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Communication Protocols and Channel Assignment Algorithms for High-Bandwidth Multichannel Networks: Design and Analysis

Ming Chen

A THESIS
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in
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Abstract

With the rapid progress in Wavelength Division Multiplexing (WDM) techniques, high-bandwidth multichannel networks with many concurrent gigabit channels over a single-mode fiber become feasible. In this thesis, we investigate communication protocols and channel assignment problems in these high-bandwidth multichannel networks. We propose a dynamic channel sharing protocol to accommodate both connection-oriented real-time traffic and connectionless non-real-time traffic efficiently. We further design a multiconnection protocol to allow each originating station to set up connections with multiple terminating stations and allow each terminating station to keep connections with multiple originating stations. To support these protocols, we also propose some dynamical channel assignment algorithms for connectionless traffic. These algorithms are further extended to two-layer high-bandwidth multichannel networks. The proposed channel assignment algorithms are compared with some existing algorithms found in the literature. Analytical performance results in terms of blocking probability and channel utilization for the proposed protocols, and also efficiency and computational complexity for the proposed algorithms, are given. Results show superior performance of these proposed protocols and algorithms.

Acknowledgments

I would like to express sincere gratitude, first and foremost, to Professor N. D. Georganas, my research supervisor, for his gentle and patient guidance throughout this research; also, my great respect for his competence and diligence. Without his support, this thesis would not have been possible.

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Acronyms

ATM:	Asynchronous Transfer Mode
CCITT:	The International Telegraph and Telephone Consultative Committee
CF:	Cluster First algorithm
CSMA:	Carrier Sense Multiple Access
DAP:	Dynamic Assignment Protocols
DRR:	Dynamic Round Robin algorithm
DT-WDMA:	Dynamic-Time Wavelength Division Multiaccess
FDDI:	Distributed Queue Dual Bus
HBMN:	High-Bandwidth Multichannel Networks
HOL:	Head-Of-Line
LAN:	Local Area Networks
LS:	Local Scheduling algorithm
MAN:	Metropolitan Area Networks
MRS:	Maximum Remaining Sum algorithm
MT-RAP:	Multi-Transceiver Random Access Protocols
NAL:	Node Activity List
N-R-T:	Non-Real-Time
OS:	Optimal Selection algorithm
RA:	Rotating Assignment algorithm
RC:	Receivers
RCA:	Receiver Collision Avoidance
RS:	Random Scheduling algorithm
R-T:	Real-Time
SAP:	Static Assignment Protocols
SCM:	Subcarrier Multiplexing
SDR:	System Distinct Representative algorithm
SMDS:	Switched Multimegabit Data Services

SMR:	Selecting Maximum Rank algorithm
SP:	Selective Priority algorithm
ST-RAP:	Single-Transceiver Random Access Protocols
TDM:	Time Division Multiplexing
TX:	Transmitters
WAN:	Wide Area Networks
WDM:	Wavelength Division Multiplexing
WDMA:	Wavelength Division Multiaccess

Chapter 1

Introduction

1.1 A Brief Review of Computer Network Evolution

Computer communication networks that facilitate the information delivery have undergone dramatic changes over the past two decades in terms of ever-increasing information transmission rates, the sophistication of services and applications, the integration of traffic, and interconnection of different and remote networks. The evolution in physical-level technology of these networks has classified computer networks into three generations [44]. In the age of first-generation networks, optical communication technology has not been explored. Copper-wire and microwave-radio were used as transmission media. Many existing networks are examples of first-generation networks, such as IEEE 802.3 Ethernet, IEEE 802.4 token bus, IEEE 802.5 token ring, ARPANET [89]. Information rates of these networks for coax-based networks are from a few kilobits per second to a few megabits per second. Second-generation networks employ fiber to replace the copper, co-ax and microwave media, such that the transmission rate is updated to the magnitude of a hundred megabits per second and communication quality is improved. Typical examples are the long-haul wide area networks using single wavelength on fibers, Fiber Distributed Data Interface (FDDI) ring networks and IEEE 802.6 Distributed Queue Dual Bus (DQDB) for local area networks (LANs)/Metropolitan area networks (MANs). Broadband Integrated Services Digital Networks (BISDN) is another example of the second-generation computer networks for WAN applications. The replacement of copper-wire and microwave-radio by fibers greatly increases the information rates, and reduces error rates and electromagnetic interference. The transmission rate of second-generation networks, however, is limited by the processing speed of electronic interfaces. Electrical/Optical and Optical/Electrical (E/O or O/E) conversions in second-generation networks result in speed bottlenecks. In third-generation networks, the number of E/O and O/E conversions is minimized by exploiting the unique properties of fibers. Signals in communication subnets, i.e., within networks, can remain in the optical domain and thus electronic bottlenecks can be avoided. The third-generation networks are therefore often entitled "all-optical" networks and are expected to be a key carrier backbone for future communications to support gigabit applications and novel services [43] - [45].

The optical bandwidth available on current single-mode fibers in the 1200 to 1700 *nm* low-loss windows is about 50 to 60 THz. Theoretical estimates indicate that such a bandwidth could carry over three orders of magnitude more capacity than today's transmission rates. The key in designing third-generation networks is to divide the huge bandwidth of a single mode fiber into many subchannels, where each subchannel has enough capacity to accommodate the peak rate of electronics. The communication capacity of the whole system can be thus largely increased and, in the mean time, flexible multiaccess and packet-switched communication architectures can be achieved by the use of these concurrent subchannels. In all-optical networks, these subchannels and multiaccess functions can be implemented by Wavelength Division Multiplexing (WDM) techniques [7], where each subchannel refers to a different wavelength and runs at a gigabit rate. The implementation of WDM networks needs several key optical components, such as optical amplifiers, tunable lasers and filters, optical star couplers, coupling and splitting devices. Some prototypes and demonstrations have been reported in recent years [7], [31], [41]. Full interconnectivity among multiple users is achieved by tuning tunable lasers or filters to their desired wavelength channels. Noticing the major feature of WDM networks in providing many concurrent high-bandwidth multichannels, we call these networks, in this thesis, high-bandwidth multichannel networks.

Progress in high-bandwidth multichannel networks has been paralleled by the equally impressive development in computer and video technologies. Market projection has indicated that the service volume on computer networks will increase dramatically in the next decade. This increase will be primarily due to growth in data traffic transfer from applications such as local area network (LAN) interconnects, image and file transfer, and later, supercomputer access and internetworking, interactive image communications, visualization, distributed computing, multimedia teleconferencing, and other advanced applications. Future services such as "fiber to the office" and "fiber to the home" are possible to provide hundreds and thousands of users with cross connections. High-performance full-motion color-graphics workstations with interactive client-server and graphic-oriented services requiring gigabits bandwidth will gradually enrich our working and living environment. These increasing roles for new emerging applications have facilitated the investigation of high-bandwidth multichannel networks.

1.2 Thesis Content

In chapters 2 and 3, we introduce some possible architectures and topologies as well as communication protocols for high-bandwidth multichannel networks. We classify architectures of high-bandwidth multichannel networks into all-optical and quasi-all-optical classes. Protocols for single-hop multichannel networks are emphasized and categorized into four types respectively as 1) single-transceiver random access protocols, 2) multi-transceiver random access protocols, 3) static assignment protocols, and 4) dynamic assignment protocols.

In chapter 4, we design and analyze a multiwavelength network with dynamic channel sharing. Traffic in the network is categorized into a connection-oriented real-time class and a connectionless non-real-time class according to CCITT recommendations for broadband services. Channels are first nominally assigned to each of these classes of traffic, and then are allowed to be shared whenever traffic of one class becomes heavy while the other does not. The network performance with and without channel sharing in terms of blocking and mean packet delay for real-time traffic and channel utilization for non-real-time traffic are analyzed and compared. Results show that the proposed dynamic channel sharing protocol can substantially improve the network performance without increasing the implementation complexity. The proposed network has the following features: 1) fully dynamic channel sharing between the two classes of traffic and among all real-time connections at a terminating station; 2) many-to-one connections; 3) small and bounded packet delay for real-time traffic; and 4) high network throughput.

In chapter 5, we study a multiconnection problem in multichannel networks and propose an efficient multiconnection protocol for establishing and releasing connections with dynamic bandwidth requirement. Each originating station in the network is allowed to set up connections with several terminating stations, and send packets to these terminating stations in different slots in a TDM fashion. Similarly, each terminating station is allowed to keep connections with several originating stations, and receive packets from these originating stations in different slots in a TDM fashion. Efficient slot allocation schemes for multiple connections at the same station, as well as conditions for strict-sense acceptance and rearrangeable acceptance are found.

In chapter 6, we propose three algorithms for channel assignment in high-bandwidth multichannel networks. The objective of these algorithms is to select as many packets as possible for transmission without destination conflict. Performance in terms of maximum achievable throughput, mean packet delay, and computational complexity is analyzed and compared with two typical algorithms found in the literature. Our results show that the proposed algorithms enjoy close performance and much lower computational complexity as compared to the existing optimal algorithms.

In chapter 7, we study channel assignment in two-layer multichannel networks. A two-layer multichannel network includes a backbone layer and a cluster layer, and can significantly increase the bandwidth utilization. Though, in general, channel assignment algorithms for one-layer multichannel networks can be applied to two-layer networks, the algorithm efficiency is degraded because an efficient algorithm for one-layer networks may no longer be efficient for two-layer networks. We propose two channel assignment algorithms for two-layer multichannel networks. Simulation and analysis show that the proposed algorithms perform very closely to the optimal solution while their complexity is much lower.

Finally, we summarize this thesis and suggest future research in chapter 8.

1.3 Summary of Thesis Contributions

In this thesis, we propose two multiaccess protocols and several channel assignment algorithms for high-bandwidth multichannel networks. We also model and analyze the performance of these protocols and algorithms. The major contribution of this thesis can be summarized as follows.

- We propose the dynamic channel sharing multiaccess protocol which is shown to be very efficient in accommodating different classes of traffic. Channels are first nominally assigned to every class of traffic according to its traffic prediction. These nominally assigned channels are then allowed to be dynamically shared by different classes of traffic according to their actual traffic loads.

- We study the tradeoff in choosing a scheduling or a non-scheduling algorithm in designing a multiaccess protocol. Results show that the scheduling algorithm has much better performance than the non-scheduling algorithm for certain range of network size. For very large network size, however, the execution time of the scheduling algorithm may become an intolerable protocol overhead which results in even poorer performance than a non-scheduling algorithm.

- We propose a multiconnection protocol for connection-oriented traffic. Unlike existing protocols found in the literature, the proposed protocol allows an originating station to have multiple connections with terminating stations, and allows a terminating station to connect with multiple originating stations. We find constraints on these multiple connections, as well as strict-sense acceptance conditions and rearrangeable acceptance conditions, for connection requests. We propose efficient bandwidth allocation schemes to allocate slots to multiple connections with different bandwidth requirements.

- To support the proposed protocols, we also propose some channel assignment algorithms. The K-HOL algorithm computes the first K packets of each input buffer to produce a conflict-free channel assignment for all stations. Since K is a parameter and can be selected, this algorithm has the flexibility in trading-off complexity and efficiency. This is an important feature of the algorithm when dealing with a network with a critical processing time requirement. We also proposed two other channel assignment algorithms, the Dynamic Round Robin (DRR) algorithm and the Selective Priority (SP) algorithm. DRR is simpler than SP, while SP performs better than DRR in terms of delay and throughput. All of these proposed algorithms are heuristics and are shown to be much simpler than optimal algorithms while the performance is shown to be comparable.

- We study the channel assignment problem in two-layer high-bandwidth multichannel networks, where the first layer of the network is called the cluster layer and the second layer is called the backbone layer. We propose the Selecting Maximum Rank (SMR) algorithm and the Rotating Assignment algorithm (RA) for channel assignment in two-layer multichannel networks. Though existing algorithms for one-layer multichannel networks are considered applicable for two-layer multichannel networks, results show that their performance is much poorer as compared with the above proposed algorithms.
- We employ Markov Modulated Bernoulli Processes (MMBP) to model different classes of traffic of the network. The change of the Markov state indicates the change of traffic classes. Under uniform traffic conditions and the fair channel assignment feature of our algorithms, we decompose multiple queues of the network into individual ones. We find the blocking probability for connection-oriented traffic and the channel utilization for connectionless traffic. We also find the maximum achievable throughput, the computational complexity, and the mean departure rate of packets of input queues under the proposed channel assignment algorithms. With these results, we obtain the mean packet delay. These analytical results are shown close to simulation results.

1.4 Publications Resulting from This Thesis

1. M. Chen and N. D. Georganas, "Design of a multiwavelength network with dynamic channel sharing," submitted to *IEEE/ACM Trans. Networking*, 1994. A short version in *Proc. IEEE Globecom94*, session 54, San Francisco, CA, Nov. 1994.
2. M. Chen and N. D. Georganas, "Channel assignment in high-bandwidth multichannel networks: algorithms and their performance," submitted to *IEEE Trans. Communications*, 1994.
3. M. Chen and N. D. Georganas, "Channel assignment in two-layer multichannel networks," submitted to *Journal of Computer Networks & ISDN Systems*, 1994.
4. M. Chen, N. D. Georganas, and Oliver Yang, "A fast algorithm for multi-channel/port traffic assignment," *Proc. IEEE Supercom/ICC'94*, New Orleans, Louisnana, May 1994, pp.96-100.

5. M. Chen and N. D. Georganas, "Communication protocols for SCM-WDM networks," *Proc. IEEE Globecom'93*, Houston, Texas, pp.154-158.
6. M. Chen and N. D. Georganas,"Multimedia data delivery in high-speed multichannel networks," *Proc. CCECE'93*, Canadian Conf. on Electrical & Computer Engineering, Sept. 1993, Vancouver, pp.636-639.
7. M. Chen and N. D. Georganas, "Multiconnection over multichannels," accepted by *IEEE Infocom'95*, Boston, MA, 1995.

Chapter 2

High-Bandwidth Multichannel Networks: Architectures

The most popular approach to implement a high-bandwidth multichannel network is to divide the vast bandwidth spectrum of a single mode fiber into many concurrent subchannels using wavelength division multiplexing (WDM) techniques [21]. Each subchannel refers to a different wavelength and thus transmitters (lasers) and receivers (optical filters) can work on different wavelength channels accordingly without interfering with each other. Different architectures of multichannel networks can be realized depending on the use of different optical components, network topologies, and different routing strategies. Each architecture has its own features and suited to specific applications. These architectures can be identified as single-hop, multihop, broadcast-and-select, and wavelength routing [77]. In this thesis, we generally categorize high-bandwidth multichannel networks into two classes: *all-optical* networks and *quasi-all-optical* networks. An all-optical network is defined as a network where lightwave signals once injected into the network will not be converted to electrical forms in intermediate nodes to get routing information before reaching their destinations. A quasi-all-optical network, on the other hand, refers to a network where lightwave signals need to be converted to electrical versions at intermediate nodes before being forwarded to the next node. All-optical networks obviously have higher throughput and smaller network response time than quasi-all-optical networks as there is no electronic speed bottleneck inside the network. These features are paid, however, by employing more critical and expensive optical devices. Quasi-all-optical networks, though transmission rates are limited by electronic speed, are more flexible and easier to be implemented because routing decisions made in quasi-all-optical networks are in the electrical domain which is much more controllable. Quasi-all-optical networks will co-exist with all-optical networks, before they are finally replaced by all-optical networks, just like the replacement of copper and microwave radio by optical fibers.

Throughout this thesis, we interchange the use of terminologies of nodes and stations. Both of them refer to user terminals where data is generated and received. A station transmitting data is called an originating station, and a station receiving data is called a terminating station. A station can be both an originating station and a terminating station simultaneously, if it has transmission and reception demand at the same time.

2.1 All-Optical Multichannel Networks

Current achievable all-optical multichannel networks are defined on the subnets excluding end users. All-optical multichannel networks can be realized by the use of passive star couplers, directional couplers, as well as some multiplexers and demultiplexers to perform routing functions. We sequentially introduce star, bus, ring, routing and linear all-optical networks in the following subsections.

2.1.1 Star Networks

A multichannel network using a passive star-coupler is shown in Fig. 2.1. Basic hardware includes an optical passive star coupler and fibers connected with N stations, where each station works at a different wavelength. The coupler reflects incoming wavelength signals to all the stations. Each station can be equipped with one or many optical transmitters (lasers) and receivers (optical filters) for transmission and reception of data as well as control signals. These transmitters and receivers can be either tunable or fixed-tuned, depending on particular communication protocols used. Full interconnectivity is achieved by tuning their transmitters and receivers to desired wavelength channels.

If we regard each transmitter and receiver of a station as an input port and output port of the star-coupler respectively, the star architecture can be used as a photonic switch fabric [40]. To

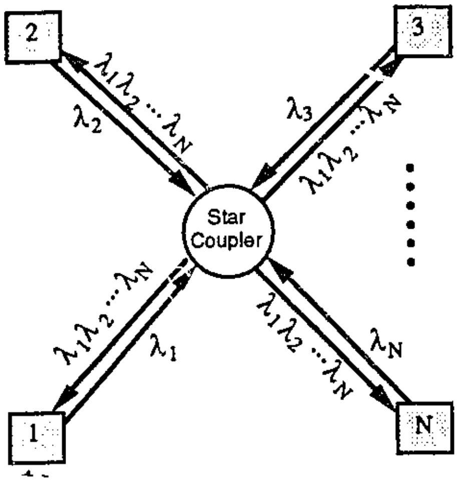


Figure 2.1: *A passive star network*

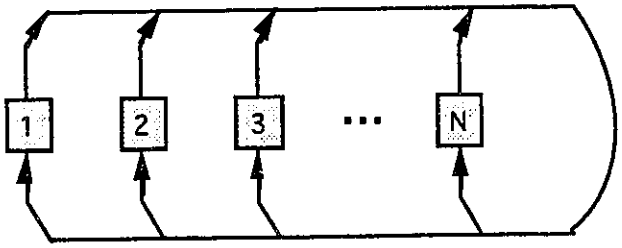


Figure 2.2: *A unidirectional bus network*

realize fast packet-switching, fast tunable lasers and/or tunable filters are needed.

2.1.2 Bus Networks

A bus multichannel network using directional couplers is displayed in Fig. 2.2. Each station on the bus network accesses two bus lines through taps of directional couplers. One bus line is for transmission and the other is for reception. Traffic is first injected into the transmission bus, and then broadcast to receivers on the reception bus. The bus topology is attractive for electrical high-speed networks, because it provides sequential ordering and collision-free multiaccess for all users. Comparing to the star networks, however, the bus structure has a serious power budget problem. If each station transmits a power P, then, in a star network, each station can receive a power of P/N, while in a bus network, only P/N^2 power can be received at each station [49].

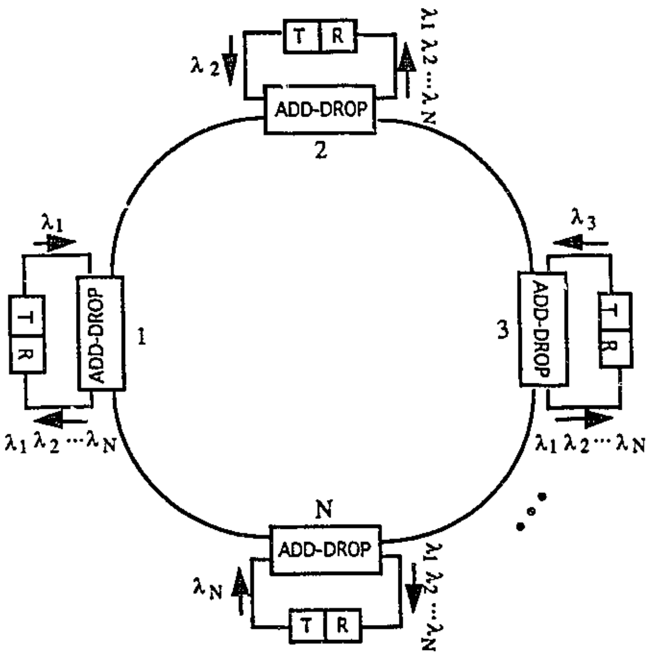


Figure 2.3: *A ring network*

2.1.3 Ring Topology Networks

In designing an optical multichannel ring network one may encounter two problems. The first is the power budget problem as that in a bus network, and the second is that optical signals circulating on a ring must be removed after they finish a complete circle such that there is no interference between current optical signals and old optical signals. To get around these problems, an ADD-DROP device has been proposed to replace splitting taps for WDM ring architectures as shown in Fig. 2.3 [54]. With this device, optical signals can be injected into the ring from each station instead of using slitting couplers which have larger power loss. In the meantime, the ADD-DROP device can filter out residual optical power after one complete circle.

A ring architecture has advantages in saving the number of fibers and being suitable for some particular geographic requirements, as compared with the star networks where each station needs a fiber connected to the central star coupler. The maximum number of active stations in a ring network, however, is much smaller than that in a star network because of splitting loss

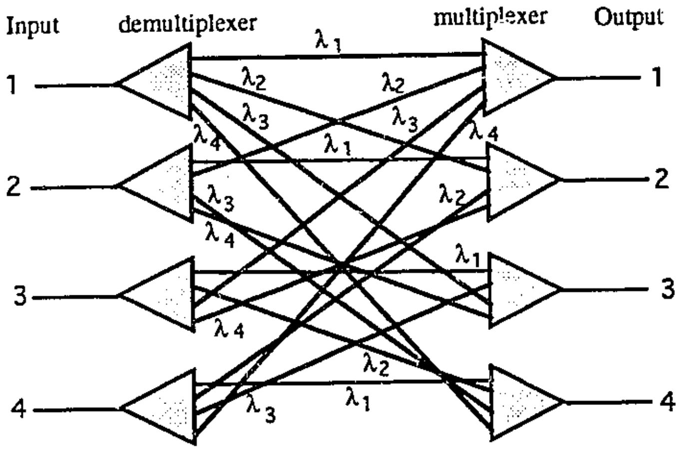


Figure 2.4: *A static routing network*

of taps or insertion loss of the ADD-DROP devices. This makes the ring and bus multichannel networks less popular than the star multichannel networks.

All these three types of architectures are often called *broadcast-and-select*, because signals injected from stations are broadcast to all stations and selected by the destination stations, no matter whether the network is a star, bus or ring. The common problems with these broadcast-and select networks are the splitting loss and the lack of wavelength reuse. They are preferred therefore for local area or metropolitan area applications, but not for wide area networks where hundreds or even thousands of wavelength channels are needed for a large population of users. The routing networks discussed below are considered more suitable for wide area cases.

2.1.4 Static Wavelength Routing Networks

Wavelength routing networks are proposed to make reuse of wavelengths in networks [44]. In a wavelength routing network, a routing node consists of a passive wavelength-selective-structure which is implemented by a fixed-tuned grating multiplexer/demultiplexer. They are static because the multiplexer and demultiplexer are fixed-tuned. Routing can be dynamic, if photonic switches are employed in routing nodes. A 4×4 static routing node structure is shown in Fig. 2.4 [44], where signals arriving at different input ports will be routed onto different output ports according to wavelengths. In Fig. 2.4, input signals at wavelength λ_1 at input 1 will be routed

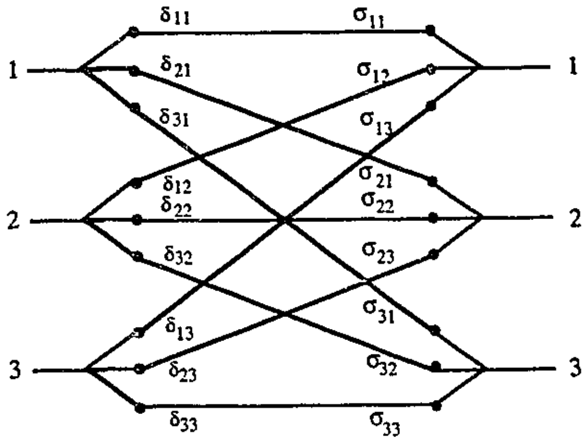


Figure 2.5: *A linear routing node structure*

to output port 1; input signals at wavelength λ_2 at input 1 will be routed to output port 2; and so on. Tunable transmitter and/or tunable receivers are therefore needed at originating and termination nodes respectively to change path patterns, due to the static routing characteristics.

2.1.5 Linear Lightwave Networks

Static routing is inflexible because each path accommodates only one wavelength. A more flexible approach is to allow each incoming signal coupled onto every output ports with preassigned fraction of coefficient such that input ports working on different wavelengths can share the same output port at the same time. This is called linear lightwave structure [44]. A 3 × 3 linear lightwave node structure is displayed in Fig. 2.5, where $\sum_{i=1}^{3} \delta_{ij} \leq 1$ for every j and $\sum_{j=1}^{3} \sigma_{ij} \leq 1$ for every i because of conservation of energy. An arbitrary fraction of the input energy thus can be routed to any output. A simple example can show the advantage of the linear lightwave network. Let us look at a routing network in Fig. 2.6 where each node has a degree of two.

Let each node be a 2 × 2 switch. Assume that station 1 is talking to station 1' through nodes $A-C-E$ and. At the same time, suppose that station 3 wants to talk to station 3'. The path for the connection of station 3 and 3' must include $C-E$. This is impossible because node C is already set to the pattern that input signals from node B will be switched to the link to node D, because of the 2 × 2 switch characteristics. If we replace those switches by linear routing

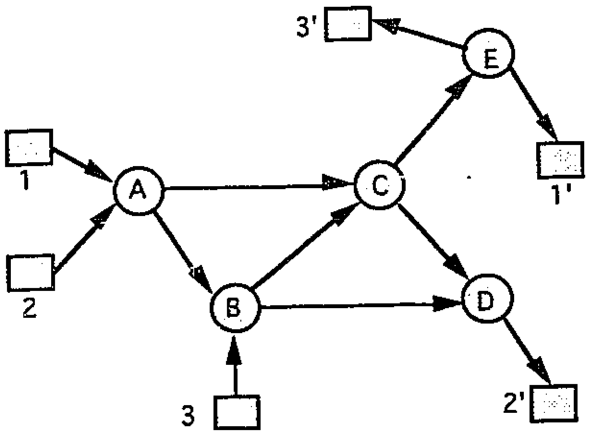


Figure 2.6: *A linear routing network*

devices, signals from node B can thus be coupled to node E and/or node D depending on the setting of the coupling coefficient at node C. Station 3' can therefore receive optical signals from station 3, provided station 1 and 3 use different wavelengths.

2.2 Quasi-All-Optical Multichannel Networks

Quasi-all-optical multichannel networks need O/E and E/O conversions in their intermediate nodes to forward packets to the next node. Typical quasi-all-optical networks are multihop lightwave networks and lightwave networks with E/O and O/E conversions within the network subnet.

2.2.1 Shuffle Multihop Networks

An eight-node shuffle multihop network is shown in Fig. 2.7. Each node has two optical transmitters and two optical receivers, as well as a data source and sink (omitted in the figure). Each transmitter works at a different wavelength. These nodes are interconnected in a perfect shuffle manner. Transmitters of nodes 5 to 8 are connected to receivers of nodes 1 to 4. Four nodes in the most right side are drawn in the figure for convenience and, in fact, they represent nodes 1 to 4 themselves. Packets in a shuffle network are transported in a store-and-forward fashion. A node will forward an arrived packet to the next node, if the packet is not destined to it, and the node will put the arrived packet into its receiving buffer otherwise.

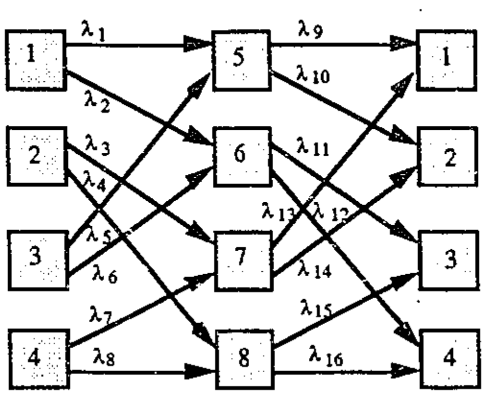


Figure 2.7: *A shuffle multihop network*

The key attraction of shuffle networks is that tunable transmitters and receivers are not needed. This makes the network much easier to build since fast tunable lasers and filters are far more complicated than fixed-tuned devices. The disadvantages of this approach are that the actual network capacity is discounted because packets need to travel for many hops before reaching their destinations and, in each hop, O/E and E/O conversions are performed.

2.2.2 Logic Multihop Networks

Multihop networks actually can be realized in any topology of networks by retuning transmitters and/or receivers at each node. This is also a feature of multihop networks, i.e., reconfiguring the logic topology of the network. Take Fig. 2.1 as an example for $N = 4$. If we use two fixed-tuned transmitters and two receivers in each node, we can obtain a shuffle multihop network with logic topology as shown in Fig. 2.8. In this figure, nodes 1 and 2 can transmit only to node 3 and 4 at wavelengths λ_1, λ_2 and λ_3, λ_4 respectively. Nodes 3 and 4 can transmit only to nodes 1 and 2 at wavelengths λ_5, λ_6 and λ_7, λ_8 respectively.

In [18] a logic multihop application is studied based on a star-coupler WDM network. Some waiting packets in buffers are routed to intermediate nodes which are not their destinations to balance packet queue lengths among all nodes. This strategy proves to be efficient in reducing buffer size for given packet loss probabilities.

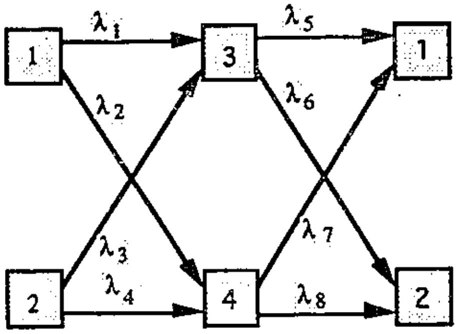


Figure 2.8: *A logic multihop example*

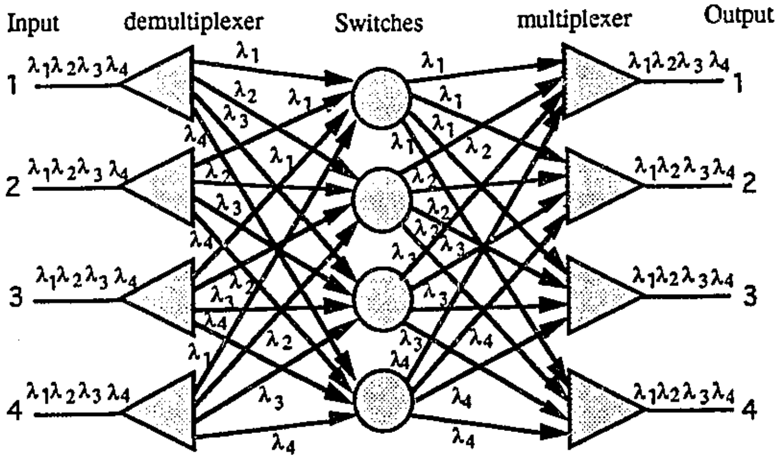


Figure 2.9: *A dynamic routing network*

2.2.3 Dynamic Routing Networks

Static routing networks are all-optical but require either tunable transmitters or receivers at ends to vary paths. A dynamic routing network is proposed to switch traffic internally such that no external tunable transmitter and receiver are needed. The switching function is performed by photonic switches as shown in Fig.2.9.

Packets at input ports can be transported to any of the output ports by using the internal switches. Since practical photonic switches cannot be implemented as all-optical at the current stage, the dynamic approach has therefore a throughput limit caused by electronic processing speed. In Fig. 2.9, all the signals at λ_1 after the demultiplexer are sent to the first 4×4 photonic switch, all the signals at λ_2 are sent to the second switch, and so on. Each output of a switch goes to a different output port, where signals at different wavelengths are multiplexed into a fiber. In this way, an arriving signal at any particular wavelength can be routed to any of the output ports and the signal remains on the same wavelength.

Chapter 3

High-Bandwidth Multichannel Networks: Protocols

The essential problem in designing a communication protocol for high-bandwidth multichannel networks is to find out a distributed control scheme to coordinate data transmission and reception. For example, if station A wants to send a data packet to station B, then at least two things need to be done: 1) station A needs to inform station B of its intention, and 2) station B needs to know where, i.e., on which wavelength channel, to receive the incoming packet. Namely, the transmitter and receiver must be tuned to the same wavelength channel to establish a connection. This is a problem in the absence of a central controller in multichannel networks. In most of the proposed protocols, one or more control channels are used to exchange control information. As discussed before, the single-hop passive star-coupler based architecture is most popular for high-bandwidth multichannel networks because of its efficient power distribution. In this chapter, we review existing protocols found in the literature for single-hop multichannel networks using a passive star-coupler. We categorize protocols for these networks into four classes: single-transceiver random access protocols, multi-transceiver random access protocols, static assignment protocols, and dynamic assignment protocols. They are defined as follows.

- Single-Transceiver Random Access Protocols (ST-RAP): only one transmitter and one receiver are employed at each node for transmission/reception of both data and control packets. Successful transmission/reception of data packets is not guaranteed.

- Multi-Transceiver Random Access Protocols (MT-RAP): successful transmission and reception are not guaranteed, but each node is equipped with more than one transmitters and receivers. Tranceivers for data and control information transmission are normally separated.

- Static Assignment Protocols (SAP): Wavelength channels are statically assigned to stations in space-division-multiplexing and/or time-division-multiplexing fashions. Users transmit data packets in their pre-assigned space (wavelength channels) and/or time (slots), and hence there is no transmission collision or conflict.

- Dynamic Assignment Protocols (DAP): Wavelength channels are dynamically assigned to stations on demand basis. A station is assigned a channel only when it has transmission demand. Each station maintains global traffic information of the network. Transmission and reception of data packets for all stations are scheduled intelligently such that there is no channel collision and destination conflict.

Data packets failed in transmission usually need retransmissions under random access protocols. The failure of transmissions is caused by channel collisions (on both data and control channels), destination conflict, and receiver collision. SAP and DAP protocols do not have collision and conflict problems because transmission/reception is either pre-coordinated or dynamically scheduled, so that collision and conflict can be avoided. Channel collision, destination conflict, and receiver collision are defined respectively as follows.

- Channel collision: Two or more packets are sent on the same channel at the same time, and therefore all are destroyed.

- Destination conflict: Two or more packets from different channels arrive at the same receiver at the same time, and therefore at most one can be received successfully, assuming an optical receiver can tune to at most one wavelength channel at the same time.

- Receiver collision: A data packet cannot be received at a receiver, if this receiver is working on another wavelength channel, even if there is no channel collision and destination conflict.

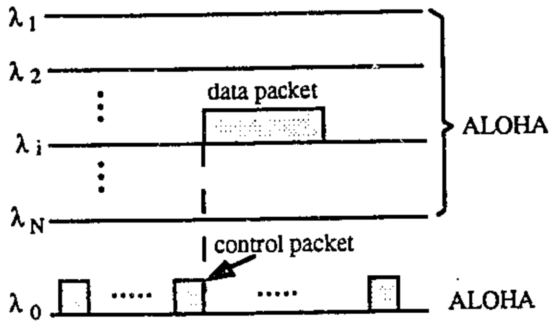


Figure 3.1: *ALOHA/ALOHA protocol*

3.1 Single-Transceiver Random Access Protocols

Random access protocols differ in the number and types of optical devices used, and access strategies on both control channels and data channels. A control channel refers to a channel (a wavelength) which is purely for control purposes. A data channel, on the other hand, is a channel for data transmission and reception. Using more tranceivers, and/or employing both tunable transmitters and receivers at each node can improve transmission efficiency, but on the other hand increases the system cost.

The simplest requirement for random protocols can be that each node is equipped with a tunable transmitter and a fixed-tuned receiver. If a node, say node i, wants to send a packet to another node, say node j, it simply tunes to the receiver's wavelength channel and sends out the packet. Control channels thus may not be needed. This simple scheme, however, has very poor efficiency because all the collisions and conflicts defined above can occur under this protocol. One more practical random protocol using a single-transceiver was proposed in [47]. Instead of using a fixed-tuned receiver, the protocol in [47] employs a tunable receiver as well as a tunable transmitter. Besides, a common control channel is shared by all nodes. Access to the control channel is provided by the use of three random access protocols: ALOHA, slotted ALOHA, and carrier sense multiple access (CSMA). ALOHA, CSMA and N-server switch mechanisms are used for data channels. Various combinations of these protocols were studied in this paper. Under ALOHA/ALOHA protocol, i.e., ALOHA protocols applied to both data and control channels, if node i wants to send a packet to node j, it first transmits a control packet on the

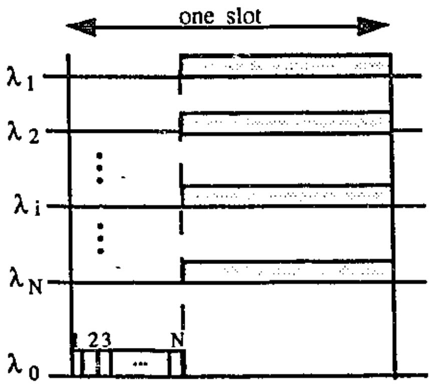


Figure 3.2: *Slotted-ALOHA/ALOHA protocol*

control channel at a random time, and then immediately transmits the data packet on a data channel, say channel k. A control packet here includes source address i, destination address j, and the wavelength channel number k. This process is illustrated in Fig. 3.1. This protocol has low efficiency since both control and data packets may encounter collisions. The slotted-ALOHA/ALOHA protocol is similar except that access to the control channel is through a slotted-ALOHA protocol which is well known to be better than the pure ALOHA protocols mentioned above. Other protocol combinations such as ALOHA/CSMA, CSMA/ALOHA were also discussed in [47]. One disadvantage of these CSMA protocols is that they need fast feedback which is difficult in high-speed networks, because of the relative large propagation delay.

An improved protocol on [47] is studied in [70], where slotted-ALOHA for the control channel, ALOHA and the N-server mechanism for data channels are examined. The idea in [70] is that a node will transmit on a data channel only after it learns that its control packet sent on the control channel is successful. As a result, better efficiency can be achieved as compared to that in [47].

Slotted ALOHA protocols are further extended in [85], and two sets of slotted ALOHA protocols and one set of reservation ALOHA protocols are proposed in this paper. One of the first set protocols can be illustrated in Fig. 3.2. This protocol is parallel to those in [47], in the sense that a packet is immediately transmitted after a control packet transmission. In Fig. 3.2, the length of a slot accommodates a control header and a data packet. The control header on the

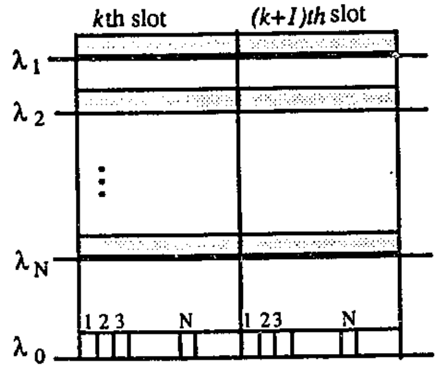


Figure 3.3: *Improved slotted-ALOHA/ALOHA protocol*

control channel is divided into N control minislots. A node that has data backlog will randomly select a control minislot to transmit its control packet (a destination address). If the selected control minislot is the kth minislot, wavelength λ_k will be assigned to the node and the node will transmits its data packet after N minislots. An obvious drawback of this protocol is that the first N minislots on every data channel as well as the time period a slot minus N minislots on the control channel are wasted. To improve the protocol efficiency, a variation as shown in Fig. 3.3 is proposed. Now a slot is reduced to accommodate only one data packet. Control packets are still transmitted on the N control minislots and data channels are identified by the positions of minislots selected. Since there is only a single transceiver at each node, a node that transmits a control packet in slot k transmits its data packet on a data channel in the consecutive slot, i.e., slot $(k+1)$. Based on this protocol, an asynchronous transmission of data packets is also proposed in [85], where a data packet is transmitted immediately after a control packet is transmitted without waiting for one slot. The second set of protocols in [85] parallels those approaches in [70], because both of these protocols employ delayed feedback strategy. Here a data packet can be sent out only after the node learns that the control packet transmission is successful. Reservation ALOHA protocols are designed in [85] for circuit-switched traffic.

Further improvements on the above protocols can be found in [86] [87]. In [86], the wasted channel capacity, as illustrated in Fig. 3.2, is used to transmit control information. Thus the structure becomes a multiple control channel system with control packet transmission over all channels as shown in Fig. 3.4. Now the additional control channel is no longer needed. Control

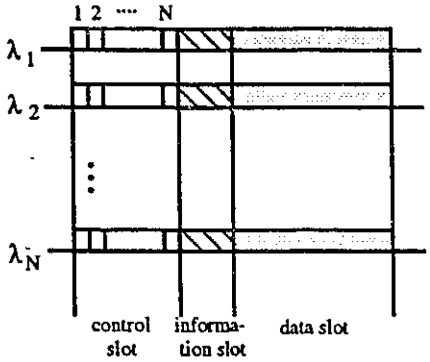


Figure 3.4: *Multiple control channel slotted-ALOHA/ALOHA protocol*

minislots on each channel are pre-assigned to each node. A node wishing to send data packets must transmit its control packet to the destination node via a pre-assigned control minislot. A data packet is transmitted on a randomly selected channel. Data channel collisions exist when more than two nodes select the same channel for data packet transmission. In [87], a strategy is proposed to avoid data channel collisions. Here a slot is expanded to consist of a information slot besides the control slot and data slot as shown in Fig. 3.4. The receiver can feedback information to the sender through this information slot to tell the sender if the control packet is correctly received. In the following data slot, data packet transmissions can thus avoid collisions. This feedback strategy, however, has value only for small propagation delay networks.

In the above mentioned protocols, the tuning time of tunable receivers is assumed to be zero. This is impossible in practical systems. Real tunable tranceivers will introduce tuning overhead. In addition, receiver collisions exist when packets are sent to a busy receiver. In [58], [88], the effect of receiver collisions on network performance is studied. The Receiver Collision Avoidance (RCA) protocol proposed in [58] assumes a non-zero tuning time. Here a tunable transmitter and tunable receiver as well as a control channel is the same as that of slotted-ALOHA/ALOHA protocol. Each slot on the control channel, however, is further divided into control slots and each of these control slots is assigned a fixed wavelength to avoid channel collisions. For example, node i will send a packet to node j via channel k, if its control packet is successfully sent out in the kth control slot. In addition, each node maintains a Node Activity List (NAL) to keep track of history on the control channel for a time period. In this way, nodes can check if a destination node is busy and therefore receiver collisions can be avoided. A node can send out a control packet only if it knows that the destination node is idle.

Single-transceiver random access protocols have low transmission efficiency not only because of channel collisions and destination conflicts but also due to the overhead of control information transmission which is indispensable in single-transceiver systems. A slot needs to be always longer than the length of a data packet to accommodate control overhead. In the following section, we describe some random access protocols using separate data and control tranceivers.

3.2 Multi-Transceiver Random Access Protocols

In [36], another slotted ALOHA protocol and a random TDMA protocol are proposed. Different from other slotted ALOHA protocols, here each node uses one tunable transmitter and multiple fixed-tuned receivers. The feature of this structure is that a node can thus simultaneously receive signals from more than one node, provided channel collisions do not happen. In addition, these protocols assume limited tuning range of tunable transmitters used. Let $T(i)$ and $R(i)$ be two sets of wavelengths, over which node i is able to transmit and receive. The slotted ALOHA protocol here defines that node i with a data packet destined to node j will randomly select a wavelength channel from the intersection of $T(i)$ and $R(j)$ and transmits a data packet on this channel. Clearly, channel collision occurs when other nodes also choose the same channel for data transmission at the same time. The random TDMA protocol avoids channel collision by assigning a different destination to each node for data packet transmission in each slot. This pre-assignment is randomly formed such that assignment can be fair to all nodes. If, however, a node is assigned a destination to which the node has no transmission demand, this assigned slot is wasted. Moreover, maintaining an identical random number at each node for channel selection is difficult in distributed networks.

The protocol in [66] further extends those ALOHA/ALOHA access protocols in [47] by introducing multiple control channels. In this protocol, there are $2N$ channels for N stations. Each node receives control information on its own wavelength and transmits packets on another fixed wavelength. This protocol is simple for implementation but obviously requires twice as many wavelengths as other protocols do. POPSMAC [67] is another ALOHA/ALOHA random access protocol using multi-control channels, but a more flexible scheme is achieved by allowing the number of control channels to be less than the number of data channels. This protocol assumes the following. Each node has a fixed-tuned transmitter and a tunable receiver for data packet transmission and reception. Each node also has a tunable transmitter and a fixed-tuned receiver for control packet transmission and reception. M nodes share one control channel and therefore there are N/M control channels for N nodes. A node transmits a control header packet on the control channel first, and then transmits a data packet on its data channel. There is no data packet collision since each packet is sent on a "Home" data channel. A packet transmission is successful so long as the transmission of a control header packet is successful. As other ALOHA/ALOHA protocols, network wide synchronization is not required.

DT-WDMA [20] also uses separate tranceivers for data and control transmissions. Here each node is equipped with a fixed-tuned transmitter and a tunable receiver for data transmission and reception, and another fixed-tuned transceiver for control information exchange. The same as in other protocols, one wavelength is used as a control channel for all nodes. All channels are slotted into slots. The control channel is further partitioned into minislots which are assigned respectively to each node. Node i sends a control packet only in minislot i. A node with packets to send will first transmit a control packet on the assigned minislot on the control channel, and then transmits a data packet in the following slot. The feature of this protocol is that there is no channel collision since each node transmits data packet on its "Home" channel. Moreover, a node can transmit a data packet immediately in the next slot after sending out a control packet on the control channel without waiting for feedback. The control scheme of this protocol is simple and the transmission efficiency is much better than those ALOHA protocols because channel collisions are avoided. A data packet transmission, however, may not be successful because of destination conflict when another node also sends a data packet to the same destination in the same slot. The unsuccessful packet has to be retransmitted later. Under our definition, this protocol still falls into the class of random access protocol in the sense that the probability of a successful packet transmission is less than 1. With the use of additional tranceivers for the control channel, the slot length can be more compact. Time slots on data channels are thus used to transmit only data packets without control overhead.

Channel No	slot 1	slot 2	slot 3
0	(1,2)	(1,3)	(2,1)
1	(2,3)	(3,1)	(3,2)

Figure 3.5: *Static assignment protocol*

3.3 Static Assignment Protocols

Static assignment protocols avoid channel and receiver contention by statically assigning channels to each node. Each node transmits on an assigned wavelength or slot channel without interfering with others. Transmissions are guaranteed to be successful. Static assignment has the usual drawback of any static assignment technique, i.e., it is insensitive to varying bandwidth requirements and some bandwidth is wasted when channels are assigned to those nodes without bandwidth demand. Also, the access delay in the network is significantly high.

Typical examples of static assignment protocols can be found in [25], [35]. Here channels are slotted into cycles and each cycle is divided into slots. In each slot, a transmission-reception assignment for a source-destination is assigned over each channel. Each node is equally assigned to access channels. After one cycle, each node will be assigned a chance to transmit a data packet to each of the other node. The assignment pattern for each cycle is fixed and repeated in every cycle. An example for two channels and three nodes is shown in Fig. 3.5. We can see from the figure that nodes 1 and 2 are assigned to send to node 2 and 3 respectively in slot 1. In slot 2, nodes 1 and 3 are assigned to send to nodes 3 and 1, and so on. After three slots, every node has got one slot to send to each of the other two nodes. This protocol can have a few variations such as Destination Allocation (DA) and Source Allocation (SA). Under the DA protocol, the number of nodes assigned to transmit to a receiver in one slot is more than one. This is attractive for light traffic networks because a node assigned for transmission may not have data backlog and at least one packet transmission can be successful. Channel collision, however, is introduced since the number of transmitters is more than the number of channels in the network. Under SA protocol, the number of nodes allowed to transmit in a slot equals the number of channels, but each node is allowed to transmit to any other node.

The above work has been extended in [37], where each node is equipped with multiple tranceivers which are assumed to be able to tune over all channels. A scheduling algorithm is designed to minimize the tuning time and packet transmission duration, given a traffic demand matrix. The work in is further extended in [38], where nodes are grouped according to their traffic patterns. Nodes within a group are connected by a local WDM star, and nodes in different groups can communicate via a remote WDM star. An algorithm is also given to determine time-wavelength traffic schedule.

In the protocol in [64], a limited tuning range is considered for optical tranceivers. A network is divided into groups of subnetworks. Each node in this protocol is equipped with a transmitter module which is assumed to be tunable in a small wavelength range. Different transmitter modules form a concatenation of wavelengths group. Each node is also equipped with a receiver module which is a wavelength demultiplexer and can be switched to different wavelength group by fast optical space switches. A full interconnectivity of all nodes in the network is achieved by reconfiguring, i.e., assigning transmission permissions to a different group of nodes in every slot.

3.4 Dynamic Assignment Protocols

In dynamic assignment protocols, channels are assigned to those nodes which have data backlog and, in the mean time, data transmission and reception are scheduled such that channel collisions and destination conflicts are avoided. In all the above mentioned protocols, each node works independently without being aware of traffic situations of other nodes in the network. Dynamic assignment protocols make nodes more intelligent such that they are able to know and remember other nodes' traffic status and therefore can most efficiently schedule traffic to avoid collisions and to increase transmission efficiency.

A dynamic assignment conflict-free protocol is proposed in [17]. The network structure is the same as that in [20], i.e., there are N nodes and $N+1$ wavelength channels. N channels are for data transmission and one is for control information exchange. Arriving data packets at each node are reported in the shared control channel and are recorded as a traffic backlog matrix

Matrix B

0	2	0	5
4	0	1	3
1	2	0	0
0	4	8	0

(a)

Matrix T

	1		
			1
1			
		1	

(b)

Figure 3.6: *Traffic matrix in a dynamic assignment protocol*

B at each node. Fig. 3.6 (a) is an example of the traffic backlog matrix with four nodes and four channels, where an entry $b_{ij} = d > 0$ indicates that node i has d packets destined to node j, and $b_{ij} = 0$ otherwise. Since all arrived packets at every node are announced through the control channel, every node can thus maintain an identical backlog matrix B. B is updated whenever a new packet arrives or a packet leaves the system. An identical distributed algorithm, the Maximum Remaining Sum (MRS) algorithm, is used at each node to make a near-optimal schedule in every slot. The MRS scheduling algorithm is shown to have a very close performance to an optimal algorithm, the System distinct Representative (SDR) algorithm, but a very low computational complexity comparing with the SDR algorithm. The MRS algorithm is used to compute the backlog matrix B to produce an assignment matrix T in every slot. An example of the T matrix is shown in Fig. 3.6 (b). The T matrix in Fig. 3.6 (b) indicates the following transmission assignment:

$$\begin{bmatrix} \text{node } 1 \Longrightarrow \text{node } 2 & \text{node } 2 \Longrightarrow \text{node } 4 \\ \text{node } 3 \Longrightarrow \text{node } 1 & \text{node } 4 \Longrightarrow \text{node } 3 \end{bmatrix}$$

In [18], the work in [17] is further extended to accommodate unbalanced traffic and, in the mean time, to save buffers. Note that in [17] tranceivers in each slot may not be all assigned to transmit or receive packets because there may exist some destinations to which no packet is destined. In [18], these idle tranceivers are used to route some packets from congested queues to some light traffic queues. These packets are temporarily stored in light traffic stations and wait for further forwarding to their destinations in later slots. This protocol is therefore a combined single-hop and multihop protocol under which conflict-free packets are sent to their destinations in one hop while routed packets reach their destinations via multiple hops. Since packet routing is done using idle tranceivers, the network throughput is not affected and in fact is shown a little improved in this paper. By the use of this strategy, this study shows that buffer sizes can be greatly reduced for given blocking probability requirements.

Another protocol called Dynamic Allocation Scheme (DAS) is proposed in [23] and its detailed performance is studied in [24]. This protocol is similar to [17] except for the channel assignment algorithm. Here, a destination is randomly selected and assigned to each node. If the node does not have data backlog destined for that destination, another randomly selected destination is again assigned to the node. This procedure is repeated until the node is assigned a destination to which it has packet to send, or all the possible destinations have been checked through. Once a node is assigned a destination, the destination receiver will tune to the wavelength channel for data reception. Comparing to the MRS algorithm in [17], this random selection algorithm has a higher complexity since each random selection in the algorithm needs to be compared to previous selections to make sure that there is no assignment conflict.

In [27], an optical delay line is proposed at the receivers to buffer the collided packets. A receiver extracts a buffered packet when it is not receiving packets from a channel. If the delay line is long enough, all data packets can be finally received by their destination nodes, otherwise packet loss occurs. Channels are dynamically assigned to each node, and destination conflicts are solved at the receivers. This scheme is shown to have a much better throughput than the one in [20]. This feature is achieved under the assumption of an ideal optical buffer (optical delay line). The practical optical buffer, however, introduces unexpected power loss at a splitting tap used to couple power out when light circulates in a fiber cycle. If an optical amplifier is used to increase the power, noise is also introduced and amplified. These limitations make the application of optical buffer not practical in the current stage.

In [52], three classes of traffic are identified. Channels are dynamically assigned to user nodes for connection-oriented traffic in demand-assignment fashion. For datagram traffic, however, nodes still need to contend on control channels. This protocol assumes N nodes with $2N$ wavelength channels, where N channels are control channels. Each node has a fixed tuned transmitter and a tunable receiver for data transmission and reception, and a tunable transmitter and a fixed-tuned receiver for control purposes. Each node is assigned a different control channel. In this way, processing complexity of each node can be reduced in the sense that the information rate on the control channel is much lower than when one control channel is shared by all nodes as described in [20] [17]. This protocol accommodates three classes of traffic and mainly profits connection-oriented traffic. Burst datagram traffic can be accepted randomly on channels not occupied by connection-oriented traffic.

In [63], a protocol called STARNET divides traffic into packet streams and circuit streams. All these streams are generated at each node and transmitted by a fixed-tuned transmitter. Circuit streams are decoded by tunable receivers, while packet streams are transmitted to the next node in a logic way such that an optical ring topology like FDDI can be formed. Since tunable tranceivers are only for circuit-switched packet streams, fast tunable devices are not required under this protocol.

To simplify the end node structure for dynamic channel assignment protocols, the protocol in [56] suggests that traffic assignment and control functions be carried out in the network rather than in end nodes. Thus a channel controller is utilized for slot assignment on each wavelength channel. Channels are partitioned into frames in which end nodes send their transmission requests and data packets, and controllers transmit their slot assignments. Channel collisions are avoided through slot assignments. Since slot assignments are done at controllers, these assignments need to be sent back to end nodes, and thus control overheads are introduced in each frame. Moreover,the idea of using a fraction of slots on data channels to transmit control information is similar to the multi-control channel strategy proposed in [86].

Generally speaking, random access protocols are simple, and the minimum access delay is small, because of immediate transmission of data packets. The drawbacks of the random access protocols are that the network efficiency is low due to transmission collisions and destination conflicts, and they can applied only to local area networks (LANs) because detection of collision suffers from the propagation delay in high speed networks. In single-transceiver random access protocols (ST-RAP), control information and data packets have to be transmitted in sequence, and thus introduce an overhead penalty. ST-RAP protocols have lower network efficiency and longer access delay than multi-transceiver random access protocols (MT-RAP). Static assignment protocols (SAP) are also simple and have better network efficiency than random access protocols for balanced traffic. For unbalanced traffic, however, the efficiency of SAP protocols and the efficiency of random access protocols vary case by case. Also, SAP protocols have moderate access delay since a packet needs to wait for the assigned slot. SAP protocols can be applied to both local and metropolitan area networks (MANs) for balanced traffic. Dynamic assignment protocols (DAP) assign channels to nodes on demand and therefore are most efficient and can be applied to LANs and MANs. DAP protocols, however, usually have higher complexity than random access and static assignment protocols. Also, because DAP protocols need "thinking" even for light traffic, the access delay to the network is larger than random access protocols. These characteristics of different types of protocols are summarized in table 3.4.

	efficiency	complexity	access delay	applications
ST-RAP	low	low	small	LANs
MT-RAP	moderate	low	small	LANs, some for MANs
SAP	moderate	low	moderate	LANs & MANs for balanced traffic
DAP	high	high	moderate	LANs & MANs

Chapter 4

A Dynamic Channel Sharing Protocol

In order to accommodate different types of traffic, CCITT has defined four classes of traffic for broadband integrated services networks [1]. These four classes of traffic are classified according to combinations of three basic parameters, i.e., the packet (or cell) delay, the variable (or constant) bit rate, and the connection mode. Typical representatives of these traffic classes are connection-oriented real-time and connectionless non-real-time traffic. Other classes such as connection-oriented non-real-time traffic and variable bit rate traffic can be easily included into the above two classes of traffic in high-bandwidth multichannel networks, where transmission capacity is not a problem. In this chapter, we simply denote connection-oriented real-time traffic and connectionless non-real-time traffic by R-T traffic and N-R-T traffic respectively. Examples for R-T traffic can be found in video telephony, video conference, video surveillance, video-on-demand. Examples for N-R-T traffic can be messaging traffic, such as video mail and computer data transfer. The switched Multimegabit Data Services (SMDS) support N-R-T traffic. A bearer service capable of integrating these classes of traffic has been drawing much attention in high-bandwidth computer networks.

Various architectures and protocols for high-bandwidth multichannel networks have been studied in recent years as discussed in Chapters 2 and 3. For applications of multiple classes of traffic, we proposed a protocol to accommodate R-T and N-R-T multimedia traffic in [14], where R-T traffic has guaranteed channel allocation, while N-R-T traffic is scheduled for transmission so long as the required channels are available. In [52], three classes of traffic, i.e., connection-oriented with guaranteed bandwidth, connection-oriented without guaranteed bandwidth, and datagram traffic, are considered. This protocol assumes N stations with $2N$ wavelength channels, where N channels are used for control and are assigned to N stations respectively. Time on both data and control channels is divided into equal frames, where each frame is further slotted into m slots on the control channel and n slots on data channels ($n \leq m$). Up to m stations can simultaneously have connections with a station, but at most n of them can transmit data to the station in a TDM fashion. Once a control slot and a data slot are occupied by class 1 traffic, other traffic cannot use these slots. Connections are set up subjected to access contention. Class 3 data packets are sent by randomly selecting one of the m control slots and one of the n data slots in a frame, and hence suffer possible collisions on both data and control channels as well as destination conflict.

In this chapter, we propose a dynamic channel sharing protocol for a high-bandwidth multichannel network supporting the above mentioned two classes of traffic. Wavelengths and slots are dynamically shared by the two classes of traffic and multiple connections. We also establish tradeoffs between the implementation complexity and the network performance requirement in order to choose a transmission coordination scheme with or without traffic scheduling, as well as to determine the number of control channels needed in a practical design. The blocking performance for R-T traffic and the channel utilization performance for N-R-T traffic with and without channel sharing are analyzed and compared.

4.1 Network Description

In constructing a high-bandwidth multichannel network, a star topology with a passive star coupler as shown in Fig.2.1 is most popular because of its efficient distribution for optical power in comparison with bus, ring, and tree topologies [73] [77]. In addition, using tunable laser transmitters and fixed-tuned optical filters gives simplicity in designing an access protocol as transmitters with data backlog can directly tune to receivers instead of informing receivers to be tuned to the transmitters. In this chapter, we design a high-bandwidth multichannel network structure according to these preferences. A logical network model can be shown in Fig. 4.1.

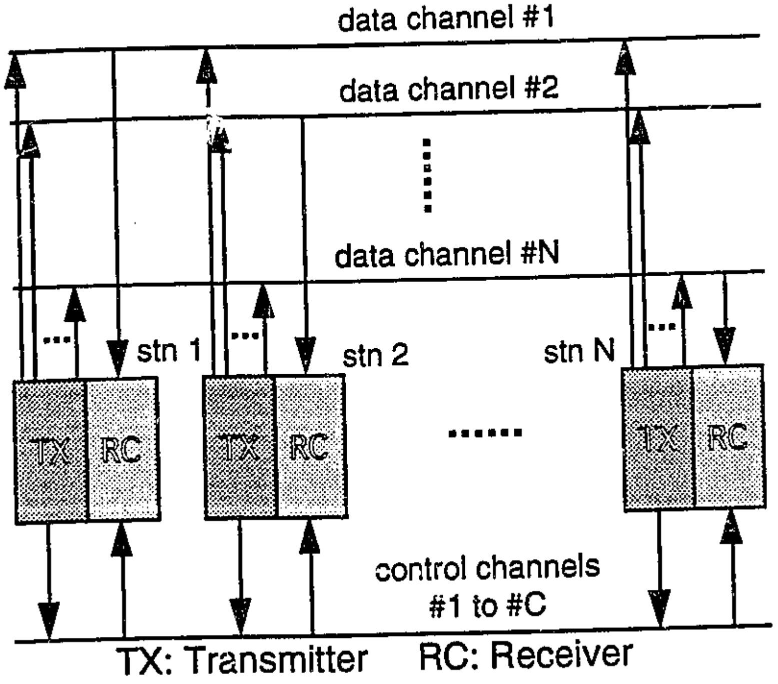


Figure 4.1: *A logical multichannel network structure*

Let there be N stations and $(N+C)$ channels $(1 \leq C \leq N)$. N channels are for data transmission/reception and are assigned in a fixed fashion to N receivers at stations respectively. Let the channel assigned to the receiver at station j be numbered channel j. The other C channels are used for control information exchange (C to be determined). Let each station have a tunable transmitter (TX) and a fixed-tuned receiver (RC) to transmit and receive data. Moreover, we use both time-division-multiplexing (TDM) and subcarrier-multiplexing (SCM) techniques (also see chapter 7) to partition the C wavelength channels into N TDM-SCM control channels in order to accommodate a large scale network. Let L be the number of subcarriers used at one wavelength channel. We divide the N stations into C groups with each group sharing one wavelength channel and the same L subcarriers. To transmit control information, let each station be equipped with a fixed-tuned RF transmitter working at one of the L subcarriers. L of these RF transmitters are connected to a fixed-tuned laser transmitter located in a convenient place in the network. To receive control information from all other stations, let each station be equipped with a wavelength demultiplexer with C outputs, each output followed by a subcarrier demultiplexer with L outputs (RF local oscillators) . Let time on both data and control channels

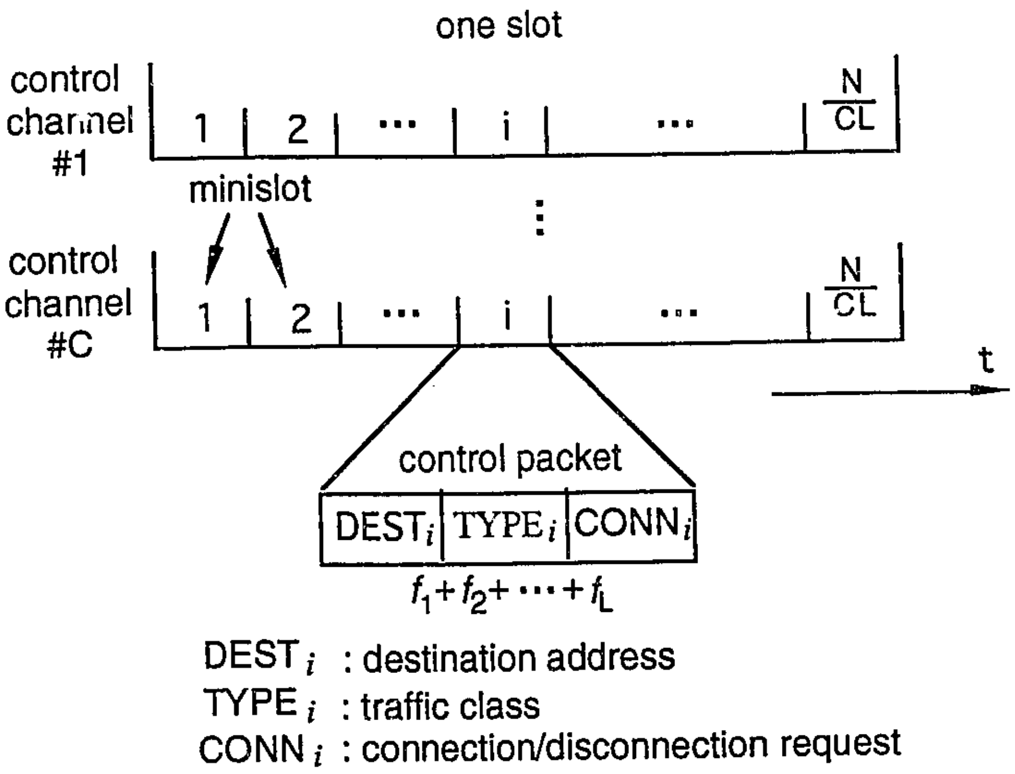


Figure 4.2: *Slot format on control channel*

be slotted into equal size slots. The slot size must be equal to or larger than the packet size (assume equal packet size in the network). Let slots on the C control wavelength channels be further divided into $N/(C \cdot L)$ minislots. Each of these minislots is assigned to L stations which work at different subcarrier frequencies. Thus a total of N distinct TDM-SCM control channels are achieved and are assigned in a fixed fashion to each station in the network. Each station announces the arrival of packets on its assigned TDM-SCM control channel. Fig.4.2 shows the slot format on control channels. Fields of control packets on each slot will be explained later.

4.2 Channel Sharing Policies

After receiving traffic information from control channels, each station can thus maintain identical global traffic information of the network in a set of information matrices as defined below.

4.2.1 Traffic Matrices

- R-T Traffic Status Matrix: $S = \{s(ij)\}_{N \times N}$, $i, j = 1, 2, \cdots, N$, where $s(ij) = 0, 1$. $s(ij) = 1$ indicates that station i has R-T connection to station j, and $s(ij) = 0$ otherwise.
- N-R-T Traffic Arrays: $D_k = \{d(ik)\}$, $(i = 1, 2, \cdots, N$ and $k = 1, 2, \cdots)$ where $d(ik) = j > 0$ indicates that the kth N-R-T packet in the input queue of station i is destined for station j, and $d(ik) = 0$ indicates an empty buffer slot.
- Transmission Array for N-R-T traffic: $T = \{t(i)\}$, $i = 1, 2, \cdots, N$, where $t(i) = j > 0$ indicates that station i is assigned to transmit a N-R-T packet to station j.
- Reception Array for N-R-T traffic: $R = \{r(i)\}$, $i = 1, 2, \cdots, N$, where $r(j) = i > 0$ indicates that station j is assigned to receive a N-R-T packet from station i.

Obviously, the reception array contains redundant information which is completely defined by the transmission array. We introduce it to simplify the operations of the protocol which will be discussed later. Connections need to be set up for R-T traffic to send connection-oriented data streams, while N-R-T traffic is carried over the network on a slot basis and therefore connection set up is not required.

4.2.2 Connection Policy for Transmission

Let each originating station be able to connect to at most one terminating station for R-T traffic. A station with R-T data backlog destined to multiple terminating stations must establish connections to these stations one by one. This policy specifies the following constraint on the R-T traffic matrix:

$$\sum_{j=1}^{N} s(ij) \leq 1, \ i = 1, 2, \cdots, N \tag{4.1}$$

4.2.3 Connection Policy for Reception

Let each terminating station be able to listen to multiple originating stations. Multiple stations send data packets to the same terminating station in a time division multiplexing fashion. This can be easily implemented by tuning each transmitter to the same wavelength channel of the receiver. This feature makes wavelength channels with gigabit bandwidth be shared by multiple stations simultaneously. On the other hand, to guarantee the bandwidth demand for R-T traffic, the number of originating stations connected to the same terminating station must be limited. Let R_{sim} be the maximum number of originating stations that can have simultaneous connections to a terminating station. Furthermore, to avoid that all channels may be occupied by R-T traffic at the same time, while other stations with N-R-T traffic cannot make any transmission, the number of channels occupied by R-T connections in the network needs to be also limited. Let R_{nom} wavelength channels be nominally assigned to R-T traffic (and so $(N - R_{nom})$ channels are nominally assigned to N-R-T traffic). These nominal assignment policies set constraints on the traffic matrix as follows.

$$\sum_{j=1}^{N} \left(\sum_{i=1}^{N} s(ij) \right)^{+} \leq R_{nom} \tag{4.2}$$

$$\sum_{i=1}^{N} s(ij) \leq R_{sim} \quad j = 1, 2, \cdots, N \tag{4.3}$$

$$\sum_{k=1}^{N} (t_k)^{+} = \sum_{k=1}^{N} (r_k)^{+} \leq N - R_{nom} \tag{4.4}$$

where the function $()^{+}$ is defined as $(x)^{+} = 1$ if $x > 0$ and $(x)^{+} = 0$ otherwise, for any real number x.

4.2.4 Dynamic Channel Sharing Policy

For light and regular traffic, equations (4.2) to (4.4) satisfy traffic requirements. In practice, especially with bursty traffic situation, traffic may not be balanced and therefore channels or slots nominally assigned to one class of traffic or connections may not be sufficient, while channels and slots assigned to other stations or connections may be plentiful. To dynamically adapt to unbalanced traffic, we define a dynamic channel sharing scheme with the following three features.

For feature 1, we allow R-T traffic to overflow to channels nominally assigned to N-R-T traffic. The $(N - R_{nom})$ channels nominally allocated to N-R-T traffic can be utilized by R-T connections provided that there is no N-R-T traffic carried on these channels. This policy modifies (4.2) to

$$\sum_{j=1}^{N}\left(\sum_{i=1}^{N}s(ij)\right)^{+} \leq R_{nom} + max\left\{ N - R_{nom} - \Omega,\ 0 \right\} \tag{4.5}$$

where Ω is the number of terminating stations to which there exist N-R-T data destined:

$$\Omega = \left\{ \sum_{j=1}^{N}[r(j)]^{+} \,\middle|\, \forall j = d(ik) > 0; i = 1,2,\cdots,N; k = 1,2,\cdots \right\} \tag{4.6}$$

For feature 2, we allow slot sharing among different R-T connections at a terminating station. Let time slots on a data channel shared by multiple connections be grouped into cycles. The cycle size is nominally set to the number of stations having simultaneous connections to that station. The slot sharing policy allows it to vary dynamically cycle by cycle, from 1 to R_{sim}, depending on the actual number of active stations which are sending data. An originating station, say station i, is assigned a slot, say slot $(n+k)$, in a cycle and can move ahead to slot n in the next cycle if k stations, which have been assigned k slots in the previous cycle before station i, do not claim on control channels that they have packets to send in the next cycle. Similarly, station $(i+1)$, station $(i+2)$, $\cdots$ will move forward in the cycle accordingly. For example, let 5 stations have connections to one destination station. The nominal cycle size is then 5. In a cycle, let stations 2 and 3 have no packet to send, and so they report this information on their control channels. In the following cycle, the cycle size is changed to 3 and each of the 3 slots are respectively assigned to stations 1, 4, and 5. This scheme adapts to real traffic situations and hence can reduce the mean packet delay for R-T traffic.

For feature 3, we allow N-R-T traffic to overflow to channels nominally assigned to R-T traffic as long as these channels are not fully occupied. In addition, since the cycle size for R-T connections changes with traffic situations, slots at a terminating station when cycle sizes become zero can be thus utilized by N-R-T traffic. In other words, in slots when one or multiple stations connected to a terminating station are silent (no packet to send but still have connections), N-R-T packets destined to the same terminating station are allowed to be transmitted to that terminating station. Thus the nominal condition in (4.4) is modified to

$$\sum_{k=1}^{N}(t_k)^+ = \sum_{k=1}^{N}(r_k)^+ \leq \left[N - \sum_{j=1}^{N}\left(\sum_{i=1}^{N} s(ij)\right)^+\right] + \left[\sum_{i=1}^{N} TYPE_i \cdot (1 - CONN_i')\right] \qquad (4.7)$$

where $CONN_i' = 0$ if $CONN_i = (0,0)$, and $CONN_i' = 1$ otherwise. $CONN_i$ indicates the arrival information of R-T data packets and will be defined in the next section. The first term of (4.7) is the number of channels not occupied by R-T traffic and the second term indicates the number of channels over which R-T connections are established but there is no data transferred. With the above channel sharing policies, wavelength channels and slots can be shared dynamically. The implementation of the channel sharing policies will not introduce additional computations except for some simple monitoring on traffic status and control channels. Feature 1 can be achieved by checking the traffic arrays of N-R-T traffic. Features 1 and 2 can be achieved through monitoring the traffic matrix and traffic reports on control channels .

4.2.5 Channel Assignment Policy for N-R-T Traffic

According to the channel sharing policy, slots on all channels can be used to carry N-R-T traffic when these slots are not occupied by R-T packet transmission. The full use of channels will not affect R-T connections since N-R-T packets are transmitted slot by slot. The transmission of N-R-T packets to destinations will be stopped if there exists data destined to the same destination. Since N-R-T connectionless traffic is time insensitive and does not need to set up connections, we manage to schedule it in such a way that a maximum set of N-R-T packets could be selected in every slot for transmission without destination conflict. This strategy, obviously, can increase the network throughput and reduce the mean packet delay. It, however, also introduces scheduling complexity which in return affects the actual network performance. We will study the tradeoff later. Assume that each packet is destined to at most one destination. Then for conflict-free transmissions,

$$t(i) \neq t(j), r(i) \neq r(j) \quad i \neq j, \;\; i,j = 1,2,\cdots,N. \qquad (4.8)$$

4.3 Protocol Description

Arriving traffic information at each station is carried by control packets on the station's assigned control TDM-SCM channel. A control packet from a station, say station i, includes three fields: the destination address field $DEST_i$, the traffic type field $TYPE_i$, and the connection status field $CONN_i$, where

$$DEST_i = \begin{cases} j > 0 & \text{a connection request, or a packet arrived} \\ & \text{at station } i \text{ destined to station } j; \\ 0 & \text{otherwise.} \end{cases} \tag{4.9}$$

$$TYPE_i = \begin{cases} 0 & \text{indicate N-R-T traffic;} \\ 1 & \text{indicate R-T traffic.} \end{cases} \tag{4.10}$$

$$CONN_i = \begin{cases} (0,1) & \text{connection request;} \\ (1,0) & \text{disconnection request;} \\ (1,1) & \text{R-T data transmission;} \\ (0,0) & \text{no R-T data transmission.} \end{cases} \tag{4.11}$$

For example, that $DEST_i = j > 0$, $TYPE_i = 1$, and $CONN_i = (0,1)$ indicates that station i requests a R-T connection to station j, and so on.

4.3.1 Connection and disconnection for R-T Traffic

If an originating station, say station i, wants to set up a R-T connection to a terminating station, say station j, it sends out a control packet with $DEST_i = j$, $TYPE_i = 1$ and $CONN_i = (0,1)$ on its assigned control channel. Since control channels are assigned in a fixed fashion to all stations, station i is easily identified by other stations by recognizing its position on control channels. The request will be accepted if the following non-blocking conditions are satisfied, and will be rejected otherwise.

$$\sum_{i=1}^{N} s(ij) \leq R_{sim}, \quad \text{if} \quad \sum_{i=1}^{N} s(ij) > 0, \text{ or,} \tag{4.12}$$

$$\sum_{j=1}^{N} \left(\sum_{i=1}^{N} s(ij) \right)^{+} < R_{nom}, \quad \text{if} \quad \sum_{i=1}^{N} s(ij) = 0, \text{ or,} \tag{4.13}$$

$$d(ik) \neq j, \ \forall i,k, \quad \text{if} \quad \sum_{i=1}^{N} s(ij) = 0 \text{ and } \sum_{j=1}^{N} \left(\sum_{i=1}^{N} s(ij) \right)^{+} \geq R_{nom} \tag{4.14}$$

where equation (4.12) indicates that though the receiver is busy, the total number of R-T connections at it is less than R_{sim}; equation (4.13) indicates that the receiver is idle and the total number of R-T connections in the network is less than R_{nom}; and equation (4.14) indicates that the receiver is idle and the total number of R-T connections in the network is larger than or equal to R_{nom}, but there are no N-R-T data destined to station j. When station i finishes its transmission of R-T data, it sends a control packet to disconnect the connection to station j on its control slot containing $DEST_i = j$, $TYPE_i = 1$ and $CONN_i = (1,0)$. Station j then releases slots assigned to station i.

4.3.2 Data Transmission for R-T Traffic

After a connection is set up from station i to station j, station i tunes its transmitter to channel j. If station j is listening to multiple stations, the cycle size on data channel j is then increased by one and station i will be assigned a slot which is numbered in the order of the stations involved. Data packets from station i will be sent to station j on data channel j once in a cycle. If the cycle size is 1 at the terminating station, i.e., only station i has connection with station j, station i then transmits data packets on the data channel in continuous slots as long as it has data generated. The cycle size changes dynamically with the number of active stations connected to station j subjected to the upper bound of R_{sim}.

4.3.3 Data Transmission for N-R-T Traffic

N-R-T data packets can be carried over the network with either a scheduling scheme or a non-scheduling scheme, depending on whichever results in better performance without violating the implementation complexity. An example for the non-scheduling scheme is given in [20]. The scheduling scheme employs traffic scheduling before sending out data packets such that packets selected for transmission from every station are destined to different destinations, and so the selected data packets are conflict-free [17] [24]. If the scheduling time cannot be covered in a packet length period, the slot size has to be enlarged to cover the scheduling and therefore the actual network throughput will be reduced. Given a network throughput requirement, there clearly exists a tradeoff in determining if a scheduling or non-scheduling scheme is preferable.

Different scheduling algorithms may result in large differences in scheduling complexity and efficiency. In this thesis, we propose an efficient and simple algorithm, called the K-HOL algorithm, for traffic scheduling. The detailed performance of the K-HOL algorithm will be studied in chapter 6. The K-HOL algorithm computes the first K traffic arrays, i.e., $D_1 \cdots D_K$, to produce a transmission array T and a reception array R. Since K is a variable, it therefore can be selected according to practical complexity constraints. For $K = 1$, the algorithm performance corresponds to non-scheduling scheme. Unlike the general version of the K-HOL algorithm as described in chapter 6, the K-HOL algorithm used in this chapter deals with two classess of traffic, and is presented as follows.

The K-HOL Algorithm

```
Step 1: Let $T := 0$; $R := 0$; $k := 1$;
Step 2: update T:
        $t(i) = d(ik)$ and $r[d(ik)] = i$ if $\{t(i) = 0 \ \& \ r[d(ik)] = 0 \ \& \ CONN_{(i_1,\cdots,i_s)} = (00)\}$,
        $i = ROT, ROT+1, \cdots, N, 1, 2, \cdots, ROT-1$;
Step 3: $ROT := MOD(ROT+1, N)$;
Step 4: repeat steps 2 to 3 until $k = K$
```

Note that the third condition in the algorithm, $CONN_{(i_1,\cdots,i_s)} = (00)$, indicates that N-R-T data packets can be transmitted to a terminating station even if this station has R-T connections with s originating stations which have no data to send in this slot. Note also that the use of the reception array R here obviously simplifies the processing to identify a free receiver. Without R, we need to examine elements of T one by one to find free receivers. The updating of T

starts from the ROTth element of T and thus keeps fair transmission assignment among all stations since ROT is a rotated variable. Once an element in T is assigned a non-zero value, it cannot be updated in the next steps since $(t(i) > 0)$ in this case. This constraint guarantees a FIFO discipline for all packets from a given station destined to the same destination. For each $t(i) = j > 0$, station i is assigned to transmit a packet to station j by tuning its transmitter to channel j, and station j receives the packet on channel j.

4.4 Determining the Number of Control Channels

Let $INFO\_BIT$ be the number of bits of a control packet. Let SLOT be the number of bits of a slot. Let R_T and R_C be transmission rates of data and control information respectively. Given a network size N, and the number of available subcarriers L at a wavelength channel, we can find the minimum time period to accommodate a control packet as $(N/LC) \times (INFO\_BIT) \times (1/R_C)$, where C is the number of wavelength channels. To be accommodated in a slot, this time period must be less than or equal to $SLOT \times (1/R_T)$, the time period of a slot. Consequently, the minimum number of control channels needed is

$$C = \left\{ C^* \middle| \quad \frac{N \times INFO\_BIT}{L \times C^* \times R_C} \leq \frac{SLOT}{R_T} \right\} \tag{4.15}$$

For our protocol, a control packet contains $\lfloor log_2 N \rfloor$ bits for the $DEST$ field, 1 bit for the $TYPE$ field, and 2 bits for the $CONN$ field, i.e., $INFO\_BIT = 3 + \lceil log_2 N \rceil$ bits, where the operation $\lceil x \rceil$ takes the minimum integer that is larger than or equal to x. Let $R_T = R_C$, $L = 10$, and $SLOT = 53 \times 8 = 424$ bits (the size of an ATM cell). Then we need $C = 1$ for $N = 100$, and $C = 4$ for $N = 1000$. If the control transmission rate is less than the data transmission rate, say $R_T = 2R_C$, then we need $C = 2$ for $N = 100$ and $C = 8$ for $N = 1000$. Fig. 4.3 displays the number of control channels needed, varying with N and R_T/R_C. These results show that the number of control channels can be far less than the number of data channels, if both TDM and SCM techniques are used on control channels. SCM is a mature technique and has fast tuning speed and low cost [44].

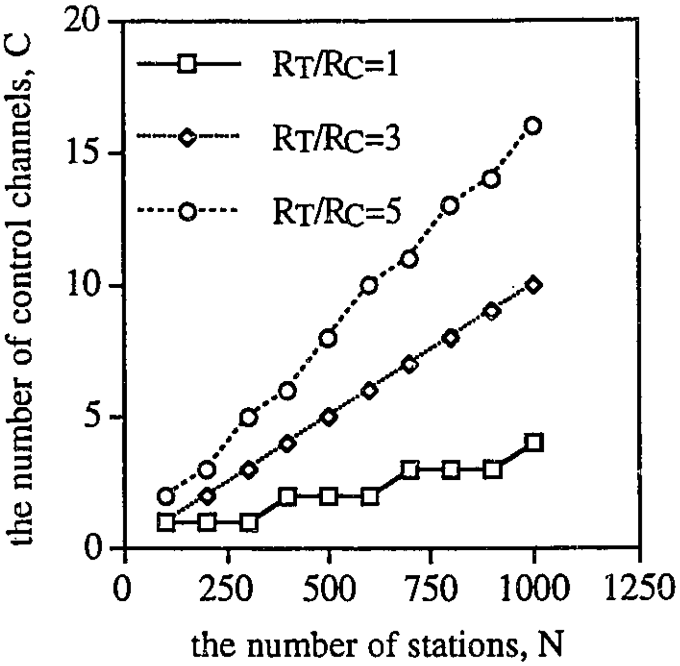


Figure 4.3: *The number of control channels varies with N*

4.5 Choosing a Scheduling or a Non-Scheduling Scheme

In this section, we study the tradeoff in choosing a scheduling or a non-scheduling scheme for N-R-T traffic. Let a scheduling algorithm have computational complexity of C_SCH scalar operations and let each scalar operation need W bits of time to execute. For one slot assignment, a scheduling algorithm thus needs $W * C\_SCH$ bits to be finished. Let PKT be the number of bits of a packet. Let $TUNE$ be the number of bits needed for a tunable filter to tune from one wavelength channel to another. A slot size needs to accommodate the packet size and $TUNE$, or the scheduling time period, whichever is larger, i.e.,

$$SLOT = \max\left\{PKT + TUNE, \quad W * C\_SCH\right\} \tag{4.16}$$

Let THR_SCH be the maximum network throughput under a scheduling scheme. Since tuning time for both of the two schemes cannot be avoided (unless multiple data transmitters are equipped), the actual network throughput under the scheduling scheme, denoted as THR_SCH^* will be discounted if $W \times C\_SCH/R_C > (PKT + TUNE)/R_T$. It is given by

$$THR\_SCH^* = THR\_SCH \times min\left\{\frac{(PKT + TUNE) \times R_C}{SLOT \times R_T}, \ 1\right\} \tag{4.17}$$

For the K-HOL algorithm, the computational complexity is found in [11] as

$$C\_SCH = (4K - 2)N + 2K - 2 \quad \text{scalar operations} \tag{4.18}$$

Fig. 4.4 shows the ideal throughput THR_SCH and the actual throughput THR_SCH^* under the K-HOL algorithm varying with different K and N for $N = 50$, $W = 1$, $R_T = R_C$, $TUNE = 0$ (refer to chapter 6 for the throughput calculation of the K-HOL algorithm). Note that $K = 1$ corresponds to a non-scheduling algorithm and the throughput is around 0.63. The ideal

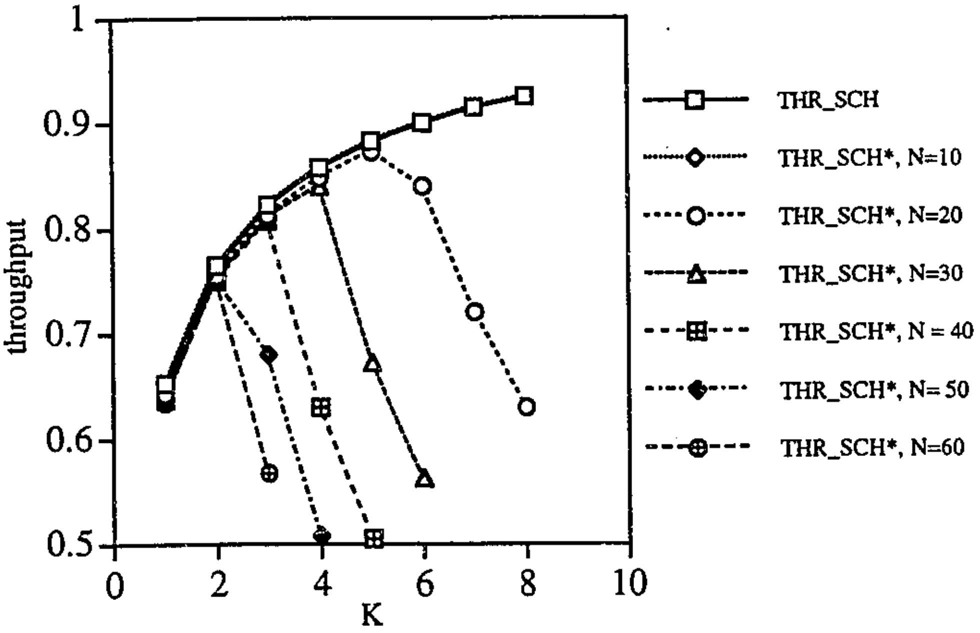


Figure 4.4: *Tradeoff between scheduling and non-scheduling schemes*

throughput THR_SCH is insensitive to network sizes N and is shown by the upper most curve. The actual throughput THR_SCH^* is shown in the lower part of the figure for different N. We can see from the figure that the scheduling scheme with the use of the K-HOL algorithm can greatly improve the network throughput for a certain range of K and N. For example, the throughput can achieve 0.83, an increase of 32% comparing with non-scheduling case, for $K = 3$ and $N = 40$. For $N = 10$ there is no overhead penalty, i.e., THR_SCH=THR_SCH^*, for $K \leq 8$ in the figure. As K and N become large, however, the actual throughput will decrease due to the introduced computational complexity. The scheduling becomes meaningless when the actual network throughput becomes less than 0.63, the maximum network throughput for non-scheduling scheme.

4.6 Network Performance Analysis

Let traffic at each station be modeled as identical Markov Modulated Bernoulli Processes (MMBP) with three states representing the two classes of traffic and an idle state: state 1 denotes R-T traffic, state 2 denotes N-R-T traffic, and state=0 indicates an idle period of a station. We assume that a failed R-T connection setup will not be retried because of the real-time nature. The transition diagram of the three-state Markov process is depicted in Fig. 4.5, where p_{ij} $(i, j = 0, 1, 2)$ is the transition probability from state i to state j. The time period in each state obeys a geometric distribution with parameter $(1 - p_{10} - p_{12})$, $(1 - p_{20} - p_{21})$, and $(1 - p_{01} - p_{02})$ respectively. Let p_r and p_d be the probabilities that a R-T and a N-R-T data packet are generated in a slot in state 1 and state 2 respectively. Let π_I, π_r and π_d denote the steady state probabilities of the three states. They can be found by solving the following matrix equation:

$$\begin{cases} \mathbf{\Pi_x} = \mathbf{P}^T \cdot \mathbf{\Pi_x} \\ \pi_I + \pi_r + \pi_d = 1 \end{cases} \tag{4.19}$$

where $\mathbf{\Pi_x} = \{\pi_I, \pi_r, \pi_d\}^T$, and $\mathbf{P} = \{p_{ij}\}_{3\times3}$. The average rates of R-T traffic and N-R-T traffic are then given by $p_r\pi_r$, and $p_d\pi_d$ respectively. For simplicity, let $p_r = p_d = 1$ in the following analysis.

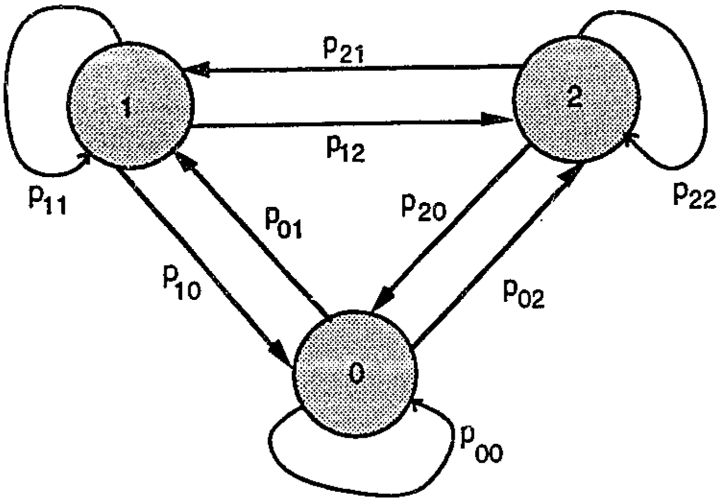


Figure 4.5: *Transit diagram of a three-state Markov process*

Because of the blocking, the actual arrival rate of R-T traffic to the network will be decreased to $\pi_r(1-P_B)$, where P_B is the blocking probability. As P_B is unknown, it will result in cumbersome iterations in calculating the actual number of existing R-T connections as well as the final P_B. In the following section 4.6.1 and section 4.6.2, we will ignore P_B. Our simulations show that this is a good approximation for reasonable range of P_B. For large P_B, the analytical results give upper bounds.

4.6.1 Mean Packet Delay of R-T Traffic

We define the packet delay as the time period between the time when a packet arrives at a station and the time it reaches its destination. Assume each packet arrives at a station at the beginning of a slot. Let D_{RT} and D_{max} be the mean packet delay and the maximum packet delay in slots for R-T traffic. Assume the propagation delay is identical among all stations in the network and let it be D_P. The maximum packet delay can be counted as 1 slot for traffic report, $R_{sim}-1$ slots before transmission, and D_P slots of propagation, i.e.,

$$D_{max} = D_P + R_{sim} \quad \text{(slots)} \tag{4.20}$$

Clearly D_{max} is bounded since R_{sim} is the maximum number of stations that can be connected to a station simultaneously. A R-T packet may not suffer the maximum delay because of the dynamic slot sharing among all connections to a station. Let $Z(k)$ be the probability that there are k simultaneous connections at a station ($k = 1, 2, \cdots, R_{sim}$). In practical situations, many-to-one connections often take place at some hot-spot stations to which multiple originating stations intend to set up connections. Let a station, say station i, be such a hot-spot station. Let γ_i be the probability that any of other stations with R-T traffic wants to set up a connection with station i. The probability that k stations request R-T connections to station i, denoted as $P(k)$, is

$$P(k) = \sum_{n=k}^{N-1} \begin{pmatrix} N-1 \\ n \end{pmatrix} \pi_r^n (1-\pi_r)^{N-n} \begin{pmatrix} n \\ k \end{pmatrix} \gamma_i^k (1-\gamma_i)^{n-k} \tag{4.21}$$

$Z(k)$ thus can be given by

$$Z(k) = \begin{cases} P(k), & k < R_{sim} \\ \sum_{i=R_{sim}}^{N} P(i), & k = R_{sim} \end{cases} \tag{4.22}$$

The mean number of connections at station i, denoted by $\overline{k}$, is then

$$\overline{k} = \sum_{k=1}^{R_{sim}} k \cdot Z(k) \tag{4.23}$$

The mean packet delay is therefore equal to

$$D_{RT} = D_P + \overline{k} \quad \text{(slots)} \tag{4.24}$$

Fig. 4.6 shows D_{RT} varying with the traffic load of R-T traffic π_r for $N = 100$, $R_{sim} = 10$, and

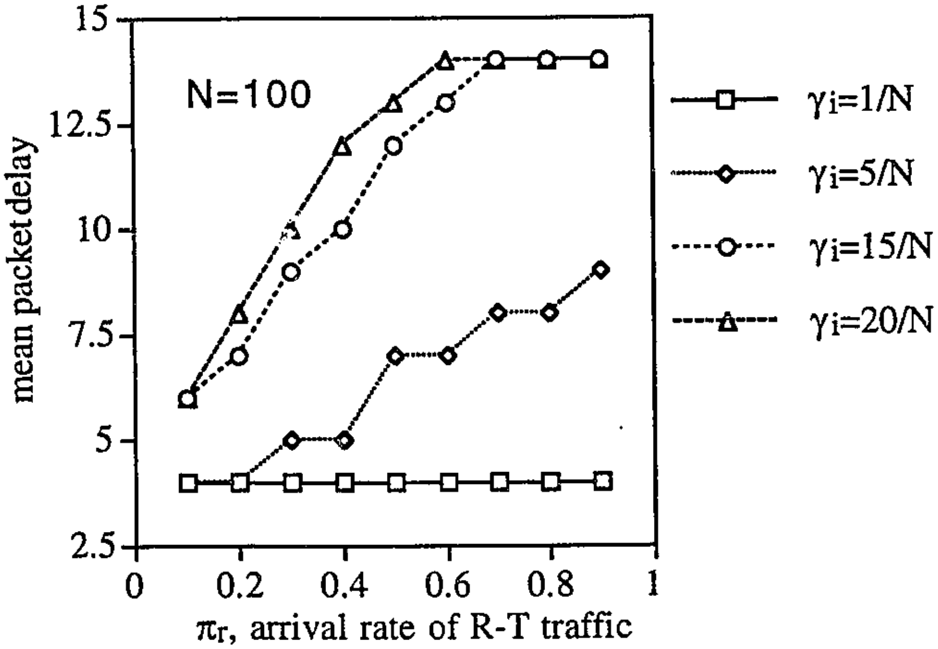


Figure 4.6: *Mean packet delay for real-time traffic*

$D_P = 4$ slots. For $\gamma_i = 1/N$, i.e., a uniform traffic distribution, the mean number of connections to station i in every slot is less than 1 in our example and therefore the mean packet delay is the propagation delay. As γ_i increases, more R-T traffic goes to the hot-spot station, and the mean number of connections increases. For $\pi_r = 0.7$ and $\gamma_i = 20/N$, the mean number of connections to station i is about 10, the same as R_{sim}, and the mean packet delay is hence 14 slots, which is the delay upper bound (D_{max}).

4.6.2 Blocking Performance of R-T Traffic

Let a station, say station i, request a R-T connection to another station, say station j. The request will be blocked if either no channel is available or station j is fully occupied. To compare with protocols without channel sharing, we will study blocking performance under the following three schemes:

- *scheme 1*: R-T traffic is not allowed to overflow to channels nominally assigned to N-R-T traffic, and $R_{sim} = 1$, $R_{nom} \geq 1$;
- *scheme 2*: the same as scheme 1 except that R-T traffic is allowed to share channels nominally assigned to N-R-T traffic, provided these channels are not currently demanded by N-R-T traffic;
- *scheme 3*: the same as scheme 2 except that $R_{sim} > 1$, i.e., a terminating station is allowed to connect to multiple originating stations for R-T traffic;

A connection request under scheme 1 will be blocked if there existed R_{nom} connections in the network, or an originating station has connected to station j. The connection request under scheme 2 will be blocked if 1) there have existed R_{nom} or more than R_{nom} R-T connections and N-R-T packets destined to station j, or, 2) one originating station has connected to station j; The request under scheme 3 will be blocked if 1) there have existed R_{nom} or more than R_{nom} R-T connections and N-R-T packets destined to station j in the network, or, 2) R_{sim} originating stations have connected to station j. Let blocking probabilities for these three schemes be denoted by P_{B1}, P_{B2}, and P_{B3} respectively. Let $P(R)$ be the probability that there exist R out of $(N-1)$ stations in the network bearing R-T traffic. Let $P(D)$ be the probability that D out of the R stations address different destinations. Under schemes 1 and 2, at most D R-T connections can be accommodated due to $R_{sim} = 1$, given R stations with R-T traffic. Under the uniform traffic assumption and the fair channel assignment policy among all stations, traffic arrival and departures at every station can be therefore treated as identical and independent with each other. $P(R)$ is thus given by the following binomial distribution:

$$P(R) = \begin{pmatrix} N-1 \\ R \end{pmatrix} \pi_r^R (1-\pi_r)^{N-R-1}, \quad R = 0, 1, 2, \cdots, N-1. \tag{4.25}$$

Given R, $P(D|R)$ equals the probability that R balls tossed randomly into N boxes occupy D boxes and can be found in [34] as:

$$P(D|R) = \begin{pmatrix} N \\ N-D \end{pmatrix} \sum_{j=0}^{D} (-1)^j \begin{pmatrix} D \\ j \end{pmatrix} \left(1 - \frac{N-D+j}{N}\right)^R, \quad D = 0, 1, 2, \cdots, R. \tag{4.26}$$

Unconditioning on R, we have,

$$P(D) = \sum_{R=D}^{N-1} P(D|R)P(R), \quad D = 0, 1, 2, \cdots, N-1. \tag{4.27}$$

P_{B1} can be thus formulated as

$$P_{B1} = \sum_{D=1}^{R_{nom}-1} P(D)\frac{D}{N} + \sum_{D=R_{nom}}^{N-1} P(D), \tag{4.28}$$

where D/N is the probability that station j is occupied by one of the existing D connections. The second term is the probability that there exist R_{nom} R-T connections in the network.

To find P_{B2}, we divide the domain of D into two ranges: $(1 \leq D \leq R_{nom} - 1)$ and $(R_{nom} \leq D \leq N - 1)$. For the first range, all D R-T connections can be accepted. The probability that a new request will be blocked, if station j is occupied by one of the D connections, is equal to the first term of P_{B1}. For the second range, at least R_{nom} R-T connections can be accepted in the network. Any of the rest $(D - R_{nom})$ R-T connections can be accepted only if there does not exist N-R-T data in the network addressed to the same destination. Let G_2 be the probability that a destination is addressed by N-R-T data in the network under scheme 2. G_2 will be determined in the next section. Let $\Phi(M, D, G_2)$ be the probability that $(M - R_{nom})$ out of the $(D - R_{nom})$ R-T connections have been accepted in the network. $\Phi(M, D, G_2)$ is given by

$$\Phi(M, D, G_2) = \begin{pmatrix} D - R_{nom} \\ M - R_{nom} \end{pmatrix} (1 - G_2)^{M - R_{nom}} {G_2}^{D-M}, \qquad M = R_{nom}, \cdots, D \tag{4.29}$$

Including the blocking probability for the first range, P_{B2} can be found as

$$\begin{aligned} P_{B2} &= \sum_{D=1}^{R_{nom}-1} P(D) \frac{D}{N} + \sum_{D=R_{nom}}^{N-1} P(D) \\ &\cdot \left[\sum_{M=R_{nom}}^{D} \Phi(M, D, G_2) \left(\frac{M}{N} + G_2 \frac{(N - M)}{N} \right) \right], \end{aligned} \tag{4.30}$$

where M/N is the probability that station j is occupied by one of the M connections, and $G_2 \cdot (N - M)/N$ is the probability that station j is addressed by N-R-T data, given it is not occupied by any existing R-T connections.

To find P_{B3}, we divide the domain of R into two ranges: $(R_{sim} \leq R \leq R_{nom})$ and $(R_{nom} < R \leq N - 1)$. Assume that $R_{nom} > R_{sim}$, and thus there is no blocking for $R < R_{sim}$. Since multiple originating stations are allowed to connect to one terminating station, the active number of originating stations in the network must be larger than or equal to the number of active terminating stations. Let $(n+m)$ be the total number of active originating stations and m be the number of active terminating stations. We model the occupancy distribution of these $(m+n)$ connections with the m terminating stations as that n balls are tossed randomly over m boxes, given that each of the m boxes has contained one ball in advance. Let $\Gamma(n,m)$ be the probability that the n balls tossed into the m boxes result in that an indicated box contains at least $(R_{sim}-1)$ balls $(n \geq R_{sim})$, and is given by

$$\Gamma(n,m) = \sum_{i=R_{sim}-1}^{n} \begin{pmatrix} n \\ i \end{pmatrix} (1/m)^i (1-1/m)^{n-i}. \tag{4.31}$$

For the first range of R $(R_{sim} \leq R \leq R_{nom})$, the blocking probability, denoted by P'_{B3}, for a new R-T request is equal to

$$P'_{B3} = \sum_{R=R_{sim}}^{R_{nom}} \sum_{D=1}^{R} \frac{D}{N} \Gamma(R-D,D) P(D|R) P(R), \tag{4.32}$$

where D/N is the probability that station j is one of the existing active terminating stations and $\Gamma(R-D,D)$ is the probability that station j has been fully occupied by R_{sim} originating stations. If station j is not occupied, there is no blocking since R has not exceeded the nominally assigned number of channels in the first range of R.

For the second range of R $(R_{nom} < R \leq N-1)$, we further divide the domain of D into two ranges: $(1 \leq D < R_{nom})$ and $(R_{nom} \leq D < R)$. For $(1 \leq D < R_{nom})$, though the number of active originating stations exceeds R_{nom}, the actual number of channels occupied is only D which is less than R_{nom} in this range. Consequently, the blocking probability for this range of D, denoted by P''_{B3}, is similar to P'_{B3} and is given by

$$P''_{B3} = \sum_{R=R_{nom}+1}^{N-1} \sum_{D=1}^{R_{nom}-1} \frac{D}{N} \Gamma(R-D,D) P(D|R) P(R). \tag{4.33}$$

For the second range of D ($R_{nom} \leq D < R$), some of the D R-T connections may not exist in the network, because of the possible request rejection. Let $\Phi(M, D, G_3)$ be the probability that M out of D connections have been accepted in the network, and it is given by (4.29) by replacing G_2 with G_3. G_3 is the probability that a destination is addressed by N-R-T data in the network under scheme 3, and will be determined in the next section. Similarly, there are $(R-D)$ connection requests which are then modeled as $(R-D)$ balls tossed into D boxes while each of the D boxes contains one ball in advance. Let $\Psi(R, D, M, V)$ be the probability that V out of $(R-D)$ balls are in an indicated group with M balls, and is given by

$$\Psi(R, D, M, V) = \begin{pmatrix} R-D \\ V \end{pmatrix} (\frac{M}{D})^V (1-\frac{M}{D})^{R-D-V}, \qquad V = 0, 1, 2, \cdots, R-D, \tag{4.34}$$

With $\Gamma(V, M)$, $\Psi(R, D, M, V)$, and $\Phi(M, D, G_3)$, the blocking probability for ($R_{nom} < R < N-1$) and ($R_{nom} \leq D < R$), denoted by $P_{B3}^{'''}$, can be formulated as

$$\begin{aligned} P_{B3}^{'''} &= \sum_{R=R_{nom}+1}^{N-1} \sum_{D=R_{nom}}^{R} \sum_{M=R_{nom}}^{D} \Phi(M, D, G_3) P(D|R) P(R) \\ &\quad \cdot \left[\sum_{V=R_{sim}-1}^{R-D} \frac{M}{N} \Gamma(V, M) \Psi(R, D, M, V) + \frac{(N-M)}{N} G_3 \right] \end{aligned} \tag{4.35}$$

where $(M/N)\Gamma(\cdot)\Psi(\cdot)\Phi(\cdot)$ is the probability that station j is one of the existing active destinations and has been fully occupied by R_{sim} connections. $[G_3(N-M)/N]$ is the probability that station j is outside the existing active destinations but there is N-R-T data destined to station j. Summing up $P_{B3}^{'}$, $P_{B3}^{''}$, and $P_{B3}^{'''}$, we find the final equation for P_{B3} as

$$P_{B3} = P_{B3}^{'} + P_{B3}^{''} + P_{B3}^{'''} \tag{4.36}$$

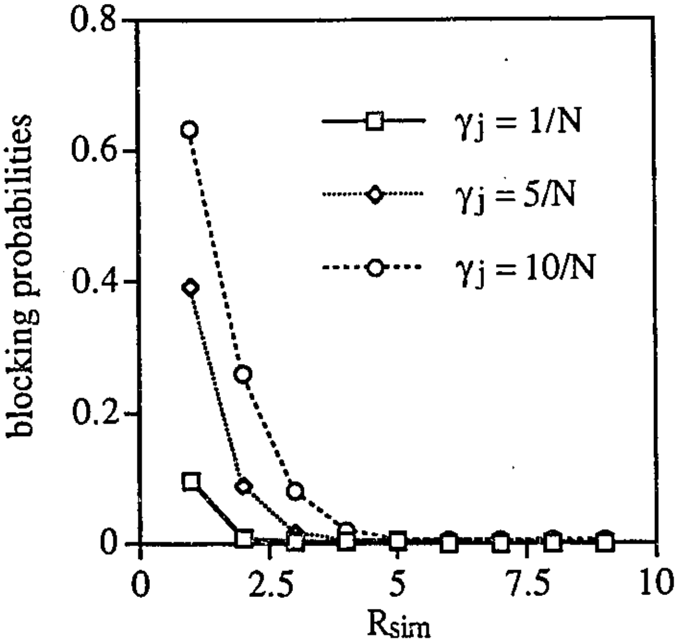


Figure 4.7: *Blocking probabilities at a hot-spot station, N=100,* $\pi_r = 0.1$.

We proceed to consider that station j is a hot-spot station. An originating station requests a R-T connection to station j with probability γ_j (see section 7.1). The blocking probability at the hot-spot station is clearly much higher than at other terminating stations. A new request of a connection to the hot-spot station will be blocked if the hot-spot station is fully occupied by R_{sim} R-T connections. Let P_{BH} be the blocking probability at a hot-spot station. P_{BH} equals

$$P_{BH} = \sum_{R=R_{sim}}^{N-1} \sum_{k=R_{sim}}^{R} \begin{pmatrix} R \\ k \end{pmatrix} {\gamma_j}^k (1-\gamma_j)^{R-k} P(R) \tag{4.37}$$

Fig.4.7 shows the improvement of the blocking performance by allowing a terminating station to connect to multiple originating stations for $N = 100$ and $\pi_r = 0.1$. As the request probability to a hot-spot station increases, the blocking probability becomes large obviously for small values of R_{sim}. For $R_{sim} \geq 5$, however, all blocking probabilities approaches 0 in our example.

4.6.3 Channel Utilization for N-R-T Traffic

The channel utilization for a multichannel network is the percentage of successfully selected packets for transmission out of the number of nonempty queues. For N-R-T traffic under uniform traffic and fair transmissions, it equals the probability that a N-R-T packet in a nonempty queue can be selected for transmission in a slot. Let this probability be ψ_i ($i = 1, 2, 3$ for the three schemes as defined above). The channel utilization reveals the transmission capacity of the network for different traffic loads, and is also needed to obtain G_2 and G_3 as defined in the previous section. In chapter 6, the network throughput under saturation traffic load for the K-HOL algorithm will be found. Here we consider a general traffic load. Because of the fair transmissions and balanced traffic assumption for all queues, we can hence decompose the N queues into N independent queues each with the same arrival rate π_d and service rate ψ_i. Obviously, ψ_i is a function of π_d.

Recalling the operation of the K-HOL algorithm, packets denoted by traffic arrays D_1, D_2, $\cdots$, D_K are scheduled from D_1 to D_K, where each packet is completely defined by an element of an array. The difficulty here for an arbitrary arrival rate is that the number of non-zero elements in D_k ($k = 1, 2, \cdots, K$) is a random variable. Let $P_i(H_k)$ be the probability that there are H_k non-zero elements in D_k under scheme i. We define N *truncated queues* with parameter k by ignoring the first (k-1) packets of every queue (if a queue has less than k packets, the truncated queue will be empty). Each truncated queue thus has a head packet which is either the kth packet of the original queue, or empty. We denote a truncated queue with parameter k as queue-k. All head packets in the N queue-ks are thus defined by D_k. Let q_k be the probability that a packet will depart from a nonempty queue-k in a slot. Let $\rho_k = \pi_d / q_k$ be the queue utilization factor for queue-k. $P_i(H_k)$ can be expressed as

$$P_i(H_k) = \begin{pmatrix} N \\ H_k \end{pmatrix} (\rho_k)^{H_k} (1 - \rho_k)^{N - H_k}$$
$$H_k = 0, 1, 2, \cdots, N \tag{4.38}$$

We are next to find q_k for $k = 1, 2, \cdots, K$. If queue-$(k-1)$ is nonempty, then $q_k = \psi_i$ because any packet being selected for transmission from the original queue (i.e., queue-1) will result in a packet departure from nonempty queue-k. If queue-$(k-1)$ is empty, then $q_k = 1$ because an arrived packet will immediately leave queue-k and join queue-(k-1). Thus q_k can be formulated as the following recursive equation:

$$\begin{aligned} q_k &= \frac{\pi_d}{q_{k-1}}\psi_i + (1 - \frac{\pi_d}{q_{k-1}}) \\ &= 1 - (1 - \psi_i)\frac{\pi_d}{q_{k-1}} \end{aligned} \tag{4.39}$$

where π_d/q_{k-1} is the nonempty probability of queue-$(k-1)$. It can be easily shown that $q_k \geq \psi_i$:

$$\begin{aligned} q_k &= \frac{\pi_d}{q_{k-1}}\psi_i + (1 - \frac{\pi_d}{q_{k-1}}) \\ &\geq \frac{\pi_d}{q_{k-1}}\psi_i + (1 - \frac{\pi_d}{q_{k-1}})\psi_i \\ &= \psi_i \end{aligned}$$

Using mathematic induction, we proceed to show that $q_{k+1} \geq q_k$ $(k = 1, 2, \cdots)$. For $k = 2$, noting that $\psi_i = q_1$ and $\pi_d/q_1 < 1$, we have

$$\begin{aligned} q_2 &= 1 - (1 - \psi_i)\frac{\pi_d}{q_1} \\ &\geq 1 - (1 - \psi_i) \\ &= q_1 \end{aligned} \tag{4.40}$$

Assuming $q_j \geq q_{j-1}$. For $k = j + 1$,

$$q_{j+1} = 1 - (1 - \psi_i)\frac{\pi_d}{q_j}$$

$$\begin{aligned} &\geq 1-(1-\psi_i)\frac{\pi_d}{q_{j-1}} \\ &= q_j \end{aligned} \tag{4.41}$$

So, $q_{k+1} \geq q_k$. This agrees to the fact that queue-k is empty with higher probability than queue-(k-1). To find ψ_i, we further define the following probability distributions:

- $P_i(J)$: the probability that there exist J R-T connections in the network (J transmitters are occupied by R-T traffic) under scheme i;
- $P_i(L|J)$: the conditional probability that L terminating stations are occupied, given that there are J R-T connections ($L \leq J$ since $R_{sim} > 1$);
- $P_i(M_k)$: the probability that M_k N-R-T packets can be selected for transmission after $D_1, D_2, \cdots, D_k$ are computed with the use of the K-HOL algorithm under scheme i;
- $P_i(V_k|J, M_{k-1}, H_k)$: the conditional probability that V_k out of the H_k non-zero packets in D_k are located at the $(N - J - M_{k-1})$ stations with free transmitters under scheme i, given J, M_{k-1}, and H_k;
- $P_i(U_k|J, L, M_{k-1}, V_k)$: the conditional probability that U_k out of the V_k packets are destined to the $(N - L - M_{k-1})$ terminating stations with free receivers under scheme i.

Given M_{k-1}, there must be $(J + M_{k-1})$ transmitters and $(L + M_{k-1})$ receivers which have been reserved by the J R-T connections and the M_{k-1} N-R-T transmission assignments. There are only $(N - J - M_{k-1})$ transmitters and $(N - L - M_{k-1})$ receivers in the network available for packets represented by D_k. Since a station has a connection set up for R-T traffic with probability $(1 - PB_i)$ ($i = 1, 2, 3$ for the three schemes), the actual arrival rates of R-T traffic become $\pi_r(1 - P_{Bi})$, and $P_i(J)$ is hence given by

$$P_i(J) = \begin{pmatrix} N \\ J \end{pmatrix} [\pi_r(1 - P_{Bi})]^J [1 - \pi_r(1 - P_{Bi})]^{N-J}$$
$$J = 0, 1, 2, \cdots, N. \tag{4.42}$$

$P_i(L|J)$ equals the probability that J balls tossed into N boxes occupy L boxes and is given in [34] as

$$P(L|J) = \begin{pmatrix} N \\ N-L \end{pmatrix} \sum_{j=0}^{L} (-1)^j \begin{pmatrix} L \\ j \end{pmatrix} \left(\frac{L-j}{N}\right)^J$$
$$L = 1, 2, \cdots, J, J \neq 0 \tag{4.43}$$

$P_i(J) = 1$ for $L = 0$ and $J = 0$; $P_i(J) = 0$ for $L > 0$ and $J = 0$. $P_i(H_k)$ is given by (4.38). Let $\Omega(i, j, m)$ be the probability that i balls, each of them occupying a different box in a total of N boxes, result in that and j out of i balls happen to be in indicated m boxes. Then

$$\begin{aligned} &\Omega(i, j, m) \\ &= \begin{pmatrix} i \\ j \end{pmatrix} \frac{m(m-1)\cdots(m-j+1)\cdot(N-m)(N-m-1)\cdots(N-m-i+j+1)}{N(N-1)\cdots(N-i+1)} \\ &= \begin{pmatrix} i \\ j \end{pmatrix} \frac{(N-i)!m!(N-m)!}{N!(m-j)!(N-m-i+j)!} \end{aligned} \tag{4.44}$$

$P_i(V_k)$ can be found as

$$P_i(V_k|J, H_k, M_{k-1}) = \Omega(H_k, V_k, N - J - M_{k-1})$$
$$V_k = 0, 1, 2, \cdots, H_k \tag{4.45}$$

Having $P_i(V_k)$, we can find $P_i(U_k)$ as

$$P_i(U_k|L, H_k, M_{k-1}, V_k) = \begin{pmatrix} V_k \\ U_k \end{pmatrix} \left(\frac{N-L-M_{k-1}}{N}\right)^{U_k} \cdot \left(\frac{L+M_{k-1}}{N}\right)^{V_k-U_k}$$
$$U_k = 0, 1, 2, \cdots, V_k \tag{4.46}$$

$P_i(M_k|J, L, H_k, M_{k-1}, V_k, U_k)$ is the probability that U_k balls tossed randomly into $(N - L - M_{k-1})$ boxes occupied $(M_k - M_{k-1})$ boxes, and is given in [34] as

$$\begin{aligned} &P_i(M_k|J, L, M_{k-1}, H_k, V_k, U_k) \\ &= \begin{pmatrix} N - L - M_{k-1} \\ N - L - M_k \end{pmatrix} \sum_{j=0}^{M_k - M_{k-1}} (-1)^j \begin{pmatrix} M_k - M_{k-1} \\ j \end{pmatrix} \left(\frac{M_k - M_{k-1} - j}{N - L - M_{k-1}} \right)^{U_k} \\ &\qquad W_k = 0, 1, 2, \cdots, N - J \end{aligned} \tag{4.47}$$

Unconditioning on M_{k-1}, H_k, J, L, V_k, and U_k, we obtain $P_i(M_k)$ as follows.

$$\begin{aligned} P_i(M_k) \;=\; & \sum_{J=0}^{N-M_k} \sum_{L=0}^{J} \sum_{M_{k-1}=0}^{M_k} \sum_{H_{k+1}=M_k-M_{k-1}}^{N-J} \sum_{V_k=M_k-M_{k-1}}^{H_{k+1}} \sum_{U_k=M_k-M_{k-1}}^{V_k} \\ & P_i(M_k|J, L, M_{k-1}, H_k, V_k, U_k) P_i(U_k|L, H_k, M_{k-1}, V_k) \\ & P_i(V_k|J, H_k, M_{k-1}) P_i(H_K) P_i(M_{k-1}) P_i(L|J) P(J) \\ & \qquad M_k = 0, 1, 2, \cdots, N \end{aligned} \tag{4.48}$$

Having $P_i(M_k)$, ψ_i can be found as

$$\psi_i = \sum_{H_1=1}^{N} \frac{1}{H_1} \sum_{M_K=1}^{H_1} M_K P_i(M_K) P(H_1) \tag{4.49}$$

where M_k varies with H_1 since M_1 is a function of H_1. Note that ψ_i found here is a function of ψ_i itself. Calculating $P_i(M_k)$ recursively from $k = 1$ to K, we can find ψ_i by iterations for each particular arrival rate π_d and blocking probability P_{Bi}. The probability that a station is addressed by N-R-T data, G_i under scheme i, is thus given by

$$\begin{aligned} &G_i = \frac{1}{N} \sum_{M_K=1}^{N} M_K P_i(M_K) \\ &\qquad i = 2, 3. \end{aligned} \tag{4.50}$$

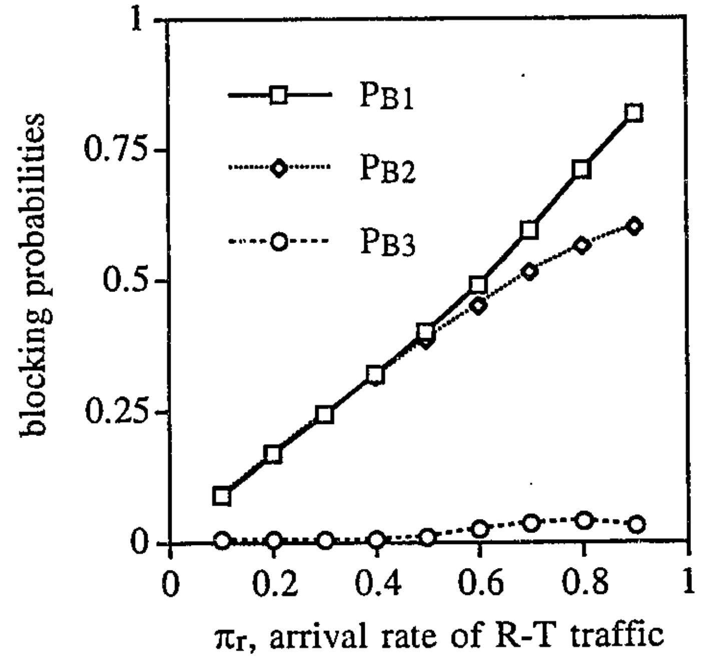


Figure 4.8: *Traffic interactive effect on blocking probabilities of R-T traffic for varying N-R-T traffic load,* $\pi_d = 1 - \pi_r$.

where K is the parameter of the K-HOL algorithm. Substitute G_2 into (4.30) and G_3 into (4.36), we can find P_{B2} and P_{B3} by iterations.

4.6.4 Results

Let $N = 12$, $R_{nom} = 7$, $R_{sim} = 4$, and $K = 3$ in the following numerical examples. Fig.4.8 shows the interactive effect of traffic on the blocking performance, where each station bears either R-T or N-R-T traffic, i.e., $\pi_d = 1 - \pi_r$, and $\pi_I = 0$. With the fully dynamic channel sharing scheme, P_{B3} is much smaller than that of the other two schemes. At $\pi_r = 0.8$, for example, $P_{B1} = 0.71$, $P_{B2} = 0.56$, and $P_{B3} = 0.038$. Scheme 2 results in a deduction of blocking probability from

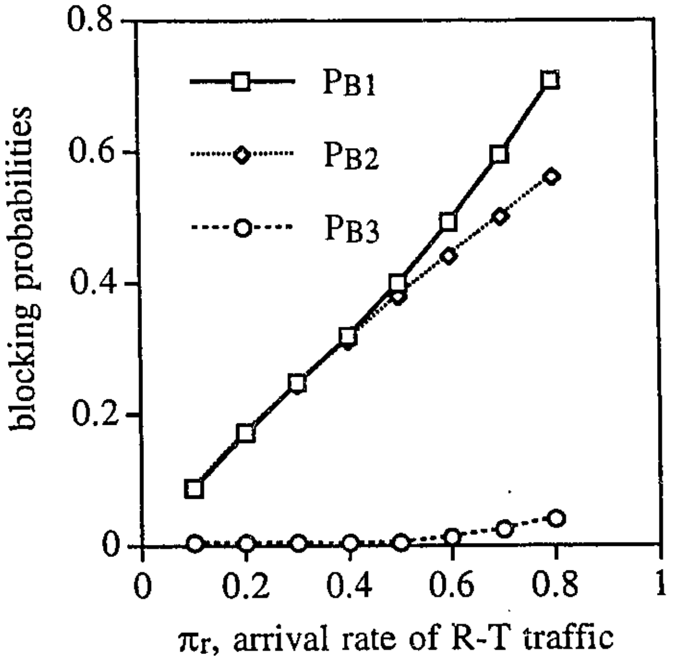


Figure 4.9: *Blocking probabilities of real-time traffic for constant non-real-time traffic load,* $\pi_d =$ 0.2.

that of scheme 1 by 21%, while scheme 3 achieves a deduction from P_{B2} by 94%! Note also that P_{B3} decreases for $\pi_r > 0.8$. This interesting result is due to the following observation: as π_r increases, π_d will decrease and more channels nominally assigned to N-R-T traffic will be occupied temporarily by R-T traffic. New R-T requests addressing those destinations currently used by R-T traffic may still be accepted since $R_{sim} > 1$. For some combinations of π_r, π_d, R_{nom}, and R_{sim}, there thus exists a critical point of π_r, such that the blocking probability will drop down after this point. Our numerical results also show that P_{B3} will go up again when π_r increases extensively. P_{B1} and P_{B2} do not have this feature because $R_{sim} = 1$ for those two schemes. Fig.4.9 shows the case when π_d is fixed at 0.2. The blocking performance in this figure is close to that in Fig.4.8 except that P_{B3} will not decrease in this case because π_d does not change with the changes of π_r.

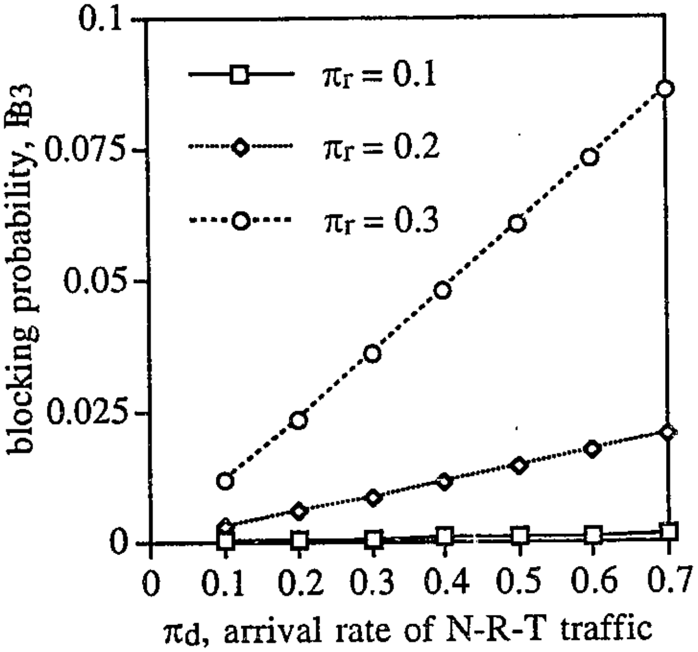


Figure 4.10: *Effect of non-real-time traffic load on blocking performance of real-time traffic,* $K = 3$.

Fig.4.10 shows the effect of N-R-T traffic load on the blocking probability of R-T traffic under scheme 3. As π_d increase, R-T traffic has less opportunities to overflow to channels nominally assigned to N-R-T traffic and therefore P_{B3} increases accordingly. Under heavy N-R-T traffic loads, such as $\pi_d = 0.7$ in our example, R-T traffic is restricted to their assigned channels and the overflow channel sharing strategy loses its advantages for R-T traffic.

Fig.4.11 reveals the channel utilization as a function of π_d with $\pi_r = 0.1$. The straight line denotes the network throughput and the intersections of these curves with the straight line show the maximum network throughputs that the network can support. The three schemes in our example show very close channel utilizations because the blocking probabilities under all schemes for the above selected parameters, though having significant difference, are too small

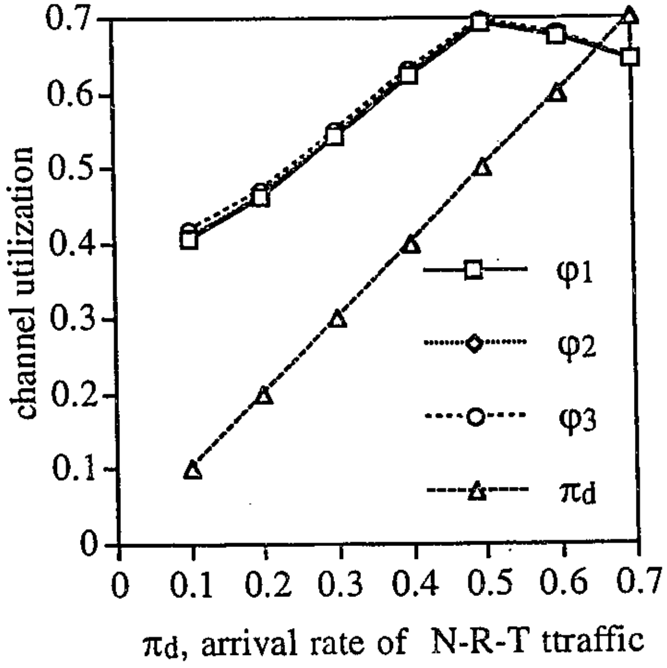


Figure 4.11: *Channel utilization as a function of* π_d *with traffic interaction,* $\pi_r = 0.1$, $K = 3$

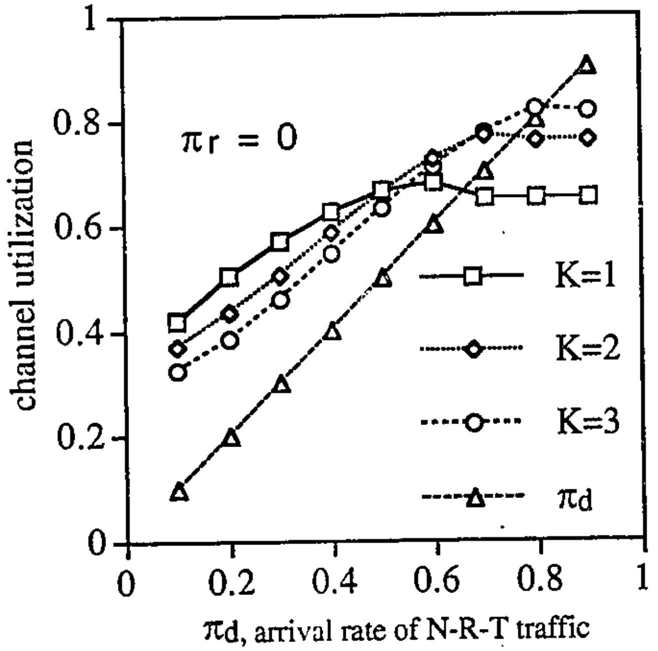


Figure 4.12: *Channel utilization as a function of π_d with only non-real-time traffic,* $\pi_r = 0$

to affect the channel utilization.

Fig.4.12 shows the channel utilization (ψ_i) varying with the arrival rate π_d with only N-R-T traffic ($\pi_r = 0$). The straight line denotes the network throughput which is equal to π_d. Again, the intersections denote the maximum network throughputs that the network can accommodate. For $K = 1$ under the K-HOL algorithm, the maximum throughput is about 0.63 in our example. With $K = 2$ and $K = 3$, the maximum throughputs can be increased by 20% and 32% to 0.75 and 0.83 respectively. Another interesting result is that, for light traffic loads, the channel utilization with a smaller K is a little higher than that with a larger K. This is due to the following negative feedback effect: the larger the K, the more packets can be selected, the higher the channel utilization, and the higher the probability that a queue will be empty. As the number of nonempty queues reduces, on the other hand, the number of packets which can be selected becomes smaller and thus the utilization is reduced. For some values of π_d, therefore, the channel utilization with smaller K can be larger than that with larger K. These critical values are clearly displayed in Fig.4.12. For higher traffic loads, however, the algorithm with larger K shows much larger accommodation capacity because the maximum network throughput can be greatly increased for large K.

4.7 Summary

In this chapter, we aimed at designing an efficient and practical multichannel network. A dynamic channel sharing protocol was proposed to accommodate both connection-oriented real-time and connectionless non-real-time traffic. Channels are first nominally assigned to different classes of traffic and then are allowed to be shared by them. Analysis was given to find the blocking probability and mean packet delay for real-time traffic, as well as the mean channel utilization for non-real-time traffic under different channel assignment schemes. Practical issues such as the minimum number of control channels needed for a given network size and the selection of a scheduling or a non-scheduling scheme were studied. Results show that the proposed channel sharing protocol is very efficient and can greatly reduce the blocking probability of real-time traffic without degrading the network throughput of non-real-time traffic. In addition, traffic burden at hot-spot stations can be largely relieved . These features are achieved through simply monitoring the traffic status without increasing the implementation complexity.

Chapter 5

A Multiconnection Protocol

Traffic projections showed that connection-oriented traffic and real-time traffic would be most important in future communication networks because bandwidth in high-performance fiber networks is no longer a problem as compared to conventional packet-switched networks. Advanced applications such as video communications and other time critical traffic require connection-oriented services, as suggested in CCITT recommendations [1]. We focus on the connection-oriented traffic in this chapter. In various protocols discussed before, an originating station is normally restricted to set up at most one connection with other terminating stations. In practice, an originating station may need to keep more than one connection simultaneously with other terminating stations, even though it does not send packets to every terminating station in every slot. On the other hand, that a station is allowed to set up multiple connections with other stations makes the connection problem complicated. We call it the multiconnection problem in this thesis. Let us consider a scenario as shown in Fig.5.1. A, B, C, and D indicate four stations in the network. Each of the four stations has multiconnections with other stations. A connection from station i to station j is expressed by a line with an end arrow pointing to station j and a bandwidth parameter c_{ij}. Let station A have two connections to stations B and C with bandwidths c_{AB} and c_{AC} respectively. Assume that $c_{AB} = 2c_{AC}$ and therefore connection (A-B) is assigned twice as many slots as that of connection (A-C) in the transmission TDM period at station A as shown in the figure. If station A wants to set up a new connection with station D with bandwidth c_{AD} as shown by a dotted line in the figure, it needs to assign slots to the new connection in its TDM period, and therefore the TDM period at station A will be increased in accommodating an additional connection, i.e., the transmission rate is reduced at station A for

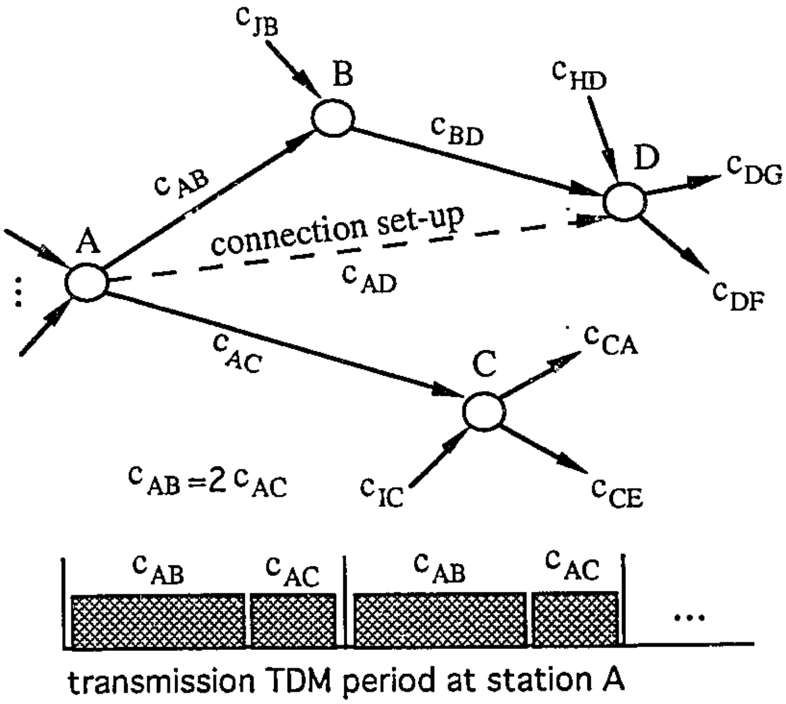


Figure 5.1: *A multiconnection example*

connections (A-B) and (A-C). This, however, causes a chain reaction problem: 1) the reduced rate may no longer be satisfied for connections (A-B) and (A-C), 2) the TDM period at station D needs to be changed to the same as that in station A to maintain an identical transmission rate, 3) as station A and station D have simultaneous connections to stations B, C, G, and F respectively, these involved stations may also need to adjust their reception TDM periods. Consequently, all involved stations may need to change their TDM periods. We can see from this example the sophistication of the multiconnection problem in multichannel networks.

5.1 Network Structure

We use two control channels in the multichannel network structure as shown in Fig.4.1 for $C = 2$. Let there be N stations in the network (N transmitters and N receivers for data transmission). Let there be N data channels, on which data packets are transferred, and two control channels on which control information is exchanged among all stations. Let time on both data channels and control channels be divided into equal slots. Let a slot be able to accommodate a data packet and all overhead control information. An example for the packet overhead is the tuning time which is needed for transmitters to change from one channel to another. Let each station be equipped with a tunable transmitter and a fixed-tuned receiver for data packet transmission. Each receiver is strictly assigned a data channel. Let the channel assigned to the receiver of station i be numbered i. Let each station be equipped with two fixed-tuned transmitters and two fixed-tuned receivers for the transmission of control information. More control channels can be employed for larger scale networks [52] [13]. Station i can transmit packets to station j so long as the transmitter is tuned to channel j. Detailed information for tunable transmitters and receivers can be found in [44].

5.2 Network Parameter Definitions

Let each connection be characterized by two parameters: a bandwidth requirement and a destination address. Let c_{ij} be a normalized bandwidth requirement such that $0 \leq c_{ij} \leq 1$ and $c_{ij} = \lambda > 0$ indicates that station i requires a connection to station j with bandwidth λ, where λ takes the unit of packets per slot. The maximum bandwidth (transmission rate) in the network for a connection is thus 1, i.e., one packet per slot. Let $C = \{c_{ij}\}_{N\times N}$ be a *connection matrix* where an element c_{ij} is as defined above. We further define a set, called a *relation set*, containing all elements that, for any of the elements in the set, there must exist another element in the set such that they are located in the same row or column. A connection matrix can contain more than one relation sets. The following connection matrix is an example for $N = 5$, where elements c_{12}, c_{32}, c_{42}, and c_{34} form a relation set and elements c_{21}, c_{23}, c_{25} and c_{53} form another relation set.

0	c_{12}	0	0	0
c_{21}	0	c_{23}	0	c_{25}
0	c_{32}	0	c_{34}	0
0	c_{42}	0	0	0
0	0	c_{53}	0	0

We also refer to the transmitter and/or the receiver of a station as in a relation set if the transmitter and/or the receiver correspond to an element in the set. The transmitter and the receiver at the same station can be in different relation sets. Taking the above connection matrix as an example, the transmitter of station 2 belongs to the set $\{c_{21}, c_{23}, c_{25}, c_{53}\}$, while the receiver of station 2 belongs to the other set $\{c_{12}, c_{32}, c_{42}, c_{34}\}$. A connection matrix is insufficient to present connection status in the network because it cannot express the time relationship among connections over which packets are transferred in different TDM slots for different connections. Therefore, we further associate each active transmitter with a TDM period, called a T cycle, and associate each active receiver with a TDM period, called a R cycle. We define the following parameters:

- T_i: the total number of slots in a T cycle of station i.
- R_j: the total number of slots in a R cycle of station j.
- T_i^c: the number of connections maintained in the T cycle of station i with bandwidths $c_{il_1} \leq c_{il_2} \leq \cdots \leq c_{il_{T_i^c}}$ respectively.
- R_j^c: the number of connections maintained in the R cycle of station j with bandwidths $c_{i_1j} \leq c_{i_2j} \leq \cdots \leq c_{i_{R_i^c}j}$ respectively.
- W_i^j: the number of slots assigned in a T cycle to the connection with bandwidth requirement c_{ij}. Different connections can be assigned a different number of slots in a cycle.

In addition, we call a connection a T connection if it is used to transmit packets, and a R connection if it is used to receive packets. Two stations thus have a full-duplex connection if there are a T connection and a R connection at one of the two stations addressing the other. Thus c_{ij} is the required bandwidth of a T connection from station i to station j, and $W_i^j / \sum_{m=1}^{T_i^c} W_i^m$ is the assigned bandwidth of the connection. $\sum_{j=1}^{T_i^c} c_{ij}$ is the total bandwidth of all T connections at station i. With these definitions, we specify the following constraints on these parameters.

(1). An assigned bandwidth must be larger than or equal to the required bandwidth, i.e.,

$$\frac{W_i^j}{\sum_{m=1}^{T_i^c} W_i^m} \geq c_{ij}, \quad j \in [1, T_i^c]; \tag{5.1}$$

(2). The total bandwidth of all T connections at a station must be less than or equal to the maximum bandwidth of the network, i.e.,

$$\sum_{j=1}^{T_i^c} c_{ij} \leq 1, \quad i \in [1, N]; \tag{5.2}$$

(3). Slots assigned to a T connection or a R connection are reserved and are not allowed to be used by other connections.

5.3 Bandwidth Allocation

We always allocate one slot to the connection with the smallest bandwidth requirement c_{ij_1}, and allocate $W_i^{j_m}$ slots to the connection with bandwidth requirement c_{ij_m} $(m \in [2, T_i^c])$, where $W_i^{j_m}$ is found as

$$W_i^{j_m} = \begin{cases} \lfloor \frac{c_{ij_m}}{c_{ij_1}} \rfloor + 1, & \text{if } \lfloor \frac{c_{ij_m}}{c_{ij_1}} \rfloor < \frac{c_{ij_m}}{c_{ij_1}} \\ \lfloor \frac{c_{ij_m}}{c_{ij_1}} \rfloor, & \text{otherwise} \end{cases} \tag{5.3}$$

where the function $\lfloor x \rfloor$ takes the largest integer which is less than or equal to x. Though constraints (5.1) and (5.2) imply that we need to check bandwidth constraints for every connection, the following theorem gives that we only need to check for one connection.

Theorem 5.1: The bandwidth requirements for all connections at a station can be satisfied, if the smallest bandwidth c_{ij_1} can be satisfied.

Proof: Given that c_{ij_1} is satisfied. We know $c_{ij_m}/c_{ij_1} \leq W_i^{j_m}$ by the definition of W_i^m in (5.3). For $m \in [2, T_i^c]$,

$$\begin{aligned}
c_{ij_m} &\leq W_i^{j_m} c_{ij_1} && \text{(due to (3))} \\
&\leq W_i^{j_m} \frac{W_i^{j_1}}{\sum_{j=1}^{T_i^c} W_i^j} && \text{(due to (1))} \\
&= \frac{W_i^{j_m}}{\sum_{j=1}^{T_i^c} W_i^j} && \text{(due to } W_i^{j_1} = 1\text{)}
\end{aligned}$$

Q.E.D.

We proceed to study the relation among different T cycles and R cycles.

Lemma 5.1: Let T_A and T_B be the sizes of cycle A and cycle B respectively, and $T_A \neq T_B$. Let both cycles repeat themselves on the same time axis. Then, each slot in cycle A meets with each slot in cycle B once and only once within $T_A T_B$ slots.

Proof: Let slots in cycle A be denoted by $(n_1, n_2, \cdots, n_{T_A})$, and slots in cycle B be denoted by $(m_1, m_2, \cdots, m_{T_B})$. Let cycle A repeat T_B times and cycle B repeat T_A times. We have the following slot arrangement:

$$\begin{array}{c}
(n_1, n_2, \overbrace{\cdots, n_{T_A}), \cdots\cdots, (n_1, n_2,}^{T_B \text{ cycle-As}} \cdots, n_{T_A}) \\
(m_1, m_2, \underbrace{\cdots, m_{T_B}), \cdots\cdots, (m_1, m_2,}_{T_A \text{ cycle-Bs}} \cdots, m_{T_B})
\end{array} \tag{5.4}$$

If we choose a slot in the same position of each cycle-A to form a new cycle, while keep the cycle-Bs unchanged, we have T_A new cycles with each new cycle T_B slots:

$$\begin{gathered}(n_1, n_1, \overbrace{\cdots, n_1), \cdots\cdots, (n_{T_A}, n_{T_A},}^{T_A \text{ new cycles}} \cdots, n_{T_A}) \\ (m_1, m_2, \underbrace{\cdots, m_{T_B}), \cdots\cdots, (m_1, m_2,}_{T_A \text{ cycle-Bs}} \cdots, m_{T_B})\end{gathered} \tag{5.5}$$

Obviously, both arrangements in (5.4) and (5.5) contain the same number of slots, i.e., $T_A T_B$ slots. Let (n_i, m_j) $(i \in [1, T_A], j \in [1, T_B])$ denote a slot combination of a slot in cycle A with a slot in cycle B. In the arrangement of (5.5), obviously, all $T_A T_B$ combinations, i.e., (n_1, m_1), (n_1, m_2), $\cdots$, (n_{T_A}, m_{T_B}), are listed. On the other hand, with the arrangement in (5.4), we have the same set of $T_B T_A$ combinations as in (5.5). Since $T_B T_A$ is the smallest period for these combinations, each combination pair cannot repeat within $T_B T_A$ slots. Since each combination is unique in $T_B T_A$ slots and $T_B T_A$ slots contain the complete set of combinations, any slot in cycle A must meet any slot in cycle B once and only once in $T_B T_A$ slots, given that cycles A and B repeat on the same time axis.

Q.E.D.

Remark: This Lemma implies that two T connections (or R connections) with different cycle sizes at a station must have conflict slots in which more than one packet need to be transmitted (or received) in a single slot.

Lemma 5.2: A nonzero element of matrix C is in a relation set with more than one element, if and only if there exists at least another nonzero element which is in the same relation set and lies in the same row or in the same column.

Proof: Let c_{ij} be a nonzero element in C. Let X be a relation set with set size larger than 1. If there is another nonzero element of C in row i, denoted by c_{ik} $(k \neq j)$, being in relation set X, then $c_{ij} \in X$ by the definition of the relation set. Similarly, if there is a nonzero element of C in column j, denoted by c_{lj} $(l \neq i)$, being in relation set X, then $c_{ij} \in X$. Conversely, if $c_{ij} \in X$, and assume that there is no other nonzero element in row i and in column j, then X must have only one element c_{ij}, which is contradictory to the assumption that X contains more than one element.

Q.E.D.

Lemma 5.3: If two elements in matrix C are in the same row or the same column, their corresponding transmitters and receivers must have the same sizes of T and R cycles.

Proof: Consider two elements c_{ij} and c_{il} in row i. These two elements indicate that station i has two connections respectively with station j and station l. If station i transmits packets to station j with cycle size T_j and transmits packets to station l with cycle size T_l and $T_j \neq T_l$, then there must exist conflict slots in which packets to both destinations need to be transmitted at the same time due to Lemma 5.1. This is contradictory to constraint (3). Therefore the T cycle sizes must be the same, and this results in that the R cycle sizes at stations j and l are the same. Similarly, consider two elements c_{mj} and c_{nj} in column j. The two elements indicate that station j simultaneously keeps two connections with stations m and n. If station j receives packets from station m with cycle size T_m and receives packets from station n with cycle size T_n and $T_m \neq T_n$, then there must exist conflict slots where more than one packet need to be received in the same slot due to Lemma 5.1. This is contradictory to constraint (3). The T cycle sizes at stations m and n must be the same, because of the identical reception cycle sizes.

Q.E.D.

Theorem 5.2: T and R cycles of all transmitters and receivers in the same relation set must have the same size.

Proof: Consider element c_{ij} in relation set X. According to Lemma 5.2, c_{ij} must result in at least one nonzero element c_{ik} or c_{lj} being in the same relation set, where $k \neq j$ and $l \neq i$. According to Lemma 5.3, T and R cycles of transmitters and receivers corresponding to these three elements must have the same size. Since i and j can be any number in $[1, N]$, T and R cycles of all transmitters and receivers corresponding to the same relation set must have the same size.

Q.E.D.

5.4 Connection Acceptance Policies

Let us consider a connection request from station i destined to station j with bandwidth requirement c_{ij}. Before the request arrives, let the number of slots in the T cycle of station i be T_i, and the number of slots in a R cycle of station j be R_j. If the transceiver pair is already in the same relation set, then $T_i = R_j$ due to theorem 5.2. Let slots in the T cycle be denoted as $(s_1, s_2, \cdots, s_{T_i})$ and slots in the R cycle be denoted as $(r_1, r_2, \cdots, r_{R_j})$. Let $s_l = 0$ indicate that the lth slot in the T cycle is free and $r_l = 0$ indicate that the lth slot in the R cycle is free.

5.4.1 Strict-Sense Acceptance

Strict-sense acceptance is defined as that a connection is accepted without changing the slot allocation for existing connections. We consider the following two cases.

Case 1. The transmitter and the receiver are in the same relation set X before the request arrives and therefore $T_i = R_j$. The request can be accepted in strict-sense if the following three conditions are satisfied.

$$\sum_{l=1}^{T_i^c} W_i^l \leq T_i - W_i^j \tag{5.6}$$

$$\sum_{k=1}^{R_j^c} W_{i_k}^j \leq R_j - W_i^j \tag{5.7}$$

$$s_{l_1} = s_{l_2} = \cdots = s_{l_{W_i^j}} = r_{l_1} = r_{l_2} = \cdots = r_{l_{W_i^j}} = 0, \qquad \forall l_k \in [1, T_i] \tag{5.8}$$

where conditions (5.6) and (5.7) indicate that there exist at least W_i^j free slots in the T cycle and the R cycle respectively. Condition (5.8) indicates that W_i^j slots in the T cycle and W_i^j slots in the R cycle are in the the same positions, i.e., the free slots in the T cycle match the free slots in the R cycle. If the three conditions are not satisfied, the request can still be accepted if the following condition can be satisfied.

$$\frac{min\{W_i^{l_1}, W_i^j\}}{W_i^j + T_i} \geq min\{c_{il_1}, c_{ij}\}. \tag{5.9}$$

$$\frac{W_k^{l_1}}{W_i^j + T_k} \geq c_{kl_1}. \qquad k \neq i, c_{kl_1} \in X \tag{5.10}$$

Conditions (5.9) and (5.10) indicate that if conditions (5.6) -(5.8) cannot be satisfied, we can increase T cycles and R cycles by W_i^j slots at all involved transceivers to accommodate the new connection, provided the changed cycle size can be satisfied for the connection with the smallest bandwidth at all involved transmitters due to theorem 5.1.

Case 2. The transmitter is in relation set X and the receiver is in relation set Y, and X and Y are mutually exclusive. If $T_i = R_j$, the strict-sense acceptance conditions are the same as for case 1. If $T_i \neq R_j$, we need combine the two sets into one set with set size $max\{T_i, R_j\}$. The request can be accepted if the following conditions can be satisfied.

$$\frac{min\{W_m^{l_1}, W_n^{l_1}, W_i^j\}}{max\{T_i, R_j\}} \geq min\{c_{ml_1}, c_{nl_1}, c_{ij}\} \quad \forall c_{ml_1}, c_{nl_1} \in X \cup Y \tag{5.11}$$

$$\textit{Condition (5.8) can be satisfied.} \tag{5.12}$$

Condition (5.11) indicates that bandwidth requirements for all connections in the new relation set can be satisfied if the cycle is set to $max\{T_i, R_j\}$. After the cycle size is adjusted, every station in the same relation set has a sufficient number of free slots to accommodate the new connection, but these free slots must match as expressed in (5.8).

5.4.2 Rearrangeable Acceptance

We define rearrangeable acceptance of a request as the following: if there exist sufficient W_i^j free slots in the transceiver's T and R cycles but not all of them match, we can create matching slots by rearranging these slots, i.e., reallocating slots among connections. Rearrangement is under the assumption that all involved transceivers are in the same relation set, and different sets will be combined to the same set as described above otherwise. Let W_i^j free slots in the R cycle be denoted by a set $D = \{d_1, d_2, \cdots, d_{W_i^j}\}$. Let W_i^j free slots in the T cycle be denoted by a set $S = \{s_1, s_2, \cdots, s_{W_i^j}\}$. Suppose k slots in S do not match with D. Let $S^1 = \{s_1^1, s_2^1, \cdots, s_k^1\}$ be a slot set with all slots occupied. Let $S^0 = \{s_1^0, s_2^0, \cdots, s_k^0\} \in S$ do not match with D. The relationship of these sets is depicted in Fig.5.2. We thus need to exchange slots in S^1 with slots in S^0 to make all W_i^j free slots in S matching with free slots in D.

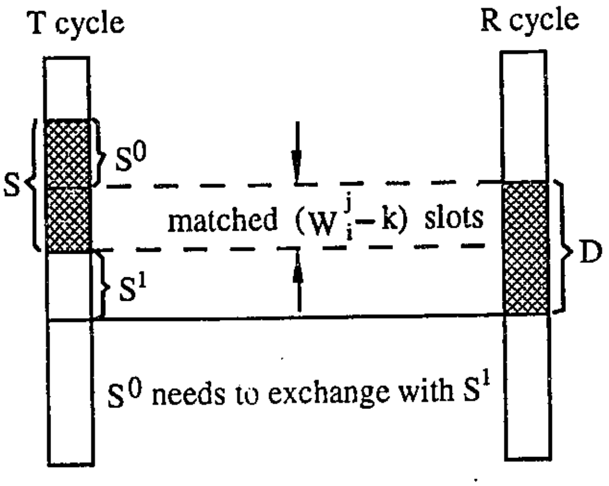


Figure 5.2: *Slot exchange in T cycle to match slots in R cycle*

Lemma 5.4: Slots needed to be exchanged at all transmitters and receivers are in the same set of positions.

Proof: After S^1 exchanges with S^0, k terminating stations which are using slots in S^1 to receive packets from station i need to exchange their slots in positions covered by S^1 with slots covered by S^0 accordingly. Similarly, originating stations which have T connections with these k terminating stations need to exchange their slots in positions covered by S^1 and S^0. This process will repeat until either the reallocated slots are free or all stations in the same relation set finish the exchange of their slots. All of these slots exchanged therefore are in positions corresponding to sets S^1 and S^0.

Q.E.D.

Lemma 5.5: If a transmitter or a receiver needs to exchange its slots, it needs to do so only once. We call this process an irreversible process.

Proof: From Lemma 5.4, we know that each transmitter or receiver exchanges its slots in the same set of slots. If a transmitter or receiver exchange slots twice for a connection request, it implies that there exists a receiver which receives packets from two transmitters in the same set of slots. This is contradictory to constraint 4 as defined before. Therefore slot exchange at each transceiver is at most once, i.e., irreversible.

Q.E.D.

Theorem 5.3: A connection can be rearrangeably accepted if and only if there exist at least W_i^j free slots at both the T cycle at the originating station and the R cycle at the terminating station.

Proof: If there exist W_i^j slots in the originating station and the terminating station, these two stations have enough bandwidth to accommodate the newly requested connection. To match these slots, we need to exchange some slots with some other slots in the T cycle. The slot exchange, however, will result in other transceivers in the same relation set exchanging their slots accordingly. Lemmas 5.4 and 5.5 guarantee that the slot exchange at all transceivers is within the same set of slots and the exchange process is irreversible. Therefore the number of times for slot exchange is bounded and a slot exchange process terminates when the desired slots in an intermediate transmitter or receiver are free, or all the involved transceivers finish their slot exchanges. Conversely, if a connection is rearrangeably acceptable, but the number of free slots in the T and R cycles at the required transceiver is less than W_i^j, the connection will be either rejected, or accepted by enlarging the cycle size. Since both of them are contradictory to the assumption for rearrangeable acceptance, there must exist at least W_i^j free slots in the T and R cycles if the connection is rearrangeably acceptable.

Q.E.D.

5.5 Protocol Description

We consider connection set-up, packet transfer and disconnection between a station, say station i, and another station, say station j, with bandwidth requirement c_{ij}. The connection can be either full-duplex or half-duplex.

5.5.1 Control Channel Signalling

There are two control channels in the network as mentioned in Section 5.2. Let each slot in control channel 1 be further partitioned into N equal minislots assigned strictly to N stations respectively. Let each slot in control channel 2 be further divided into M equal minislots where each minislot can accommodate the address and bandwidth requirement information of a connection (to be discussed below). Station i first sends a minipacket, called a flag minipacket, on its assigned minislot on control channel 1. A flag minipacket from station i contains one

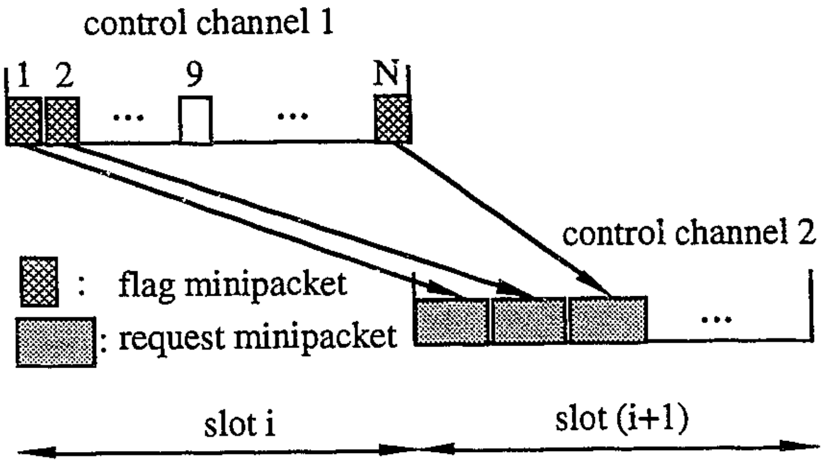


Figure 5.3: *Combined fixed and dynamic control scheme*

field, denoted by α_i, which can be expressed in two bits as follows.

$$\alpha_i = \begin{cases} (0,1) & \text{full-duplex connection request} \\ (1,0) & \text{half-duplex connection request} \\ (1,1) & \text{disconnection request} \\ (0,0) & \text{no request} \end{cases} \tag{5.13}$$

Since minislots on control channel 1 are fixedly assigned to stations, other stations can easily identify the address of a station which send out a flag minipacket by finding the position of the flag minipacket in a slot. After station i sends out a flag minipacket α_i on control channel 1, it follows by sending another minipacket, called a request minipacket, containing a destination address and a bandwidth requirement on control channel 2. Assume that m flag minipackets have been sent out before station i sends its flag minipacket in the same slot. Station i sends a request minipacket in the $(m+1)$th minislot on control channel 2 in the next slot. Denote a request minipacket from station i by $(\beta_i\gamma_i)$ where $\beta_i = j$ is the destination address and $\gamma_i = c_{ij}$ is the bandwidth requirement. Fig.5.3 shows an example of sending flag minipackets and request minipackets. In slot i, stations 1, 2 and N send out three flag minipackets in their assigned minislots, while other stations are silent. In slot $(i+1)$, the three stations send out three request minipackets sequentially on control channel 2.

Here we employ a combined scheme of both static assignment and demand assignment on control channels 1 and 2 respectively. Since a flag minipacket contains only two bits, a large number of stations can be accommodated on control channel 1. Recognizing that not all stations in the same slot need to set up or release a connection, the number of minislots on control channel 2 can be much smaller than the network size N. This combined scheme thus can accommodate more stations as compared to a pure fixed assignment. Let H be the maximum number of minislots on control channel 2 in one slot. In case that the number of requests in one slot is larger than H, the un-served requests are assigned priority and queue up to wait for minislots in the next slot. Control information is thus exchanged among stations dynamically.

5.5.2 Half-Duplex Connection

Station i sends a flag minipacket $\alpha_i = (0,1)$ on control channel 1, and then sends a request minipacket on control channel 2 in the next slot. After receiving the connection request on control channel 2, each station in the network checks whether the request can be accepted according to the rules introduced above. If the request is accepted, each station updates its connection matrix C and related T cycles and R cycles. After a connection is set up, station i can transmit packets to station j in assigned slots in its T cycle whenever it has packets destined to station j. Station j receives packets in the corresponding slots in its R cycle. When station i finishes its packet transmission, it sends a disconnection request on control channel 1 and the destination address on control channel 2. Then all transceivers in the same relation set will reduce their cycle size if there exist free and matching slots in their cycles.

5.5.3 Full-Duplex Connection

Station i sends a flag minipacket $\alpha_i = (1,0)$ on control channel 1, and then sends a request minipacket on control channel 2 in the next slot. A full-duplex connection request is equivalent to two half-duplex connections from station i to station j and *vice versa*. After receiving the full-duplex request on control channel 2, each station examines the acceptance conditions for the two connections. If the request is accepted, two connections are set up and the connection matrix and related T and R cycles will be updated accordingly. Packets are transmitted over the two connections until connections are terminated.

The above protocol is a distributed control protocol. Each station receives control information on control channels and does the same algorithm computation. We can easily modify it to a central control version, where a station is assigned as a master controller. Only the controller needs to receive control information and compute acceptance algorithms. The controller sends back acceptance information to stations using either its assigned minislots on control channels 1 and 2, or an additional control channel.

5.6 Summary

Multiconnection in multichannel networks was shown to be a complicated problem due to the interaction among connections at some shared transmitters and receivers. In this chapter, we found constraints on network parameters and connection policies for the multiconnection problem. Effective schemes for slot allocation for multiple connections at a station were given and conditions for strict-sense and rearrangeable acceptance were obtained. A multiconnection protocol with a combined control scheme using both fixed and dynamic assignments on control channels was proposed.

Chapter 6

Channel Assignment Algorithms

In a WDM multichannel network, packets generated at originating stations will be directed through multiple concurrent channels to their destinations [7]. In digital satellite communication networks, an on-board processor is assumed to switch data packets from upbeams to their destinations through downbeams [91] following certain channel assignment rules. In input-buffered ATM switching systems, packets (cells) from inputs are switched to their desired outputs under the control of a traffic assignment controller [53]. These networks and systems can be in general modeled as a multichannel network where packets arriving at originating stations queue up in input buffers and are assigned for transmission to their terminating stations through multichannels. Channel assignment is therefore an important issue for efficient coordination of transmission and reception of packets. The objective of a channel assignment algorithm in this case is to select as many packets as possible for transmission in every slot subjected to a complexity constraint. To exchange traffic information, one or more control channels are normally employed [20] [66].

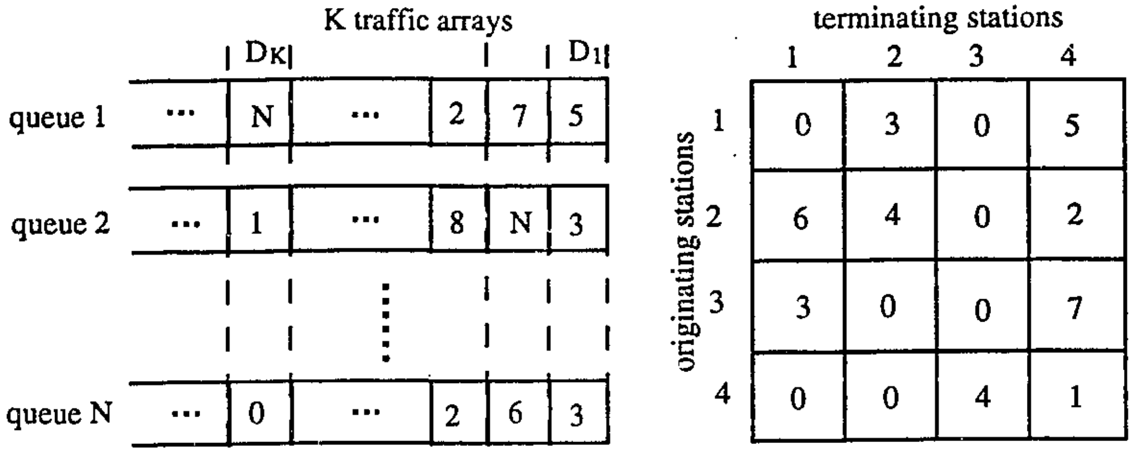
A common problem limiting the channel assignment efficiency is destination conflict, also called the Head-of-line (HOL) blocking in switching systems, which arises when multiple packets from different originating stations are destined to the same terminating station. A few assignment algorithms to solve this problem can be found in the literature. The MRS algorithm [17] is a near-optimal algorithm which selects a set containing as many conflict-free packets as possible for transmission in every slot. The Random Scheduling (RS) algorithm [24] is a heuristic which selects source-destination pairs randomly for possible conflict-free transmission. The SDR algorithm [48] is an optimal algorithm using a back-tracking method. In [51], another optimal algorithm using depth-first search for maximum matchings is proposed. Optimal algorithms are normally more complex than near-optimal or heuristic algorithms, while their performance is found comparable with these heuristics for our applications. We will thus not compare optimal algorithms with our algorithms in this chapter. Three algorithms are proposed and studied in this chapter, namely the K-HOL algorithm, the Dynamic Round Robin (DRR) algorithm, and the Selective Priority (SP) algorithm for channel assignment in multichannel networks, where the K-HOL algorithm is the general version of the K-HOL algorithm discussed in chapter 4. Maximum achievable throughput, mean packet delay, and computational complexity of these proposed algorithms are analyzed and compared with the MRS and RS algorithms respectively.

6.1 Problem Formulation

We still consider the logic multichannel network as shown in Fig.4.1. Channels are divided into data channels, on which data packets are transmitted, and control channels on which traffic information is exchanged among all stations. Let time on all channels be divided into equal slots. Let a slot accommodate a data packet and all overhead control information. Full interconnection is achieved by using either tunable or fixed-tuned transmitters and/or receivers at each station. Thus an originating station can send data packets to a terminating station so long as the involved transmitter and receiver are tuned to the same channel. Packets arrived at each station are stored in a buffer (if the transmitter is busy) waiting to be sent to their destinations following channel assignment policies slot by slot.

Let there be N stations in the network (N transmitters and N receivers). Assume that traffic information is dynamically reported on control channels such that each station maintains identical traffic information of the network. We employ two different data structures as the following traffic matrices and arrays.

- Traffic Arrays $D_k = [d_{k1}, d_{k2}, ..., d_{kN}]^T$ ($k = 1, 2, \cdots$), where an element $d_{ki} = j > 0$ indicates that the kth packet in the input queue of station i is destined to station j, and $d_{ki} = 0$ otherwise.
- Traffic Matrix $B = \{b_{ij}\}_{N\times N}$, where each row of the matrix denotes an originating station



(a). traffic arrays (b). B matrix

Figure 6.1: *Examples of the traffic arrays and the traffic matrix*

and each column denotes a terminating station. An element $b_{ij} = n > 0$ indicates that there are n packets in the queue of station i destined to station j, and $b_{ij} = 0$ otherwise.

- Transmission Array $T = \{t_1, \cdots, t_N\}$, where $t_i = j > 0$ indicates that station i is assigned to transmit a packet to station j, and $t_i = 0$ otherwise.

- Reception Array $R = \{r_1, \cdots, r_N\}$, where $r_j = i > 0$ indicates that station j is assigned to receive a packet from station i, and $r_i = 0$ otherwise.

Ignoring packet sequence, the traffic arrays D_ks and the traffic matrix B contains the same traffic information of the network. Again, we introduce R here to simplify the operations of the algorithms. Fig.6.1 (a) and (b) illustrate the traffic arrays and traffic matrix respectively. To achieve conflict-free transmission and reception, the transmission and reception arrays must satisfy the following constraints.

1. $t_i \neq t_j, \forall i \neq j,\ i, j = 1, 2, \cdots, N$.
   {at most one packet can be sent to a terminating station in one slot}

2. $r_i \neq r_j, \forall i \neq j,\ i, j = 1, 2, \cdots, N$.
   {an originating station can transmit at most one packet in a slot}

A station is equipped with a transmitter and a receiver for data transmission. Given a set of traffic arrays or a traffic matrix, the channel assignment problem is to select elements in D_ks or B to formulate a transmission array T with as many nonzero elements as possible, while keeping computational complexity low.

6.2 The Algorithms

The three algorithms, the K-HOL algorithm, the DRR algorithm , and the SP algorithm, are defined as follows.

6.2.1 The K-HOL Algorithm

The K-HOL algorithm is employed to assign channels to non-real-time traffic in chapter 4. Without considering multiple classes of traffic, we redefine and analyze a general K-HOL algorithm in this chapter. The underlying philosophy behind the K-HOL algorithm is from the following observation. For light traffic, the number of nonempty arrays D_is is small and thus the performance difference between an efficient algorithm and an inefficient algorithm can be neglected. For heavy traffic loads, however, only a small number of D_is could have significant contribution to satisfied performance. With only part of the traffic information, a channel assignment algorithm with lower complexity can be expected. The algorithm is described as follows.

The K-HOL Algorithm:

```
(1). initialization: $T := 0$, $R := 0$, and $ROT := m$;
(2). $k := 1$;
(3). check $D_k$: from $i := ROT$ to $N$ and from $i := 1$ to $ROT - 1$, if ($t_i = 0$, $d_{ki} > 0$, $r_{d_{ki}} = 0$),
     then $t_i := d_{ki}$ and $r_{d_{ki}} := i$;
(4). $k := k + 1$ and $ROT := MOD(ROT + 1, N)$;
(5). repeat steps (2) to (4) until $k = K$.
```

where $T := 0$ and $R := 0$ initialize the process. ROT is a rotating variable which is initially set to a random integer m ($m \in [1, N]$), and will be updated in $[1, N]$ in rotating fashion to keep fair assignments among all stations. The condition $t_i = 0$ indicates that station i is not assigned to transmit any packet, and the condition $r_{d_{ki}} = 0$ indicates that station d_{ki} is not assigned to receive any packet. The two conditions thus guarantee a conflict-free packet transmission/reception. Note that the use of reception array R here simplifies the process of identifying destination conflicts. Without R, we need to check elements of T sequentially to find out if any of the destina [illegible] been assigned. In addition, the algorithm has a FIFO discipline for packets from the same originating station to the same terminating station, since an assignment t_i for a packet in a position with a smaller k cannot be overwritten for a packet with a larger k.

K is the parameter of the algorithm and its value directly affects the network throughput performance. Let ψ^*_{max} be a given maximum network throughput requirement. We can find a $K = K^*$ such that

$$K = \left\{ K^* \left| \frac{1}{N} \sum_{m=1}^{N} m P_{K^*}(m) \geq \psi^*_{max} \right. \right\} \tag{6.1}$$

where $P_{K^*}(m)$ is the probability that a maximum of m conflict-free elements can be selected from D_1 to D_{K^*} by the use of the K-HOL algorithm with parameter K^*, and hence $\frac{1}{N}\sum_{m=1}^{N} m P_{K^*}(m)$ is the maximum throughput. $P_{K^*}(m)$ will be derived later. For light traffic, we may not need K^* arrays of D_ks. Denote the longest queue length at time t by $W(t)$, then a dynamic selection of the parameter K for any traffic condition can be expressed as

$$K = \begin{cases} K^* & \text{if } W(t) \geq K^* \\ W(t) & \text{otherwise.} \end{cases} \tag{6.2}$$

We can implement this by introducing a counter in each station to keep track of $W(t)$.

6.2.2 The Dynamic Round Robin (DRR) Algorithm

The round robin scheme is well known for its fair assignment to users. Token ring/bus LANs as defined by the IEEE 802 series are examples of the application of the round robin scheme where a token is circulated in a ring or a logic ring. A station can access the network only after it seizes the token. Fixed assignment [25] is another example of round robin where each station is periodically assigned to access the network in a TDM fashion. These typical applications of the round robin scheme were designed for shared medium networks to avoid channel collisions. In fixed assignments, bandwidth assigned to stations without transmission demand is wasted. The idea of the DRR algorithm is to use a round robin scheme to select both originating stations and terminating stations for packet transmission over concurrent multichannels. Transmission permissions are assigned only to those stations with bandwidth demand, and which are not in conflict with other assignments. The algorithm is dynamic in the sense that it can sense bandwidth demand of stations and avoid assignment conflicts. The algorithm works with the traffic matrix B and is defined as follows.

The DRR Algorithm:

```
(1). initialization: $T := 0$; $R := 0$; $ROT_1 := m$; $ROT_2 := n$;
(2). $i := ROT_1$;
(3). from $j := ROT_2$ to $j := N$ and from $j := 1$ to $j := ROT_2$: if ($r_j = 0$ and $b_{ij} > 0$), then
         $t_i := j$, $r_j := i$, $ROT_2 := MOD(ROT_2 + 1, N)$, and goto (4);
(4). $i := MOD(i + 1, N)$;
(5). repeat steps 2 to 4 until $i = ROT_1 - 1$.
```

Similarly, m and n are two initial random integers in $[1, N]$ assigned to the two rotating variables ROT_1 and ROT_2. The DRR algorithm starts from row ROT_1 and column ROT_2 of the matrix, and searches to select a nonzero element, say, b_{ij}. If the destination j has not been assigned to receive a packet, then station i is assigned to transmit a packet to station j in the next slot. The algorithm continuously checks every row of B until all N rows are examined.

6.2.3 The Selective Priority Algorithm

The idea of the SP algorithm is from the following observation. The more the number of nonzero elements in a row of the B matrix is, the higher the probability that a nonzero element can be selected. If we select a packet indicated by an element in B with a value larger than 1, i.e., an element indicating more than one packet, the number of nonzero elements in this row will not be reduced. To well balance the distribution of nonzero elements in a row, the SP algorithm selects each time the largest element which has no conflict with other selections. The algorithm is presented as follows.

The SP Algorithm:

```
(1). initialization: $T := 0$; $R := 0$; $ROT1 := n$; $ROT2 := m$; $J := 0$; $i := ROT1$;
(2). $temp := 0$; $J := 0$;
(3). from $j := ROT2$ to $j := N$ and from $j := 1$ to $j := ROT2 - 1$: if ($r_j = 0$ and $b_{ij} > temp$)
      then $J := j$ and $temp := b_{ij}$;
(4). if ($J > 0$), then $t_i := J$; $r_J := i$ and $ROT2 := MOD(ROT2 + 1, N)$;
(5). $i := MOD(i + 1, N)$;
(6). repeat steps 2 to 5 until $i = ROT1 - 1$.
```

Again, $ROT1$ and $ROT2$ are rotating variables updated in $[1, N]$ to keep fair assignments among all stations.

6.3 Performance Analysis

Let the size of all packets in the network be equal. Let each slot cover exactly the length of a packet (packet header overhead is neglected in this analysis). Let traffic generated at each input queue obey an identical Bernoulli process with arrival rate p. Let each packet be destined to any of the N destinations with equal probability $1/N$ unless specifically indicated. For the sake of argument, we call elements in D_ks conflict-free elements if each of them takes a different value. Clearly, elements $d_{1i}, d_{2i}, \cdots$ of D_ks represents the packet queue at station i. We denote it by queue i. The selection of a packet from a queue is hence equivalent to the selection of the corresponding element in D_ks. In the B matrix, row i also refers to queue i. Since an element, say b_{ij}, in B corresponds to more than one packet for $b_{ij} > 1$, we further define N *miniqueues* for each queue, where miniqueue j of queue i contains all packets at station i destined to station j. Miniqueue j in this sense is referred to as being destined to station j. The queue length of miniqueue j of queue i is thus equal to b_{ij}.

6.3.1 Maximum Achievable Throughput

The network throughput equals the arrival rate before the arrival rate reaches the upper bound in a lossless network. This upper bound reveals the maximum achievable network throughput (the maximum arrival rate) that a network can support. Let the maximum achievable network throughput for the three algorithms be $\gamma_{max}(K-HOL)$, $\gamma_{max}(DRR)$, and $\gamma_{max}(SP)$ respectively.

$\gamma_{max}(K-HOL)$:

Assume that all queue lengths under heavy load are longer than K. There are hence $N \times K$ nonzero elements in $(D_1, \cdots, D_K)$. The K-HOL algorithm selects conflict-free elements from D_1 to D_K. Let $P_k(m)$, ($k = 1, 2, \cdots, K$ and $m = 1, 2, \cdots, N$) be the probability that m elements can be selected from $D_1, \cdots, D_k$. We proceed to find a recursive equation for $P_{k+1}(m)$ given $P_k(m)$. $P_1(m)$ is the probability that all N elements in D_1 take m different values, which equals the probability that N balls tossed randomly into N boxes occupy m boxes. It is given in [34] by

$$P_1(m) = \begin{pmatrix} N \\ N-m \end{pmatrix} \sum_{j=0}^{m} (-1)^j \begin{pmatrix} m \\ j \end{pmatrix} \left(\frac{m-j}{N}\right)^N, \quad m = 1, 2, \ldots, N. \tag{6.3}$$

Given $P_k(m)$, $P_{k+1}(m)$ can be expressed as

$$P_{k+1}(m) = \sum_{M=1}^{m} P_{k+1}(m|M) P_k(M), \quad m = 1, 2, \ldots, N. \tag{6.4}$$

Given that M elements are selected in previous k selections, M transmitters (M positions in D_{k+1}) and M receivers (M destinations) are reserved. Therefore, there are only $(N-M)$ elements in D_{k+1} which are available for further selection. Let $P(R|M)$ be the probability that R out of the $(N-M)$ elements in D_{k+1} address those destinations where receivers are not reserved. $P(R|M)$ can be expressed by (4.45) as

$$P(R|M) = \Omega(N-M, R, N-M). \tag{6.5}$$

Let $P(m|R)$ be the probability that $(m-M)$ out of the R elements are conflict-free, i.e., they are destined to different destinations. $P(m|R)$ equals the probability that R balls randomly tossed into $(N-M)$ boxes occupy $(m-M)$ boxes:

$$P(m|R) = \begin{pmatrix} N-M \\ N-m \end{pmatrix} \sum_{i=0}^{m-M} (-1)^i \begin{pmatrix} m-M \\ i \end{pmatrix} \left(\frac{m-M-i}{N-M}\right)^R. \tag{6.6}$$

$P_{k+1}[m|M]$ thus equals

$$P_{k+1}(m|M) = P(m|R)P(R|M). \tag{6.7}$$

Substituting (6.7) into (6.4), $P_{k+1}(m)$ can be found as

$$\begin{aligned} P_{k+1}(m) &= \sum_{M=1}^{m} \sum_{R=m-M}^{N-M} P(m|R)P(R|M)P_k(M) \\ &= \sum_{M=1}^{m} \sum_{R=m-M}^{N-M} \sum_{i=0}^{m-M} (-1)^i \begin{pmatrix} N-M \\ N-m \end{pmatrix} \begin{pmatrix} m-M \\ i \end{pmatrix} \Omega(N-M, R, N-M) \\ &\quad \cdot (\frac{m-M-i}{N-M})^R P_k(M), \\ &\quad m = 1, 2, \cdots, N. \end{aligned} \tag{6.8}$$

Using (6.3) and (6.8), we can therefore find the probability distribution $P_k(m)$ for any given parameter k by recursive calculation. The maximum network throughput for the parameter K is then given by

$$\gamma_{max}(K-HOL) = \frac{1}{N}\sum_{m=1}^{N} mP_K(m). \tag{6.9}$$

$\gamma_{max}(DRR)$:

For the DRR algorithm, the heavy traffic load allows the queue length of each station to be arbitrarily long for an infinite buffer size. The maximum achievable throughput is thus equal to 1, since N conflict-free packets can always be found in the N queues with infinite number of packets. What of interest here is to find the maximum achievable throughput for finite buffer size. Let L be the buffer size for all stations.

Under the DRR algorithm, each station will be assigned transmission priority periodically in each period of N slots. The average throughput of a station over N slots equals the average throughput over all stations in one slot. Let $P_k(m)$ ($m = 0, 1, 2, \cdots, k$) be the probability that m conflict-free packets are selected after the first k rows of B (k queues) are examined. Clearly, $P_1(1) = 1$ and $P_1(0) = 0$ because there are always L packets in a queue. Assume that the L packets are uniformly distributed over all N miniqueues in a queue. $P_{k+1}(m)$, for $m = 1, 2, \cdots, (k+1)$, can be found as

$$P_{k+1}(m) = P_k(m-1)\left[1-(\frac{m-1}{N})^L\right] + P_k(m)(\frac{m}{N})^L, \tag{6.10}$$

where $P_k(m-1)$ or $P_k(m)$ indicate that $(m-1)$ or m packets have been selected from the examined k queues and hence $(m-1)$ or m destinations have been reserved by these selections. $[1-((m-1)/N)^L]$ is the probability that at least one of the L packets is not destined to the $(m-1)$ reserved destinations. $(m/N)^L$ is the probability that all the L packets are destined to the m reserved destinations. Thus we can obtain $P_N(m)$ recursively from $k = 1$ to N, and the

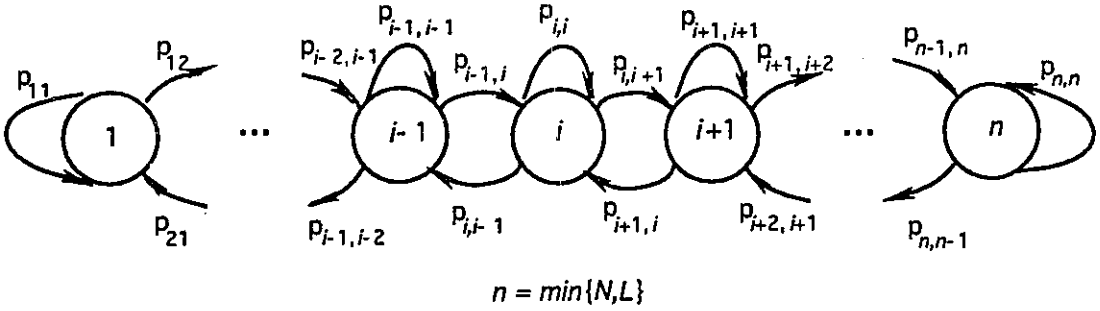


Figure 6.2: *Birth-and-death transition diagram for the number of nonzero elements in a row*

maximum achievable throughput of the DRR algorithm is

$$\gamma_{max}(DRR) = \frac{1}{N}\sum_{m=1}^{N} mP_N(m). \tag{6.11}$$

$\gamma_{max}(SP)$:

Similar to the above, we study the maximum throughput of the SP algorithm for a finite buffer size L. As we select each time a packet from the longest miniqueue in a queue, the probability distribution of the number of the nonempty miniqueues in a queue is smoothed and is no longer binomial. Recognizing the geometric traffic characteristics and the fact that the change of the number of nonempty miniqueues in a queue in each slot is at most one, we can therefore model the number of the nonempty miniqueues as a birth-and-death process. Let state i denote i nonempty miniqueues in a queue. Let $p_{ij}(k)$ be the transition probability from state i to state j ($|i-j| \leq 1$) for queue k. Fig.6.2 depicts the state transitions of the birth-and-death process. Let $Q_k(i)$ be the steady-state probability of state i for queue k. Since there are always L packets in a queue, a new packet will be generated and join one of N miniqueues randomly if a packet is selected from the queue. A birth will occur to a queue when the new packet joins an empty miniqueue, if a packet is selected from a miniqueue with more than one packet. A death will occur to a queue if a packet is selected from a miniqueue with only one packet, and a newly generated packet joins a nonempty miniqueue in the queue. We proceed to find $p_{ij}(k)$, $Q_k(i)$, and $P_k(m)$. For $k = 1$, $P_1(1) = 1$, $P_1(0) = 0$, and

$$
\begin{aligned}
p_{i,i-1}(1) &= \begin{cases} 0 & i < L \\ \dfrac{i-1}{N} & i = L \end{cases} \\
p_{i,i+1}(1) &= \begin{cases} \dfrac{N-i}{N} & i < L \\ 0 & i = L \end{cases} \\
p_{i,i}(1) &= 1 - p_{i,i-1} - p_{i,i+1}
\end{aligned} \tag{6.12}
$$

For $i = L$ in $p_{i,i-1}(1)$, there must be i nonempty miniqueues each with only one packet for $k = 1$. After one packet is selected, there are $(i-1)$ nonempty miniqueues left in the queue, and a death occurs if a newly generated packet joins one of the $(i-1)$ miniqueues (with probability $(i-1)/N$). For $i < L$ in $p_{i,i+1}(1)$, there must exist at least one miniqueue with more than one packet. The selection must happen to one of such miniqueues according to the assignment policy of the SP algorithm, and therefore a birth occurs when a new packet joins an empty miniqueue (with probability $(N-i)/N$).

We proceed to consider For $1 < k \leq N$. By equation (4.45), the probability that j out of i nonempty miniqueues in a queue are destined to those destinations which are reserved by previous m selections is $\Omega(i,j,m)$, where $j \leq min\{i,m\}$. Moreover, we model state i as that i of the L packets fill i of N empty miniqueues, and the rest $(L-i)$ packets distribute randomly in these nonempty miniqueues. Assume that each of the $(L-i)$ packets joins any of the i nonempty miniqueues independently with identical probability $1/i$. A death will occur if a miniqueue with a single packet is selected and a new arrival goes to a nonempty miniqueue. We can find the transition probabilities as follows for $1 < k \leq N$.

$$
\begin{aligned}
p_{i,i-1}(k) &= \begin{cases} \displaystyle\sum_{m=1}^{k-1} \sum_{j=0}^{min\{i-1,m\}} \Omega(i,j,m)(\frac{j}{i})^{L-i}(\frac{i-1}{N})P_{k-1}(m) & i < L \\ \displaystyle\sum_{m=1}^{k-1} \sum_{j=0}^{min\{i-1,m\}} \Omega(i,j,m)(\frac{i-1}{N})P_{k-1}(m) & i = L \end{cases} \\
p_{i,i+1}(k) &= \begin{cases} \displaystyle\sum_{m=1}^{k-1} \sum_{j=0}^{min\{i-1,m\}} \Omega(i,j,m)\left[1-(\frac{j}{i})^{L-i}\right](\frac{N-i}{N})P_{k-1}(m) & i < L \\ 0 & i = L \end{cases}
\end{aligned}
$$

$$p_{i,i}(k) \quad = \quad 1 - p_{i,i-1} - p_{i,i+1} \tag{6.13}$$

Since $\Omega(i,j,m)$ is the probability that j of the i miniqueues are destined to j destinations which are part of the m reserved destinations, therefore only $(i-j)$ miniqueues are possible for further selection in queue k. $(j/i)^{L-i}$ is the probability that all the $(L-i)$ packets join the j miniqueues whose destinations are reserved, and so a selection must take place in a miniqueue with only one packet for $j < i$. $(i-1)/N$ is the probability that a new packet goes into a nonempty miniqueue. Similar arguments apply to $p_{i,i+1}(k)$. The steady-state probabilities $Q_k(i)$ $(k = 1, 2, \cdots, min\{L, N\})$ can be found by solving the following equations recursively from $i = 1$ to $min\{L, N\}$:

$$\begin{cases} Q_k(i) = Q_k(i+1)p_{i+1,i}(k) + Q_k(i)p_{i,i}(k) + Q_k(i-1)p_{i-1,i}(k) \\ \sum\limits_{i=1}^{min\{L,N\}} Q_k(i) = 1 \end{cases} \tag{6.14}$$

With $Q_k(i)$, $P_k(m)$ can be found as

$$\begin{aligned} P_k(m) \quad = \quad & \sum_{i=1}^{min\{N,L\}} \Big[P_{k-1}(m-1)[1 - \Omega(i,i,m-1)u(m-i-1)] \\ & + P_{k-1}(m)\Omega(i,i,m)u(m-i) \Big] Q_k(i), \end{aligned} \tag{6.15}$$

where $P_{k-1}(m-1)$ or $P_{k-1}(m)$ indicates that $(m-1)$ or m destinations have been reserved by previous $(k-1)$ selections. $[1 - \Omega(i,i,m-1)u(m-i-1)]$ is the probability that at least one of the i nonempty miniqueues in queue k is destined to a destination which has not been reserved, and $\Omega(i,i,m)u(m-i)$ is the probability that all the i nonempty miniqueues in queue k are destined to the m reserved destinations for $i \leq m$. $u(m-i)$ is a step function defined as $u(m-i) = 1$ if $m > i$ and $u(m-i) = 0$ otherwise. For $k = 1$ to N, we can find $P_N(m)$ recursively. The maximum achievable network throughput for the SP algorithm is

$$\gamma_{max}(SP) = \frac{1}{N} \sum_{m=1}^{N} m P_N(m). \tag{6.16}$$

6.3.2 Mean Packet Delay

Because of the uniform traffic and the fair assignment properties of the algorithms, each queue can be treated independently as a *Geom*/*Geom*/1 queue with arrival rate p. Let $\psi(X)$ be the service rate of a queue for algorithm X. $\psi(X)$ equals the probability that a packet can be selected from a nonempty queue of a station under algorithm X. Given p and $\psi(X)$, the mean packet delay, defined as the average waiting time of a packet plus the transmission time of the packet, is a special case of the discrete-time queueing [71] for geometric distribution of the service time. Let $D(X)$ be the mean packet delay for algorithm X, then

$$D(X) = \frac{1-p}{\psi(X)-p}. \tag{6.17}$$

We proceed to find $\psi(X)$, i.e., $\psi(K-HOL)$, $\psi(DRR)$, and $\psi(SP)$, for the three algorithms respectively. For the delay analysis, we assume infinite buffer size for all stations.

$\psi(K-HOL)$:

Recalling the operation of the K-HOL algorithm, packets are selected based on the selection of elements from arrays D_1, D_2, $\cdots$, D_K. To identify the number of nonzero elements in D_k, we define N *truncated queues* with parameter k by ignoring the first (k-1) packets of every queue. If a queue has less than k packets, the truncated queue will be empty. Each truncated queue thus has a head packet which is either the kth packet of the original queue, or empty. We denote a truncated queue with parameter k as queue-k. All head packets in the N queue-ks are thus defined by D_k. Let q_k be the probability that a packet will be selected from a nonempty queue-k in a slot. Clearly, $q_1 = \psi(K-HOL)$. Let $\rho_k = p/q_k$ be the queue utilization factor for queue-k. Let $P(H_k)$ be the probability that there are H_k non-zero elements in D_k. $P(H_k)$ can be expressed as the following binomial distribution:

$$P(H_k) = \begin{pmatrix} N \\ H_k \end{pmatrix} (\rho_k)^{H_k}(1-\rho_k)^{N-H_k},$$
$$H_k = 0,1,2,\cdots,N. \qquad (6.18)$$

We proceed to find q_k for $k = 1,2,\cdots,K$. If queue-(k-1) is nonempty, then a packet in queue-k will join queue-(k-1), provided a packet from queue-1 is selected (with probability ψ(K$-HOL$)). If queue-(k-1) is empty, then an arriving packet will immediately leave queue-k and join queue-(k-1). Thus q_k equals

$$q_k = \rho_{k-1}\psi(K - HOL) + (1 - \rho_{k-1}), \qquad (6.19)$$

where ρ_{k-1} is the nonempty probability of queue-(k-1). We proceed to find $\psi(K - HOL)$. The probability that M_1 elements can be selected from D_1 given H_1 nonzero elements in D_1, denoted by $P(M_1|H_1)$, equals the probability that H_1 balls randomly placed in N boxes occupy M_1 boxes:

$$P(M_1|H_1) = \begin{pmatrix} N \\ N - M_1 \end{pmatrix} \sum_{j=0}^{M_1}(-1)^j \begin{pmatrix} M_1 \\ j \end{pmatrix} (\frac{M_1 - j}{N})^{H_1}. \qquad (6.20)$$

Let $P(M_k)$ be the probability that M_k elements are selected from D_1,D_2,$\cdots$,D_k. Given M_{k-1}, M_{k-1} transmitters and M_{k-1} receivers in the network are reserved by the M_{k-1} selections. In other words, there are at most $(N - M_{k-1})$ elements in D_k available for further selection. Let $P(V_k|M_{k-1}, H_k)$ be the probability that V_k out of the H_k nonzero elements in D_k are available for selection, given M_{k-1} and H_k. It is given by

$$P(V_k|H_k, M_{k-1}) = \Omega(H_k, V_k, N - M_{k-1}), \quad V_k = 0,1,2,\cdots,min\{H_k, N - M_{k-1}\}, \quad (6.21)$$

where function $\Omega(\cdot)$ is defined in (4.44). Let $P(U_k|M_{k-1}, V_k)$ be the probability that U_k out of the V_k elements in D_k address those destinations where receivers are not reserved. $P(U_k)$ is given by

$$P(U_k|H_k, M_{k-1}, V_k) = \begin{pmatrix} V_k \\ U_k \end{pmatrix} \left(\frac{N - M_{k-1}}{N}\right)^{U_k} \left(\frac{M_{k-1}}{N}\right)^{V_k - U_k}, \qquad U_k = 0, 1, 2, \cdots, V_k. \tag{6.22}$$

$P(M_k|H_k, M_{k-1}, V_k, U_k)$ is the probability that U_k balls tossed randomly into $(N - M_{k-1})$ boxes occupy $(M_k - M_{k-1})$ boxes:

$$\begin{aligned} &P(M_k|M_{k-1}, H_k, V_k, U_k) \\ &= \begin{pmatrix} N - M_{k-1} \\ N - M_k \end{pmatrix} \sum_{j=0}^{M_k - M_{k-1}} (-1)^j \begin{pmatrix} M_k - M_{k-1} \\ j \end{pmatrix} \left(\frac{M_k - M_{k-1} - j}{N - M_{k-1}}\right)^{U_k} \\ &\qquad W_k = 0, 1, 2, \cdots, N \end{aligned} \tag{6.23}$$

Unconditioning on M_{k-1}, H_k, V_k, and U_k, we obtain $P(M_k)$ as follows.

$$\begin{aligned} P(M_k|H_1) = &\sum_{M_{k-1}=0}^{M_k} \sum_{H_k = M_k - M_{k-1}}^{H_1} \sum_{V_k = M_k - M_{k-1}}^{H_{k+1}} \sum_{U_k = M_k - M_{k-1}}^{V_k} \\ &P(M_k|M_{k-1}, H_k, V_k, U_k) P(U_k|H_k, M_{k-1}, V_k) \\ &P(V_k|H_k, M_{k-1}) P(H_k) P(M_{k-1}) \\ &\qquad M_k = 0, 1, 2, \cdots, N \end{aligned} \tag{6.24}$$

Then the probability that a nonzero element can be selected from a row equals the following average:

$$\psi(K-HOL)=\sum_{H_1=1}^{N}\sum_{m=1}^{H_1}\frac{mP(M_N=m|H_1)}{H_1}P(H_1) \tag{6.25}$$

Calculating $P(M_k)$ recursively from $k=1$ to K, we can find $\psi(K-HOL)$ by iterations for each particular arrival rate p.

$\psi(DRR)$:

Let ψ' be the probability that a packet can be selected from a nonempty miniqueue. The probability that a miniqueue is nonempty, denoted by ρ', is $p/(N/\psi')$. The probability that a queue is empty equals the probability that all miniqueue in this queue is empty. Then

$$(1-\frac{p}{N\psi'})^N=1-\frac{p}{\psi(DRR)}. \tag{6.26}$$

Solving for ψ', we have

$$\psi'=\frac{p}{N[1-(1-p/\psi(DRR))^{1/N}]} \tag{6.27}$$

Let R_j be the probability that there are j nonempty miniqueues in a queue, which is binomial due to the fair assignment policy, and is given by

$$R_j=\begin{pmatrix} N \\ j \end{pmatrix}(\rho')^j(1-\rho')^{N-j}. \tag{6.28}$$

Let $P_k(m)$ $(m \leq k)$ be the probability that m packets have been selected in the first examined k queues. For the first examined queue $(k=1)$, $P_1(1)=1-(1-\rho')^N$ which is the probability that there is at least one nonempty miniqueue, and $P_1(0)=1-P_1(1)$. For $1<k\leq N$, $P_k(m)$ can be found by the following recursive equation:

$$P_k(m) \;=\; \sum_{j=1}^{N} \left[P_{k-1}(m-1)[1-(\frac{m-1}{N})^j] + P_{k-1}(m)(\frac{m}{N})^j \right] R_j, \tag{6.29}$$

where $P_{k-1}(m-1)$ or $P_{k-1}(m)$ indicate that $(m-1)$ or m packets have been selected from previously examined $(k-1)$ queues, and thus $(m-1)$ or m destinations have been reserved. $[1-((m-1)/N)^j]$ is the probability that at least one of the j nonempty miniqueues is not destined to the reserved $(m-1)$ destinations, and $(m/N)^j$ is the probability that all the j nonempty miniqueues are destined to reserved destinations. We can obtain $P_1(m), P_2(m), \cdots, P_N(m)$ recursively using (6.29). Let ψ_k be the probability that a packet can be selected from the kth examined nonempty queue. ψ_k can be found as

$$\psi_k = \sum_{m=1}^{k} P_k(m) P_{k-1}(m-1), \tag{6.30}$$

where $P_0(0) = 1$ for $m = 1$ and $k = 1$, and $P_0(m) = 0$, $m \neq 0$. and finally

$$\psi(DRR) = \frac{1}{N} \sum_{n=1}^{N} \psi_k. \tag{6.31}$$

$\psi(SP)$:

Similarly, let ψ' be the probability that a packet can be selected from a nonempty miniqueue. We can establish the same relationship between ψ' and $\psi(SP)$ as shown in (6.27) with $\psi(SP)$ replacing $\psi(DRR)$. Let ρ' be the probability that a miniqueue is nonempty. Then $\rho' = p/(N\psi')$. Using the truncated queue concept as presented in section 4.2.1, let q_k be the probability that a packet can move out a truncated queue, queue-k, in a miniqueue. Let ρ_k be $p/(Nq_k)$. So a miniqueue contains only one packet implies that the truncated queue, queue-2, is empty. This probability is $(1-\rho_2)\rho_1$, where q_2 equals ψ' if the miniqueue is nonempty, and 1 otherwise. Consequently,

$$q_2 = \frac{p}{N\psi'}\psi' + (1 - \frac{p}{N\psi'}) \;=\; 1 - \frac{p(1-\psi')}{N\psi'} \tag{6.32}$$

Following the method in finding $\gamma_{max}(SP)$, we model the number of nonempty miniqueues in a queue as a birth-and-death process as shown in Fig.6.2 except that the left most state is 0 and the right most state is N. The transition probabilities for $k = 1$ are

$$\begin{aligned}
p_{i,i+1}(1) &= \begin{cases} \left[1-(1-\rho_2)^i\right](\frac{N-i}{N})p & i>0 \\ p & i=0 \end{cases} \\
p_{i,i-1}(1) &= \begin{cases} (1-\rho_2)^i\left[(\frac{i-1}{N})p+(1-p)\right] & i>0 \\ 0 & i=0 \end{cases} \\
p_{i,i}(1) &= 1-p_{i,i+1}(1)+p_{i,i-1}(1)
\end{aligned} \tag{6.33}$$

In $p_{i,i+1}(1)$, $[1-(\rho_2)^i]$ is the probability that at least one of the i nonempty miniqueues contains more than one packet. The selection therefore must happen to one of these miniqueues. $(N-i)p/N$ is the probability that a newly arriving packet joins an empty miniqueue. Similar explanations apply to $p_{i,i-1}(1)$. To see the change of states in queue k, let us consider i nonempty miniqueues containing l packets ($l > i$). Assume that the $(l-i)$ packets are uniformly distributed in the i nonempty miniqueues. With this assumption, we have transition probabilities for a general k:

$$\begin{aligned}
p_{i,i+1}(k) &= \begin{cases} \sum_{m=0}^{k-1}\left[\sum_{j=0}^{min\{m.i-1\}} \Omega(i,j,m)(1-(1-\rho_2)^{i-j})+\Omega(i,i,m)u(m-i)\right] & \\ \cdot p(\frac{N-i}{N})P_{k-1}(m) & i>0 \\ p & i=0 \end{cases} \\
p_{i,i-1}(k) &= \begin{cases} \sum_{m=0}^{k-1}\sum_{j=0}^{min\{m.i-1\}} \Omega(i,j,m)(1-\rho_2)^{i-j} & \\ \cdot\left[p(\frac{i-1}{N})+(1-p)\right]P_{k-1}(m) & i>0 \\ 0 & i=0 \end{cases} \\
p_{i,i}(k) &= p_{i,i+1}(k)+p_{i,i-1}(k)
\end{aligned} \tag{6.34}$$

where $\Omega(i,j,m)$ and $u(\cdot)$ are defined as before. For $p_{i,i+1}(k)$, $P_{k-1}(m)$ indicates that m destinations are reserved in previous $(k-1)$ selections. $(1-(1-\rho_2)^{i-j})$ is the probability that at least one out of the $(i-j)$ nonempty miniqueues which are not destined to the reserved destinations contains only one packet. $(N-i)p/N$ is the probability that a newly arriving packet fills an empty miniqueue. $(i-1)p/N$ is the probability that a newly arriving packet fills a nonempty miniqueue. $(1-p)$ is the probability of no arrival. Given ψ', we can find these transition probabilities and the steady-state probability for nonempty miniqueues in queue k by (6.14) and $P_k(m)$ by (6.15) with summation range $[0,N]$ instead of $[1, min\{L,N\}]$. Let ψ_k be the probability that a packet can be selected in the kth examined queue. With $P_k(m)$, ψ_k can be found by equation (6.30), and $\psi(SP)$ can be found by equation (6.31).

6.3.3 Computational Complexity

The computational complexity of an algorithm refers to the execution time needed for an algorithm. Let $C(X)$ be the computational complexity of algorithm X. For the K-HOL algorithm, the computation of the algorithm includes $(2N+1)$ assignment operations in step 1 to initialize T, R, and ROT, $3N$ comparison operations and $2N$ assignment operations in step 2 to update T, and 1 comparison and 1 subtraction operations for the update of ROT in step 3. The algorithm needs at most $(K-1)$ cycles from steps 2 to 3. Let an assignment operation, or a comparison operation, or a subtraction operation be a scalar operation. Then the total computational complexity of the K-HOL algorithm can be found as

$$C(K-HOL) \leq (5K-3)N + 2K - 1. \tag{6.35}$$

For the DRR algorithm, the computation needs 4 assignments in step 1, at most $2N$ comparison operations and 2 assignment operations to update T and R, 1 comparison and 1 subtraction to update ROT_2. The process will be repeated N times to examine all rows of B. Thus the total computational complexity of the DRR algorithm is

$$C(DRR) \leq 2N^2 + 4N + 4. \tag{6.36}$$

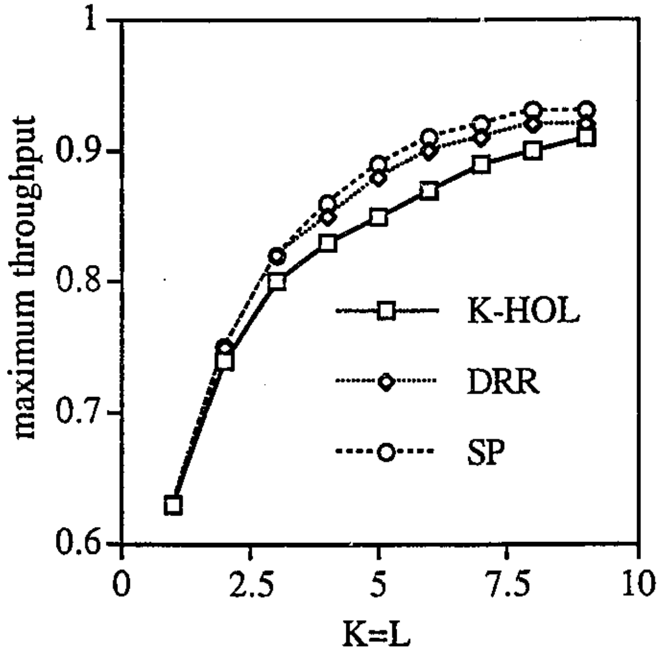


Figure 6.3: *Maximum achievable throughput for* $N = 10$

For the SP algorithm, the algorithm needs 4 assignments for the initialization, 2 assignments for the variable *temp* and J, at most 2N comparisons, 2N assignments, 1 comparison and 2 assignments and for choosing a maximum element in a row, and 1 comparison and 1 subtraction for the update of *ROT*2. The algorithm needs to examine N rows of B. The total complexity is

$$C(SP) = 4N^2 + 5N + 6. \tag{6.37}$$

6.3.4 Results

Fig.6.3 shows the maximum achievable network throughputs for the three proposed algorithms, varying with L ($K = L$ for K-HOL). The throughputs for $L < 3$ are very close for the three algorithms. For $3 < L < 8$, SP shows an improved throughput performance than DRR, and DRR performs better than K-HOL. The largest difference appears at $L = 5$, where SP

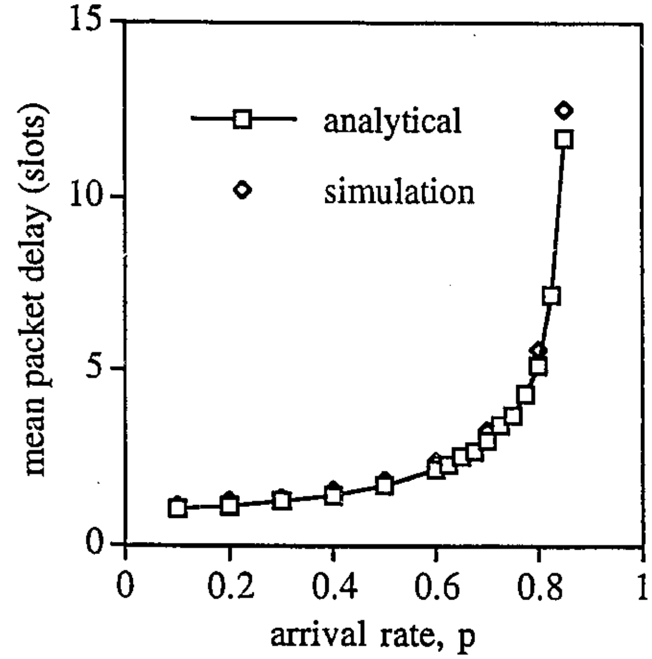


Figure 6.4: *Mean packet delay versus arrival rate under the K-HOL algorithm, $N = 10$, $K = 7$*

achieves about 3.6% throughput increase comparing to K-HOL. For $L > 8$, the maximum throughputs for the three algorithms get close again and approach 1 for infinite L. This is intuitive because N conflict-free packets in a buffer with infinite buffer size can be always found under a heavy traffic load, no matter which of the three algorithms is employed. Note also that the maximum throughputs for the three algorithms increase more rapidly for small L than they do for larger L. For example, the maximum throughput of K-HOL increases by 17.4% from $K = 1$ to $K = 2$, while the increase is only 1.5% from $K = 6$ to $K = 7$. This validates the idea of the K-HOL algorithm that only a small number of K can contribute significantly to the performance improvement. With a small K, the algorithm is much simpler than other algorithms (see complexity comparison results). Table 6.3.4 shows the detailed results of the maximum achievable throughputs of the three algorithms for various values of N. As N increases, the throughputs decline slightly.

Fig.6.4, Fig.6.5 and Fig.6.6 show the mean packet delay under K-HOL, DRR and SP respectively. The delay difference for the traffic loads less than 0.8 is very small among the three algorithms.

		$L = 1$	2	3	4	5	6	7
K-HOL ($K = L$)	N=10	0.6305	0.7301	0.7957	0.8308	0.8537	0.8737	0.8901
	20	0.6149	0.7259	0.7839	0.8216	0.8466	0.8671	0.8798
	30	0.6136	0.7234	0.7820	0.8187	0.8445	0.8630	0.8799
	40	0.6120	0.7206	0.7787	0.8159	0.8420	0.8627	0.8775
	50	0.6104	0.7186	0.7778	0.8151	0.8418	0.8605	0.8763
	60	0.6100	0.7185	0.7768	0.8142	0.8410	0.8614	0.8757
DRR	10	0.6210	0.7540	0.8157	0.8545	0.8789	0.8966	0.9071
	20	0.6172	0.7422	0.8045	0.8439	0.8709	0.8886	0.9044
	30	0.6135	0.7366	0.8006	0.8402	0.8666	0.8871	0.9013
	40	0.6117	0.7341	0.7976	0.8360	0.8653	0.8843	0.9002
	50	0.6109	0.7333	0.7954	0.8361	0.8619	0.8825	0.8980
	60	0.6107	0.7324	0.7940	0.8337	0.8612	0.8812	0.8972
SP	10	0.6317	0.7553	0.8199	0.8595	0.8865	0.9079	0.9208
	20	0.6176	0.7418	0.8059	0.8465	0.8735	0.8961	0.9105
	30	0.6144	0.7370	0.8013	0.8410	0.8684	0.8904	0.9048
	40	0.6110	0.7349	0.7990	0.8381	0.8658	0.8862	0.9025
	50	0.6109	0.7337	0.7965	0.8351	0.8630	0.8823	0.8996
	60	0.6108	0.7322	0.7953	0.8346	0.8611	0.8811	0.8975

Table 6.1: *Maximum achievable network throughput*

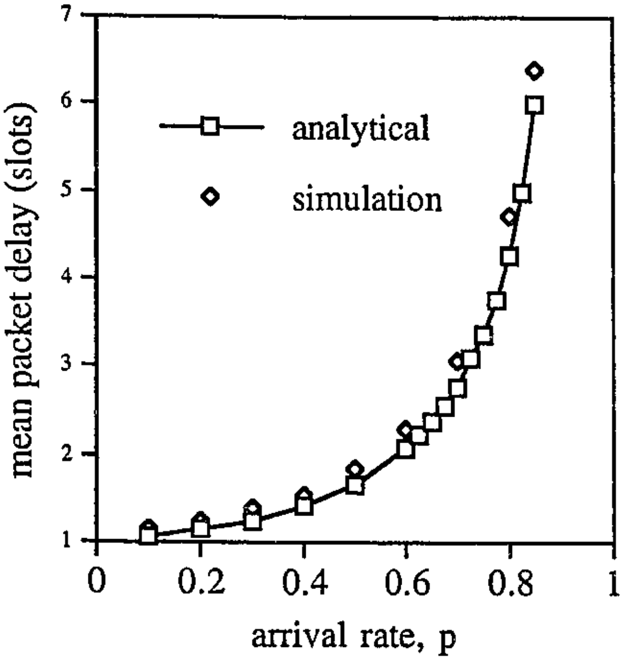


Figure 6.5: *Mean packet delay versus arrival rate under the DRR algorithm,* $N = 10$

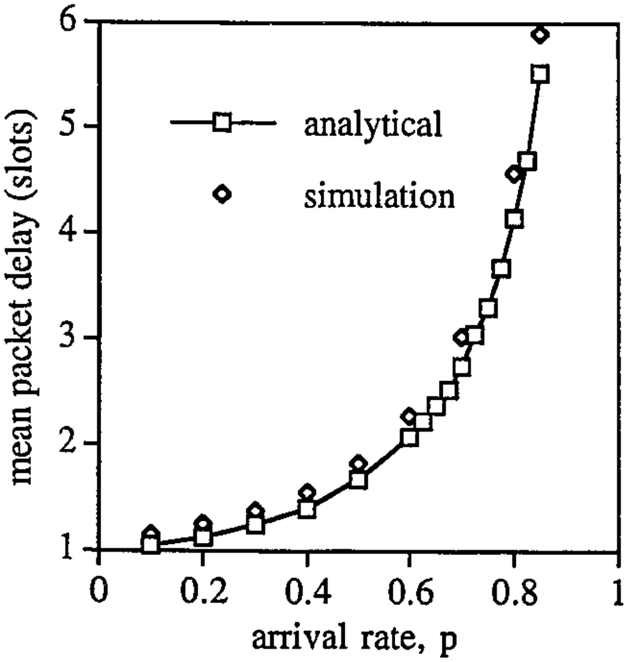


Figure 6.6: *Mean packet delay versus arrival rate under the SP algorithm,* $N = 10$

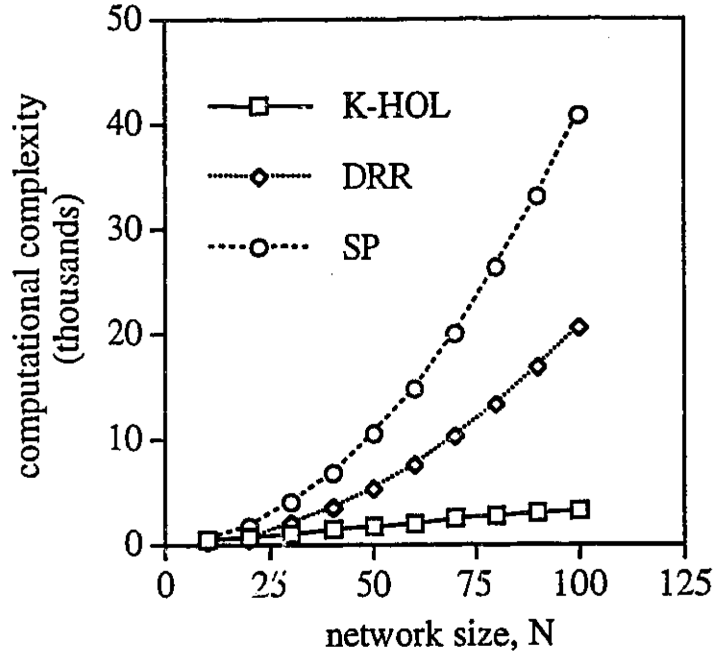


Figure 6.7: *Computational complexity versus network size, $K = 7$*

At $p = 0.85$, the mean packet delay for SP is found slightly smaller than that of DRR, while both of them are about a half of that of the K-HOL algorithm. In addition, simulation presents close results to analytical solutions as shown in these figures. Numerical results also show that the mean packet delay for the three algorithms will slightly increase as N increases. This agrees to the results of the maximum throughput, where throughput for a smaller N is higher than that for a larger N.

Fig.6.7 compares the computational complexity of the three algorithms. The results reveal the tradeoff between delay-throughput performance and the algorithm complexity. Although K-HOL is not as good as the other two algorithms in delay-throughput performance for heavy traffic loads, its complexity is much lower than that of DRR and SP. For network size $N = 100$, the complexity of SP is about twice as that of DRR, and is about 12.6 times more than that of K-HOL. As N increases, the situations for DRR and SP become worse, because the complexity of these two algorithms is $O(N^2)$, while it is $O(N)$ for the K-HOL algorithm. We can see from these results that performance improvement of these algorithms is achieved with a price. Each of them has its own strengths and we need to consider all factors in order to choose an algorithm to satisfy implementation requirements for practical applications.

6.3.5 Comparison with Others

To further evaluate the performance of the proposed three algorithms, we compare them, by simulation, with two typical algorithms, the RS algorithm [24] and the MRS algorithm [17].

Complexity Comparison

The computational complexities of the MRS algorithm $C(MRS)$ is given in [17] as

$$C(MRS) \leq 12N^2 - 2N \tag{6.38}$$

The complexity of the RS algorithm varies for different implementation schemes. If, for instance, each random selection is from all originating and terminating stations without recording the ones which have been examined in previous steps, the maximum number of operations of the algorithm is not bounded. Let us consider the scheme that each station is selected among the remaining stations which have not been examined before. Assume in a step that $(i-1)$ originating stations have been checked. To choose the ith station randomly, we need to generate a random integer number in $[1, N-i+1]$. This needs 1 multiplication operation and at most $(N-i+1)$ subtraction operations for a modular operation [76]. A total of at least $(N-i+2)$ scalar operations thus is needed in choosing an originating station. After an originating station, say station i, is selected, a destination will be randomly selected. Assume in a step that $(j-1)$ destinations have been checked and all of them are not selected. To select the jth destination randomly, we need similar processes as in selecting an originating station, i.e., $(N-j+2)$ scalar operations. In addition, we need to compare the selected destination with previous $(i-1)$ assignments to avoid destination conflicts, as well as to scan the packet backlog to select a packet destined to the selected destination (N comparisons). If the selected destination is reserved, another process of selecting a destination is repeated. Consequently, these processes result in the following operations:

	K-HOK (K=7)	DRR	SP	RS	MRS
N=10	0.333	0.244	0.453	55.9	1.18
20	0.653	0.884	1.703	1.4×10^3	4.76
30	0.973	1.92	3.75	9.7×10^3	10.7
40	1.29	3.36	6.6	3.9×10^4	19
50	1.61	5.2	10.3	1.1×10^5	29.9
60	1.93	7.44	14.7	2.8×10^5	43.1
70	2.25	10	19.9	6×10^5	58.6
80	2.57	13.1	26	1.1×10^6	76.6
90	2.89	16.5	32.8	2×10^6	97
100	3.21	20.4	40.5	3.5×10^6	119
200	6.4	80.8	161	1.1×10^8	479
300	9.61	181	361	8.2×10^8	1.1×10^3

Table 6.2: *Computational complexity comparison (thousands)*

$$C(RS) \le \sum_{i=1}^{N}\sum_{j=1}^{N}(N-i+2)(N-j+2)(N+i-1) \tag{6.39}$$

This is a complexity of $O(N^3)$. Table 6.3.5 compares the computational complexity of the proposed three algorithms with the RS and the MRS algorithms ($K = 7$ for K-HOL). Obviously the complexity of the proposed three algorithms is reduced dramatically comparing to the other two algorithms. Taking $N = 80$ for an example, $C(RS) = 143,603C(MRS) = 423,076C(SP) = 839,694C(DRR) = 4,280,155C(K-HOL)$, and $C(MRS) = 2.94C(SP) = 5.84C(DRR) = 29.8C(K-HOL)$.

Delay Comparison

Fig. 6.8 shows the simulation results of the mean packet delay for $N = 10$ and $K = 7$ under uniform traffic conditions. Although performance of MRS is the best among all algorithms, the difference of the delay is very small for normal traffic loads such as $p < 0.8$. At $p = 0.85$,

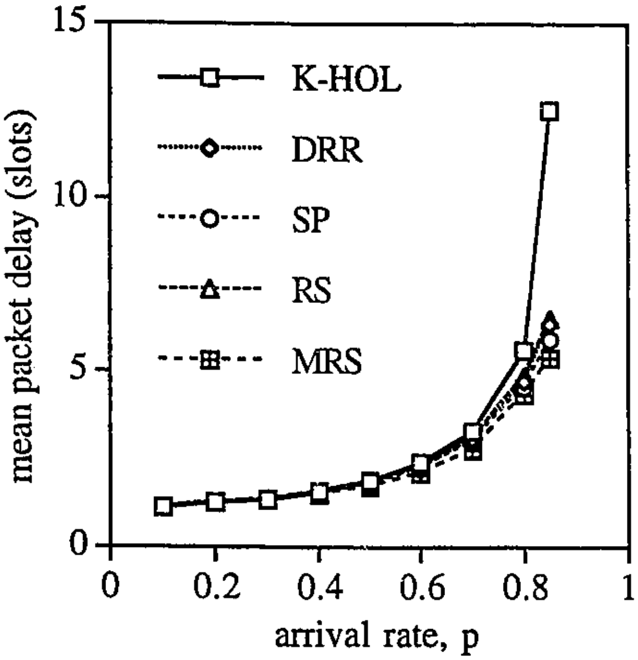


Figure 6.8: *Mean packet delay under uniform traffic by simulation,* $K = 7$*,* $N = 10$

the mean delay for K-HOL is about twice as that of the others, while the largest difference for the rest four algorithms is less than 1.1 slot. It is interesting to note that the delay for RS is about 3% more than that of DRR and is about 10% more than that of SP at $p = 0.85$. This result reveals that the random selection scheme, besides its super high complexity, has a poorer performance than the round robin scheme for uniform traffic. This makes RS algorithms less significant in assignment algorithms. Fig. 6.9 displays the mean packet delay result for nonuniform traffic, where traffic arrival rates to stations are still identical, but the probability that a packet is destined to a different destination is different. In our examples, packets address to a half of stations with twice the probability than they address the other half stations. Because of the unbalanced traffic distribution, the maximum arrival rate that the network can support is reduced from a value close to 1 to a value less than 0.8. For $p \leq 0.6$, the delay performance for all algorithms is nearly the same, but diverges for $p > 0.6$. At $p = 0.7$, the delay for K-HOL is still about twice as that of others. The difference among the rest four algorithms is less than one slot. It is worth noting that the delay of MRS is closer to that of others in this case than it is for uniform traffic as shown in Fig.6.8. This shows that MRS is more sensitive to

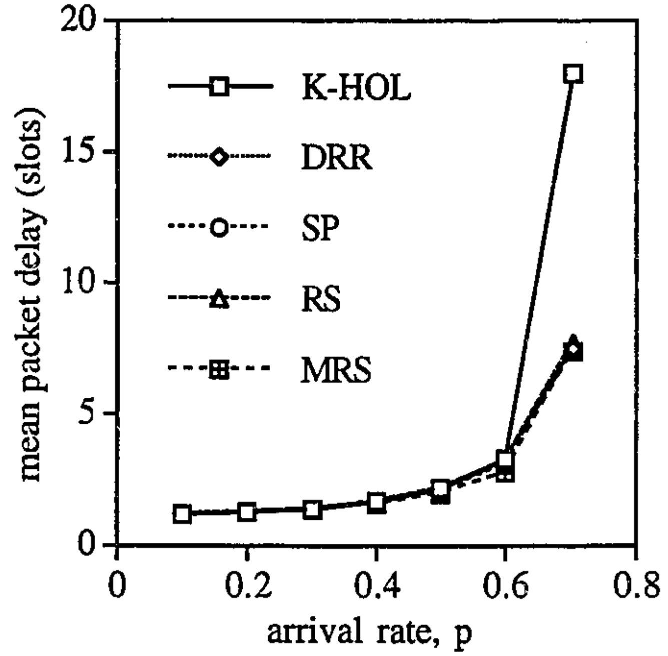


Figure 6.9: *Mean Packet delay under nonuniform traffic by simulation,* $K = 7$*,* $N = 10$

traffic distributions. Table 6.3.5 shows a more detailed delay performance of all these algorithms including their delay variances under 95% simulation confidence intervals. The variances for light traffic are almost the same, while they differ slightly for heavy traffic.

6.4 Summary

We proposed and studied three algorithms or channel assignment in multichannel networks. Computational complexity, maximum achievable network throughput, and mean packet delay were analyzed. Recursive equations were derived to measure their performance. Results show that each of the three algorithms has its own merits. K-HOL has the lowest computational complexity among all algorithms. In addition, it has the flexibility to revise its complexity to tradeoff performance requirements and implementation constraints. Dynamic Round Robin (DRR) is simpler than Selective priority (SP), while SP performs better than DRR in terms of delay and throughput. We also compared these three algorithms to two typical algorithms, MRS and RS algorithms, found in the literature. Comparisons reveal that the round robin scheme as

		Mean Packet Delay±Confidence Interval (95%) (Delay Variance)				
		$p = 0.2$	0.4	0.6	0.7	0.85
uniform traffic	K-HOL ($K = 7$)	1.25±0.01	1.56±0.01	2.4±0.02	3.33±0.03	12.51±1.6
		(0.47)	(0.27)	(0.29)	(0.4)	(9.6)
	DRR	1.25±0.01	1.54±0.01	2.28±0.02	3.06±0.03	6.41±0.1
		(0.47)	(0.26)	(0.26)	(0.35)	(0.75)
	SP	1.25±0.01	1.54±0.01	2.29±0.02	3.03±0.02	5.9±0.1
		(0.47)	(0.26)	(0.27)	(0.33)	(0.69)
	RS	1.25±0.01	1.55±0.01	2.3±0.02	3.1±0.03	6.54±0.1
		(0.47)	(0.26)	(0.28)	(0.35)	(0.85)
	MRS	1.25±0.01	1.52±0.01	2.08±0.01	2.8±0.02	5.4±0.1
		(0.46)	(0.25)	(0.21)	(0.25)	(0.66)
non-uniform traffic	K-HOL	1.27±0.02	1.71±0.01	3.34±0.04	15.1±2.0	–
		(0.5)	(0.33)	(0.8)	(15)	
	DRR	1.27±0.02	1.69±0.01	3.15±0.03	8.87±0.04	–
		(0.5)	(0.31)	(0.66)	(0.38)	
	SP	1.27±0.02	1.68±0.01	3.12±0.03	8.22±0.04	–
		(0.49)	(0.31)	(0.68)	(0.44)	
	RS	1.27±0.02	1.68±0.01	3.19±0.03	8.56±0.04	–
		(0.5)	0.31)	(0.7)	(0.49)	
	MRS	1.27±0.02	1.64±0.01	2.88±0.03	8.40±0.05	–
		(0.49)	(0.29)	(0.58)	(0.88)	

Table 6.3: *Delay and delay variance comparison by simulation,* $N = 10$

used in DRR has slightly better efficiency and is much simpler than the random selection scheme as used in the RS algorithm. Comparing to the MRS algorithm, the proposed three algorithms achieve much lower complexity and similar delay-throughput performance.

Chapter 7

Channel Assignment in Two-Layer Multichannel Networks

In order to support more stations and share the gigabit bandwidth in multichannel networks, the conventional RF/microwave FDM technology can be employed to build up a two-layer multichannel network [78]-[82]. Bit-streams from each station in the first layer are modulated by different RF carrier frequencies before being sent into the second layer. These bit-streams, after being selected at the receiving part, are demodulated by tunable RF local oscillators (LOs). Similar applications of the two-layer network structure can be found in digital satellite communications networks. The two-layer structure may play an important role in providing Fiber-To-The-Home (FTTH) and Fiber-To-The-Office (FTTO) services in the near future because of its flexible bandwidth allocation property. Some papers have explored both theoretical and practical problems of two-layer multichannel techniques [78] [28]. Though channel assignment algorithms found in the literature could be directly applied to two-layer multichannel networks, we will show that the algorithm performance will be discounted. In this chapter, we propose and study two channel assignment algorithms for two-layer multichannel networks. The proposed algorithms will be compared with some existing algorithms, as well as an optimal solution.

7.1 Network Structure

Fig.7.1 shows a two-layer multichannel network structure. The first layer is called backbone layer connecting many clusters of nodes. The second layer is called cluster layer including stations

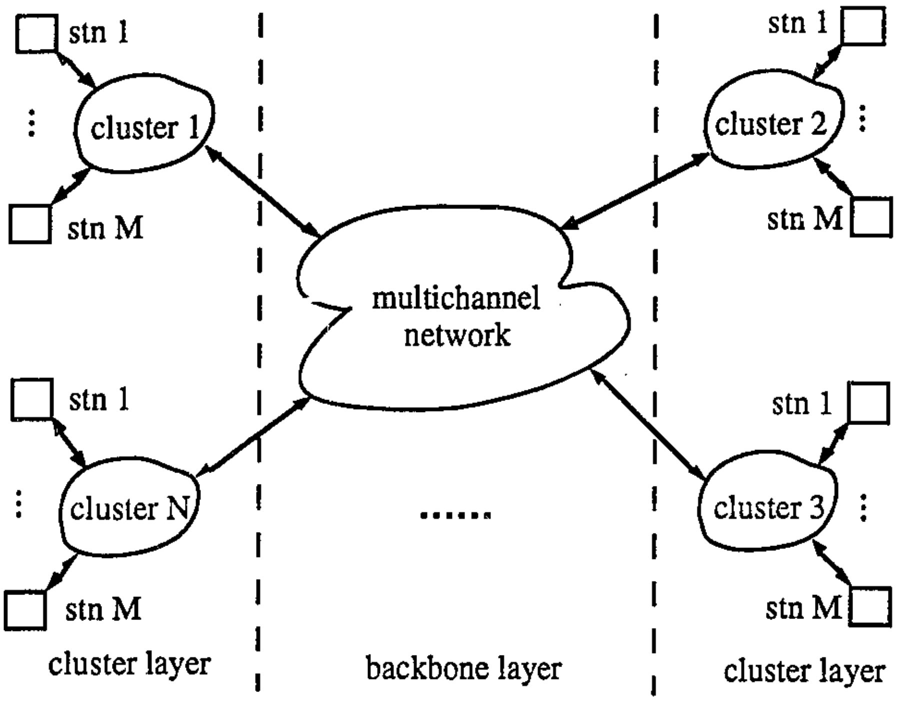


Figure 7.1: *A two-layer multichannel network structure*

located around their respective clusters. Let there be N clusters of nodes, with each cluster node containing M stations. Let there be N channels in the backbone layer and M channels in the cluster layer. We call channels in the backbone layer B-channels and channels in the cluster layer C-channels. Stations within a cluster use different C-channels and stations in different clusters reuse the same set of B-channels. This channel reuse strategy clearly improves the channel utilization in a multichannel network. A path in the network for a connection or a transmission between two stations can thus be uniquely identified by a B-channel and a C-channel. Denote (B_i, C_j) as the path including B-channel i and C-channel j. A full interconnectivity among all stations can be achieved by making tranceivers of cluster nodes and/or stations work on their desired channels. This is achieved in WDM networks by means of tunable transmitters and receivers which can be tuned to work on different channels. The choosing of tunable or fixed-tuned transmitters and/or receivers of stations and/or clusters depends on implementation preferences. Let all channels be divided into slots and each slot accommodate a data packet including its control overhead. Assume that each station maintains identical traffic information of all stations in the network. This can be easily implemented with the use of one or more control channels in the network [20]. Stations announce their traffic information dynamically on these control channels.

7.2 Data Structure

Let stations maintain traffic information of the network with the following data structures.

- Local Backlog Matrix B_{ij}: B_{ij} is an $M \times M$ matrix, where $i, j = 1, 2, ..., N$. The element $b_{ij}(m, n) = d > 0$ $(m, n = 1, 2, ..., M)$ indicates that there are d packets in the mth station of cluster i destined for the nth station at cluster j, and $b_{ij}(m, n) = 0$ otherwise;
- Network Backlog Matrix B: B is an $N \times N$ block matrix, where each block element is a local backlog matrix B_{ij};
- Cluster Assignment $T = \{T_1, ..., T_N\}$: where $T_i = j > 0$ indicates that cluster i is assigned to connect to cluster j on B-channel i, and $T_i = 0$ otherwise.
- Station Assignment Array $R_{iT_i} = \{R_{iT_i}(1), ..., R_{iT_i}(M)\}$ $(i = 1, 2, ..., N)$: where $R_{ij}(m) = n > 0$ indicates that station m at cluster i is assigned to transmit a packet to station n at cluster j, and $R_{ij}(m) = 0$ otherwise.

The following are examples of the data structure for $N = M = 3$:

$$B = \begin{bmatrix} B_{11} & B_{12} & B_{13} \\ B_{21} & B_{22} & B_{23} \\ B_{31} & B_{32} & B_{33} \end{bmatrix} = \begin{bmatrix} \begin{pmatrix} 2 & 5 & 0 \\ 1 & 0 & 0 \\ 2 & 0 & 4 \end{pmatrix} & \begin{pmatrix} 0 & 5 & 0 \\ 2 & 0 & 1 \\ 0 & 2 & 3 \end{pmatrix} & \begin{pmatrix} 3 & 3 & 0 \\ 1 & 6 & 0 \\ 0 & 0 & 3 \end{pmatrix} \\ \begin{pmatrix} 0 & 3 & 0 \\ 4 & 0 & 0 \\ 0 & 1 & 3 \end{pmatrix} & \begin{pmatrix} 3 & 7 & 0 \\ 0 & 0 & 1 \\ 0 & 1 & 3 \end{pmatrix} & \begin{pmatrix} 5 & 3 & 0 \\ 1 & 2 & 0 \\ 2 & 0 & 4 \end{pmatrix} \\ \begin{pmatrix} 2 & 0 & 0 \\ 6 & 3 & 0 \\ 1 & 0 & 3 \end{pmatrix} & \begin{pmatrix} 0 & 2 & 0 \\ 2 & 0 & 7 \\ 3 & 1 & 0 \end{pmatrix} & \begin{pmatrix} 1 & 2 & 3 \\ 1 & 4 & 0 \\ 3 & 0 & 2 \end{pmatrix} \end{bmatrix}. \qquad (7.1)$$

A possible cluster assignment array is

$$T = \left[\begin{array}{c} T_1, T_2, T_3 \end{array}\right] = \left[\begin{array}{c} 1,3,2 \end{array}\right],$$

and the corresponding station assignment arrays can be

$$\left[\begin{array}{l} R_{11}(1) = 2, R_{11}(2) = 1, R_{11}(3) = 3; \\ R_{23}(1) = 1, R_{23}(2) = 2, R_{23}(3) = 3; \\ R_{32}(1) = 2, R_{32}(2) = 3, R_{32}(3) = 1; \end{array}\right].$$

Take cluster 2, i.e., the second row of matrix B, as an example. The element B_{23} shows that there are 5 packets at station 1 of cluster 2 destined for station 1 at cluster 3, and there are 3 packets at station 1 of that cluster destined for station 2 at cluster 3, and so on. The element $R_{23}(1) = 1$ shows that station 1 at cluster 2 is assigned to transmit a packet to station 1 at cluster 3, and $R_{23}(2) = 2$ shows that station 2 of cluster 2 is assigned to transmit a packet to station 2 at cluster 3 and so on.

7.3 The Algorithms

The objective of channel assignment algorithms in multichannel networks is to select as many packets as possible from different stations at different clusters for transmission under the following conflict-free constraints:

- $T_i \neq T_j, \forall i \neq j, i, j = 1, 2, \cdots, N;$

  {Each cluster can be assigned each time at most one B-channel}

- $R_{ij}(m) \neq R_{ij}(n), \forall m \neq n, m, n = 1, 2, \cdots, M.$

  {Each station can be assigned each time at most one C-channel}

Since channel assignment algorithms for one-layer multichannel networks select only B-channels, intuitively, they may not be effective for the two-layer multichannel networks. This can be shown by the following simple example. Let matrix B have the following values for $N = M = 3$,

$$B = \left[\begin{array}{ccc} \begin{pmatrix} 0 & 5 & 0 \\ 0 & 0 & 0 \\ 0 & 2 & 0 \end{pmatrix} & 0 & 0 \\ 0 & \begin{pmatrix} 3 & 1 & 1 \\ 0 & 0 & 0 \\ 0 & 0 & 0 \end{pmatrix} & \begin{pmatrix} 0 & 3 & 0 \\ 0 & 0 & 1 \\ 1 & 0 & 0 \end{pmatrix} \\ 0 & 0 & \begin{pmatrix} 0 & 0 & 0 \\ 0 & 0 & 0 \\ 0 & 2 & 1 \end{pmatrix} \end{array} \right]. \tag{7.2}$$

Apparently, an algorithm for one-layer networks will select B_{11}, B_{22}, B_{33} to formulate channel assignments which results in the maximum number of 3 clusters. Since at most one station can be selected from the selected three clusters, the total number of stations which can be selected for conflict-free transmissions is 3. If we select B_{11} and B_{23}, instead, we cannot choose any other B_{ij} in the last row of B according to our conflict-free transmission constraints. However, with this selection of smaller number of clusters, the maximum number of stations for conflict-free transmission is increased to 4, since B_{23} has 3 conflict-free elements. This example shows that the maximum number of B-channel assignments may not give the maximum number of conflict-free station assignments. This illustrates an essential difference between channel assignment algorithms for one-layer and two-layer multichannel networks. For a two-layer multichannel network, we notice the fact that the number of stations assigned could be possibly a maximum if each selection of stations corresponding to each B_{ij} is a maximum. We use the terminology of System Distinct Representative (SDR) [48] to denote in a matrix a maximum set of nonzero elements from distinctive rows and columns. A SDR set in a B_{ij} thus corresponds to a set of conflict-free stations according to the conflict-free constraints defined above. The channel assignment therefore includes two steps for station and cluster assignments respectively:

1. Select an SDR from each B_{ij} respectively. We call this process the selection of $\{b_{ij}(n,m)\}$.
2. Select clusters such that the corresponding SDR sets include as many nonzero elements as possible. We call this process the selection of $\{B_{ij}\}$.

We further define an operation $M(\cdot)$ which carries out the first step of the assignment algorithm such that $M(B_{ij})$ produces an SDR assignment $\{R_{ij}(1), ..., R_{ij}(M)\}$. An optimal or heuristic algorithm for $M(\cdot)$ operation can be found in [17] [11] [24] [48] . We cannot, however, directly input the matrix $B_{NM \times NM}$ into the operation $M(\cdot)$, because a cluster could be assigned to connect to more than one clusters in this way and this violates the conflict-free constraints for clusters. In addition, if we select clusters by comparing SDR sets found in step 1 for each cluster, we have N choices for cluster 1, and $(N-1)$ choices for cluster 2 and so on for conflict-free assignments. Consequently, we can choose one from $(N!)$ possible combinations. Clearly, at least one of these combinations is an optimal solution. This optimal solution, however, is with the price of a high computational complexity $O(N!)$. We proceed to propose two heuristic algorithms for the channel assignment algorithms. Define a symbol $\| \cdot \|$ to represent the number of nonzero elements of an array, e.g., $\| R_{ij} \|$ gives the number of nonzero elements of R_{ij}. The value of $\| R_{ij} \|$ is automatically given in $M(\cdot)$ operation after $\{b_{ij}(n,m)\}$ is selected, and hence does not introduce additional complexity.

7.3.1 Algorithm A : Rotating Assignment (RA) Algorithm

First, use $M(\cdot)$ to find $R_{ij}(k)$ and $\| R_{ij} \|$, where $i, j = 1, 2, ..., N$ and $k = 1, 2, ..., M$. Secondly, we employ a rotating variable $1 \leq i' \leq N$ in the second step to keep fairness among the N clusters by starting the second step at $i = i'$. For each i, we select a column T_i, corresponding to a B_{iT_i} with a maximum number of nonzero elements and is in conflict-free with all previous selections $(T_{i'}, T_{i'+1}, ..., T_{i-1})$. This process can be formulated as:

1. find $\{R_{ij}(1), R_{ij}(2), ..., R_{ij}(M)\} = M(B_{ij})$, and
   $\| \{R_{ij}(1), R_{ij}(2), ..., R_{ij}(M)\} \| = \| M(B_{ij}) \|$, $i, j \in [1, N]$, $k \in [1, M]$.

2. find $(T_1, T_2, ..., T_N)$ such that

$$T_i = \left\{ k \;\middle|\; \begin{array}{l} \| M(B_{ik}) \| \geq \| M(B_{ij}) \|, \\ \forall j \neq k \text{ and } k, j \neq T_{i'}, \cdots, T_N, T_1, \cdots, T_{i'-1} \end{array} \right\}.$$

This algorithm therefore yields the following assignments for $i = 1, 2, \cdots, N$:

$$\left\{ R_{iT_i}(1), R_{iT_i}(2), \cdots, R_{iT_i}(M) \right\} = \left\{ M(B_{iT_i}) \right\}.$$

Taking (1) as an example, we can follow this algorithm to obtain possible assignments as follows:

$$\begin{aligned}
&\{T_1, T_2, T_3\} = \{1, 3, 2\} \\
&\{R_{11}(1), R_{11}(2), R_{11}(3)\} = M(B_{11}) = \{2, 1, 3\}, \\
&\{R_{23}(1), R_{23}(2), R_{23}(3)\} = M(B_{23}) = \{1, 2, 3\}, \\
&\{R_{32}(1), R_{32}(2), R_{32}(3)\} = M(B_{32}) = \{2, 3, 1\}.
\end{aligned}$$

7.3.2 Algorithm B: Selecting Maximum Rank (SMR) Algorithm

The number of nonzero elements in an array is known as the rank of the array. Matrix $\{\| M(B_{ij}) \|\}_{N\times N}$ therefore has $N \times N$ elements which equals the ranks of assignment arrays. The SMR algorithm has the same first step as given in algorithm A and, in the second step, selects a maximum element in matrix $\{\| M(B_{ij}) \|\}_{N\times N}$ and deletes the corresponding row and column accordingly. The position of each selected element is recorded as (m_k, n_k), where m_k denotes a row, n_k denotes a column and $k = 1, 2, ..., N$. This process can be formulated as

1. the same as that of algorithm A.

2. find (m_k, n_k) such that

$$(m_k, n_k) = \left\{ (i^*, j^*) \;\middle|\; \begin{array}{l} \| M(B_{i^*j^*}) \| \geq \| M(B_{ij}) \|, \\ \forall i, i^* \neq m_1, m_2, \cdots, m_{k-1}, \\ j, j^* \neq n_1, n_2, \cdots, n_{k-1}, \\ k = 1, 2, \cdots, N. \end{array} \right\}.$$

3. $T_{m_k} = n_k$, $k = 1, 2, ..., N$.

Taking (2) as an example, we have

$$\begin{bmatrix} \| M(B_{11}) \| & \| M(B_{12}) \| & \| M(B_{13}) \| \\ \| M(B_{21}) \| & \| M(B_{22}) \| & \| M(B_{23}) \| \\ \| M(B_{31}) \| & \| M(B_{32}) \| & \| M(B_{33}) \| \end{bmatrix} = \begin{bmatrix} 1 & 0 & 0 \\ 0 & 1 & 3 \\ 0 & 0 & 1 \end{bmatrix},$$

$$\begin{bmatrix} (m_1, n_1) = (2, 3) \\ (m_2, n_2) = (1, 1) \\ (m_3, n_3) = (0, 0) \end{bmatrix} \text{ and } \begin{bmatrix} T_1 = 1 \\ T_2 = 3 \\ T_3 = 0 \end{bmatrix}.$$

7.4 Performance

Assume that traffic obeys an identical Bernoulli process at all stations with arrival rate p. We study the channel utilization and mean packet delay of the network under the RA algorithm and the SMR algorithm respectively. Packet overhead is ignored in the performance studies in this study. We choose $N = 12$ and $M = 4$ for all our examples studied.

7.4.1 Channel Utilization

Since traffic is uniform and the assignment is fair to all stations, we simply decompose the local backlog matrices B_{ij}s and refer to each element $b_{ij}(m, n)$ of B_{ij} as the queue size of a *miniqueue* with arrival rate $p/(NM)$. Although the two selection schemes of the two algorithms should result in different queueing performance, the channel utilizations for the two algorithms can be the same under the assumption that selections of each $b_{ij}(n, m)$ as well as each B_{ij} are independent. This is because the selection of of B_{ij}s in both of the proposed algorithms depends on the number of nonzero rows in B_{-r}, which obeys the same binomial distribution under the assumption of independent selection. Our simulation results (see the mean packet delay comparison in the latter part of this section) show that the proposed two algorithms in fact perform very closely. Let ψ_c be the channel utilization at the cluster layer. ψ_c is the probability that a packet (refers to $b_{ij}(n, m)$) can be selected for transmission from a nonempty miniqueue, given that the corresponding cluster (refers to B_{ij}) in the backbone layer is selected. Let ψ_b be the probability that a cluster (B_{ij}) can be selected. Let ψ be the probability that a packet can be selected from a nonempty miniqueue. We have

$$\begin{aligned}
\psi_c &= \Pr\left[\begin{array}{l|l} b_{ij}(n,m) \text{ of } B_{ij} & B_{ij} \text{ is selected, } \forall n,m \in [1,M] \\ \text{is selected} & \text{and } \forall i,j \in [1,N] \end{array}\right] \\
&= \frac{\Pr[B_{ij} \text{ and } b_{ij}(n,m) \text{ are selected}]}{\Pr[B_{ij} \text{ is selected}]} \\
&= \frac{\Pr[B_{ij} \text{ is selected} \mid b_{ij}(n,m) \text{ is selected}] \cdot \Pr[b_{ij}(n,m) \text{ is selected}]}{\Pr[B_{ij} \text{ is selected}]} \\
&= \frac{\Pr[b_{ij}(n,m) \text{ is selected}]}{\Pr[B_{ij} \text{ is selected}]} \\
&= \frac{\psi}{\psi_b} \qquad (7.3)
\end{aligned}$$

where the probability that B_{ij} is selected given that $b_{ij}(n,m)$ is selected equals 1, because B_{ij} must be selected given that $b_{ij}(n,m)$ is selected. We define a binomial distribution $\beta(N,n,\gamma)$ as the probability that an iid event with occurring probability γ occurs n times in N Bernoulli trials. Then

$$\beta(N,n,\gamma) = \begin{pmatrix} N \\ n \end{pmatrix} \gamma^n (1-\gamma)^{N-n}. \qquad (7.4)$$

Let ρ be the probability that a miniqueue is nonempty ($b_{ij}(n,m) > 0$), then

$$\rho = \frac{p}{MN\psi} = \frac{p}{MN\psi_c\psi_b}. \qquad (7.5)$$

Let α be the probability that a row of B_{ij} is zero ($\sum_{m=1}^{M} b_{ij}(n,m) = 0$, $\forall n \in [1,M]$ and $\forall i,j \in [1,N]$), then

$$\alpha = (1-\rho)^M. \qquad (7.6)$$

Let θ be the probability that $B_{ij} = 0$ ($\forall i,j \in [1,N]$), then

$$\theta = (1-\rho)^{M^2}. \tag{7.7}$$

Thus the probability that a row of B is zero equals θ^N. Let $P_h(H)$ be the probability that there are H nonzero rows in B, then

$$P_h(H) = \beta(N, H, \theta^N) \tag{7.8}$$

Let $P_k(m|H)$ $(m = 0, 1, 2, \cdots, H)$ be the probability that m B_{ij}s in B are selected from the first examined k nonzero rows. For $k = 1$, clearly, $P_1(1) = 1$ and $P_1(m) = 0$ for $m \neq 1$. Given $P_{k-1}(m|H)$, $P_k(m|H)$ can be found recursively as

$$\begin{aligned} P_k(m|H) &= \Big[1 - \beta(N-m+1, 0, 1-\theta)\Big] P_{k-1}(m-1|H) \\ &\quad + \beta(N-m, 0, 1-\theta) P_{k-1}(m|H) \end{aligned} \tag{7.9}$$

where $1 - \beta(N-m+1, 0, 1-\theta) P_{k-1}(m-1|H)$ is the probability that at least one out of $(N-m+1)$ B_{ij}s is nonempty given $(m-1)$ B_{ij}s are selected in the first examined $(k-1)$ selections, and $\beta(N-m, 0, 1-\theta) P_{k-1}(m|H)$ is the probability that all $(N-m)$ B_{ij}s are empty given that m B_{ij}s are selected in the first examined k selections. Using the above recursive equation, we find $P_H(m|H)$, the probability that m B_{ij}s can be selected in B given H rows of B are nonempty. The channel utilization for the backbone layer ψ_b equals the mean number of selected B_{ij}s divided by the mean number of nonzero B_{ij}s:

$$\psi_b = \sum_{H=1}^{N} \sum_{m=1}^{H} \frac{m P_H(m|H) P_h(H)}{H}. \tag{7.10}$$

From (7.3), the channel utilization is $\psi = \psi_c \psi_b$, where ψ_c depends on the algorithm used for the

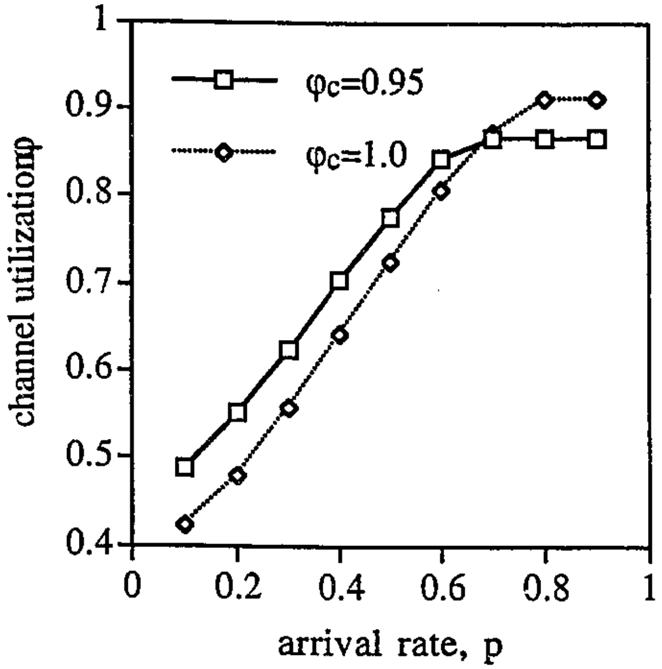


Figure 7.2: *Channel utilization versus arrival rate*

cluster layer channel assignment and can be found for a given algorithm. We can thus find ψ by iterations because ψ_b is a function of itself. Fig.7.2 shows the channel utilizations versus arrival rate under the assumption that ψ_c equals constants. For for $\psi_c = 1.0$, $\psi = \psi_b$ and ψ increases as the arrival rate grows. For heavy traffic ($p > 0.8$) it becomes saturated. The saturation point also reveals the maximum network throughput that the network can support. Beyond this point, packet queues become unstable with unlimited queue length, because the departure rate of packets cannot increase with the increase of the arrival rate in the saturation situation. For $\psi_c = 0.95$, ψ is larger than that for $\psi_c = 1.0$ in the light traffic load range. This is because that smaller ψ_c results in longer queues, and therefore more packets can be selected. For heavy traffic loads, however, the channel utilization become saturated earlier than that for $\psi_c = 1.0$. The maximum network throughput is decreased by about 5% in this case.

7.4.2 Mean Packet delay

We proceed to study the mean packet delay performance of the network under the RA and SMR algorithms by simulation. We also compare these two algorithms to three other possible algorithms defined as follows:

1. *Local Scheduling (LS) Algorithm*: Each station does not have traffic backlog of the network. Each cluster selects a B-channel such that a maximum number of packets from the local cluster can be selected. If there are other clusters choosing the same B-channel in the same time, only one cluster can be successful. This algorithm is similar to the DT-WDMA algorithm proposed in [20].
2. *Cluster First (CF) Algorithm*: Each station maintains traffic backlog of the network. A maximum number of clusters is selected for the cluster assignment. Station assignment is assigned lower priority. This algorithm is similar to the conflict-free algorithm for WDMA networks proposed in [17].
3. *Optimal Selection (OS) Algorithm*: As described in section 4, this algorithm searches an optimal solution in all ($N!$) possible combinations.

We use the MRS algorithm proposed in [17] to assume the $M(\cdot)$ operation, and let the round trip propagation delay between any station pair, denoted by d, be 5 slots in our examples.

We first consider a balanced traffic distribution. Each packet is destined for any station or cluster with an equal probability. Fig.7.3 shows the mean packet delay comparison among the five algorithms. The proposed RA and SMR algorithms demonstrate very comparable performance with that of the optimal algorithm. From the figure, we can also see that SMR is slightly better than RA, and the optimal OS is slightly better than SMR. The maximum delay difference among the three algorithms, however, is found to be less than one slot, while the complexity of SMR and RA is much simpler than that of OS, as the latter does ($N!$) times searches. In contrast, the CF and the LS algorithms show the poorest performance with a maximum network throughput about 0.4, which is about 50% less than that of the other three algorithms. This shows that algorithms for one-layer networks give poor performance for two-layer networks.

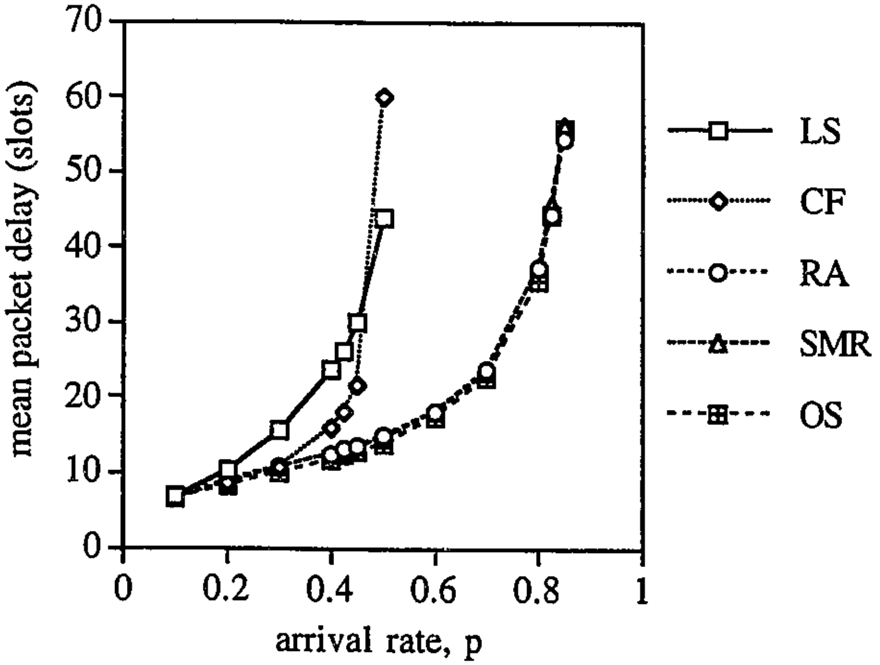


Figure 7.3: *Delay-throughput performance for uniform traffic by simulation*

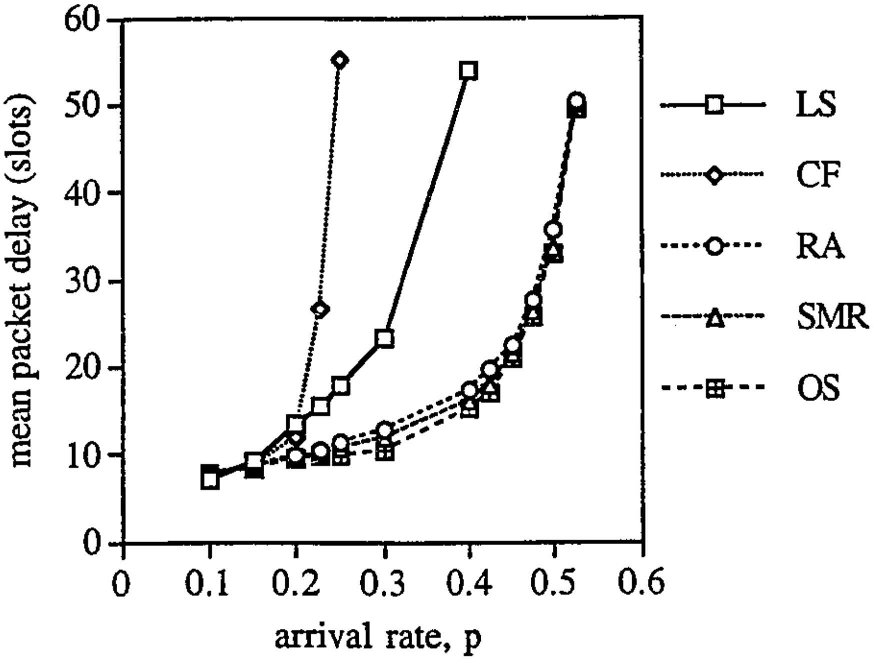


Figure 7.4: *Delay-throughput performance for non-uniform traffic by simulation*

For unbalanced traffic distribution, let a packet be destined for cluster k with the following non-uniform probability distributions:

$$d_k = \frac{k}{\sum_{i=1}^{N} i} = \frac{2k}{N(N+1)}, \quad k = 1, 2, ..., N$$

Obviously, d_2 is two times longer than d_1 and d_N is N times longer than d_1. This extreme example is used here to examine our algorithm performance for unbalanced traffic. Fig.7.4 displays the simulation results. With this unbalanced assumption of traffic, the maximum network throughput that RA and SMR algorithms can support is about 0.5, while the LS algorithm can only support a maximum throughput of 0.3. The CF algorithm shows the poorest performance for our example with a maximum network throughput of about 0.2.

7.5 Summary

Considering the particular structure of a two-layer multichannel network, we propose in this chapter two channel assignment algorithms, the Selecting Maximum Rank (SMR) algorithm and the Rotating Assignment algorithm (RA). The simulation and analytical results show that the proposed algorithms have very good channel utilization performance, and much better delay and throughput performance than algorithms for one-layer multichannel networks when they are applied to two-layer multichannel networks. The delay and throughput performance of the two algorithms are shown to be very close to an optimal algorithm, whereas the complexity is apparently much lower.

Chapter 8

Conclusions

In this thesis, we investigated problems in high-bandwidth multichannel networks and proposed two communication protocols and several channel assignment algorithms for high-bandwidth multichannel networks.

The first protocol is a dynamic channel sharing protocol to be able to accommodate both connection-oriented real-time and connectionless non-real-time traffic. Channels are pre-assigned nominally to the two classes of traffic, and then are allowed to be shared by them according to their traffic loading situations. The blocking probability and mean packet delay for real-time traffic, as well as the mean channel utilization for non-real-time traffic, were derived. Practical issues associated with this protocol such as the minimum number of control channels needed for a given network size and the selection of a scheduling or a non-scheduling scheme were also studied.

The second protocol deals with the multiconnection problem in multichannel networks, where each station is allowed to have several connections with other stations and data packets are transmitted in TDM over multiconnections. We found constraints on network parameters and connection policies. Logic relations among established connections and slot matching problems were analyzed. Bandwidth allocation schemes and strict-sense and rearrangeable acceptance conditions were obtained. A combined control scheme using both fixed and dynamic assignments on control channels was proposed in this protocol.

In the designing of channel assignment algorithms, we aimed at increasing the efficiency of algorithms and improving channel utilization in multichannel networks and two-layer multichannel networks, while reducing the computational complexity of algorithms. Results shows that these proposed algorithms have their own merits. K-HOL has the lowest computational complexity among all algorithms. In addition, it has the flexibility in trading-off complexity and efficiency. The Dynamic Round Robin (DRR) algorithm is simpler than the Selective Priority (SP) algorithm, while SP performs better than DRR in terms of delay and throughput. The Selecting Maximum Rank (SMR) algorithm and the Rotating Assignment algorithm (RA) for two-layer multichannel networks were shown to produce superior performance over those algorithms for one-layer multichannel networks when they are applied to two-layer multichannel networks. These algorithms were also compared with an optimal algorithm and found very close in their efficiency whereas the complexity is apparently much lower.

We suggest further research on this topic as follows.

Internetworking of high-bandwidth multichannel networks with themselves as well as with existing communication networks: Even after high-bandwidth multichannel networks appear in the commercial market, they must co-exist or work together with currently used networks for several years. The internetworking among them therefore is an important topic. Because of the very high speed characteristics of high-bandwidth multichannel networks, protocol layer processing needs to be a minimum. New protocols such as proposed in this thesis normally combine some protocol layers as defined in the OSI structures to reduce protocol processing overhead. These new protocols and transmission rates result in incompatibility with those of existing networks. Speed adaptation and protocol conversion are definitely needed in gateways connecting a high-bandwidth network with an existing network.

Channel management in high-bandwidth multichannel networks: as high-bandwidth multichannel networks need to work with existing networks, an interconnected network in the future will likely include two levels. Low rate communication services will be still provided by existing networks which belong to the first level network. High rate traffic and some specific services will be carried in the high-bandwidth multichannel networks which form the second level network.

Channel bandwidths in different levels of the network can range from a few kilo bits per second to many gigabits per second. How to efficiently allocate these channels or subchannels to provide good quality of service will be an interesting problem.

New protocols: protocols found in the literature normally assume traffic environments to be uniform Bernoulli arrivals or Poison arrivals. For bursty traffic, unbalanced traffic and hot-spot traffic, some special designs of protocols may perform much better than those of existing protocols. In general, many practical problems need to be taken into account in designing protocols for high-bandwidth multichannel networks.

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