing interface connected to the next hop.
- If a RELEASE message is received indicating a connection reference value which is not recognized as an active connection, or a connection setup in progress, a RELEASE CONFIRM message with cause value *Nr. 5, "Invalid connection reference value"* is sent in reply. Remain in the Closed state.
- If a SETUP SUCCESS message is received indicating a connection reference value which is not recognized as a connection setup in progress, a RELEASE CONFIRM message with cause value *Nr. 5, "Invalid connection reference value"* is sent in reply. Remain in the Closed state.

- When a SETUP message is received with a call reference value indicating an active connection or a connection in the process of being set up, the SETUP message should be ignored.

- When a RELEASE CONFIRM message is received with a call reference value which is not recognized as an active connection or a connection in the process of being set up, it should be ignored.

- Whenever an unexpected RELEASE message is received (for example in response to a SETUP message), the connection shall be released and a RELEASE CONFIRM message sent in reply. A RELEASE CONFIRM message is sent to the previous node in the connection and the Closed state is entered. If there is no cause value in the received RELEASE message, the cause value should be *Nr. 13, "Protocol error, unspecified"*.

- Whenever an unexpected RELEASE CONFIRM message is received (for example in the Established state), the connection shall be released. A RELEASE message should be sent to the previous node and the Release Received state entered. If there is no cause value in the received RELEASE CONFIRM message, the cause value should be *Nr. 3, "Normal/unspecified connection release"*.

- When a SETUP message is received which has one or more information elements missing, a RELEASE CONFIRM message with cause *Nr. 6, "Mandatory information element is missing"* shall be returned.

- When a RELEASE message is received with the cause value missing, the actions shall be the same as if a RELEASE message with cause value *Nr. 3, "Normal/unspecified connection release"* was received. Only, the RELEASE CONFIRM message to the node from which the RELEASE message was received shall contain a cause value *Nr. 6, "Mandatory information element is missing"*.

- When a RELEASE message is received with an invalid cause value, the actions shall be the same as if a RELEASE message with cause value *Nr. 3, "Normal/unspecified connection release"* was received. Only, the RELEASE CONFIRM message to the node from which the RELEASE message was received shall contain a cause value *Nr. 9, "Invalid information element contents"*.

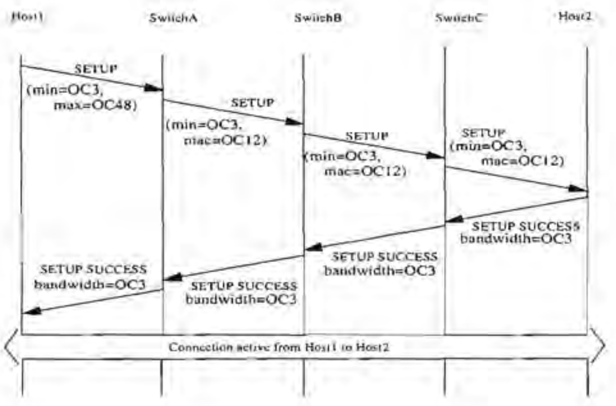

Host1
SwitchA
SwitchB
SwitchC
Host2
SETUP (min=OC3, max=OC48)
SETUP (min=OC3, mac=OC12)
SETUP (min=OC3, mac=OC12)
SETUP (min=OC3, mac=OC12)
SETUP SUCCESS bandwidth=OC3
SETUP SUCCESS bandwidth=OC3
SETUP SUCCESS bandwidth=OC3
SETUP SUCCESS bandwidth=OC3
Connection active from Host1 to Host2

Figure 3.11: Connection setup with minimum and maximum bandwidth requirements

- When a RELEASE CONFIRM message is received with a missing or invalid cause value, it will be assumed that a RELEASE CONFIRM message with a cause value *Nr. 3, "Normal/unspecified connection release"* has been received.

3.8 Maximum bandwidth selection

When an end host requests a connection, it may specify that the minimum and maximum bandwidth should be the same, in which case all the nodes along the path shall attempt to establish a connection of the requested bandwidth. If any node cannot provide the bandwidth, it shall send the RELEASE CONFIRM message as described in Section 3.6.1.

A node can also specify a range of bandwidth, specified by a minimum and a maximum. These two values (minimum, maximum) indicate to the receiver of the SETUP message that the preceding node requests a connection of bandwidth *max bandwidth*, but if the request cannot be satisfied, the requesting host will be satisfied with any bandwidth larger than or equal to *min bandwidth*. If *max bandwidth* cannot be satisfied, but another value larger than (or equal to) *min bandwidth* can, then the SETUP message shall be changed to indicate the new maximum bandwidth - which may result in the minimum and maximum bandwidths being equal somewhere along the path. The resource reservation shall continue as specified in Section 3.6.1. This may result in different bandwidth allocations along the path of the connection, but it shall be rectified at the receipt of the SETUP SUCCESS message. The end host to which the connection is being set up shall place the last value of *max bandwidth* in the *bandwidth* information element of the SETUP SUCCESS message. While the SETUP SUCCESS message traverses the network, all nodes which form part of the connection shall edit their allocations accordingly.

For example in Figure 3.11, Host 1 requests a connection with bandwidth OC48, it also indicates that it will be satisfied by a connection with bandwidth of OC3. Switch A does not have enough resources available, but it does have resources available for an OC12 connection which it reserves for the connection. Switch A goes ahead by editing the SETUP message to reflect the changes and passes the request on to Switch B. Switch B does have enough resources available for the OC12 connection, and so has Switch C. Unfortunately, Host B only has enough resources for an OC3 connection. So, it reserves the resources for the OC3 connection and sends the SETUP SUCCESS message to Switch C with the *bandwidth* information element containing OC3. Switch C changes the allocation it made previously (only reserving resources for an OC3 connection instead of an OC12 connection) and passes the message to Switch B. In this way the message traverses the network, and an OC3 connection is successfully set up between Host A and Host B.

This solution has the unavoidable drawback that resources that are not going to be used are unavailable for a short time.

3.9 Improvements to signalling protocol

3.9.1 Routing through heterogenous nodes

So far in the discussion we assumed the network is always homogeneous. Now, this assumption is relaxed and we consider the problem associated with *heterogenous networks* (which are networks consisting of switches with different cross-connect rates). First we consider a simple example illustrated in Figure 3.12. In this example end host A requests a connection with rate OC1 to end host B. Switch S1 receives this message but its immediate neighbour has a higher cross-connect rate, and hence an OC1 connection cannot be provided.

The routing protocol which is described in Chapter 4 will construct a routing table in which the "next-hop node" entry of node I may be a node that is not directly connected through a physical or logical link to node I. It is considered a "neighbour" if it is the next-hop switch that operates at the same or lower rate as node I through which a connection must be routed.

Connection setup in a heterogenous network can now be explained as follows. If the "next-hop node" entry in the routing table indicates a node that is not an immediate neighbour of node I (physically or logically) as determined by examining the connectivity

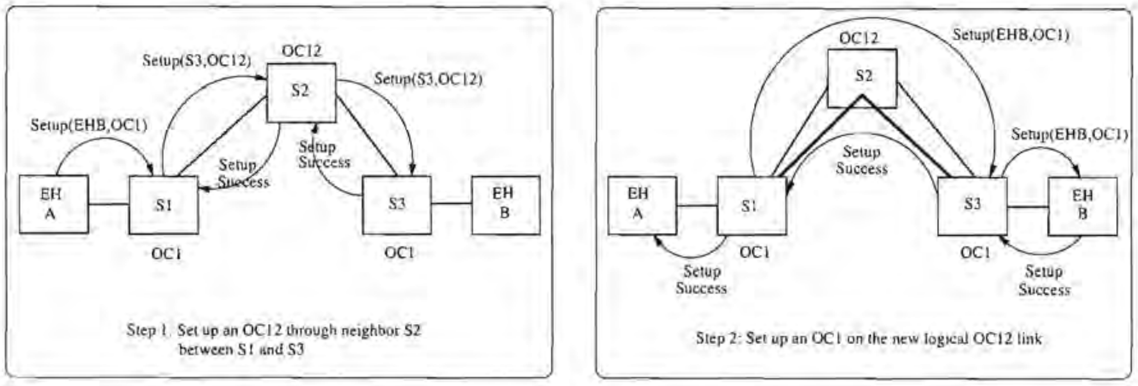


Figure 3.12: Connection setup in heterogenous network

table, the routing table is queried again to locate which immediate neighbour should be used to reach the "next-hop node" of node I. Assume that node J is the immediate neighbour and node K is the "next-hop node". Node I must determine the cross-connect rate of node J and request a connection setup at this rate between itself and node K. Once this is complete, the setting up of the lower rate connection can proceed over the newly created logical link between nodes I and K.

In the example of Figure 3.12, the routing table of switch S1 will look as follows. For data rate OC1 and destination B the "next-hop node" shall be given as S3. According to the information in the connectivity table (Table 3.5) S3 is not a neighbour of S1. So, it has to search the routing table again to look for the entry where S3 is a destination. The lowest data rate at which there is an entry for S3 shall be used as the requested rate when a connection is set up between itself and S3. Once this connection is established, nodes S1 and S3 become immediate neighbours on this newly created logical link. Switch S1 can then send the OC1 *Setup* request to switch S3 for further routing as shown in Step 2 of Figure 3.12.

Another example is given in Figure 3.13. In this example end host A starts by sending a setup request to switch S1 for a connection of rate OC12 to end host B. Switches S1, S2 and S3 can all handle the connection requirements due to their cross-connect rates being smaller or equal to the requested rate. We shall follow the progress of the connection setup from switch S3.

Switch S3 receives the setup request for an OC12 connection, to pass the connection setup request on to the next hop it queries the routing table. The routing table has an entry for data rate OC12 that indicates that switch S6 is the "next-hop node" (Table 3.15).

Destination node address	Data rate OC12			...	Data rate OC192		
	Next hop option 1	Next hop option 2	Next hop option 3		Next hop option 1	Next hop option 2	Next hop option 3
End host B	Switch S6						
Switch S6					Switch S4		

Table 3.15: Routing table of switch 3

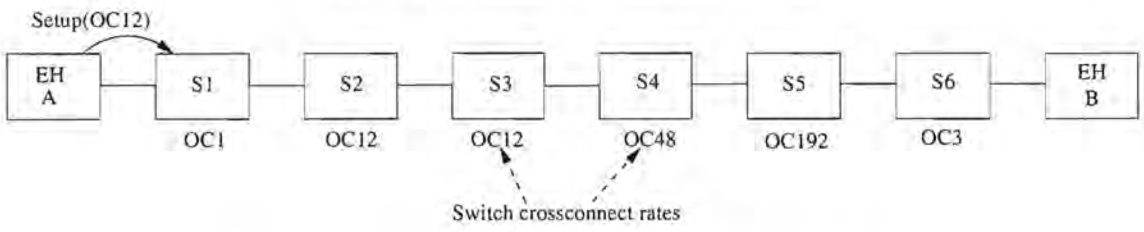


Figure 3.13: Connection setup in large heterogenous network

Querying the connectivity table, switch S3 realizes that S6 is not an immediate neighbour, so it searches the routing table again for an entry with switch S6 as a destination. Because OC192 is the highest data rate on the path to switch S6, there shall be no entry for S6 under data rate OC48, but there will be an entry with data rate OC192 which is presented in Table 3.15. According to Table 3.15 the next hop is switch S4. This switch is directly connected, so switch S3 requests a connection with rate OC192 from itself to switch S6 with the next hop being S4. After this connection is set up, S3 can send the request for an OC12 connection directly on the new logical link.

The question of when the logical link is released has not received much attention, but it should be noted that the logical link can be held open even when the connection that prompted its setup is closed. This enables the setup of other connections over this logical link without the added overhead of setting up the logical link. Considering the first example again, the logical OC12 link between switch S1 and switch S3 can be held open even if the OC1 connection closes. With the help of the routing protocol, this link can be advertised, and new OC1 connections can be set up over it. Once the last OC1 connection closes, a timer can be started, if no new requests for OC1 connections are received before the timer's expiration, the logical link can be closed.

Note that when a connection passes through a switch with a lower crossconnect rate, for example by realizing an OC12 connection as 12 OC1 connections, some mechanism of multiplexing/demultiplexing is required. These actions have been assumed to occur transparently because of the synchronous nature of the SONET network.

3.9.2 Transport of signalling messages

Transporting the signalling messages over the IP network has already introduced two important limitations. Firstly, using IP without TCP implies unreliable transmission. The network does not provide any guarantee that the signalling message will reach its destination, there is no indication if the signalling message has reached its destination and if the message reaches the destination, there is no guarantee the data is intact. As described in Section 3.7.2, the unreliable nature of the network can cause some switches to waste valuable resources. The second limitation is security. Although this is not the focus of this research, the lack of authentication mechanisms, together with the use of a public network for the transport of the signalling messages might introduce some security risks, for example denial of service attacks, to the circuit switched network. The use of a private IP network just to transport signalling messages is wasteful of resources.

The IETF Signalling Transport working group is currently working on the Stream Control Transmission Protocol (SCTP) [22]. SCTP started out as a transport protocol for PSTN signalling messages over the IP network, but the capabilities of this protocol will solve the problems introduced when only IP is used to transfer the signalling messages. SCTP provides acknowledged error-free non-duplicated transfer of user data and is resistant to flooding and masquerade attacks [22]. However, with our goal for hardware implementation, it still needs to be determined if SCTP or TCP can be implemented in hardware.

Another option is to carry the signalling messages using SONET. Inside a SONET frame there are three section overhead bytes (named D1, D2 and D3) and eight line overhead bytes (named D4 to D12). These bytes are termed the section data communications channel (DCC) and line data communication channel (DCC) respectively and are used for operations, administration, maintenance and provisioning (OAM&P). Carrying signalling messages on one of these point-to-point DCC channels will eliminate the need for an IP network.

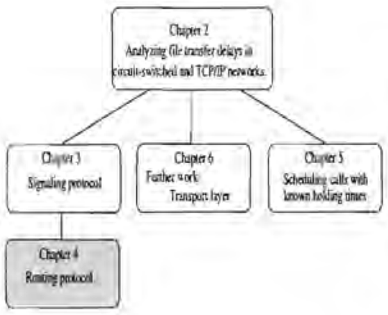
3.10 Conclusion

Signalling protocols in current TDM networks only provide low-bandwidth (DS0) circuits on-demand to one application (telephony). This chapter describes a new hardware implementable signalling protocol that is able to provide high-bandwidth on-demand circuits for applications such as bulk data transfers. The signalling protocol presented in this chapter requires a supporting routing protocol. The routing protocol that will be presented in Chapter 4 is responsible for the creation of the routing table entries that are required by the signalling protocol.

Chapter 4

Routing protocol

4.1 Introduction


Chapter 2
Analyzing file transfer delays in circuit-switched and TCP/IP networks.
Chapter 3
Signaling protocol
Chapter 6
Futher work:
Transport layer
Chapter 5
Scheduling calls with known holding times
Chapter 4
Routing protocol

An accompanying routing protocol is required for the signalling protocol presented in Chapter 3. Recall from Section 3.4 that we selected to use hop-by-hop routing and route precomputation for the signalling protocol. This means that when a request for a connection arrives at a node, the routing table should contain all the information required to produce a next-hop node address for any address reachable through itself.

The routing table produced by the routing protocol should also contain enough information to be able to accommodate connection setup in a heterogenous network. The "next-hop node" entry, J, in the routing table of node I may be a node that is not *directly* connected through a physical or logical link to node I. In order for node J to be considered a "next-hop node" by node I, node J *must* operate at the same or lower rate as node I. As a result, on a physical path from node I to the next-hop node J there may be one or more intervening nodes that are not regarded as next-hop nodes. If the "next-hop node" entry in a routing table indicates a node that is not an immediate neighbour of node I (physically or logically), a second entry is needed indicating which immediate neighbour should be used to reach the "next-hop node". Section 4.2 starts by introducing the addressing scheme that will be used by the routing protocol, Section 4.3 continues with the design decisions, and Section 4.4 describes the routing protocol.

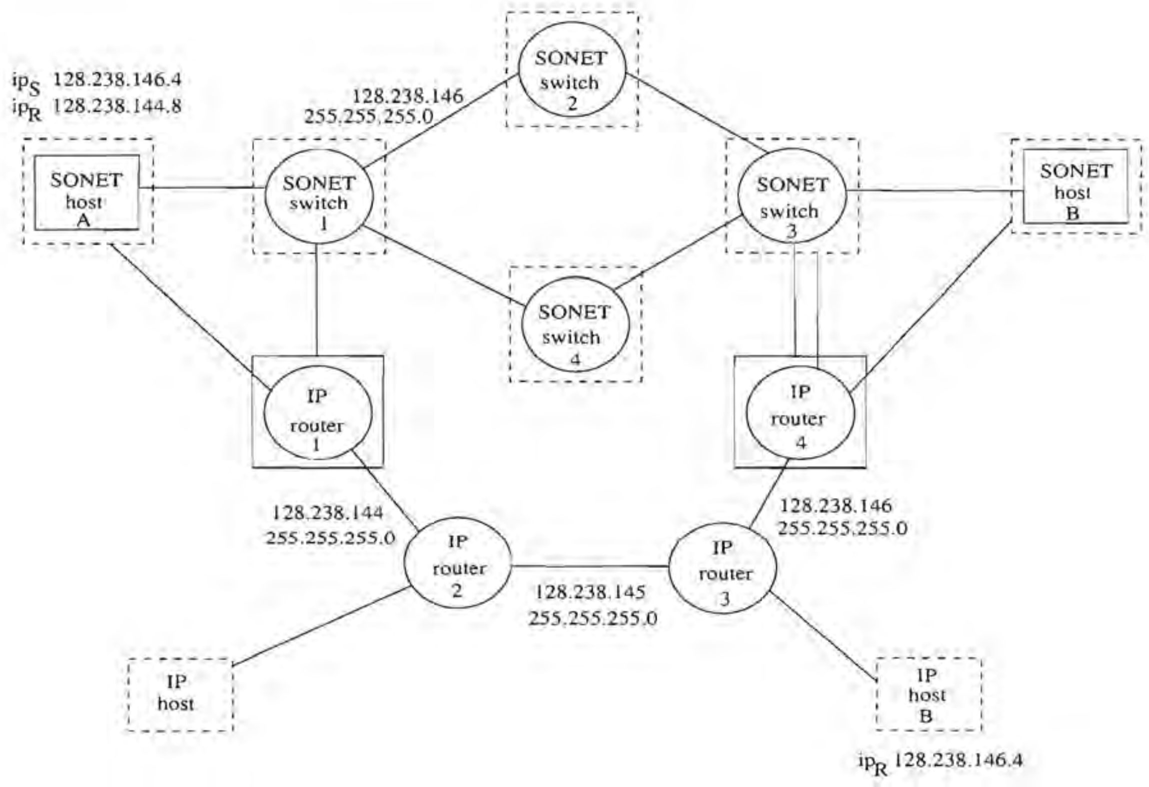


Figure 4.1: Routing network

4.2 Addressing

For an efficient, low overhead routing protocol, the need exists for an addressing scheme that allows for summarization. Address summarization refers to the use of address prefixes to represent a collection of end host addresses that begin with the same prefix. Only the address prefixes need to be distributed between nodes participating in the routing protocol and represent the reachability of all end hosts. This section presents an addressing scheme for use in SONET networks. It is proposed that a second set of IP addresses be used to identify all nodes forming part of the SONET network, an addressing scheme that lends itself to address summarization without reducing the number of IP addresses available to the IP network

A benefit of using IP addresses is that recent work on route lookups in hardware, [8] and [16]-[18], can be leveraged for the first action of route determination needed in call setup.

Figure 4.1 shows an example network in which the signalling protocol presented in Chapter 3 will be implemented. In the figure all IP hosts are indicated by a dashed rectangle and all SONET hosts are indicated by a solid rectangle. This example network, where only some interconnections are indicated, shows the co-existence of the IP and SONET networks. All SONET hosts and SONET switches are IP hosts, and some IP routers are hosts of the SONET network. It is assumed that all SONET hosts and switches will have IP interfaces, for example in the form of Ethernet interfaces. For example, in Figure 4.1 the link between SONET host A and SONET switch 1 is a SONET interface, while the link between SONET host A and IP router 1 could be an Ethernet interface. IP routers are hosts of the SONET network in the Internet today where there are provisioned dedicated channels set up between routers interconnecting large sites.

With the above network design, the decision to make use of IP addresses for signalling and routing in the SONET network becomes plausible. Although every SONET switch and host already has an IP address, the usage of these addresses will introduce an immense strain on the routing protocol by increasing the sizes of the routing databases considerably. The reason for this is that in the construction of the network, it cannot be guaranteed that two neighbouring (connected by a point-to-point link) SONET nodes will be part of the same IP subnet. As a consequence, each SONET node has to be identified by its complete 32 bit IP address in the routing protocol, preventing the usage of address summarization.

For the routing protocols to make use of IP addresses while retaining the capability of address summarization that is provided with the use of subnet addressing, two preconditions are met. Firstly, in the network presented in Figure 4.1, it can be seen that one node of one network type can only be a host of the other network (as opposed to a router or switch). For example, an IP node (router or end host) can only be an end host of the SONET network, not a switch. Similarly, a SONET node can only be an end host of the IP network, not a router. Secondly, an IP address is assigned to each interface, IP and SONET, forming two distinct sets of IP addresses, which do not need to be disjoint.

If ip_R indicates an IP address of an IP interface, and ip_S indicates an IP address assigned to a SONET interface, Figure 4.1 presents a possible address assignment to SONET host A. Host A has an ip_R of 128.238.144.8 assigned to its interface that is connected to IP router 1. It also has the ip_S address of 128.238.146.4 assigned to the interface connected to SONET switch 1. Figure 4.1 also shows that SONET host A and IP host B now share an IP address. Although it is the same IP address, there shall never be any conflicts due to the context in which the addresses are used. Any IP address used in an IP header will always be an ip_R address and any IP address used in the called party address field of a SONET circuit signalling message will be an ip_S address. All signalling and routing messages of the SONET network will be carried as IP packets with ip_R addresses in the headers and hence will reach the appropriate SONET destinations because they are hosts in the IP network. The SETUP message (without the IP header) contains two IP addresses that identify the source and destination nodes of the SONET network between which a connection has to be set up. These two IP addresses are always ip_S addresses. It is to find a route between this source and destination pair that a routing protocol is needed. So, the IP addresses used for the actual signalling will never be present in an IP header, only in the SETUP message. Because these addresses will never be used in the same context, it allows the SONET network to have nodes identified by IP addresses already used in the IP network.

In a network as depicted in Figure 4.1, there will always be two routing protocols running concurrently. The IP routers exchange information about reachability of nodes with ip_R addresses, and the SONET switches exchange information about reachability of nodes with ip_S addresses. These two routing protocols never exchange routing information between each other.

It is now possible to assign IP addresses to SONET interfaces independent of the IP address assigned to their IP interfaces, allowing for a SONET switch and all its neighbours to share a subnet address. Thus allowing for address summarization.

All data tables discussed in the signalling protocol description shall contain ip_S addresses. The tables concerned are the *Routing table*, the *Connectivity table* and the *State table*. This introduces the need for a new table that provides a mapping between ip_R and ip_S addresses of which Table 4.1 is an example. While participating in connection setup, a switch will always send a signalling message to a neighbour. Note that this might not be an immediate neighbour (connected via a point-to-point link) in a heterogenous network - it could also be a node to which a connection has already been set up - a *logical neighbour*. In a homogeneous network, this table can be set up by the administrator due to the fact that signalling messages will only be sent to immediate neighbours (two nodes connected with a point-to-point link). In a heterogenous network, the initial information in the *ip_S to ip_R mapping table* (which is a mapping of the addresses of a node's immediate neighbours) is entered by the administrator. To be able to handle connection setup in a heterogenous network the signalling protocol should be able to send signalling messages to logical neighbours. For this functionality, the signalling protocol needs to add one more step during connection setup. Once a connection has been set up between two switches,

ip_S	ip_R
128.238.146.4	128.238.144.8

Table 4.1: *ip_S to ip_R mapping table* kept by SONET switch 1

the source switch needs to update the *ip_S to ip_R mapping table* to include the ip_S to ip_R mapping of the destination. This mapping can be inserted upon the receipt of a SETUP messages, and removed upon receipt of a RELEASE CONFIRM message.

The signalling protocol now requires an extra step before a signalling message is sent: each time a signalling message needs to be sent, the address mapping table has to be queried to determine the IP address that needs to be placed in the IP header. An example step when SONET switch 1 receives a request for a connection from SONET host A to SONET host B. Switch 1 first queries its routing table to find the ip_S address of the next hop to which the SETUP message should be sent. Then after CAC and switch fabric configuration it queries the *ip_S to ip_R mapping table* to find the ip_R address of the next hop node. It constructs a SETUP message and requests the IP network to deliver it to the SONET node with IP address ip_R.

4.3 Design decisions

The usage of IP addresses by all the nodes of the SONET network allows for the use of an existing routing protocol, for example OSPF [19]. Using a routing protocol like OSPF will only require two changes: first, before any routing messages are sent on the IP network, the *ip_S to ip_R mapping table* has to be queried to find the ip_R address. Second, the construction of the routing table has to be changed to enable routing in heterogenous networks. OSPF even contains rudimentary support for Quality of Service (QoS), which allows the exchange of node state (real-time) parameters, for example the current available bandwidth of outgoing interfaces.

Although an existing IP routing protocol may be used, this discussion shall continue with the design of a lightweight routing protocol designed specifically to be used in conjunction with the signalling protocol described in Chapter 3.

The presence of the IP network will be ignored in the rest of the routing protocol description. The arguments for the creation of the hop-by-hop, link-state routing protocol that creates a routing table by constructing a shortest path tree is described next.

Section 3.4 briefly mentioned the decision that the options of **route precomputation** and **hop-by-hop** routing (as opposed to source routing) will be used in the routing protocol. Using SONET it is possible to make full use of route precomputation due to the granularity of the multiplexing rates. As depicted in Table 3.3, next-hop entries are only kept for the SONET rates of OC1, OC3, OC12, OC48 and OC192. This simplifies the connection setup procedure in that the route determination step is just a table lookup. There are two reasons for the decision of hop-by-hop routing. Firstly, by using hop-by-hop routing the parsing of the SETUP message is simplified at each hop allowing for hardware implementation. Secondly, when source routing is used it is possible that conditions change as connection setup proceeds from a given source node toward a destination, and by the time the setup request reaches an intermediate node, the selected source route is no longer available, in which case crankbacks are needed. Crankbacks require the management of more state information, which leads to increased hardware complexity.

The routing protocol should be executed in parallel by all the SONET switches, without segmenting the network. This allows all the nodes to have an identical copy of the routing database. The reason that the current routing protocol proposal requires a flat network structure (no segmentation) is for simplicity. The moment a network is segmented, the different "areas" can receive summarized information regarding the real time parameters in the other areas. Because the information is summarized, some mechanism similar to Generic CAC (GCAC) from PNNI needs to be applied to a connection request before it is passed on to the next hop (together with the CAC performed by the node itself). GCAC allows a node to predict the outcome of the actual CAC performed at another switching system given that node's advertised additive link metrics. Performing GCAC will add complexity to the signalling protocol, and will heighten the risk that the real CAC performed by a switch further along the path (possibly in another area) may fail. This failure will introduce crankback. A flat network structure avoids this problem.

Routing databases tend to be very large. Using a routing protocol to distribute real time parameters, for example the available bandwidth of an interface, requires the protocol to always have up to date information in the database and to respond quickly to any changes in the database. A distance-vector routing protocol such as RIP [20] would be inefficient when used in large networks (large routing databases). A distance-vector algorithm, such as the Bellman-Ford algorithm used by RIP, does not perform well when there is a large database and a topology change occurs or a change in a node's real time parameters occurs. This is because a change in the network results in a convergence period which might not be fast enough for the routing protocol to be usable.

In distance-vector algorithms, each node keeps a table with an entry for every possible destination reachable though itself. The entry indicates the distance to the destination and the next-hop node to the destination. The distances are continually compared and updated as routing update messages are received by a node. Once a topology change occurs (for example a link goes down), and the changes to the distances are distributes through the network, the routing tables will not immediately see the change because some nodes will still consider the link as active and reflect that in their advertisements. This is the cause of the convergence period that is the time for a node's routing table to reflect the correct distance values to the affected destinations.

It is thus proposed to make use of a **link state** routing algorithm. In this algorithm each node has an identical routing database containing the identity (address) and connectivity of each node in the network from which a routing table is constructed. The routing table entries are calculated by constructing a shortest path tree. By using a link state routing algorithm, any changes in the network topology or real-time parameters of nodes in the network will be received by all the nodes executing the routing protocol very quickly, thus enabling all the nodes to have the most up to date information regarding topology and real time parameters. With this information the routing protocol is able to make more accurate routing decisions than a distance-vector routing protocol.

Information distributed by the switching systems participating in the routing protocol should contain enough detail about their respective *identity* (reachability) and *capability*. Information concerning individual end hosts' *capability* (for example an end host's interface rate) cannot be distributed due to the usage of address summarization. This information (*identity* and *capability*) can also be described as *nodal information* and in this implementation it describes the node's address (the ip_S address) and the cross connect rate of the switching system. Together with the *nodal information*, the protocol also distributes *topological information*, which is information regarding the links between the switching systems. The *topological information* is a non-negative cost assigned to an outgoing interface based upon the amount of available bandwidth at the interface: the higher the available bandwidth of an interface, the lower the cost value. A lower cost value thus indicates a higher probability for this interface to be chosen.

At initialization, a node advertises that it has an available bandwidth equal to the interface rate (resulting in a low cost value). It is proposed that the distribution of subsequent topological information be based upon threshold computations. That is the node participating in the routing protocol should not send a routing update message every time a connection is admitted or released. Only when the available bandwidth reaches a certain threshold should it send out a routing update message containing a higher cost value to indicate a change in its available bandwidth.

As with the signalling protocol, the routing protocol also makes use of the IP network to distribute routing database updates. Considering the delay involved in the connectionless network, the usage real-time parameters might introduce some connection setup failures. For example, if node I passes a connection request to node I+1 based on the information in its routing database that states that node I+1 does have enough bandwidth available, and at the same time , node I+1 sends out a routing update message that it does not have enough bandwidth available - the connection request will fail. For this reason multiple next hop entries are kept in the routing table. If the connection request fails, the setup request is reconstructed and sent to the second (or third) option for a next hop node.

4.4 Protocol description

The routing protocol will be described through its four components. All the switching systems in the network have an identical **routing database** that describes the *nodal* and *topological information* of each node in the network. This database allows each node to construct a **directed graph** representing the network. From this directed graph, each switching system constructs the **shortest path tree** with itself as the root, and using this tree it is now possible to compute the **routing table** entries.

An example network is shown in Figure 4.2. This example, and all the examples in this chapter will refer to the switching systems by their names (SwitchA, SwitchB, etc.). Note that this representation is used to simplify the explanations, an implementation would use the IP (ip_S) addresses. Each directed line indicates a unidirectional connection between two switching systems. A cost is associated with each outgoing interface. The cross connect rates are also depicted because these values play an important role in the construction of the routing table.

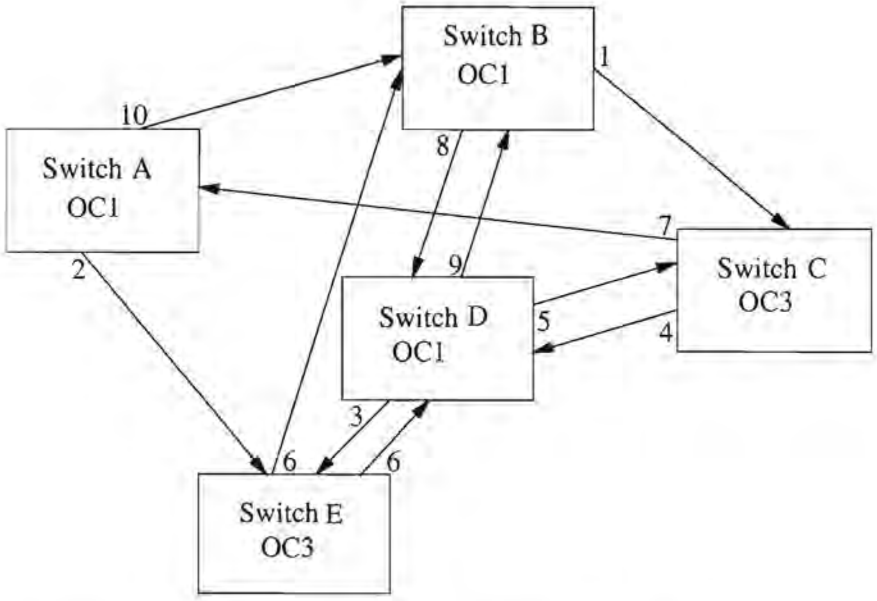


Figure 4.2: Example network configuration for routing protocol

Next hop node	Interface number	Total bandwidth	Available bandwidth	Cost

Table 4.2: Available bandwidth table including cost

4.4.1 Routing database and directed graph

For this routing protocol, each switching system needs to know to which other nodes it is directly connected. This information is currently stored in the *Connectivity* and *Available bandwidth* tables used by the signalling protocol. All connections will be unidirectional, so we are only interested in the succeeding, directly connected nodes. This information is stored in the *Available bandwidth* table. The cost associated with an interface is programmed into the *Available bandwidth* table, Table 3.4, of which Table 4.2 is a copy. The cost is a decreasing function of the available bandwidth of an interface - the higher the available bandwidth, the lower the cost.

For each node n (including itself) in the network, the routing database of each switching system in the network contains information indicated by the first column of Table 4.3.

- **Node id** refers to the ip_S address of node n.
- The **sequence number** characterizes the last routing update concerning node n.

Node id	SwitchD
Seq	x
Connectivity	SwitchB, SwitchC, SwitchE
Cost	9, 5, 3
Cross connect rate	OC1

Table 4.3: Routing database entry for node D from network depicted in Figure 4.2

This number is used to determine if a routing database update received from any node about node n does in fact contain new information, a higher sequence number indicates more recent information. The usage of the sequence number will receive more attention later.

- **Connectivity** indicates all the nodes directly connected to node n. That is, all the nodes of which node n is the preceding node in a unidirectional point-to-point physical connection. Nodes are identified by a combination of ip_S address and netmask. For example, an ip_S address of 128.238.144.0 and netmask of 255.255.255.0 indicates that all the hosts on the 128.238.144 network are directly connected to (and consequently reachable through) node n.

- **Cost** refers to the costs associated with the interfaces connecting this node (n) to other nodes. Each entry is an indication of the available bandwidth on the interface connecting node n with a node indicated by the *Connectivity* information - the lower the cost, the higher the available bandwidth.

- **Cross connect rate** is given if node n is a switching system.

With this information in the routing database it is straightforward to construct the directed graph.

If x is the highest sequence number of a routing update received concerning SwitchD, an example of an entry in the routing database for SwitchD is shown in Table 4.3. This entry will be identical in all the nodes in the network.

The cost assigned to interfaces which connect a switch to its end hosts should typically be equal. For example, if all the hosts on the 128.238.144 network are reachable through SwitchD, it shall advertise its connectivity with an ip_S address of 128.238.144.0, netmask of 255.255.255.0 and only one cost value. If there are different costs assigned to interfaces connecting hosts (or switches) that have addresses that can be summarized, the highest cost of all the interfaces will be used in all routing databases.

To limit the usage of network bandwidth a node initiates database exchange with its neighbour only on three occasions:

- At initialization.
- After any change occurs in its routing database. This includes any changes in its own connectivity, an increase over the threshold of the available bandwidth, a decrease below the threshold of the available bandwidth, or a routing update received from another node.
- After a very large time interval. It is assumed that the above two database exchange instances will guarantee database synchronization most of the time, but it is also necessary to have some error prevention mechanism, which is provided by this database exchange condition.

When node n's routing database entry changes, it should only send the new database entry to its neighbours, after which it is distributed through the network. In the other two database exchange occasions, the whole routing database is distributed among the nodes.

Routing databases are not flooded through the network. Each node shall send its routing database update message(s) to all of its neighbours that are reachable through itself (the "next-hop nodes" found in the *Available bandwidth* table) and that are participating in the routing protocol. For example, in Figure 4.2 SwitchA will send its routing database update message(s) to SwitchB and SwitchE, but these latter switches will not relay the messages on to SwitchC or SwitchD.

Only one message type is needed for this routing protocol. For each node n in the network about which a node has information in its routing database, it shall construct a message as presented in Table 4.4. It is possible to include more than one full message in an IP datagram.

The sequence number is only changed when the information it refers to has changed, and it is also only changed by the node to which the information refers. When a node sends its database to its neighbours after the large timeout it uses the same sequence number associated with its information in the database. Continuing the example this means that until a change occurs in SwitchD's information, the sequence number will continue to be x.

Information element	Size (bits)
Source node	32
Sequence number	31
Flag	1
Cross connect rate	4
Number of destinations (*nr. dest*)	12
Destination (1)	32
Netmask (1)	32
Cost (1)	16
...	
Destination (*nr. dest*)	32
Netmask (*nr. dest*)	32
Cost (*nr. dest*)	16

Table 4.4: Routing update message

The sequence number is a 31 bit unsigned integer. Each time there is a change in a node's connectivity it shall increment the sequence number associated with its information in the database and initiate the routing database update with its neighbours. Thus, the larger the sequence number, the more recent the database update. The problem with using a sequence number is deciding what to do when the sequence number has to be incremented past $2^{31} - 1$. A solution for this problem is to have an extra bit (*Flag*) in the routing update message. When a sequence number reaches $2^{31} - 1$ a node can reset the sequence number to 0 and set the extra bit to 1 indicating that this is a new routing update. All nodes in the network should now treat this message as if a message with a higher sequence number has arrived. The extra bit in the routing message should be set to 0 on all other occasions.

Using 12 bits to indicate the number of destinations allows any switching node to advertise 4095 different destinations reachable through itself. This is a worst case. The number of routes advertised by any node should typically be very low.

The 4 bit cross connect rate value is the cross connect rate of a switching system or the interface rate of an end host (although an end host should not typically participate in the routing protocol). The possible values of this information element is presented in

Cross connect/Interface rate	Value in routing message
OC1	0001
OC3	0010
OC12	0011
OC48	0100
OC192	0101
OC768	0110

Table 4.5: Possible values for the cross connect rate of a switch

		From				
		Switch A	Switch B	Switch C	Switch D	Switch E
To	Switch A			7		
	Switch B	10			9	6
	Switch C		1		5	
	Switch D		8	4		6
	Switch E	2			3	

Table 4.6: Directed graph

Table 4.5.

The directed graph for this network is represented in Table 4.6 and is identical at each node participating in the routing protocol.

4.4.2 Constructing the shortest path tree

The shortest path tree is constructed from the directed graph using the Dijkstra algorithm. The construction of the shortest path tree will be described next. The directed graph contains enough information for a switching system to construct the shortest path tree using itself as the root. We deviate from common graph terminology in this explanation by continuing to use terms from previous explanations. That is, instead of the normal usage of "vertex" or "edge" the words "node" and "link" will be used to explain the algorithm. The algorithm assumes the availability of a function w such that $w(u, v)$ returns the cost of the interface from **u** to **v** as retrieved from the directed graph. The algorithm proceeds as follows:

1. Create the four data structures. **V** is a list (or array) of all the nodes in the network (identified by ip_S address and netmask combinations) whose shortest path have yet to be found, **S** is a list (or array) of nodes whose shortest paths from the root have already been determined, **d** is a list (or array) of the current cost estimate along the shortest path to each node, and **pred** is a list (or array) of of each node's predecessor on its shortest path to the root.

2. Set **S** to empty.

3. Denote the cost to node u by $d[u]$. Set $d[u] = 0$ if u is the root and $d[u] = \infty$ otherwise.

4. Denote the predecessor of each node u by $pred[u]$. Set $pred[u] = u$ if u is the root and $pred[u] = NIL$ otherwise.

5. While there are still nodes in **V**:

   (a) Sort the nodes in **V** according to their current values in **d**.

   (b) Remove the closest node **u** from **V**.

   (c) Add **u** to **S**.

   (d) For each node, **v**, that is adjacent to **u** in the directed graph.

      i. if $d[v] > d[u] + w(u, v)$ then:

         A. $d[v] = d[u] + w(u, v)$

         B. $pred[v] = u$

Using the Dijkstra's algorithm the shortest path tree computed by SwitchB from Figure 4.2 is depicted in Figure 4.3. The shortest path from the root node to a node, n, can be found by traversing the predecessor links from n until the root node is encountered. If the node, n, has a NIL predecessor link, then there is no path from x to the root node. The predecessor list for SwitchB is presented in Table 4.7. Traversing the predecessor list to find the shortest path to SwitchE results in the path SwitchB-SwitchC-SwitchD-SwitchE. The first node in the shortest path will be used in the construction of the routing table. This will only produce one next-hop, in Chapter 3 the routing table (Table 3.3) contains three next-hop entries. To compute the additional two optional next hops that will be placed in the routing table (as shown in Table 3.3), the root node can construct two additional shortest path trees. After constructing the first shortest path tree, all links used

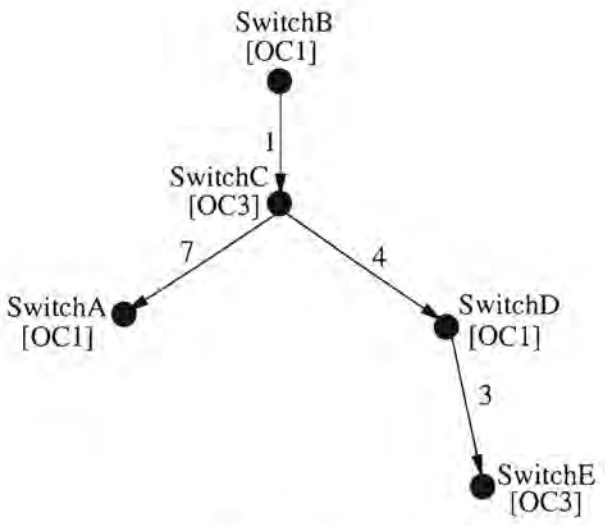


Figure 4.3: Shortest path tree constructed by node B

Node	Predecessor
SwitchA	SwitchC
SwitchB	SwitchB
SwitchC	SwitchB
SwitchD	SwitchC
SwitchE	SwitchD

Table 4.7: Predecessor list for SwitchB

in the tree are removed from the directed graph. The new directed graph is then used to construct a shortest path tree from which the second routing table entries are made. The same procedure is followed to compute the third optional next hop entries in the routing table.

4.4.3 Routing table

The shortest path tree of a node has the information for a node to decide on the next hop node to the destination. The next hop to a destination can be found by traversing the predecessor list of the shortest path tree. In a homogeneous network, the routing table can be constructed by each node *root* by considering each node n in turn. The IP address of the first hop in the shortest path from node *root* to node n becomes the entry of node n in the routing table of node *root*.

So far in the discussion it has been assumed that when a request for a connection

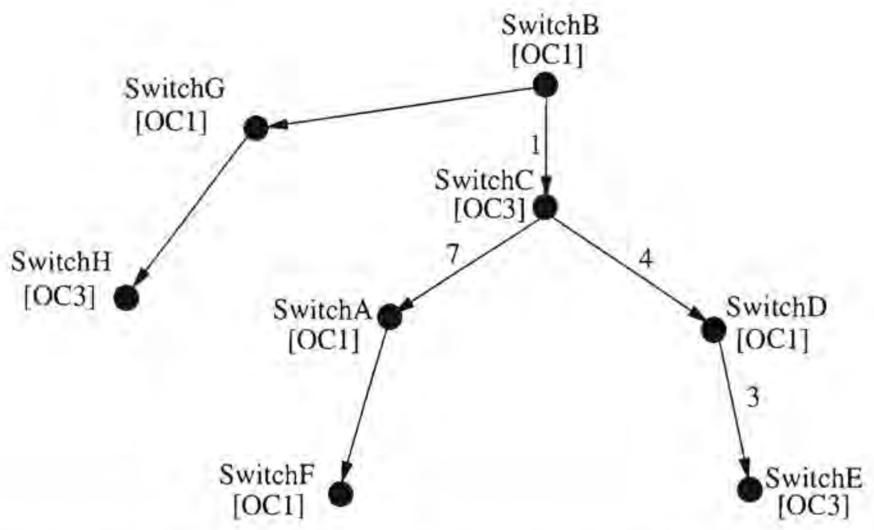


Figure 4.4: Shortest path tree constructed by node B, with extra node

arrives, the first step would be to do a route lookup in the routing table to find the next-hop node to the destination. It is also possible to first search the *Available bandwidth* table to find out if the destination is directly connected before the lookup is done in the routing table. This discussion will now continue the design where the routing table is always queried first. Thus if the destination is directly connected, the "next-hop node" entry will contain the IP address of the destination.

For this explanation, the shortest path tree in Figure 4.3 has been changed to Figure 4.4.

For this routing table to be useful in a heterogenous network, the construction of the routing table by a node (*root*) can be described as follows:

Consider the shortest path P from *root* to destination node n.

- If the next hop node *next* has the same or lower cross connect rate as node *root*, make an entry in the routing table under the section with the same cross connect rate as *next*.
- If the next hop node *next* has a higher cross connect rate than node *root*:
  - Find the closest *neighbour* node on path P that has the same or lower cross connect rate as node *root*.
    - If the *neighbour* node is found and is not the same node as n:
      - Place this node (*neighbour*) in the routing table as the "next-hop node"

used to reach node n. This entry is made at the crossconnect rate of *neighbour*.

- Search the path from *root* to *neighbour* for the node with the highest cross connect rate (*highrate*).
- Place *next* in the routing table as the next-hop node to reach *neighbour*. This entry is made at the cross connect rate section *highrate*.

* If the *neighbour* node turns out to be n or if no *neighbour* node is found:
  - Search for the node with the highest cross connect on path P
  - Place the entry for the next hop node *next* in the routing table at this cross connect rate section.

Due to the granularity of the SONET network, each entry made using the above description can be inserted at all the rates higher than its original position, if there are outgoing interfaces that operate at these or higher rates. It is impossible for a switch to connect to another node at a rate lower than its crossconnect rate, so all entries made to this respect has to be removed after the previous step has been completed.

When SwitchB constructs the routing table according to the shortest path tree depicted in Figure 4.4, it will proceed as follows. The first node it considers is SwitchC, which has a higher cross connect rate than SwitchB. As explained above, SwitchB shall search the path to SwitchC for a *neighbour*, there is no neighbour on this path, and the highest cross connect rate is OC3 - so an entry for SwitchC will be made in the routing table block under the rate OC3. Next, consider node SwitchF. Again, the next hop node (SwitchC) has a higher cross connect rate than SwitchB. Searching the path for a node that has the same or lower cross connect rate than SwitchB reveals that SwitchA is the *neighbour*, an entry is made in the routing table indicating that SwitchA is the "next-hop node" to SwitchF. Searching for the node with the highest cross connect rate on the path to SwitchA produces SwitchC. So, an entry in the OC3 section will indicate that SwitchC is the "next-hop node" for SwitchA. The resulting routing table for SwitchB in Figure 4.4 is presented in Table 4.8.

Another example network is depicted in Figure 4.5. This example network contains a path that is similar to the path describing the signalling protocol's operation in a heterogenous network given in Chapter 3. The path from Switch3 to Switch6 discussed in Section 3.9.1 can be found in Figure 4.5. The corresponding routing table, before removing the entries that Switch3 cannot accommodate due to its OC12 crossconnect rate, is given

Destination node address	Data rate OC1			Data rates OC3 to OC192		
	Next hop option 1	Next hop option 2	Next hop option 3	Next hop option 1	Next hop option 2	Next hop option 3
SwitchA				SwitchC		
SwitchC				SwitchC		
SwitchD				SwitchC		
SwitchE	SwitchD			SwitchD		
SwitchF	SwitchA			SwitchA		
SwitchG	SwitchG			SwitchG		
SwitchH	SwitchG			SwitchG		

Table 4.8: Routing table for node B

in Table 4.9.

4.5 Future work

The segmentation of the network is a functionality that is required in a routing protocol when it operates in a large network. Without network segmentation the routing databases tend to grow very large. Despite route precomputation being used, this step (route precomputation) could still consume a lot of resources, even if nodes only exchange a small number of routing updates. The first step in adding this functionality would be to specify the usage of IP addresses in the network segments, and secondly, the creation of the routing table has to be re-examined. The costs advertised between network segments will contain summaries of the cost values in the particular network segment - the next hop entry in the routing table should now contain the next hop to a network path that is most likely to be able to handle the connection based on more relaxed parameters that basic CAC.

In Section 3.9.1 an improvement to the routing protocol was briefly mentioned. The second improvement to the routing protocol concerns connections that have already been set up between switches in the form of logical links (links that have been used to set

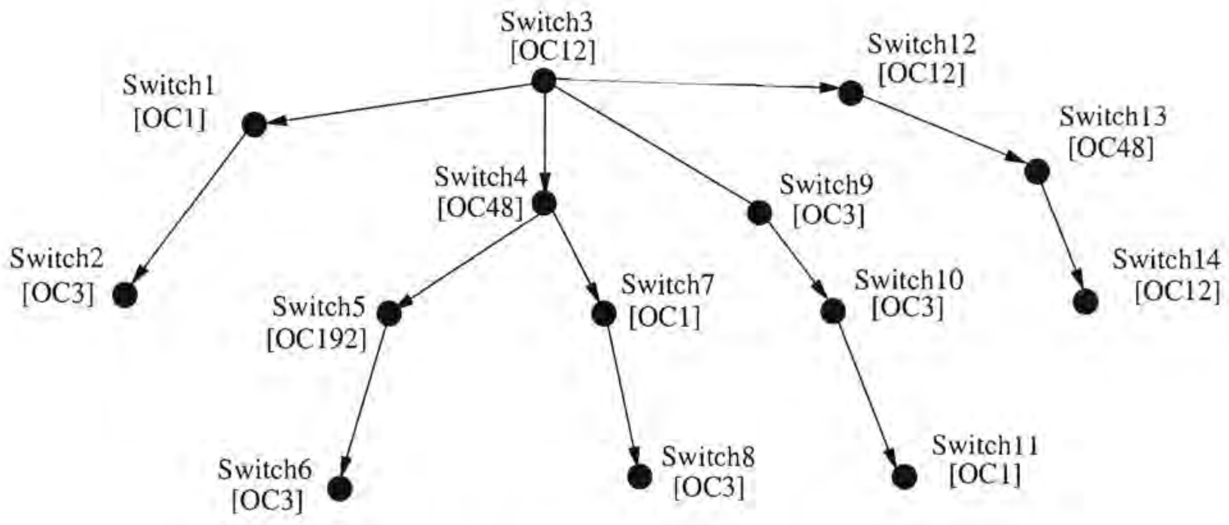


Figure 4.5: Shortest path tree constructed by Switch3

Destination node address	Data rate OC1	Data rate OC3	Data rate OC12	Data rate OC48	Data rate OC192
	Next hop option 1	Next hop option 1	Next hop option 1	Next hop option 1	Next hop option 1
Switch1	Switch1	Switch1	Switch1	Switch1	Switch1
Switch2	Switch1	Switch1	Switch1	Switch1	Switch1
Switch4				Switch4	Switch4
Switch5					Switch4
Switch6					Switch4
Switch7				Switch4	Switch4
Switch8	Switch7	Switch7	Switch7	Switch7	Switch7
Switch9		Switch9	Switch9	Switch9	Switch9
Switch10		Switch9	Switch9	Switch9	Switch9
Switch11		Switch9	Switch9	Switch9	Switch9
Switch12			Switch12	Switch12	Switch12
Switch13			Switch12	Switch12	Switch12
Switch14			Switch12	Switch12	Switch12

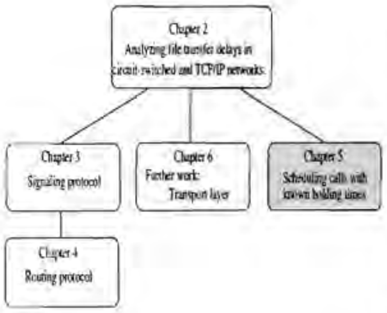
Table 4.9: Routing table for Switch3 (OC1 and OC3 rates cannot be used)

up lower rate connections over switches with high cross connect rates) and provisioned connections. An improvement to the routing protocol would be to exchange information regarding available bandwidth on connections that have already been set up, not just available bandwidth of the interfaces. For example, if a unidirectional OC12 connection is set up between two switches (from switch I to switch I+1), and only one OC1 connection is currently active on it, switch I can advertise that it is directly connected to switch I+1 (even if there are switches between them), and the cost of the link is again a function of the available bandwidth of the connection.

Chapter 5

Scheduling calls with known holding times

5.1 Introduction


Chapter 2
Analyzing file transfer delays in circuit-switched and TCP/IP networks
Chapter 3
Signaling protocol
Chapter 6
Further work:
Transport layer
Chapter 5
Scheduling calls with known holding times
Chapter 4
Routing protocol

This chapter explores the option of queueing calls in connection-oriented networks instead of blocking them when network resources are unavailable. A simple call queueing algorithm would be to hold up call setup messages at each switch along an end-to-end path until resources become available. This simple call queueing algorithm was modeled in Chapter 2, Section 2.2.1. This scheme will be shown to suffer from poor network utilization and long call queueing delays. However, if calls have known holding times, it is possible to design call scheduling algorithms that result in reduced call queueing delays and improved network utilization. We propose algorithms for such call scheduling and demonstrate the quantitative benefits of algorithms that exploit knowledge of call holding times.

In a simple connection setup scheme, a call setup message is held up at each switch sequentially along the end-to-end path until resources become available at each switch. There are two problems with this simple approach. *Firstly*, the total queueing delay incurred waiting for network resources can become rather large due to the sequential waiting period at each switch on the end-to-end path. *Secondly*, while call queueing schemes in general improve bandwidth utilization compared to call blocking schemes, the

kT_{wait} scheme will not achieve the maximum improvement possible. The kT_{wait} scheme was first used in Miyahara et al's paper [7] for the comparison between circuit-switching and CL packet-switching for large file transfers. In this scheme the signalling message requesting the connection is placed in a queue at each switch to wait for the required resources to be released. Only when the connection request reaches the head of the queue, and its requested resources are available, will the request be passed on to the next switch. Hence the term kT_{wait}. The reason an improvement in bandwidth utilization can be expected in call queueing schemes is that by having buffers to queue calls, less bandwidth is needed than for a strictly call blocking scheme. The reason why the kT_{wait} scheme does not take full advantage of this improvement is that upstream switches hold resources while waiting for downstream switches to admit a call instead of using the wait period to admit shorter calls that only traverse upstream segments.

To overcome these two problems, we started looking for ways to improve the call queueing algorithm. This led us to consider options where the switches agree upon a delayed start time for a given call c, and allow other calls sharing segments of the end-to-end path of call c to use the network resources for other calls before call c starts. This would decrease call queueing delays and allow for the utilization gains of call queueing to be realized. However, scheduling calls for a delayed start time with mutual agreement at all the switches on the end-to-end path is only possible if call holding times are known. A switch cannot guarantee that resources will be available for a new call c at some later point in time if it does not know when existing calls will complete. Hence, we focus on the call queueing/scheduling problem for calls with known holding times that can be identified when satisfying the following two characteristics:

- The data from the sending end of the call should be "stored," as opposed to "live."
- The CO network should use preventive congestion control as opposed to reactive.

We continued to design a call scheduling algorithm when call holding times are known. Such an algorithm could coexist with a traditional call admission control algorithm for calls with unknown holding times by partitioning network resources for calls with known holding times from resources for calls with unknown holding times.

This proposed method for scheduling connections can be used in any type of connection-oriented (CO) network. CO networks are networks where resources are reserved for a connection between end hosts before any data transmission may start. These networks can either be packet-switched (for example ATM or X.25 networks) or circuit-switched (for example SONET or WDM networks).

The motivating applications for such an algorithm are described in in Section 5.2. Section 5.3 describes the proposed algorithm as applied to a simple one-switch network, Section 5.4 continues by explaining some extensions to the algorithm, which includes the algorithm's behaviour in a network where multiple switches are involved. Some solutions to the distributed computing problem are proposed in Section 5.5 and Section 5.6 provides the results of simulations run to determine the feasibility of the solutions. Section 5.7 introduces some problems that still need to be solved. Conclusions are made in Section 5.8.

5.2 Motivating applications

While the motivation for this work came from our interest in supporting end-to-end large file transfers on high-speed circuit-switched networks, we considered the question of whether there are other applications for which call holding times are known. Two characteristics of calls have been identified that allows us to determine the call holding time:

- The data from the sending end of the call should be "stored," as opposed to "live."
- The CO network should use preventive congestion control as opposed to reactive.

Consider the first characteristic. Table 5.1 shows that the sending end and consuming end of any two-party (as opposed to multi-party) data transfer can each be "live" or "stored." If both ends are live and the communication is bidirectional, we classify such sessions as *interactive*. An example of this category is telephony, where the call holding time is unknown. Given that both ends are "live" and both ends can send traffic, the call could hold for any length of time. If both ends are live, but the communication is unidirectional, we refer to this case as a *live streaming* data transfer. An example is listening to/viewing a live radio/TV broadcast. The next category shown in Table 5.1 is *recording*, where the source is live, but the receiver is a storing device. In both the *live streaming* and *recording* categories, the call holding time may or may not be known since it depends on the duration of the live "event." For e.g., the duration of coverage of a live parade could be known a priori and advertised by the audio/video distributor; on the other hand, the duration of a sporting event, for e.g., a tennis match, is not known beforehand. Therefore, in general we assume that if the sending end is "live," call holding times are unknown.

Consuming end / Sending end	Live	Stored
Live	Interactive/Live streaming	Recording
Stored	Stored streaming	File transfers

Table 5.1: Classification of data transfers

However, if the sending end is "stored," call holding times are known if the CO network is used with preventive congestion control (rather than reactive). For example, transferring a stored file from a source to a receiver where it is also stored for later consumption (classified as *file transfers* in Table 5.1) on a TCP/IP network results in an unknown holding time since this network uses reactive congestion control. On the other hand, if the same file is transferred on a circuit established through a circuit-switched network, the holding time can be determined from the file size, the data rate of the circuit, and propagation delays. Similarly, applications in which stored audio or video clips (e.g., in a web-based class lecture or Video on Demand (VOD)) are consumed live (e.g., by a student listening) are classified as *stored streaming* in Table 5.1. If such data is sent on a packet-switched CO network (to take advantage of silences, packet-switched networks are better for this class of applications), and the network supports preventive congestion control, then the call holding time can be estimated with a high degree of confidence. For example, if an ATM Variable Bit Rate (VBR) connection is used rather than an Available Bit Rate (ABR) connection, the call holding time can be accurately predicted given the exact traffic characterization of the stored audio or video file.

Thus, there are a number of applications, which when used on certain types of networks, have deterministic call holding times. Hence called known holding times.

Example applications from the categories presented in Table 5.1 that may benefit from this scheduled mode of connection setup are as follows:

- A frequently used application in networks is **file transfers**, which is an example of an application that transfers data from a stored source to a destination where the data is also stored. A large file transfer has no intrinsic burstiness associated with it, and is hence best handled by a circuit switched network [5], [6] and [7]. This is relative to connectionless packet switching where the packet header overheads, acknowledgements, congestion control mechanisms, buffering etc. have an impact on the end-to-end transfer time. Knowing the file size f and the data rate *rate* of the connection, the holding time h of a connection that will be used to transfer the file(s) is given by $h = \frac{f + overhead \times f}{rate} + T_{prop}$ where *overhead* is the overhead added by the various protocol layers and T_{prop} is the propagation delay from the sending host to the receiving host. Examples of large file transfers are application downloads, downloads from web sites with significant multimedia content (even with compression, video, audio and image files tend to be large), downloads of books, technical specification documents, etc. Large file transfers also occur in the mirroring of web sites. For example, Metalab's Linux archive is reported to be mirrored by more than 70 sites across the world [23].

- A second example of an application in which calls have known holding times is **video on demand** for stored video files. If the video file is sent out from a stored source and consumed live, this is an example of a *stored streaming* application. The implication of live consumption at the receiving end is that it is more efficient to use a variable rate connection for the video transfer given that most compression techniques take advantage of changing scene conditions. For example, during video transfer it is possible that there is a time frame during which there is a still image with only audio. Using a compression algorithm, this time frame would require less capacity than a normal time frame. Now, together with the holding time, traffic patterns can be associated with time frames allowing networks accommodating variable rate connections, for example ATM networks, to manage its resources more efficiently by setting up a connection that may vary in capacity and thus suit the transfer more efficiently. That is, the CO packet-switched network that implements preventative congestion control mechanisms reserves resources for the connection that vary over time. It is now possible for the network to decide to reduce the capacity allocated to one connection during a time frame and allocate the resources to another connection only during that time frame. Thus, even variable bit rate connections could have known holding times. A scheduling algorithm can, in principle, be designed for such packet-switched networks to take advantage of this knowledge and schedule calls for later start times instead of blocking calls when it does not have the resources available at the time when the request arrives.

- Leased lines are used commonly to interconnect two routers. This means that all traffic between the two routers concerned traverse a provisioned connection (also known as leased line) set up between them. Currently, the resources allocated to such a provisioned connection (for example, bandwidth) is for the worst case, i.e. the highest expected traffic. For example, consider two routers connected through a SONET network. Assume that for the most of the time an OC12 connection is sufficient to carry the inter router traffic. But, from 11am to 3pm the traffic is predicted to increase significantly requiring an OC48 connection. To be able to handle this worst case, the provisioned connection needs to be an OC48. This has the consequence that a lot of bandwidth is wasted during the time when just an OC12 would be sufficient. Making use of scheduled connection setup, it is possible to use traffic characteristics to schedule the setup of one or more extra connections between the routers during peak usage. Continuing our example, it should be possible to provision an OC12 connection for "permanent" usage, and schedule the setup of three additional OC12 connections from 11am to 3pm every day. Knowing the duration of the highest traffic load introduces a third example application, which is **leased lines with known holding times**.

5.3 Proposed algorithm

The general idea behind this method of admitting connections with known holding times is to have switches maintain a time-varying available capacity on each of its interfaces that reflects the scheduled start times of all admitted connections. Making use of this information the connection admission control (CAC) module of each switch, which is responsible for the management of the resources available to connections passing through the switch, determines the earliest possible time when a newly arriving connection with a known holding time can become "active".

An "active" connection is a connection that is currently transferring data, while an "admitted" connection is one that is either active or scheduled to become active at some point in time. As the simplest case, we explain our CAC algorithm using a one switch network as shown in Figure 5.1. Calls are generated by an end host connected to the switch to another end host on the same switch with a given desired capacity (c) and a known holding time (h). The CAC module in the switch, as shown in Figure 5.2, keeps a time-variant available capacity $a_i(t)$ function for each outgoing interface i. Based on the destination of the connection a set of candidate outgoing interfaces (I) is selected from a routing table maintained by the routing module (see Figure 5.2). Next, based on the

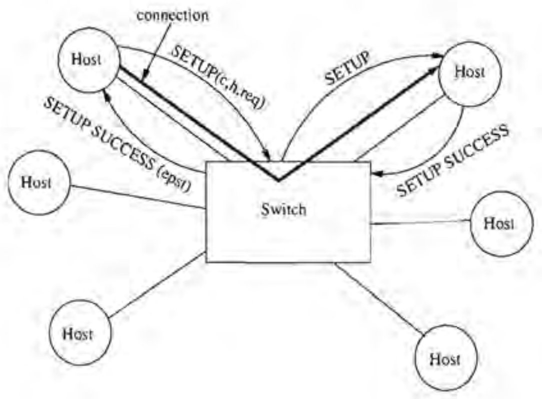


Figure 5.1: A simple one switch network

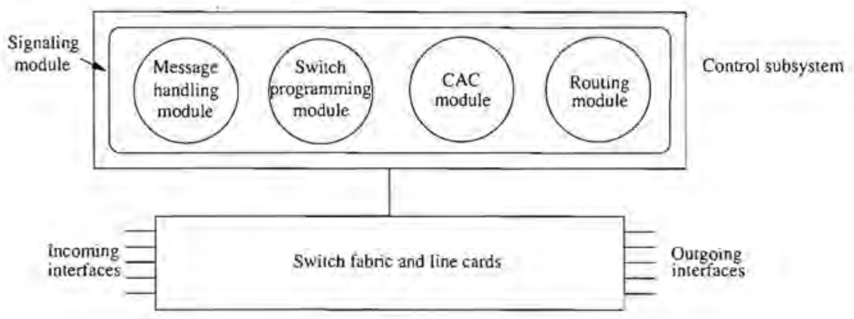


Figure 5.2: Block diagram of a switch

desired capacity (c) and holding time (h) for the connection, the earliest possible start time ($epst$) and interface (i) is determined by the CAC module using (5.1).

For each $j \in I$ determine

$$epst_j \geq \text{current time} \wedge a_j(t) \geq c, epst_j \leq t \leq epst_j + h$$

$$epst \triangleq epst_i \text{ where } epst_i = min\{epst_j \mid j \in I\} \quad (5.1)$$

$$a_i(t) \leftarrow a_i(t) - c, \text{ for } epst \leq t \leq epst + h \quad (5.2)$$

A switch is now able to respond with an earliest possible start time at which the end host that requested the connection may start transmission. The time-varying available capacity function for the chosen interface i is updated using (5.2) to reflect the newly scheduled connection. Once the connection's scheduled time arrives the switch programming module shown in Figure 5.2 programs the switch fabric, thereby activating the connection.

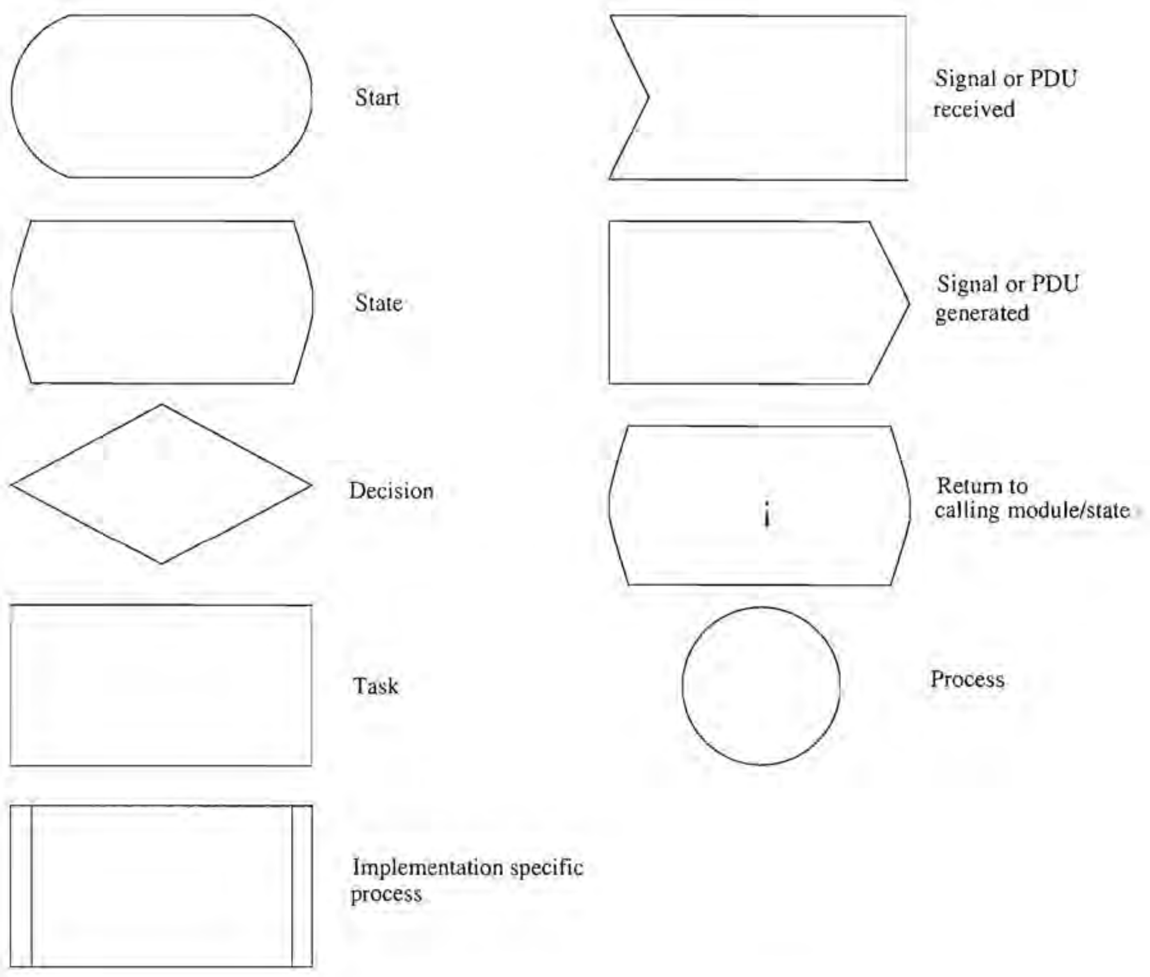


Figure 5.3: Keys used in flow diagrams

Using the keys presented in Figure 5.3, the message handling module (called from the signalling module) can be represented with the flow chart of Figure 5.4. This flow chart assumes an error-free operation. Figure 5.5 is a flow chart that indicates possible adjustments to be able to handle certain error conditions. Figures 5.6 to 5.9 depict the flow charts for processes included in Figure 5.4 and 5.5 assuming error conditions can occur.

In all these explanations, the destination host is treated as a "black box" which only returns "success" or "failure" when it replies to a connection request. That is, the switch includes the values of c,h and *epst* in the connection request sent to the destination, but the destination host does not change the value of *epst*. If the destination host can accept the connection its reply will indicate success. If it cannot accept the connection its reply shall indicate a failure. An example of the message exchanges for connection setup in a one switch network is given in Figure 5.1.

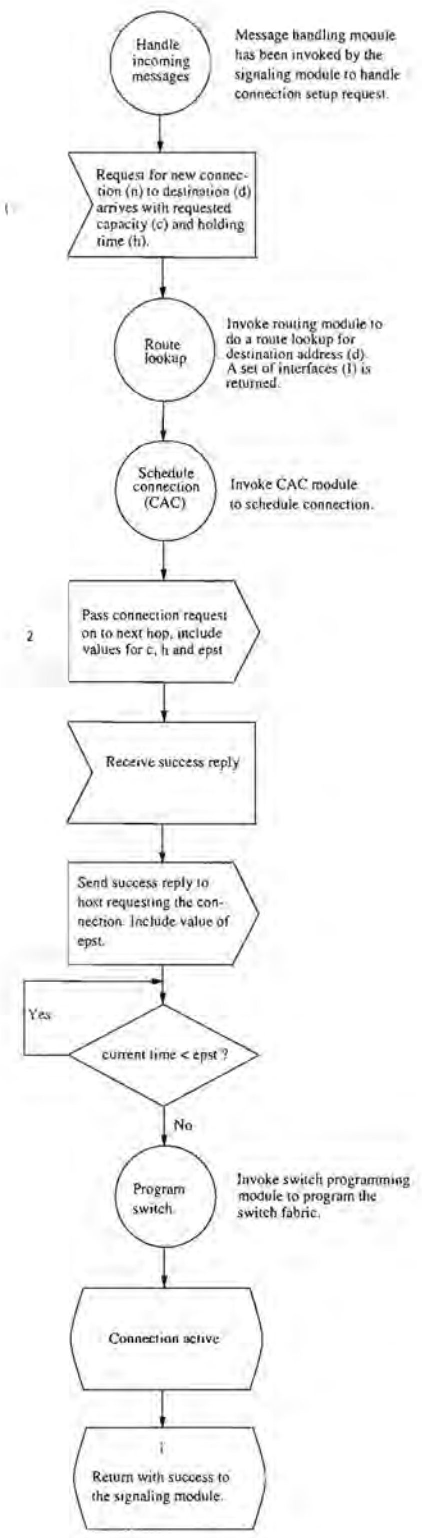

Handle incoming messages
Message handling module has been invoked by the signaling module to handle connection setup request.
Request for new connection (n) to destination (d) arrives with requested capacity (c) and holding time (h).
Route lookup
Invoke routing module to do a route lookup for destination address (d). A set of interfaces (I) is returned.
Schedule connection (CAC)
Invoke CAC module to schedule connection.
2
Pass connection request on to next hop, include values for c, h and epst
Receive success reply
Send success reply to host requesting the connection. Include value of epst.
Yes
current time < epst ?
No
Program switch
Invoke switch programming module to program the switch fabric.
Connection active
Return with success to the signaling module.

Figure 5.4: Flow chart for the message handling module in a one switch network

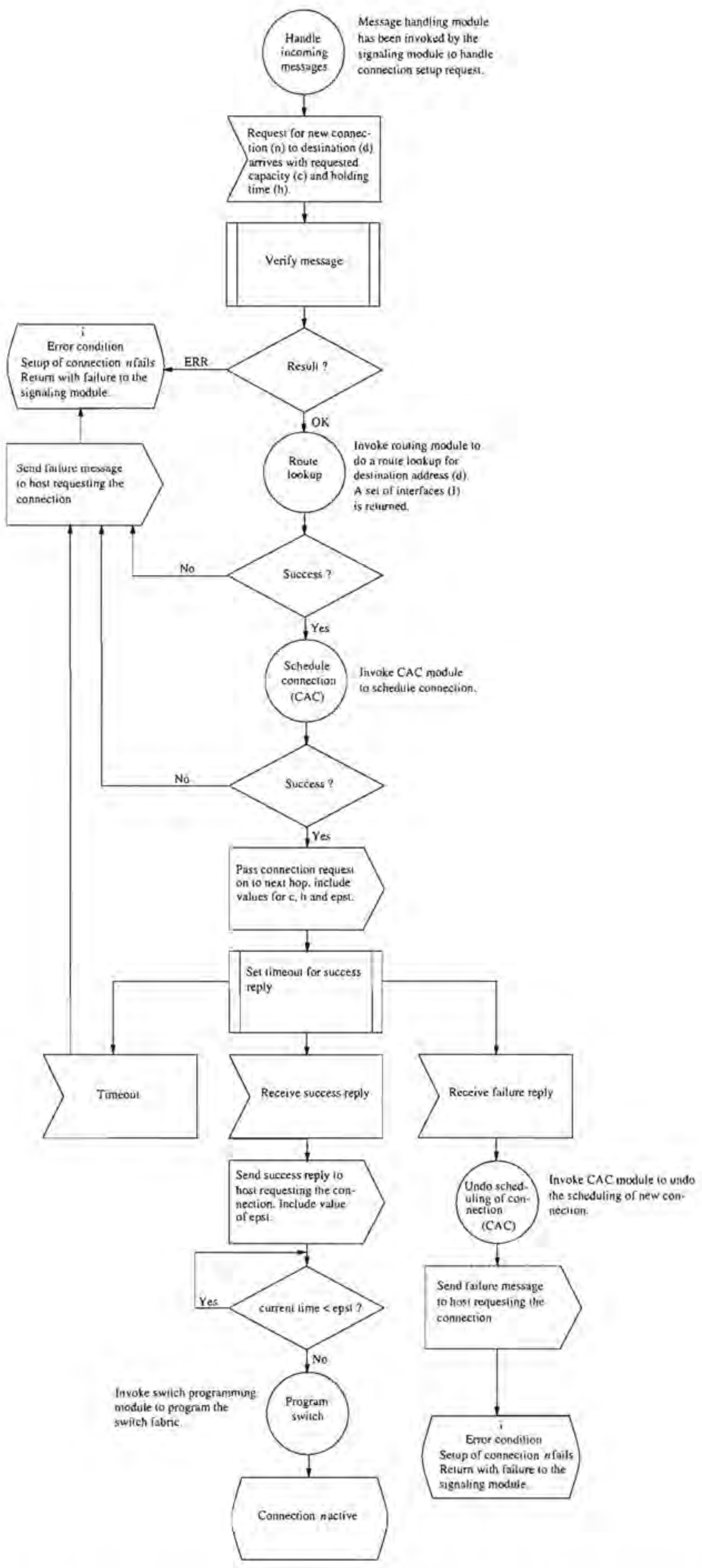

Handle incoming messages
Message handling module has been invoked by the signaling module to handle connection setup request.
Request for new connection (n) to destination (d) arrives with requested capacity (c) and holding time (h).
Verify message
Result ?
ERR
Error condition Setup of connection n fails Return with failure to the signaling module.
OK
Route lookup
Invoke routing module to do a route lookup for destination address (d). A set of interfaces (I) is returned.
Send failure message to host requesting the connection
Success ?
No
Yes
Schedule connection (CAC)
Invoke CAC module to schedule connection.
Success ?
No
Yes
Pass connection request on to next hop. Include values for c, h and epst.
Set timeout for success reply
Timeout
Receive success reply
Receive failure reply
Send success reply to host requesting the connection. Include value of epst.
Undo scheduling of connection (CAC)
Invoke CAC module to undo the scheduling of new connection.
Yes
current time < epst ?
No
Send failure message to host requesting the connection
Invoke switch programming module to program the switch fabric.
Program switch
Error condition Setup of connection n fails Return with failure to the signaling module.
Connection n active

Figure 5.5: Flow chart for the message handling module in a one switch network, with error handling

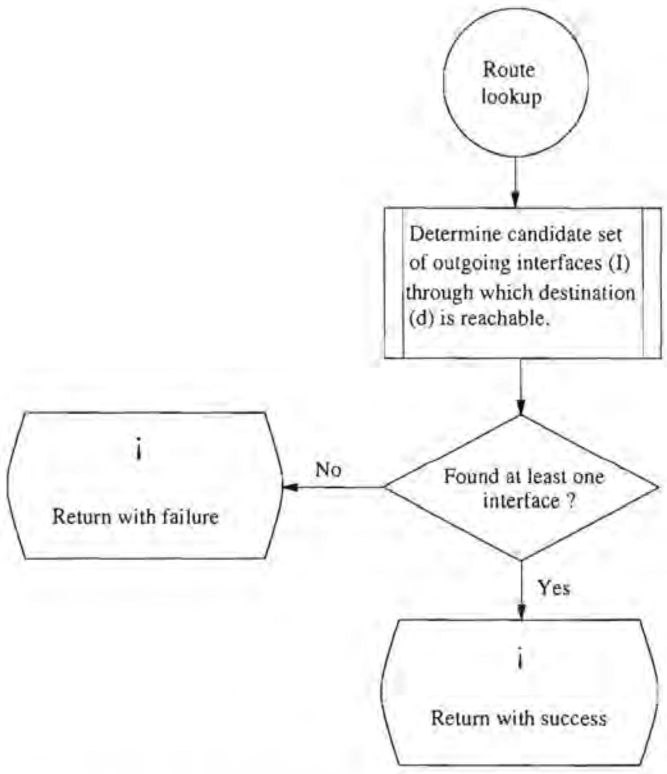

Route lookup
Determine candidate set of outgoing interfaces (I) through which destination (d) is reachable.
Found at least one interface ?
No
Return with failure
Yes
Return with success

Figure 5.6: Route lookup process included in the routing module

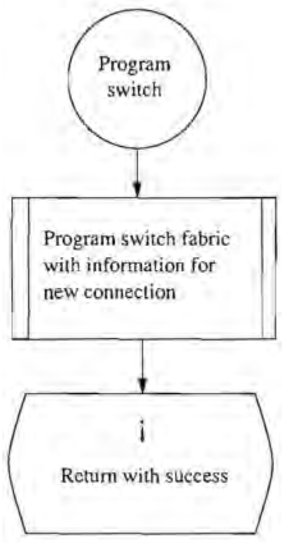


Figure 5.7: Process to program the switch fabric in the switch programming module

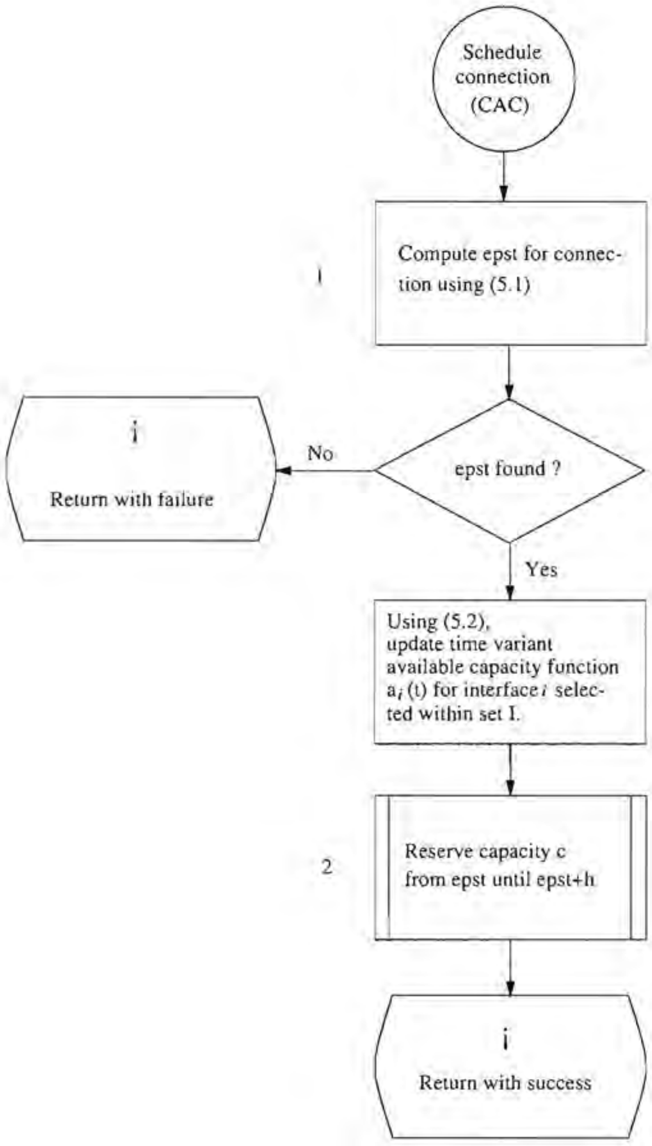

Schedule connection (CAC)
Compute epst for connection using (5.1)
1
epst found ?
No
Return with failure
Yes
Using (5.2), update time variant available capacity function a_i(t) for interface i selected within set I.
2
Reserve capacity c from epst until epst+h
Return with success

Figure 5.8: Scheduling process included in the CAC module for a one switch network

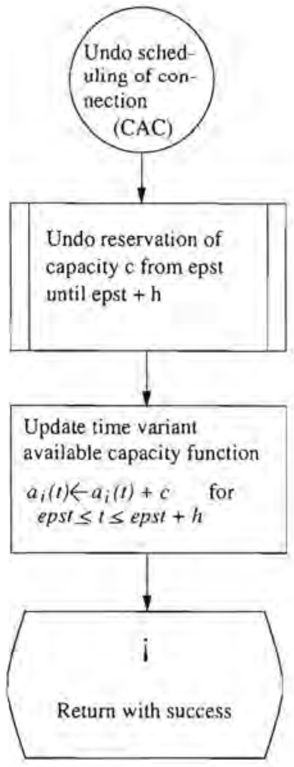

Undo scheduling of connection (CAC)
Undo reservation of capacity c from epst until epst + h
Update time variant available capacity function
$a_i(t) \leftarrow a_i(t) + c$ for $epst \leq t \leq epst + h$
i
Return with success

Figure 5.9: Process in the CAC module to undo the changes made by the scheduling process

5.4 Extensions

5.4.1 Request for start time included in connection request

Continuing with our example of a simple one switch network, an extension to the CAC algorithm is to allow the host requesting the connection to specify a desired start time, which could be different from the implied "immediate" start time in current signalling protocols. This is definitely needed in the third example application where the extra connections between the routers will only be needed during certain times. Requesting the start time could also be very useful in the other two example applications, for example only mirroring a web site from 10pm, or a customer requesting a movie to start at 8pm.

If a host is allowed to request a connection from a specific time *req*, *epst* is computed using (5.3).

For each $j \in I$ determine

$$epst_j \geq req \wedge a_j(t) \geq c, epst_j \leq t \leq epst_j + h$$

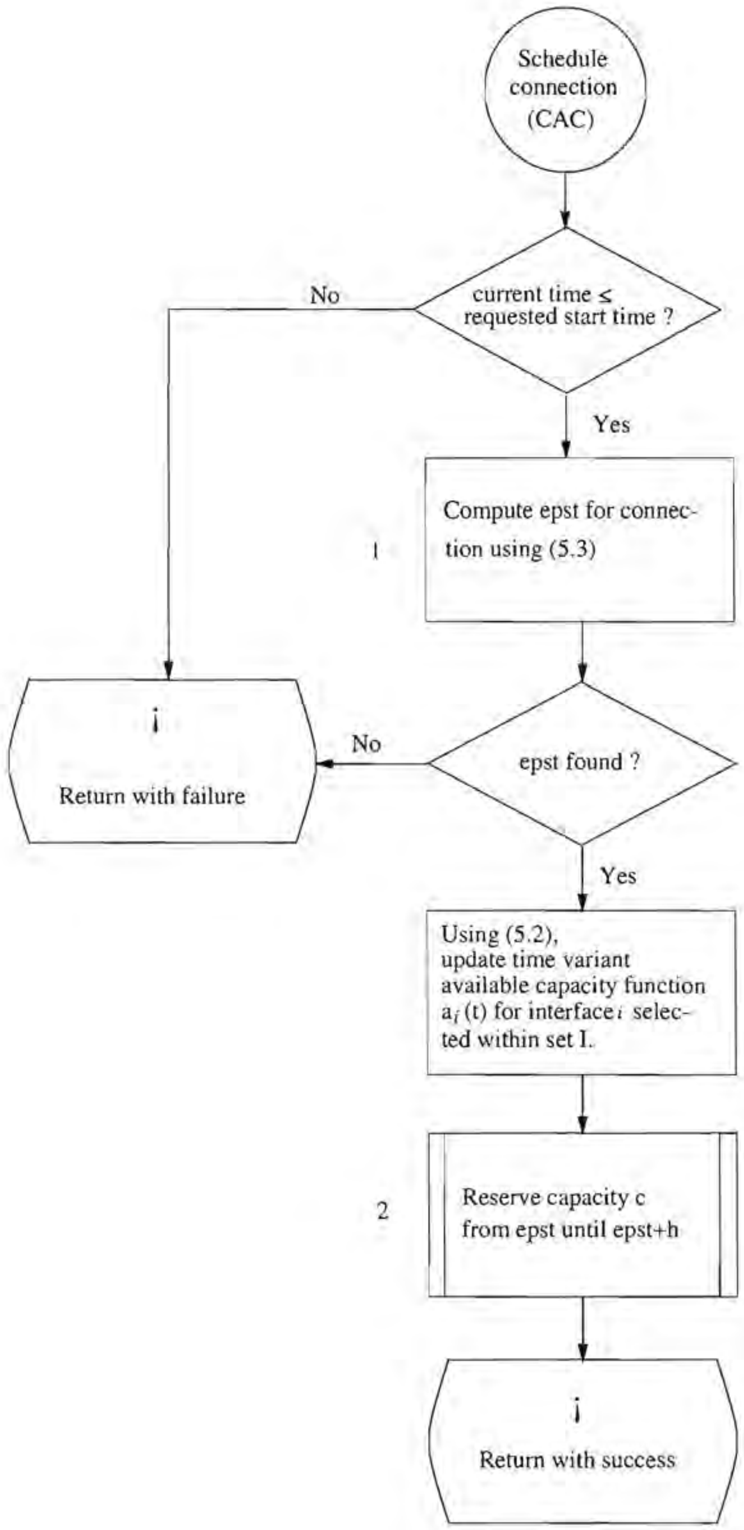
$$epst \triangleq epst_i \text{ where } epst_i = min\{epst_j \mid j \in I\} \quad (5.3)$$

The flow charts for this extension to the CAC algorithm can be depicted by changing block 1 in Figure 5.4 to "Request for new connection n to destination d arrives with requested starting time req, capacity c and holding time h" and replacing the flow chart for the scheduling process with Figure 5.10.

5.4.2 More than one switch in the network

Connections typically pass through multiple switches. A problem that arises in this case is that the switches may not compute the same value for *epst* due to the differences in their current state.

To explain the problem and its solution, we use the example shown in Figure 5.11 where Host1 requests a connection to Host2. The connection is shown to traverse through three switches. Assume the first switch, SwitchA schedules the connection and reserves resources for some time $[epst_A, epst_A + h]$, where h is the holding time and $epst_A$ is the earliest possible start time that the connection can be scheduled for at SwitchA. SwitchA needs to pass the appropriate information to the next hop switch, SwitchB, to prevent the latter from scheduling the connection for any $t < epst_A$. Now, assume SwitchB finds



Schedule connection (CAC)
current time ≤ requested start time ?
No
Yes
Compute epst for connection using (5.3)
1
epst found ?
No
Yes
Return with failure
Using (5.2), update time variant available capacity function a_i(t) for interface i selected within set I.
2
Reserve capacity c from epst until epst+h
Return with success

Figure 5.10: Scheduling process (allowing for a host to request a start time) included in the CAC module for a one switch network

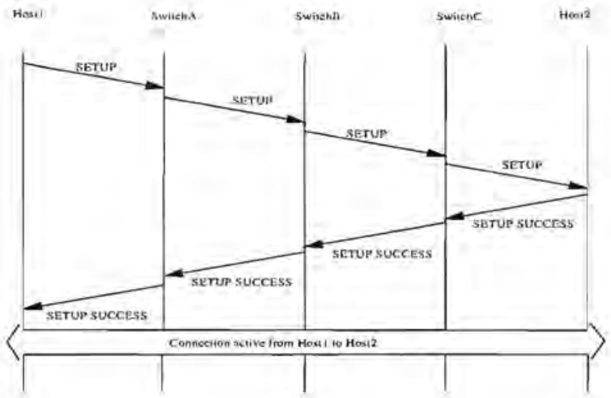


Figure 5.11: Successful setup of a connection

that it cannot schedule the connection for the same *epst* as SwitchA. Instead its *epst* is $epst_B$. This could cause a problem because SwitchA may not have resources available for this connection in the time interval $[epst_B, epst_B + h]$. This later starting time could have been allocated to some other connection at SwitchA. This is a distributed computing problem caused by needing cooperative actions at multiple switches for which we propose solutions in the next section.

5.5 Connection setup schemes

Currently, on-demand (referred to as "switched" mode of operation as opposed to "provisioned" mode) connections are set up in connection-oriented networks by using a *blocked* mode of operation. When a connection request arrives at a switch that does not have enough resources available, the request will not be *queued* until the resources become available. Instead the call is blocked and connection release procedures are initiated. The objective is to decrease the blocking probability of connections by making use of the known holding time to schedule a connection, thus providing the source that requested the connection with a later start time if resources are not immediately available.

This section compares four queued connection setup schemes. Two schemes, kT_{wait} and $kT_{wait} - T_{max}$, explore a queued mode of operation where the call holding time is not taken into account and connection setup requests are queued instead of blocked. Of all the schemes kT_{wait} is the only one that guarantees a zero blocking probability. The $kT_{wait} - T_{max}$ scheme extends the kT_{wait} scheme by allowing the source to include a latest acceptable start time for the connection in its connection request.

In the F scheme, the ingress switch selects an earliest possible start time (*epst*) for the connection such that is has resources available for the connection from *epst* to $epst+h+F$, where h is the holding time of the call and F is some large time value. It holds the resources for this connection for this large time period hoping that the downstream switches will select an *epst* within this large time period. If a large enough value of F is selected, the probability of success will be high.

In the *timeslots* scheme, the ingress switch selects multiple time ranges during which it can accommodate the call. Each downstream switch checks to see if it has available resources for one or more of these time ranges. Each switch along the path may narrow the time ranges until finally there exists at least one common time range (of length equal to the call holding time) during which the required resources are available at all switches.

An analogy can be drawn between these scheduling schemes and the scheduling of a meeting involving several people. With the meeting duration known to all involved, one way to schedule its start is for the first person to send out a memo stating that he/she has "the whole of next week" available. Another solution is for the memo to read, "I am free from 2pm to 4pm Monday, 8am to 1pm Tuesday, and the whole of the following week." Tentatively, the sender of the memo holds these time ranges for this meeting until a response arrives from all other participants. The first solution corresponds to the F scheme and the second solution corresponds to the *timeslots* scheme.

The F and *timeslots* schemes are call queueing schemes in that calls wait for resources, but they also block calls if the selected F value or number of time ranges is not large enough for success. These can be augmented with negotiations (i.e., multiple signalling exchanges on the same path), and/or searches on multiple paths to reduce call blocking probability.

5.5.1 The kT_{wait} scheme

The kT_{wait} scheme is the simplest queued connection setup scheme. In this scheme, switches do not take call holding times into account when connections are admitted. Instead, on receipt of a connection request it is placed in the queue of the interface that should be used by the connection. When the bandwidth requested by the connection request at the head of the queue becomes available, it is reserved for the connection and the connection request is passed on to the succeeding switch, or destination host. When the connection request reaches the destination host, a connection has already been set up from the source to the destination. If the destination host accepts the connection, it sends a success message back to the source. As this success message traverses the network towards the source the connection is usually marked as "established". On receipt of the success message, the source can immediately start with data transmission. If T_{wait} is the average time a connection request waits in a queue for resources to become available, and k indicates the number of switches on the connection, then the term kT_{wait} forms a large part of the total connection setup time, hence the name kT_{wait} as described in Chapter 2, Section 2.2.1.

5.5.2 The $kT_{wait} - T_{max}$ scheme

The second queued connection setup scheme is designed by slightly modifying the kT_{wait} scheme. When sending a connection request to its ingress switch, a source host might expect the connection to be established before a certain time. For example, the source host might request that a connection be established at most 50 seconds from when it requested it. This connection setup scheme also does not take the holding time into account when a connection is set up, but provides the source requesting the connection some control over the start time of the connection, hence named the $kT_{wait} - T_{max}$ connection setup scheme. The $kT_{wait} - T_{max}$ scheme works in the same manner as the kT_{wait} scheme except that the source host is allowed to include a time, T_{max}, in its connection request. The value of T_{max} indicates the latest possible time that the source would accept as a start time for this connection. As in the kT_{wait} scheme, switches place a connection request in the queue of the interface that should be used by the connection. The action taken when the connection request is at the head of the queue, and its requested bandwidth is available, differs in that the switch first checks if its current time is later than the time T_{max} included in the connection request. If the current time is later, connection release procedures are initiated.

5.5.3 The F scheme

The first queued connection setup scheme that takes advantage of the call holding time is called the F scheme. In this solution the requested resources are reserved for a longer period of time than its holding time, until reverse messages are received, during the connection setup procedure. In the forward phase of connection setup resources are held for a long enough time period that if a downstream switch selects a later value for *epst*, there is some likelihood that the earlier (upstream) switches of the path will have the resources available. Resources are held for this large time period only for the short duration it takes for the setup request to reach the destination and the reply (setup success) to return.

Using a large finish time during the scheduling of a connection works as follows: a host requests a connection from its ingress switch to a destination with the optional inclusion of a start time; the ingress switch selects a large time range (starting with the earliest time greater or equal to the requested start time) during which it is able to accommodate the connection. On receipt of a time range a switch searches for the largest time range inside the range it received from the preceding switch during which it can accommodate the connection. This time range is included in the connection request passed to the succeeding switch, or destination host.

The computation of *epst* changes in that a pre-assigned value F is added to the holding time by the ingress switch, where F is a very large extra holding time. When a host requests a connection to destination d from time *req* with capacity c and holding time h, (5.4) can now be used at the first (ingress) switch to compute *epst*. The time-varying available capacity function for the chosen interface is updated using (5.5) in which $T_{end} = epst + h + F$.

For each $j \in I$ determine

$$epst_j \geq req \wedge a_j(t) \geq c, epst_j \leq t \leq epst_j + h + F$$

$$epst \triangleq epst_i \text{ where } epst_i = min\{epst_j \mid j \in I\} \quad (5.4)$$

$$a_i(t) \leftarrow a_i(t) - c, \text{ for } epst \leq t \leq T_{end} \quad (5.5)$$

When the resources are reserved for this connection at the ingress switch, it indicates that this connection reserves capacity c from *epst* to $epst+h+F$. To enable all the switches in the connection to agree to the same value for *epst*, the value of *epst*, the holding time (h) and the time T_{end} are included in the connection request message passed to the second switch in the connection. The switch receiving this message shall handle this request as if a request for a connection from time *epst*, as included in the connection request, has been received.

The *setup request traverses* the network and each switch along the path computes a (possibly) new value for *epst* and looks for the largest time period $\leq F$ but $\geq h$, during which it will have enough resources available. If found, the requested resources are reserved for this connection (using (5.5)) for the period $(epst_{new}, epst_{new} + h + F_{new})$, and the connection request containing $epst_{new}$, $F_{new} + h$ and h are passed on to the next hop. If a switch is unable to find a time period during which it can accommodate the connection, the call is blocked and release procedures are initiated. If the destination host accepts the request for a connection from the source, the final value of *epst* is passed from switch to switch until the source is notified of the new value of *epst*.

As those *success reply messages* traverse the network to the source, all the switches change their resource reservation tables to reflect the new start time (the new value of *epst*), and the finish time to be $epst + h$. If $epst_{local}$ indicates the earliest possible start time as computed by a switch in the forward direction of the signalling procedure, and *epst* is the earliest possible start time it received in the success reply message, the time-varying available capacity function is updated using (5.6) and (5.7).

$$a_i(t) \leftarrow a_i(t) + c, \text{ for } epst_{local} \leq t < epst \tag{5.6}$$

$$a_i(t) \leftarrow a_i(t) + c, \text{ for } epst + h \leq t < T_{end} \tag{5.7}$$

The flow chart for the message handling module for the ingress switch is given in Figure 5.12. Block 1 in the scheduling process (given in Figure 5.10) changes to "Compute epst for connection n using Equation (5.4)", and block 2 changes to "Reserve capacity c from epst until epst + h + F.". The flow chart for the process "Update resource reservation and a(t)" is depicted in Figure 5.13.

Figure 5.14 shows an example of scheduled connection setup using the F scheme. For this example, the value of F is 5 hours. End host A requests a connection with a holding time of one hour starting at 6pm, switch 1 computes that it will have enough resources available from 6pm until 12pm, which it reserves. The setup request message traverses the network, and every time *epst* is increased due to the unavailability of resources. At switch 2 resources are reserved from 7pm until 12pm, and *epst* is set to 7pm. When the setup request message reaches end host B, *epst* is already 9pm. End host B accepts the connection and the success reply from switch 4 contains the final value of *epst* (9pm). As this reply message traverses the network, each node forming part of the connection now changes its respective resource reservations to indicate that the connection will be active from 9pm until 10pm.

This scheme might have the disadvantage of underutilizing the channels between switches. To be able to guarantee a low blocking probability, the ingress switch has to

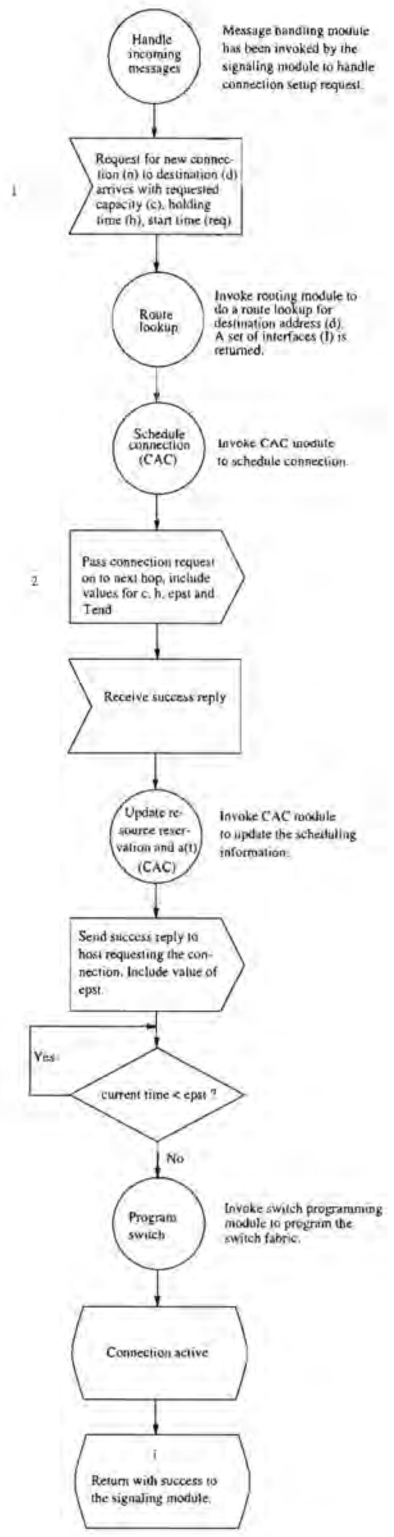

Handle incoming messages
Message handling module has been invoked by the signaling module to handle connection setup request.
1
Request for new connection (n) to destination (d) arrives with requested capacity (c), holding time (h), start time (req)
Route lookup
Invoke routing module to do a route lookup for destination address (d). A set of interfaces (I) is returned.
Schedule connection (CAC)
Invoke CAC module to schedule connection.
2
Pass connection request on to next hop, include values for c, h, epst and Tend
Receive success reply
Update resource reservation and a(t) (CAC)
Invoke CAC module to update the scheduling information.
Send success reply to host requesting the connection. Include value of epst.
Yes
current time < epst ?
No
Program switch
Invoke switch programming module to program the switch fabric.
Connection active
i
Return with success to the signaling module.

Figure 5.12: Connection setup at the ingress switch of a connection (F scheme)

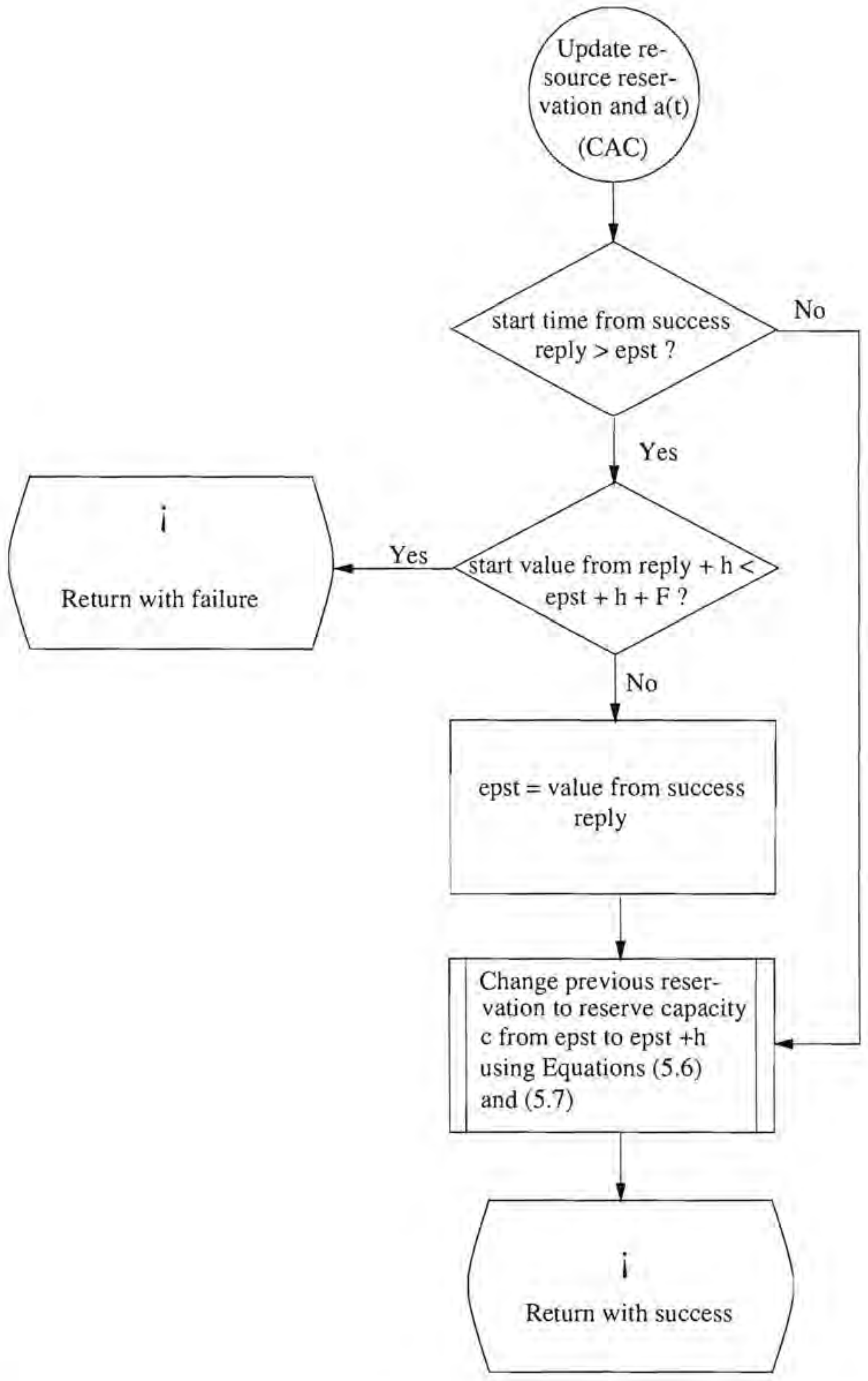

Update resource reservation and a(t) (CAC)
start time from success reply > epst ?
No
Yes
start value from reply + h < epst + h + F ?
Yes
Return with failure
No
epst = value from success reply
Change previous reservation to reserve capacity c from epst to epst +h using Equations (5.6) and (5.7)
Return with success

Figure 5.13: Process in the CAC module to update the changes made by the scheduling process in the reverse direction

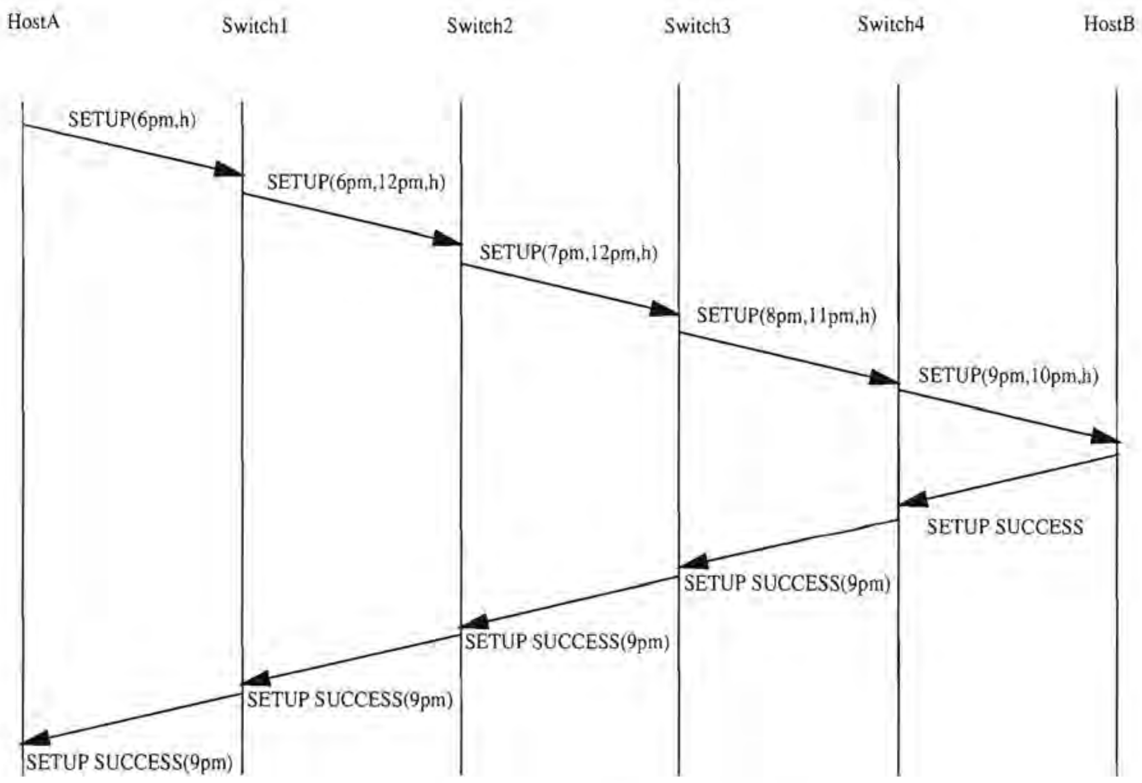


Figure 5.14: Example of scheduled connection setup

make use of a large value for F. Choosing a large value for F will increase the probability that downstream switches will find a time period corresponding to the one provided by the ingress switch during which it has enough resources available for the connection. The disadvantage of a large value for F is that the ingress switch might have resources available for the connection during time periods that are smaller than F, but large enough to handle the connection. These time periods will be ignored by this scheme, and will result in low utilization of the channels between switches. For example, Figure 5.15 describes the available bandwidth function for an OC1 interface if the current time is 2pm. Consider an F value of four hours. If the F scheme is used and a connection that has to be handled by this interface arrives, requesting an immediate start time and indicating a holding time of one hour; the earliest possible time this interface will be able to handle the connection will be at 10pm - even though resources are available from 3pm until 5pm. The next scheme, *timeslots*, shall attempt to solve this problem.

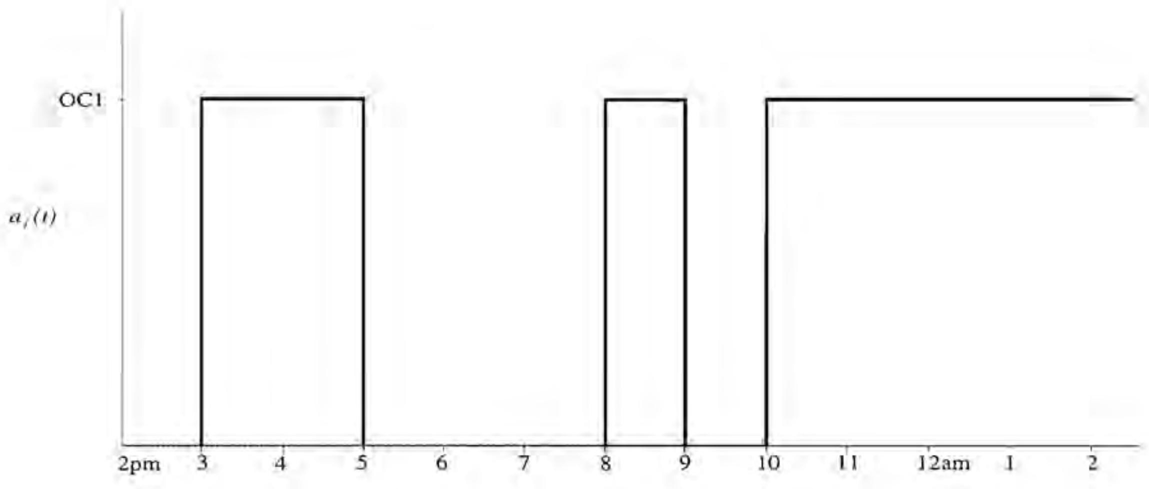


Figure 5.15: Available bandwidth of Switch1, interface i

5.5.4 The *timeslots* scheme

An improvement to the F scheme that will solve the aforementioned problem of utilization, without increasing the call blocking probability significantly, involves the use of several time ranges. Including more than one time range in the connection request will increase the expected utilization of the channels without increasing the call blocking probability significantly. In this scheme the ingress switch does not rely on a large value (F in the previous scheme) to admit a call, but rather a number n indicating the number of time ranges it should include in the connection request passed on to the succeeding switch.

A time range j is identified by a start time t_{js} that is equal to or larger than the requested time (req) and the end time t_{je} that is larger or equal to $t_{js} + h$, during which the switch has resources available to handle the connection. If I is the set of interfaces selected by the routing module as described in Section 5.3, the ingress switch selects n different time ranges by using (5.8).

Find n time ranges such that,

$$\begin{gathered} t_{1s} \geq req, \\ t_{1s} < t_{1e} < t_{2s} < t_{2e} < t_{3s} < t_{3e} < \dots \\ <_{tns}< t_{ne}, \\ t_{je} - t_{js} \geq h, \forall j, 1 \leq j \leq n, \text{and} \\ a_i(t) \geq c, t_{js} \leq t \leq t_{je}, \forall j, 1 \leq j \leq n \end{gathered}$$

$$\text{for any } i \in I \quad (5.8)$$

Upon finding n time ranges, (t_{1s}, t_{1e}), (t_{2s}, t_{2e}),..., (t_{ns}, t_{ne}), the required resources are reserved using (5.9).

$$a_i(t) \leftarrow a_i(t) - c, \text{ for } t_{js} \leq t \leq t_{je}, \text{ for } 1 \leq j \leq n \quad (5.9)$$

The n time ranges are sent in the connection setup message to the next switch along with the holding time h. On receipt of a connection request containing time ranges, an intermediate switch attempts to admit the call during each of the time ranges or any part of each range greater than or equal to the holding time. The matching time ranges are passed on to the succeeding switch. If no matching time ranges are found, the call is blocked and connection release procedures are initiated.

For example, if Figure 5.15 describes the available bandwidth function of an OC1 interface of Switch1 and a connection request arrives requesting an immediate start time for an OC1 connection lasting one hour, the *timeslots* scheme enables the switch to pass the connection request to the succeeding switch containing the time ranges [3pm,5pm], [8pm,9pm] and [10pm,∞] in the case when $n = 3$. On receipt of a connection request containing time ranges, a switch shall attempt to admit the call during each of the time ranges or any part of it greater than or equal to the holding time. The matching time ranges (of which there could now be more than n) are passed on to the succeeding switch. If no matching time ranges are found, connection release procedures are initiated. For example, Figure 5.16 shows the available bandwidth function of Switch2 before it received the connection request from Switch1. On receipt of the connection request, Switch2 shall admit the call during the timeslots [4pm,5pm], [10pm,12am] and [1am,2am]. The new time ranges are then passed to the succeeding switch.

If, after passing through all switches on the end-to-end path a time range greater than the holding time is found, the destination is offered the call. If it accepts, it creates a success message with the start time of the earliest time range in the connection request it received (*epst*). As this message traverses the network each switch releases reserved

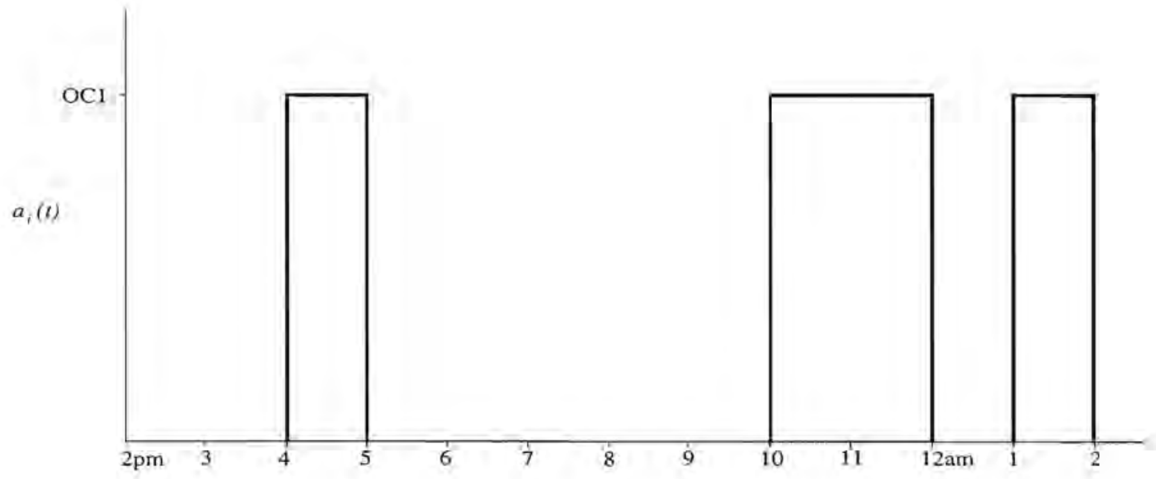


Figure 5.16: Available bandwidth of Switch2, interface i

resources during the unselected time ranges as shown in (5.10).

```
For $1 \leq j \leq n$,
    If $t_{js} \leq epst \leq t_{je}$ then
        $a_i(t) \leftarrow a_i(t) + c$ for $t_{js} \leq t < epst$ and
        $a_i(t) \leftarrow a_i(t) + c$ for $epst + h < t \leq t_{je}$
    else
        $a_i(t) \leftarrow a_i(t) + c$, for $t_{js} \leq t \leq t_{je}$    (5.10)
```

In the above description, we assumed that the number of time ranges sent from each switch is n, a constant. However, it is feasible to allow intermediate switches to create more time ranges than the number n of time ranges selected by the ingress switch. This could happen if resources are not available at an intermediate switch for the whole duration of a time range selected by the ingress switch, but instead parts of the time range have available resources at that intermediate switch. Also, the number of time ranges selected at an intermediate switch could be fewer than n if it does not have the resources available during some of the time ranges.

5.6 Simulation and results

5.6.1 Network model

A comparison of the four different connection setup schemes was done using simulation with the help of the discrete event simulator OPNET Modeler [24]. The network model studied is presented in Figure 5.17. It is a circuit-switched connection-oriented network implementing a very simplified CAC mechanism in which each channel is able to provide one "bandwidth unit", which is requested by a connection. An analogy can be drawn between this network and a SONET (Synchronous Optical NETwork) network in which all the channels are OC1 channels. The network model consists of four switches and all the links shown in Figure 5.17 are unidirectional channels providing one bandwidth unit. In general networks will have much higher channel capacities. We chose only one bandwidth unit for the interfaces to decrease the startup settling time of the simulations. Connections were set up and released between *Source* and *Dest*, and between *srcx* and *destx*. The connections between *Source* and *Dest* were studied (hence called *study traffic*), and the connections between *srcx* and *destx* were created as "interference traffic" to create a more realistic network model. Interarrival times of the connection setup requests and the holding times of the connections are assumed to be exponentially distributed. The destinations always accepted any connection, allowing us to concentrate on the CAC performed by the switches.

The mean call interarrival time and mean holding time for the study traffic was kept constant through all the simulations while the mean call interarrival time and holding times of the interference traffic was varied. This allowed us to simulate the behaviour of the different connection setup schemes' under different load conditions. The mean call interarrival time used by *Source* was 25 seconds and the mean holding time was 5 seconds. This computes to a mean load of 20% introduced to the network by *Source*.

Table 5.2 presents the combinations of mean holding time and mean interarrival time used for the interference traffic generated by *src1*, *src2* and *src3*. During each simulation run each of the sources generating interference traffic did so using the same mean holding time and mean interarrival time. With these different load conditions, the OC1 channels between switches will experience load varying from 25% to 95%, while the load on the channel between *Switch4* and *Dest* is kept constant at 20%.

Table 5.3 presents the parameters used for each scheduling scheme. There are no parameters for the *kTwait* scheme. The parameter D_{max} for the $kT_{wait} - T_{max}$ scheme

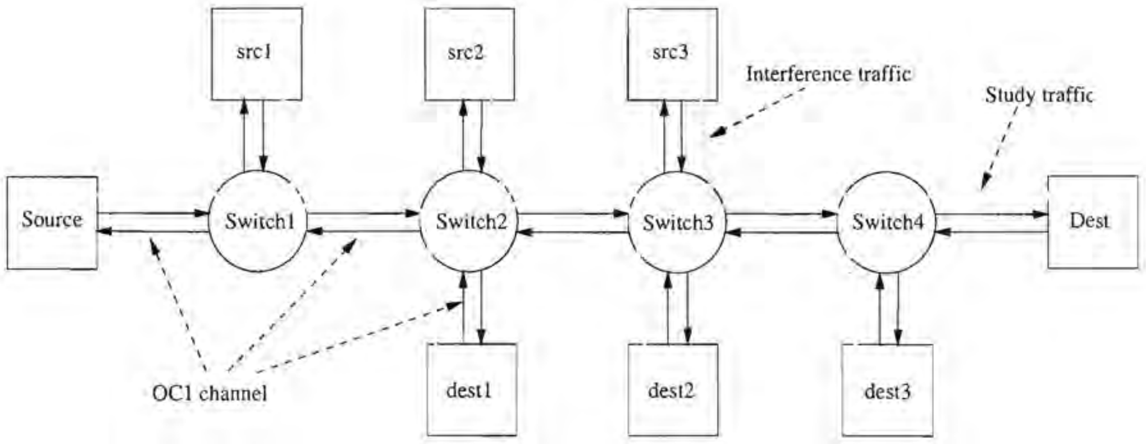


Figure 5.17: Network model

shown in Table 5.3 indicates the maximum queueing delay that is acceptable. In other words, by adding D_{max} to the time when the connection request is sent out by the source we get the value of T_{max}. Each parameter from Table 5.3 combined with each combination of mean interarrival time and mean holding time comprised the parameters for one simulation run. For example, one simulation consisted of the scheduling scheme being *timeslots*, the number of time slots being 4, with the mean interarrival time of the interference traffic of 20 seconds and the mean holding time of the interference traffic of 9 seconds. Each simulation run simulated one hour of network activity and was repeated 201 times, each time with a different seed for the random variables.

5.6.2 Results

The results of most interest are the *utilization* of the channels, the *call blocking probability*, and the *start time delay* returned by the network when a request for a connection is made. In the *timeslots* and F schemes, the channels are either free or in use by a connection. In the two kT_{wait} schemes the channels could be in one of three states: free, reserved or in use. The second state (reserved) is used to indicate that the channel has been reserved for a connection but the success reply has not yet been received, which means the channel is not "in use" (used to transfer data from the source to the destination). Hence, the utilization of the channels are measured by computing how much time a channel is "in use" by connections. The F and *timeslots* schemes also have a period in which resources are reserved but not in use, i.e., between sending the setup message and receiving the success reply. However, this time is far smaller (in the order of ms) than the wait time for

Mean interarrival time	Mean holding time	Load introduced by interference traffic
100	5	5%
50	5	10%
40	6	15%
25	5	20%
20	5	25%
20	6	30%
20	7	35%
20	8	40%
20	9	45%
10	5	50%
20	11	55%
20	12	60%
20	13	65%
10	7	70%
20	15	75%

Table 5.2: Different loads introduced by interference traffic

resources (in the order of sec), and is hence neglected.

The kT_{wait} scheme does not block any calls. For all the other schemes, the call blocking probability is estimated by computing the ratio between the number of connections requested by *Source* and the number of connections released by any intermediate switch.

The *start time delay* is the time difference between when *Source* requests a connection (*req*) and the value of *epst* as returned by the network. In the kT_{wait} and $kT_{wait} - T_{max}$ schemes, where there is no *epst* present, the *start time delay* is the difference between when the connection request is sent and the time when the network replies with success.

Parameter	Values
F (F scheme)	20, 50 and 100 seconds
t (*timeslots* scheme)	2, 3 and 4
D_{max} ($kT_{wait} - T_{max}$ scheme)	100, 150 and 200 seconds

Table 5.3: Parameters used in simulation of connection setup schemes

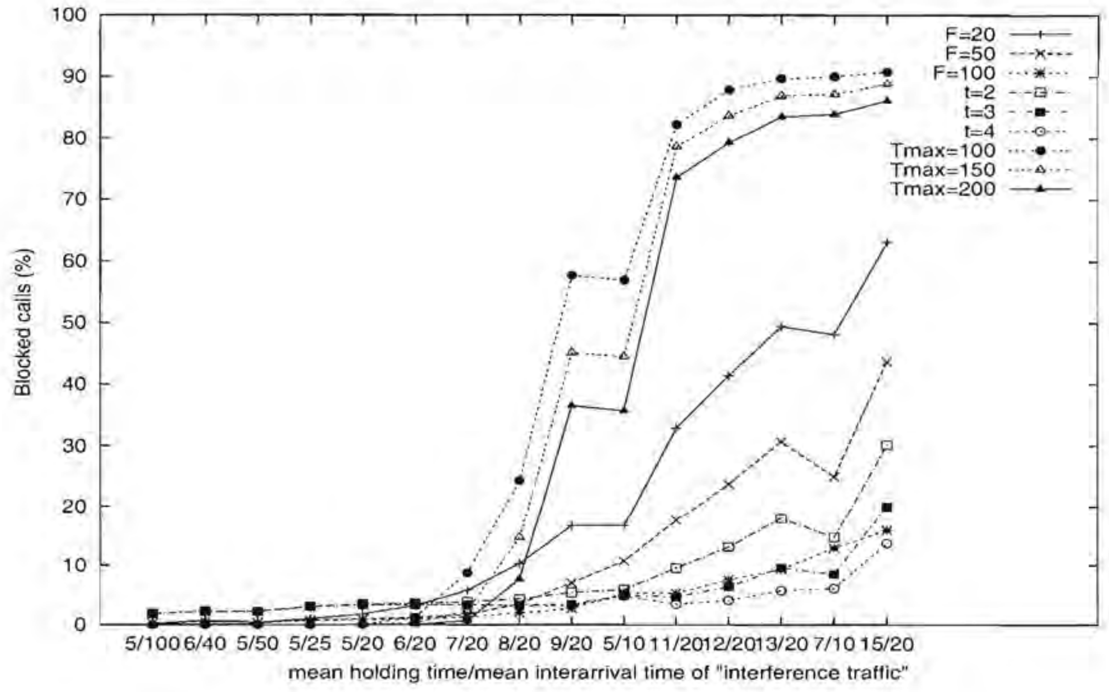


Figure 5.18: Percentage of calls blocked using different connection setup schemes

Figure 5.18 shows the percentage of calls blocked when each connection setup scheme is used. The $kT_{wait} - T_{max}$ scheme performs the worst, even with the usage of a D_{max} value of 200 seconds the blocking percentage is almost 90%. The reason for this high blocking rate can be explained by looking at the queue at *Switch1* in which connection requests requiring access to the channel that connects *Switch1* to *Switch2* are placed. With an increase in interference traffic, this queue grows very large and causes connection requests to experience delays larger than the D_{max} value used, causing the connection to be blocked. Due to the poor performance of the $kT_{wait} - T_{max}$ scheme (as shown in Fig. 5.20) we dismiss it from further comparisons.

The call blocking percentages of the F scheme proves that increasing the value of F decreases the call blocking probability significantly. With the usage of a large F value by *Switch1*, the chances that a succeeding switch can admit a call during the same time *Switch1* is able to admit it increases significantly. Using a small F value (for example 20 seconds) when the average call holding time is large (13 or 15 seconds) serves to increase call blocking.

Increasing the number of time ranges used by the *timeslots* scheme decreases blocking probability significantly. Increasing the number of time ranges selected by the ingress switch improves the chances that an intermediate switch will find an overlapping time

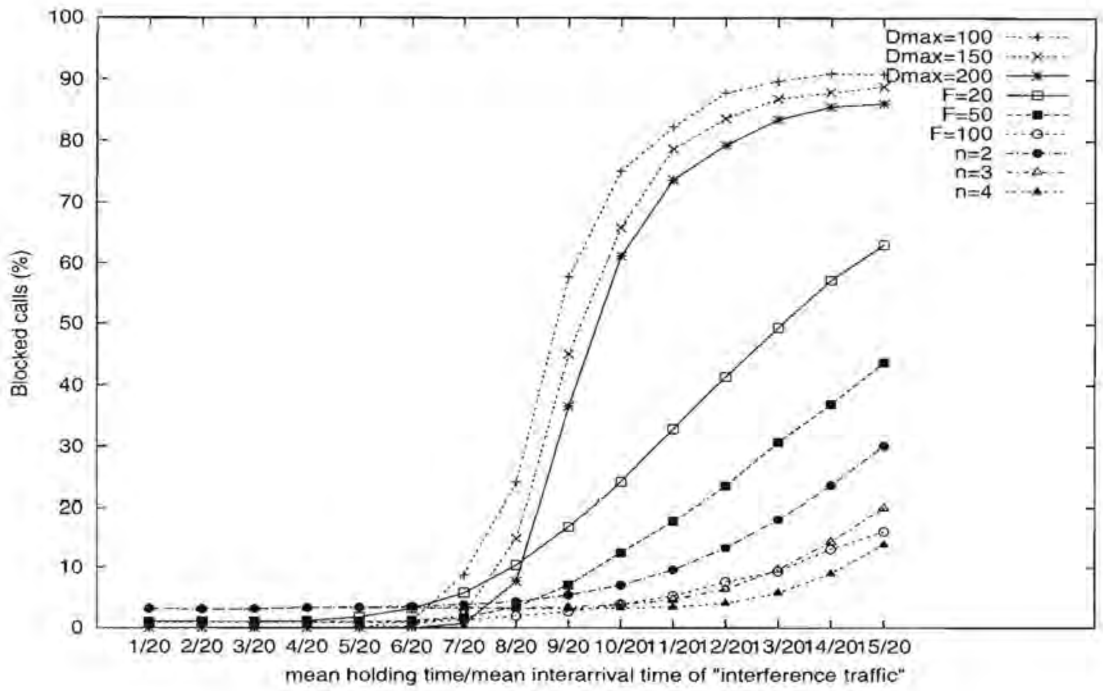


Figure 5.19: Percentage of calls blocked using different connection setup schemes, constant mean interarrival times

range during which it has the requested resources available.

The drop of call blocking probability at the points "5/10" and "7/10" in many of the graphs indicates the dependence of the various schemes on actual values of mean holding times and mean interarrival times. A rerun of the simulation where the interarrival times are kept constant, only varying the mean holding times confirms this. Figure 5.19 contains the results.

It is recognized that the blocking probabilities shown in Figure 5.18 and Figure 5.19 are rather high. By using much larger numbers of timeslots, and/or adding multiple path searches, the blocking probabilities can be reduced to more acceptable values than the currently shown 12% at 95% utilization for the $n = 4$ *timeslots* scheme.

The graph presented in Figure 5.21 shows, for all the remaining schemes, the difference in time from when *Source* requested a connection to *Dest* and the time when it is allowed to start data transmission for all successful connections. Using a large value for F in the F scheme has been proved to decrease the call blocking percentages, but now a disadvantage of a large value for F becomes apparent. Using a large value for F means that the host requesting a connection might receive a start time that is later than can in fact be handled by the network. This is because, in the case where $F = 100$ seconds, even if a connection

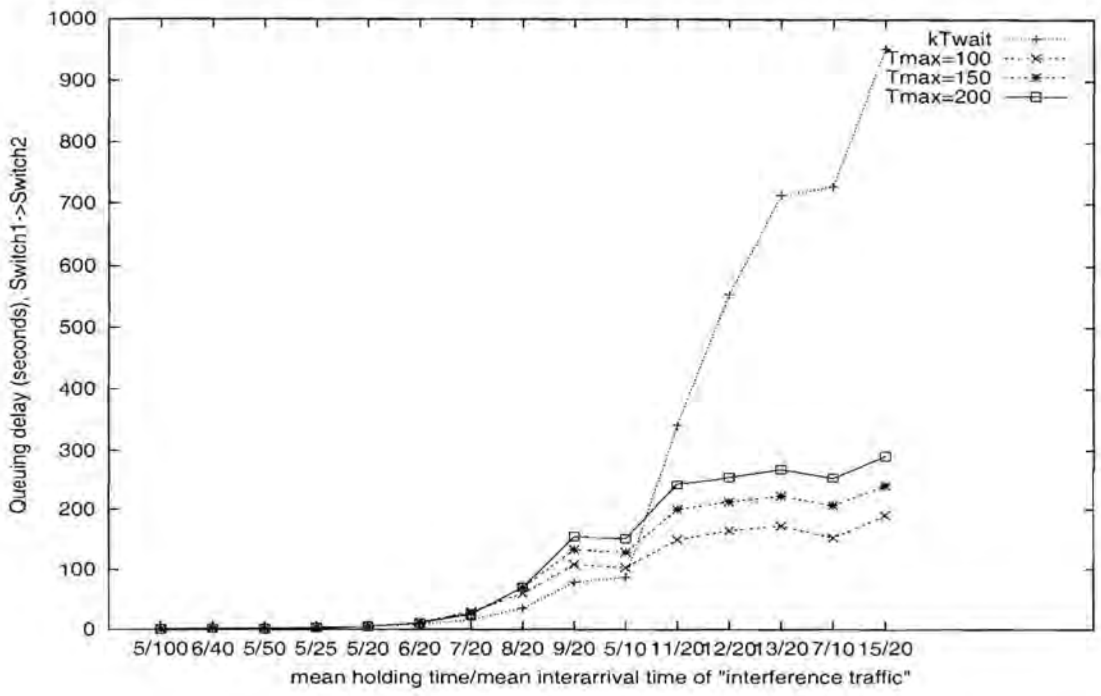


Figure 5.20: Queueing delay experienced by connection requests for connections requiring channel Switch1 -> Switch2

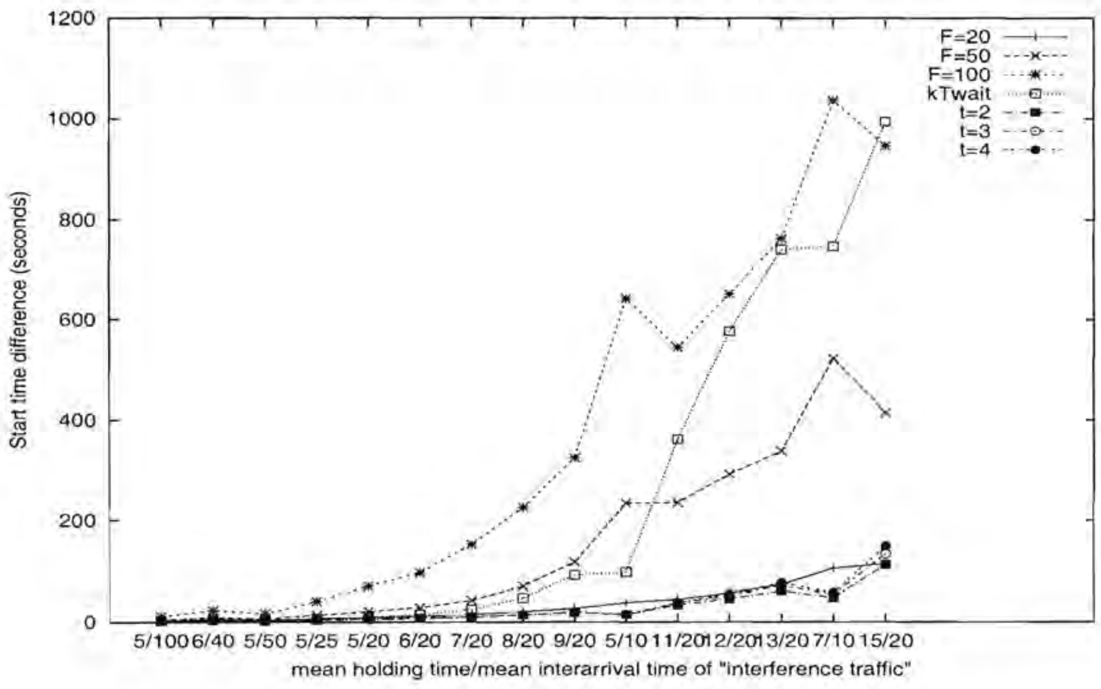


Figure 5.21: Start time delays for all connection setup schemes

will only be active for 5 seconds - the first switch will search for a time period of 100 seconds during which it has enough resources available to handle the connection. Even if the channel is free for 50 seconds when the connection request arrives - the connection will not be admitted immediately. This is the reason for the poor performance of the F scheme when a value of 100 seconds is used. Reducing the F value to 50 seconds and 20 seconds respectively reduces the start time delay considerably.

The *timeslot* scheme proves to be the best choice when a connection needs to be admitted as soon as possible. Note that the choice of different timeslots does not significantly affect the start time delay. This is because when the success reply is returned from the destination, it will always include the *epst* value as the start time of the earliest timeslot.

The kT_{wait} scheme's behaviour can be explained by looking at Figure 5.20. When the interference traffic contributes more than 50% to the total load, the queueing delay at *Switch1* grows exponentially. It appears that the kT_{wait} scheme is not able to handle a load of more than 70%. The advantage of the kT_{wait} scheme in that it does not block any connections now becomes insignificant. Even though the kT_{wait} scheme does not block connections, it becomes unstable when the load is higher than 70%. A source may be content to use the *timeslots* or F scheme to set up connections, and even if a connection request fails, the sending of a request more than once will result in less delays in these schemes than if the kT_{wait} scheme was used under high loads. Again, the dependence of the schemes on the actual values used for the mean interarrival time and mean holding time is illustrated in Figure 5.22.

Figures 5.23 and 5.24 represents the utilization of the OC1 channel between *Switch1* and *Switch2*. The *timeslot* scheme performs the best, even when the load of the network is at 95% (20% study traffic and 75% interference traffic), the first channel is utilized close to optimal. As expected, the F scheme performs poorly when a large value of F is used - although it still outperforms the kT_{wait} scheme. The moment the interference traffic passed the 50% mark, the kT_{wait} scheme's performance started dropping. The reason for this large drop is explained above.

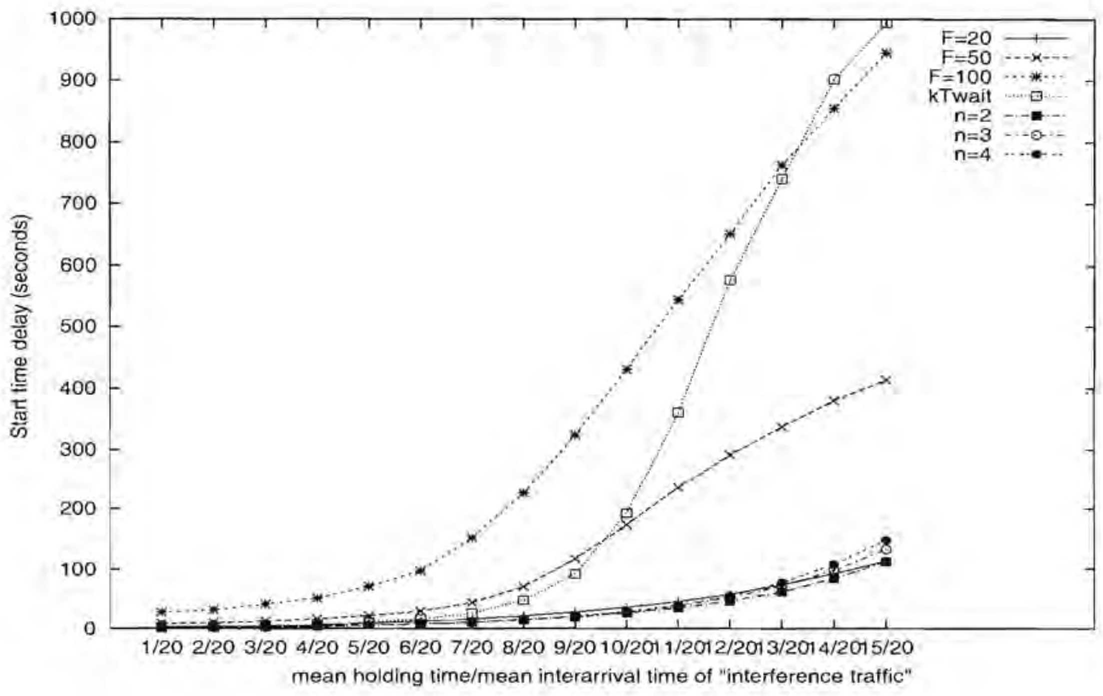


Figure 5.22: Start time delays for all connection setup schemes, constant mean interarrival times

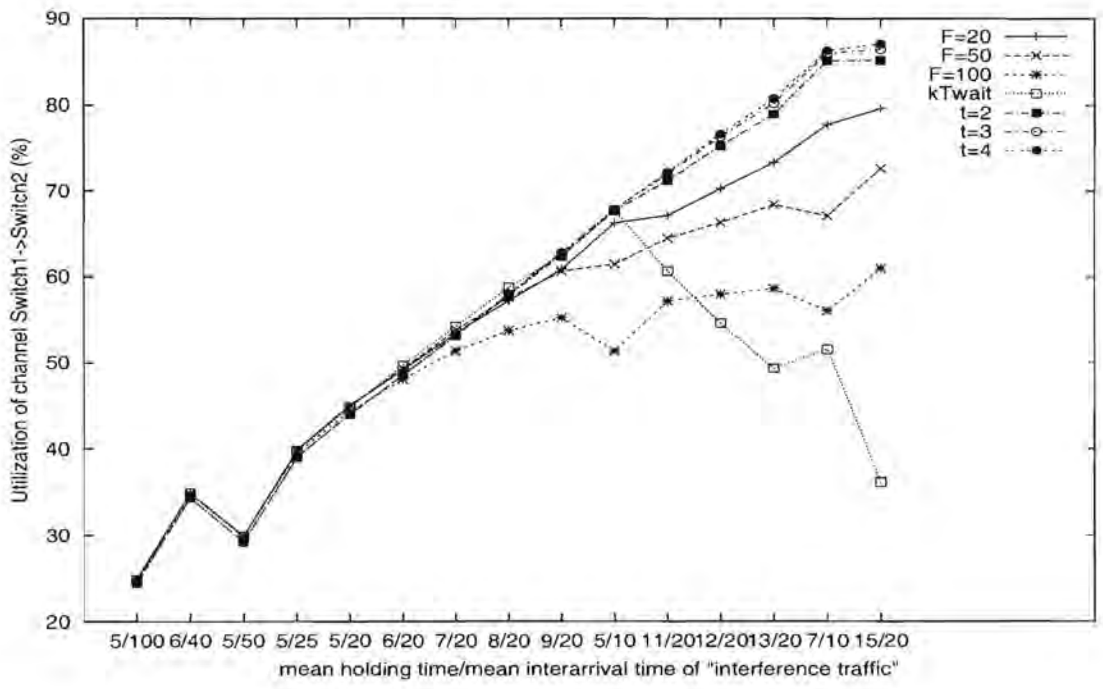


Figure 5.23: Utilization of channel Switch1->Switch2

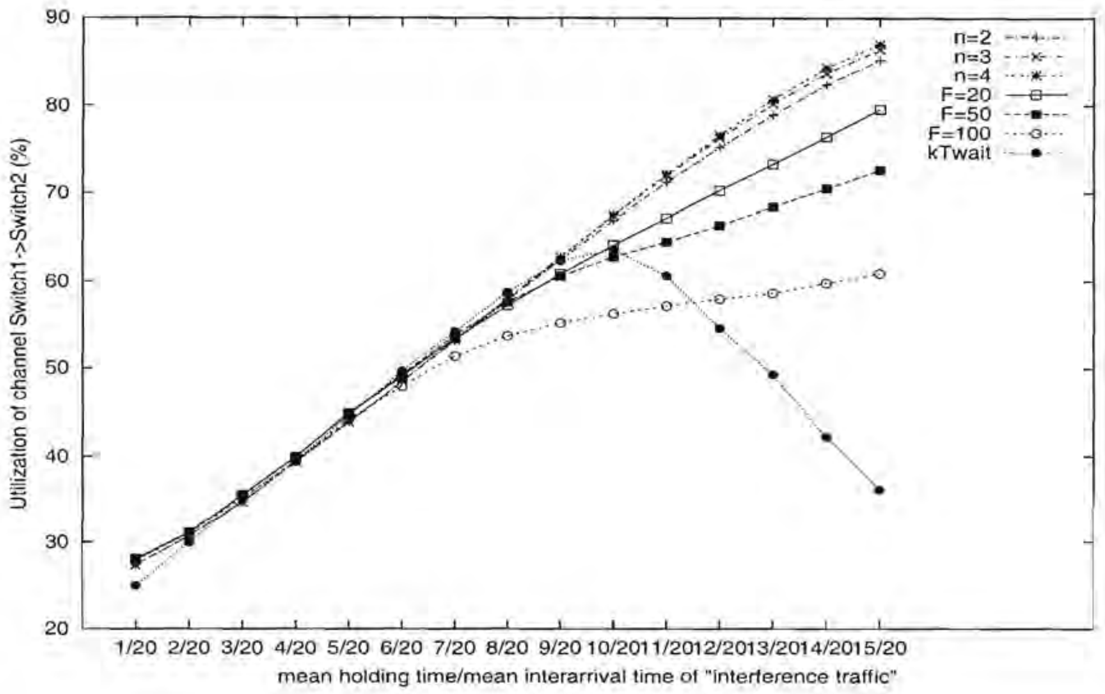


Figure 5.24: Utilization of channel Switch1->Switch2, constant mean interarrival times

5.7 Further extensions and future work

5.7.1 Switch programming delay

Switch programming is the act of setting translation or mapping tables at switches to realize circuits. In the case of circuit-switched connections (circuits), it involves the correct time slot/interface number or wavelength/interface number translations at the switches. In the case of virtual circuits as in ATM networks, it involves label mapping or virtual path identifier/virtual channel identifier mapping tables. The problem introduced by the equations discussed so far is that a switch forming part of the connection will initiate the programming of its fabric at exactly *epst*, which is the same time at which the end host (source) will start transmission. This may have the consequence that data from the source host may arrive at a switch before the resources have been reserved for it. This section considers two possible solutions to this problem.

Solving problem at switches

This solution assumes that each switch forming part of the connection has a value *prog_delay* associated with it, this is the value computed to be the delay from when a switch programming request is sent to the switch programming module until the switch fabric has been programmed with the information for the connection. Once the switch fabric has been programmed, the connection is active. Taking the value *prog_delay* into account, (5.1) changes to (5.11).

For each $j \in I$ determine

$$x_j \geq \text{ current time } \wedge a_j(t) \geq c, x_j \leq t \leq x_j + h$$

$$x \triangleq x_i \text{ where } x_i = min\{x_j \mid j \in I\} \; epst \leftarrow x + prog\_delay \tag{5.11}$$

This change to (5.1) can be applied similarly to all the relations and assignments discussed so far ((5.1) to (5.4)) replacing the variable *epst* with x, and computing *epst* with the assignment $epst \leftarrow x + prog\_delay$.

Solving problem at end host

Consider the value $prog\_delay(S)$, which indicates the delay introduced when programming the switch fabric of switch S. If the source host knows this value for all the switches along the path to the destination, it can compute the maximum of all these programming delays. Now the source host could solve the programming delay problem by only starting its transmission at *epst+ maximum programming delay along path to destination.*

5.7.2 Propagation delay computation

In the introduction to this chapter, the first example application mentioned for scheduled connections was file transfers. In that discussion it was mentioned that the holding time of a connection used to transfer a file can be computed using the equation $h = \frac{f + overhead \times f}{rate} + T_{prop}$.

The problem with this equation is the question of how T_{prop} is computed. All connections are unidirectional, so there is currently no way for two nodes in the circuit switched network to compute the propagation delay between them. The proposed solution to this problem is to choose a value P which is either an educated guess of the value of T_{prop}, or some large value which should be larger than any propagation delay in the network. If a large value is chosen for P, the connection should be released by the sending host after transmission is completed, as opposed to the finish time being used by the switch for connection release. This makes the bandwidth used by the connection available to other connections immediately after release. Bandwidth will not be wasted. There will nevertheless be a time frame from the actual release time until time P during which no connections will be scheduled. This problem resulting from the inability to compute T_{prop} needs more attention.

5.7.3 Time

All the connection setup schemes discussed so far will only succeed in a network where all the end hosts and switches are perfectly synchronized to a common clock. In the simulation environment this was easily modeled using one global clock. However, for an implementation of the connection setup scheme, some mechanism of clock synchronization has to be implemented. An extensive survey done in 1995 reported that most machines using NTP (Network Time Protocol) to synchronize their clocks are within 21ms of their synchronization sources, and all are within 29ms on average [12]. The consequences of different times at switches have not been examined, but it is expected that the scheduling mechanisms will not perform well in such an environment. This is because situations may arise when data is received before a connection is set up, or a connection is closed before data transmission is completed. Mechanisms that use relative time values or some such approach are needed to deal with these time differences.

5.8 Summary and Conclusions

This chapter described various scheduling algorithms for calls with known holding times. To support multiple-switch connections, two scheduling alternatives, the F scheme and *timeslots* scheme, were proposed. In the F scheme, resources are reserved for a long period F, longer than the call holding time, to guarantee that all switches would find a start time for the connection within this long period. In the *timeslots* scheme, multiple time ranges when the ingress switch can support the call are selected and sent downstream in the signalling message. Each intermediate switch then selects a subset from within this set, until the destination is reached. These two schemes were compared against a simple call queueing scheme, the kT_{wait} scheme, which does not assume knowledge of call holding times, and a variant of this scheme. A simulation study of the schemes showed that (a) significant gains are possible by using knowledge of known call holding times, if that is available, and (b) of the two alternatives proposed for scheduling calls with known holding times, the *timeslots* scheme proved to be the better alternative. The kT_{wait}

scheme becomes unstable when the traffic load exceeds 70%. At 70% loading, the *timeslots* algorithm that uses knowledge of holding times can offer start time delays that are 85% smaller than with the kT_{wait} that does not assume knowledge of call holding times. Also, the *timeslots* scheme can be used with traffic loads of 95%. As a result, at 70% load a channel utilization increase of 37% is possible using schemes that exploit knowledge of call holding times, when compared to the kT_{wait} scheme. The reason we consider the *timeslots* scheme the winner over the F scheme, is that at large values of F, call blocking is low, but the start time delay is high, and with small values of F, the inverse is true. On the other hand, in the *timeslots* scheme, both the call blocking probability and the start time delay are relatively low for all loads tested. Its drawback over the F scheme lies in the increased signalling overhead, which might be negligible if few time slots are used.

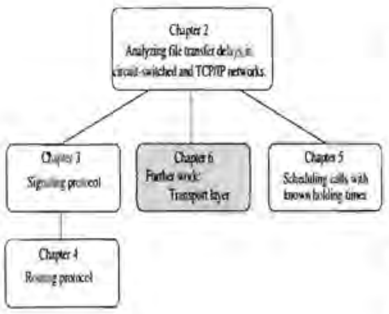
The novelty of this work lies in its introduction of call queueing and providing delayed starts for connections. It culminates in a proposal to use known call holding times while scheduling to improve both network resource utilization and call queueing delays.

Future work includes the incorporation of the *timeslots* scheduling scheme into the signalling protocol presented in Chapter 3.

Chapter 6

Further work: The transport layer

6.1 Introduction


Chapter 2
Analyzing file transfer delays in circuit-switched and TCP/IP networks.
Chapter 3
Signaling protocol
Chapter 6
Further work:
Transport layer
Chapter 5
Scheduling calls with known holding times
Chapter 4
Routing protocol

To enable an application to transfer bulk data, for example large files, over a circuit switched network, a transport layer is required to provide the application with the guarantee of a complete and error free transmission. The details of this discussion are beyond the scope of this research, and are currently used as guidelines for future work.

The large bandwidths available in a SONET network means that the product of end-to-end delay and bandwidth is very large. The "bandwidth*delay product" measures the maximum amount of data in transit between two nodes. For example, if the transport protocol TCP is used to transfer data between two SONET nodes, the "bandwidth*delay product" is the buffer space required at the sender and receiver to obtain maximum throughput on the TCP connection over the path, i.e., the amount of unacknowledged data that TCP must handle in order to keep the pipeline full [25]. Although solutions have been proposed in [25] to increase TCPs performance in networks with high "bandwidth*delay products", it is still not efficient enough to be run over a circuit switched network.

NETBLT (NETwork BLock Transfer) [26] is a transport level protocol intended for the rapid transfer of a large quantity of data between computers. The protocol features also allow it to perform very well in networks with a high "bandwidth*delay product". NETBLT was initially designed to run over IP and therefor provides a service of connection setup between the communicating nodes using a two-way handshake. In a circuit switched network, the transport layer does not need to set up a connection between the transport entities due to the fact that a (network layer) connection has already been set up between the communicating nodes.

Instead of examining more transport protocols to find an ideal protocol to be used for file transfers in a circuit switched network, the functions of transport protocols are examined. Section 6.2 briefly discusses the transport layer capabilities that are envisioned to be ideal when a circuit switched network is used to transfer bulk data. These capabilities are currently being evaluated by the CATT research group for the design of a new transport protocol.

6.2 Protocol requirements

The transport protocol is responsible for guaranteeing a reliable data transport service between two nodes connected with a uni-directional (from the source to the destination) channel. A uni-directional link limits the communication between the two transport entities, but creating a bi-directional link between the transport entities will result in a waste of bandwidth. For example, the smallest channel available in a SONET network is OC1 (51.84Mbps), which, when dedicated to communication between transport entities for the whole duration of a file transfer, will be very wasteful. Recall that all the nodes forming part of the SONET network will have interfaces connecting them to the IP network. It is therefore proposed that all messages from the destination transport entity to the source transport entity be sent over an IP network. These messages are few during a file transfer. They should not add an excessive load to the IP network.

Figure 6.1 shows the paths TPDUs (Transport Protocol Data Units) will follow between two nodes connected by a network layer connection.

The services provided by a transport protocol are:

- *connection management*, which is a mechanism needed to maintain state information regarding the data transfer at the source and destination;
- *flow control* to avoid buffer overflows at the destination;
- *acknowledgements* received by the source that indicate the reception of data (successful or unsuccessful) by the destination;
- *error handling* that includes error detection and recovery schemes;

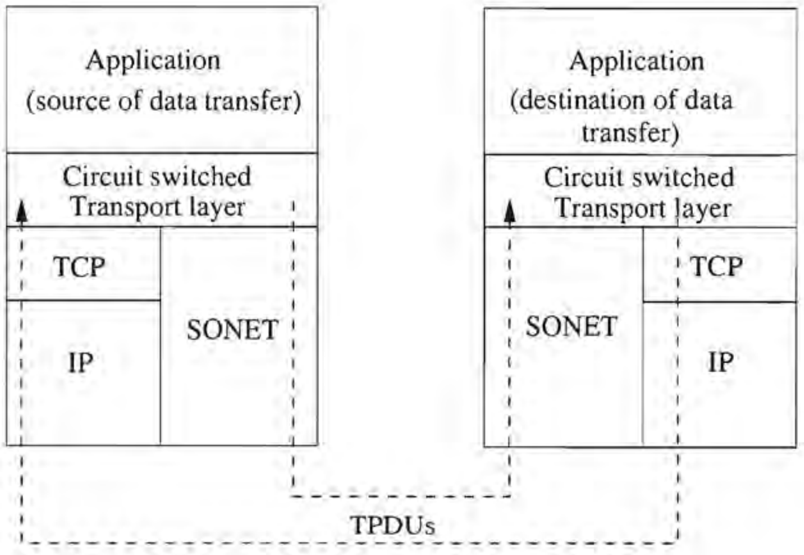


Figure 6.1: Transport layer in a connection oriented circuit switched network

- *congestion control* to handle lost packets in network routers.

Each service will be considered next, and recommendations are made for mechanisms that will be ideal for a transport protocol in a circuit switched network.

To transfer a file completely and without error, the two corresponding transport layers (at the source and destination) have to keep a record of which data has been received (at the destination), and how much data may be sent without acknowledgement (at the source). This is the *state information* of the transport protocol. The updates of state information, which should only be sent by the destination, will have to be sent out-of-band. That is, with the data being transferred over the circuit-switched network, the destination notifies the source about the progress of the transfer by sending messages over the connectionless packet-switched network (IP), as depicted in Figure 6.1.

It has already been suggested that there need not be a connection set up between the two corresponding transport layers because of an existing connection at the network layer level. The existence of a network connection allows the transport protocol to make use of an *implicit connection setup scheme* in which a transport layer connection is opened when the network layer connection is set up. The transport layer connection can now either be closed when the network layer connection is closed and/or the closing of the connection can be based on a timer when a scheduling mechanism similar to the ones discussed in Chapter 5 is used to set up connections.

The transport protocol has to bridge the gap between the services provided by the network layer, and the services required by higher layers [27]. In our target application of file transfer over a circuit switched network, it is assumed that when the (transport layer) connection is set up, the applications would already have agreed on the transfer rate acceptable to both the source and destination, which has been negotiated during the setup of the network layer connection. The decision to use an implicit transport layer connection setup scheme implies that all transport layer parameters that need to be negotiated are agreed to during network layer connection setup.

The rate at which the destination can accept data is taken into account when the circuit is set up. Because a circuit-switched network is used to transfer the data, the network will never become congested due to the reservation of resources in the network. But, once a connection is set up between the network layers of the source and destination, it is possible that, (due to variations in host tasks activity) the destination might not be able to accept data at the rate it had been able to handle when the network layer connection was set up. Thus, some *flow control* mechanism needs to be introduced to enable the destination to notify the source that it needs to limit the data transmission. For flow control, the mechanism of *rate control* is ideal. Using rate control the destination can ask the source to send data at a lower rate. For example, if an OC12 connection has been set up between the source and destination, the destination can request the source to send data at OC1 rate. This has the consequence that some resources might be wasted in the circuit-switched network during the times that the destination is not able to handle the initial data rate.

Data being transferred over a circuit-switched network will never be delivered out of sequence. This leaves the transport protocol with two responsibilities. First, all the data sent by the source has to reach the destination application free of bit errors (*error free delivery*). Second, all data transmitted by the source has to reach the destination (*complete delivery*). The usage of sequence numbers is required for error correction. Byte-based sequence numbers (as used in TCP) allows for efficient error correction by indicating the exact amount of data that is missing or containing errors, but considering the amount of bytes that could be in transit between two nodes in a SONET network, the sequence numbers will get really large. Thus a packet based sequence numbering scheme is proposed. Because the TPDUs will always arrive in sequence, a missing packet will be easily discovered. The destination can request the retransmission of a missing packet by sending a NAK (Negative Acknowledgement) to the source on the IP network. By making use of

NAKs, as opposed to positive acknowledgements (ACKs) only used for data received error free and in sequence, the signalling load is reduced considerably.

Additionally each TPDU could include a FEC (Forward Error Correction) code that is able to handle the most frequently occurring error patterns. When an error is encountered in a TPDU, a NAK is sent to the source requesting a retransmission of the packet (identified by its sequence number), similarly to the mechanism summarized in [28].

To determine if all the data transmitted by the source has reached the destination (*complete delivery*), it is proposed that all TPDUs are always the same length, except possibly the last TPDU. The last TPDU also includes an indication to report that it is the last TPDU of the transmission, thus indicating a complete transmission. Once the transport protocol receives this indication, it can signal the SONET signalling process to close the network layer connection. Given that circuits are best used with continuous traffic (rather than bursty), our thinking is that it is appropriate to disconnect the circuit when transmission is done.

6.3 Conclusion

Only a few design decisions have been considered for the transport protocol. The decisions should result in a light-weight transport protocol that allows for the complete and error free data delivery in a circuit switched network. The interfaces between the transport layer and the network and application layers still needs to be defined, as well as the detail of all the mechanisms described above.

Chapter 7

Conclusion

This research set out to find efficient mechanisms for bulk data transfer. An analytical and experimental analysis of file transfer delay in circuit-switched and TCP/IP networks suggested that circuit-switching is ideally suited for bulk data transfer.

To enable TDM networks to be used for bulk data transfer, a mechanism is required to enable the setting up of high-bandwidth on-demand circuits. This is because switched connections (as opposed to provisioned connections) are currently only available at the low rate of DS0(64Kbps) while data rates of OC1(51.84Mbps) to OC768(40Gbps) would be ideal for bulk data transfer. A signalling and supporting routing protocol that enables on-demand setup of circuits with rates of OC1 to OC192 was designed to reach this goal. For high throughput and a general increase in efficiency, the signalling protocol was designed to be implemented in hardware, and the routing tables used by it are updated by a routing protocol that was designed to be implemented in software. The signalling protocol is currently being implemented in hardware by the CATT research group.

Bulk data has the characteristic that when it is transferred over a connection-oriented network that implements preventative congestion control mechanisms (as opposed to reactive congestion control as implemented in TCP/IP networks), the duration or holding time of the connection can be determined by using the filesize, data rate and propagation delays.

Two scheduling algorithms were designed that use these known holding times. These algorithms enable a switch to reply with a later start time if it is unable to accommodate the connection request at the time of its arrival. The scheduling algorithms are simulated and compared to a simple queueing mechanism where a connection request is queued until its requested resources become available, only then being forwarded to the next hop of the connection.

The use of such scheduling is shown to improve the efficiency of bulk data transfer in connection-oriented networks. The simulations show that at 70% loading, these schemes offer start time delays that are up to 85% smaller and channel utilization of up to 37% larger than a simple queueing mode of operation where call holding times are ignored when connections are set up.

The evidence thus suggests that with appropriate signalling and scheduling, efficient bulk data transfer can be achieved on circuit-switched networks. Of course, a suitable transport protocol would be required to enable the reliable transfer of data between two hosts. While the ideal characteristics of such a protocol have been described in this dissertation, details of its design and implementation remain a matter for future research.

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