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Protocols and Performance Models for High Speed, Dual-bus, Fiber-Optic Local Area Networks

by

Bandula W. Abeysundara

A thesis
submitted to the Faculty of Graduate Studies and Research
in partial fulfillment of the requirements for the degree of
Doctor of Philosophy

Department of Computing Science

Edmonton, Alberta

SPRING 1991

ISBN 0-315-66761-3

UNIVERSITY OF ALBERTA

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NAME OF AUTHOR : Bandula W. Abeysundara

TITLE OF THESIS : Protocols and Performance Models for High Speed, Dual-bus, Fiber-Optic Local Area Networks

DEGREE FOR WHICH THIS THESIS WAS PRESENTED : Doctor of Philosophy

YEAR THIS DEGREE GRANTED : 1991

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Dated April 19, 1991

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The undersigned certify that they have read, and recommend to the Faculty of Graduate Studies and Research, for acceptance, a thesis entitled **Protocols and Performance Models for High Speed, Dual-bus, Fiber-Optic Local Area Networks** submitted by **Bandula W. Abeysundara** in partial fulfilment of the requirements for the degree of **Doctor of Philosophy**.

....
A. E. Kamal (Supervisor)

....
U. M. Maydell

....
P. Gburzynski

....
W. Krzymien

....
J. F. Hayes

Date : January 21, 1991

ABSTRACT

At high data transmission rates, the packet transmission time in a local area network (LAN) could become comparable to, or less than the medium propagation delay. Therefore, in high speed LANs, the ratio of the channel propagation delay to the packet transmission time, or the *normalized* propagation delay, **a**, could approach or even exceed 1. The performance of existing LAN schemes, for example, LANs employing the CSMA/CD protocol, degrades rapidly as a approaches 1. In a high speed environment therefore, LAN medium access protocols should be capable of yielding satisfactory performance over a wider range of a values.

In this thesis, we propose two new LAN medium access protocols, named Z-Net and X-Net, as suitable candidates for operation at high speeds. The network architecture is based on two, unidirectional fiber-optic channels. A distinct feature of the architecture is the use of active taps to minimize the waste of channel bandwidth due to collisions. When collisions occur, one transmission continues to completion, while others are aborted. The proposed medium access protocols behave as random-access schemes at light load. As load increases, the behaviour is similar to that of a controlled-access scheme with implicit token-passing. Because of this hybrid nature of the access protocols, they possess the advantages of zero medium access delay at light load and bounded delay at all loads. The protocols are completely distributed, with all stations executing the same access protocol. Thus, the vulnerability of the network to a single station failure is minimized. Further, for the execution of the protocols, stations do not require a knowledge of other station locations in the network. This reduces maintenance effort, when there are frequent changes in station locations.

The performance of the Z-Net and X-Net are evaluated using approximate analytic models. These results are then validated against the results obtained from simulation models. The channel utilization values of the analytic and simulation models are in very close agreement. Performance results show that both protocols achieve high channel utilization (with X-Net exhibiting superior

performance) even when the packet transmission time is low compared to the channel propagation delay. Therefore, the proposed schemes are suitable for operating at high channel data rates. The bounded delay property makes them suitable for supporting real-time traffic.

Acknowledgements

I wish to thank Professor Ahmed Kamal, my supervisor, for his guidance and assistance during the entire period of this work. I also wish to thank the members of my supervisory and examining committees, Professors Ursula Maydell, Tamer Ozsu, Pawel Gburzynski, Witold Krzymien and Jeremiah Hayes. Their comments were valuable in improving the thesis.

The Department of Computing Science and the Natural Sciences and Engineering Research Council (NSERC) of Canada provided me with financial assistance during the months when I needed it the most. I will always remember the kindness of the support staff of the Department of Computing Science, specifically that of Edith Drummond, Karen Lochmanetz, Debi Owens, Ollie Frost and Rod Johnson.

My good friend, Dr. Hossam Hassanein was always there to provide me with information and advice on almost anything - academic or non-academic. Finally, I am grateful to my wife, Lalitha, and my children, Gayan, Namal and Navin, for acceptin. my excuses, though unwillingly, for being away from home most of the time.

Table of Contents

List of Tables

List of Figures

List of Symbols

N	number of stations
T	packet transmission time
τ	end-to-end medium propagation delay
a	ratio of end-to-end propagation delay to the packet transmission time
$\tau_{i,j}$	propagation delay between stations i and j
U	channel utilization
α	tap ratio of a fiber-optic switch
θ	time taken by a station to detect BOC on a bus
β	time taken by a station to detect EOC on a bus
γ	time taken by a station to start its actual transmission once it decides to transmit.
$t_{st,y}^{x}$	time instant of the start of x^{th} transmission cycle by station y
$i(j)$	identity of the station that started the r^{th} $((r+1)^{th})$ transmission cycle
$k(l)$	number of stations in ready state at time $t_{st,i}^{r}$ $(t_{st,j}^{r+1})$, which are located downstream from i (j) on the R-L bus
n	cycle length of r^{th} cycle
e	identity of the station that transmitted last in the r^{th} cycle
t_c^r	cycle time of r^{th} cycle (i.e., $t_c^r = t_{st,j}^{r+1} - t_{st,i}^{r}$)
$t_{ent,x}$	time instant of the server entering station x
$t_{dep,x}$	time instant of the server departing station x
$\tau_{x,y}$	propagation delay from station x to station y
$a(b)$	type of r^{th} $((r+1)^{th})$ transmission cycle (*R-L* or *L-R*) in X-Net
$c(d)$	mode of station i (j) just before the start of r^{th} $((r+1)^{th})$ cycle (*random* or *controlled*) in X-Net

List of Abbreviations

ACL	Accumulated cycle length
AIT	Accumulated idle time
AS	Activity signal
BOC	Beginning of carrier
CFR	Cambridge Fast Ring
CSMA/CD	Carrier Sense Multiple Access with Collision Detect
DQDB	Distributed Queue Dual Bus
EC	Empty Cycles (a parameter used in X-Net)
ECC	Empty Cycle Counter
EOC	End of carrier
EOT	End of train
FDDI	Fiber Distributed Data Interface
FIFO	First in first out
HSLAN	High speed local area network
ITC	Idle time counting
LAN	Local area network
L-R	Left-to-Right
MAN	Metropolitan area network
MLAN	Multi-channel local area network
NAC	Network access controller
R-L	Right-to-Left
THT	Token holding time
TLP	Tokenless protocol
TRT	Token rotation time
TTRT	Target token rotation time
WAN	Wide area network

Chapter 1

Introduction

1.1. Local Area Networks

With the advent of cheaper and powerful microprocessors, there has been a tendency within the past two decades to move from large centralized computers towards smaller machines. Along with the wide spread use of a large number of smaller, autonomous machines, the advantages gained by providing access to an interconnection facility became apparent. Today, computer networks provide interconnection among various computers and peripheral devices (such as printers and mass storage devices), and allow sharing of information and expensive resources among different users. Local area networks (LANs) are a subset of computer networks that provide networking facilities within a limited geographical area. In addition to the short distances spanned, compared to wide area networks (WANs) LANs are characterized by the following:

(a). High data rates

LANs usually use much higher data rates (typically over 1 Mbps) than WANs.

(b). Low error rates

Local networks generally experience much fewer data transmission errors (bit error rates in the range of 10^{-8} to 10^{-11}) than long haul networks.

(c). Simpler routing

Many LANs possess the message broadcast feature and thus do not require any message routing algorithms. Even without the broadcast facility, the routing algorithms employed are usually much simpler than in WANs.

(d). Ownership by a single organization

A LAN is typically owned and used by a single organization because of its limited geographical coverage. This leads to less administrative and maintenance complexities and costs.

(e). Lower communication costs

Lower error rates, simpler (or, absence of) routing algorithms and lower administrative and maintenance costs combine to make overall communication costs in a LAN lower than that of a WAN.

Apart from the LANs and WANs, recently another category of computer networks has emerged. These are metropolitan area networks (MANs), which share some of the characteristics of both LANs and WANs. A MAN can be viewed as a very large LAN, using access protocols less sensitive to network size than those used in LANs [Mollenauer 1988]. A main objective of MANs is to provide interconnection of LANs. Thus, they typically employ very high data rates, cover larger distances than LANs and are capable of supporting data, voice and video communication services.

An early paper that gives a good introduction to LANs is [Clark et al. 1978]. Recent papers surveying local area network technology and access protocols are [Stallings 1984c], [Sachs 1988] and [Abeysundara and Kamal 1991].

1.2. LAN Performance Measures

A variety of measures have been used in evaluating the performance of LANs. Three mostly used performance measures are: *information throughput, channel utilization* and *(various forms of) delay.* Information throughput can be defined as the total number of information bits transmitted per unit time [Bux 1984]. Even though a certain number of overhead bits for addressing, error control and other administrative purposes are transmitted in addition to the information bits, these overhead bits are excluded in calculating the information throughput. Channel utilization can be defined as the fraction of time spent in transmitting information bits, as opposed to the total time spent in transmitting information as well as overhead bits [Fine and Tobagi 1984]. To achieve high channel utilization, the waiting time until the start of a successful packet transmission and the overhead associated with the transmission should be kept low. These overheads typically consist of the transmission of preambles for receiver synchronization (in asynchronous schemes), and time spent in transmitting address, error control and other protocol dependent control information.

Delay can be defined in several forms, depending on the time instants considered in the measurement of delay. One measure is the *mean transfer time* of packets. This is defined as the average time interval from the generation of a packet at the originating station until its complete reception at the destination [Bux 1981]. This time interval consists of the time spent in a queue at the originating station until the packet moves to the head of the queue (normally termed, *queueing delay*), waiting time at the head of the queue until the beginning of successful transmission of the packet (defined as the *access* or *insertion delay*), the packet transmission time, and the medium propagation delay. For stations with single buffers, the *queueing delay* defined above will be zero. This is because, when the station buffer is not empty, either the packet generation process is considered to be inhibited, or, packets generated are considered to be lost. Another measure of delay is the time interval between the start (or, end) of two consecutive successful transmissions by a station, termed as the *station cycle time.*

In addition to the performance, some of the other important considerations in the evaluation of LAN schemes are the cost, ease of implementation, fairness, reliability, maintainability and the expandability.

1.3. The need for High Speed LANs

Chlamtac and Franta discuss the rationale behind the need for ever increasing networking speeds in [Chlamtac and Franta 1990]. The dramatic increase in computer processing power over the last few years, the enormous increase in the volume of stored or processed data accompanied by the declining costs of storage and optical transmission media, and the need to interconnect low speed network segments are motivating factors for high data rate networks. A low speed local network would quickly become the bottleneck between devices needing to transfer huge amounts of data with very low delay. As an example, a LAN used as a back-end network connecting computers to other storage devices and peripherals has to operate at least as fast as the devices on it, in order to minimize buffering constraints. As the technology pushes the speeds of hard disks and optical disks further up, back-end networks need to operate even faster [Joshi and Iyer 1984]. Apart from these, advances in computer and communication technologies have widened the application areas of LANs. Distributed computing systems and office automation are examples of different areas where increasing use of LAN technology is made. The integration of different services on the same network require LANs to cater for not only data, but also voice and video. These new applications further increase demands on the performance of LANs. LANs are required to provide not only high channel throughputs but also satisfy stringent delay requirements. Therefore, in meeting these increasing demands, it is essential that future LANs should be capable of operating at much higher data rates, achieving high channel efficiencies and lower delay.

1.4. Performance Degradation of LANs at High Speed

In a high speed local area network (HSLAN), the packet transmission time T will be comparatively small due to the high data rate of the channel. Even for sufficiently large packet sizes, it is possible that T becomes comparable to, or even less than the channel propagation delay τ. If a is defined as the ratio of the end-to-end channel propagation delay to the packet transmission time (i.e., *normalized* channel propagation delay), this means that a could be close to 1 or even greater in HSLANs. This will be a key characteristic in a local network operating at very high speeds. Therefore, the architecture and the protocols used in such a network should be able to achieve acceptable performance even under high values of the *normalized propagation delay* a.

Generally, the overhead associated with a medium access protocol increases with the propagation delay. For example, in the IEEE 802.5 token ring [IEEE 1985c], assume that all stations are backlogged (i.e., having packets for transmission) and the packet transmission time is less than the round-trip propagation delay. Then, the overhead between two consecutive packet transmissions consists of the time spent since the end of a packet transmission by a station until the reception of token by the next ready station. This overhead time depends on the round-trip medium propagation delay and the propagation delay between the two stations considered. When a is small (i.e., much less than 1), the packet transmission time is dominant compared to the medium propagation delay. Therefore, the fraction of total time spent in information transmission is large. In most of the medium access protocols therefore, high channel utilizations and lower delays can be achieved when a is small.

When a is high (close to, or above 1) however, the medium propagation delay is the dominant factor, compared to the packet transmission time. Therefore, generally, the fraction of time spent in information transmission is low. This is because of the large fraction of overhead caused by the comparatively high propagation delay. This results in a rapid degradation in performance with increasing a in many LAN schemes.

In several papers [Bux 1981; Bux 1984; Stallings 1984b; Tobagi and Hunt 1980; Tobagi et al. 1983], the performance degradation of CSMA/CD bus schemes (for example, Ethernet [Metcalfe and Boggs 1976]) with increasing a is discussed. Performance results reported in [Bux 1981] and [Tobagi et al. 1983] show that the CSMA/CD bus behaves satisfactorily as long as the *normalized propagation delay* a is sufficiently low. When a increases, performance degrades rapidly. The same behaviour can be observed when the CSMA/CD protocol is used on other topologies. This is because, with decreasing ratio of packet transmission time to *slot length* (*slot length* must be at least equal to the maximum round-trip propagation delay of the system), the protocol overhead increases significantly in terms of the fraction of time lost for collisions and their resolution [Bux 1984]. Therefore, CSMA/CD schemes are not suitable for networks with high transmission rates, small-size packets, and long distances because under these circumstances a could become high.

In [Fine and Tobagi 1984], the performance of several bus schemes as a function of a is analyzed in detail. The analysis shows how the performance of different bus protocols are affected to a varying degree with increasing a. In some bus schemes considered in the paper, performance degrades rapidly with increasing a, indicating that they are not very suitable to perform as HSLANs.

1.5. Related Work

Several network schemes have been proposed in the recent past to function as HSLANs. Many of these are capable of using, or are designed to use optical fibers as the communication medium. This is due to several desirable characteristics of optical fibers. These include:

- very high bandwidth
- protection against electromagnetic interference and signal leakage
- better signal attenuation characteristics compared to coaxial cables
- small size and light weight
- more elastic than conventional twisted, shielded wire pairs and coaxial cables.

A major problem in using optical fibers is the difficulty in aligning and joining two fibers during field installations. Usually, special equipment is required for joining fibers without introducing high losses. Further, compared to metallic media, optical fibers are fragile. Therefore, special shielding may be necessary to make them sufficiently robust. This problem is being alleviated with the introduction of plastic fibers [Tangney and O'Mahony 1988].

Several papers [Finley 1984; Limb 1984; Marhic and Tobagi 1986; Matsushita et al. 1985; Nasschi et al. 1985; Personick 1985; Rhodes 1983; Wernli 1986; Maxemchuk 1988] discuss specifically the application of optical fibers and fiber-optic devices in local area networks.

In this section, the architecture, medium access protocol, and the performance characteristics of several recently proposed HSLAN schemes are reviewed briefly. These are broadly categorized under bus, ring and star architectures.[1]

1.5.1. Bus Networks

In bus networks, all stations are connected to a transmission medium that spans over the whole length of the network. The medium and the interfaces are usually passive, and therefore bus networks are considered to be reliable. A disadvantage of bus networks is that baseband signals will suffer more attenuation and distortion compared to the shorter point-to-point links of a ring or a star network.

Several LAN schemes have been proposed, based on the linear, bidirectional bus topology (Figure 1.1.a), and its variations. Ethernet [Metcalfe and Boggs 1976] is an example of a LAN scheme that uses the linear, bidirectional bus. Variations to this basic architecture are unidirectional buses with single-folded (Figure 1.1.b), double-folded (Figure 1.1.c) and dual (Figure 1.1.d) bus architectures.

A comprehensive survey of broadcast bus LANs and their performance characteristics is contained in [Fine and Tobagi 1984]. Several bus LAN schemes have been proposed in the recent past, which are designed to operate at high data rates. The majority of these protocols fall into the category of *attempt-and-defer* schemes [Fine and Tobagi 1984]. This access mechanism is implemented on unidirectional bus systems, where there is an implicit ordering of the stations. In attempt-and-defer schemes, a ready station waits until the channel becomes idle. It then starts transmission, deferring to transmissions from the upstream stations. Some of the recently proposed HSLANs using this access mechanism are reviewed in the following sections.

1.5.1.1. Expressnet

Expressnet [Tobagi et al. 1983] is a unidirectional broadcast bus system using a double-folded passive network (Figure 1.1.c). It is an *attempt-and-defer* scheme [Fine and Tobagi 1984]

[1] For a more comprehensive review, see [Abeysundara and Kamal 1991].

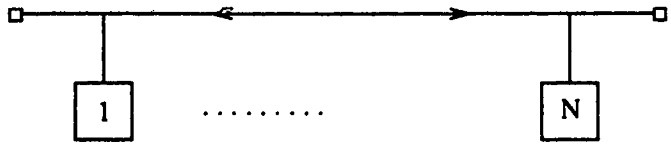
and thus its operation is based on the principle of transmission cycles. A ready station first waits for the *end-of-carrier* (EOC) event on the outbound channel (denoted by EOC(OUT)). The station then starts its packet transmission, defering to transmissions by upstream stations. A station that has transmitted successfully will not detect EOC(OUT) in the current transmission cycle again. Thus, ready stations get only one transmission opportunity per cycle, ensuring fair scheduling among stations. The transmissions in a cycle form a train of packets on the inbound channel. The end of a packet train on the inbound channel (denoted by EOT(IN)) is used by all stations to start a new round.

If all stations are idle for a long period of time, there will be no EOC(OUT) event for a station to start a new transmission or EOT(IN) event for the stations to start a new round. Therefore, to avoid this, each *alive* station has to transmit a short burst of unmodulated carrier called *LOCOMOTIVE* when it detects EOT(IN). The *LOCOMOTIVE* will ensure that EOT(IN) event take place at regular intervals.

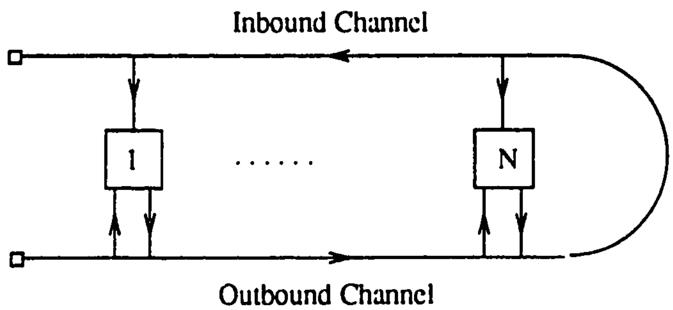
In [Marhic and Tobagi 1986], the details of a 10 Mbps fiber-optic implementation of the Expressnet are reported. The performance of Expressnet is discussed in [Tobagi et al. 1983]. In [Tobagi and Fine 1983], a detailed analysis and a performance comparison is made with the Fasnet network [Limb and Flores 1982] (to be described next). Expressnet has the advantage of achieving high channel utilization for a wider range of a values. Therefore, it appears to be suitable for operation at high speeds. A disadvantage, however, is the complexity of the access protocol, which may result in higher implementation costs [Tseng and Chen 1983].

1.5.1.2. Fasnet

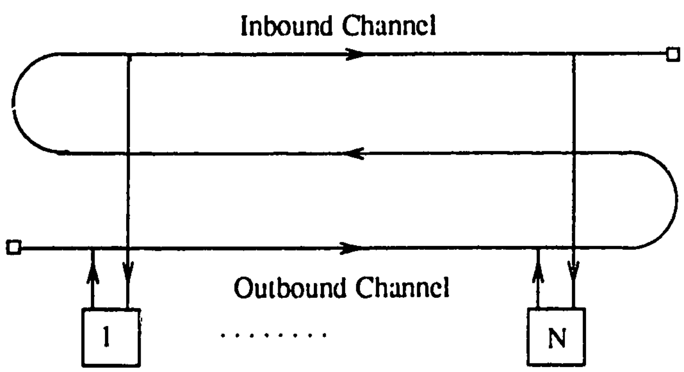
Fasnet [Limb and Flores 1982] is a conflict-free, implicit token-passing scheme utilizing two passive unidirectional buses (Figure 1.1.d). Referring to Figure 1.1.d, a station i wishing to transmit to a station j uses the R-L bus if i>j, and the L-R bus if i<j. This implies that each station should have a knowledge of the relative locations of the other stations on the network.



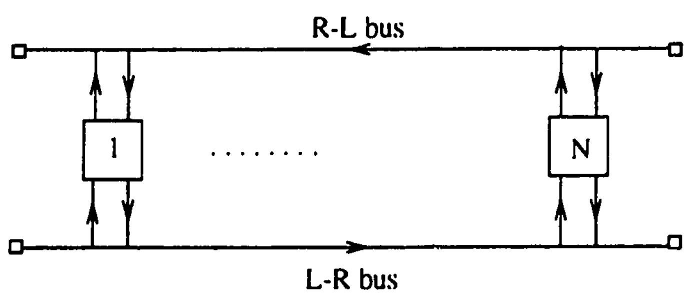
a. Bidirectional bus



b. Single-folded bus



c. Double-folded bus



d. Dual-unidirectional buses

Figure 1.1. Different bus architectures.

The operation of Fasnet is synchronous, with the time divided into *slots*. The time duration of a packet frame is one slot. The Access Control (AC) field preceding the packet contains three control bits. The START bit is set by the *head station* (the most upstream station on a bus) to indicate the start of a new cycle of transmissions. Stations read the BUSY bit to find out whether upstream stations are using the current slot for transmission. The END bit is used by the *end station* (the most downstream station on a bus) to inform the end of a cycle to the head station, so that a fresh cycle can be initiated. A normal station, after transmitting in a given frame, has to wait for the next *start of cycle* for its next transmission. This ensures the ordered fair access to the medium by all stations.

In [Fine and Tobagi 1984] and [Limb and Flores 1982], expressions are given for the channel utilization. In addition, in [Fine and Tobagi 1984], an expression for packet delay is obtained. With N continuously backlogged stations (and neglecting the length of the *access control field*), the channel utilization of Fasnet is given by $N/(N+\lceil 2a \rceil +1)$.

A disadvantage of Fasnet is that the interround overhead is at least two slots, even if the end-to-end propagation delay is very small. Because of this, the channel utilization is very low when the number of backlogged stations is small. Furthermore, unless packets are always exactly the size of a slot, some channel capacity is wasted due to the unused portion of a slot [Fine and Tobagi 1984]. Another drawback is that stations require a knowledge of the relative location of other stations on the network. Whenever there are station additions, removals or other changes in station locations, the location information in all stations therefore has to be updated accordingly.

1.5.1.3. D-Net

D-net [Tseng and Chen 1983] is a LAN scheme designed for implementation with an optical fiber transmission medium. The architecture is a single-folded unidirectional bus (Figure 1.1.b).

The medium access protocol of D-Net is very similar to that of Expressnet. The *Locomotive Generator* is centralized, in contrast to the distributed locomotive generation in Expressnet. The channel utilization and the packet delay of D-net are shown to be the same as that of Expressnet. However, D-Net has the additional advantage of simplicity. Compared to Expressnet, the topology is reduced from a double-folded to a single-folded bus. No *cold-start* procedure is required. The introduction of a separate *Locomotive Generator* simplifies the station design. A disadvantage however, is the vulnerability of the network to *Locomotive Generator* failures. Therefore, suitable redundancy measures have to be used to improve the network reliability.

1.5.1.4. Buzz-Net

The Buzznet architecture [Gerla et al. 1983] consists of two unidirectional buses (Figure 1.1.d) as in Fasnet, and is designed to operate with optical fiber as the transmission medium. The access protocol is a hybrid of random access and virtual token schemes. A special *buzz* pattern is used to signify the transition from random access to virtual token controlled mode.

A ready station is initially in the random access mode. When both buses are sensed idle, the station transmits on both buses. If one bus is sensed busy, the station waits until that bus becomes idle to start transmission. If both buses are busy, the station moves to the *controlled access* mode. Then the station starts a *buzz* signal transmission procedure, described in detail in [Gerla et al. 1983]. The purpose of the *buzz* signal is to choose a station as the starter of the next cycle. Thus, at the end of this procedure, a transmission cycle in the *controlled access* mode is established with the leftmost station in the network starting the cycle. Each ready station then transmits when the L-R (left-to-right) bus becomes idle, while defering to upstream transmissions. At the end of the transmission cycle, when both buses are sensed idle for a period of 2τ, stations switch back to the *random access* mode.

The performance of Buzznet is evaluated in [Gerla and Wang 1987] and [Gerla et al. 1987] using simulation results and approximate analytic models. Simulation results of Buzznet and several other protocols are presented in [Gerla et al. 1985]. The main advantage of Buzznet is its zero packet delay at light loads. A disadvantage of Buzznet is that, even with only two ready stations, a station may switch back and forth between the two access modes. Therefore, when load increases, the delay performance of Buzznet deteriorates rapidly. Further, the maximum achievable channel utilization (under heavy load) is given by N/(N+6a), compared to N/(N+2a) in Expressnet, where N is the number of stations.

1.5.1.5. Token-less Protocols

In [Rodrigues et al. 1984], three versions of a protocol suitable for fiber-optic LANs with a dual-unidirectional bus architecture (Figure 1.1.d) are proposed.

The operation of the basic version of the token-less protocol (TLP-1) is as follows. A ready station waits for either EOC on a bus or the time-out event indicating the *network dead* (ND) condition. If ND event occurs first, the station has to perform the initialization procedure. If EOC on bus A (denoted by EOC(A), where A could be either R-L or L-R) is detected first, the station transmits an *activity signal* AS on bus A for θ seconds (where θ is the time taken by a station to detect BOC), notifying the downstream stations of the channel busy condition. If BOC(A) is detected within this θ seconds, the station aborts transmission and allows the upstream station to continue. It then waits for the next EOC. If no BOC(A) is detected until the end of AS transmission, the station transmits its packet on both buses. At the end of the packet transmission, AS is transmitted on bus B (the other bus) until BOC(B) or the time-out ES (extreme station) occurs. This transmission of AS prevents an upstream station on bus A to transmit after detecting EOC(B). Thus, in the current round, the order of packet transmission is from the most upstream station to the most downstream on bus A. This can be viewed as a virtual token propagating from the upstream end to the downstream end on bus A in the current round.

A station where the time-out ES occurs, realizes that it is an extreme station on the bus. It then performs a *round restart* to begin the next round, allowing stations to transmit from the upstream to the downstream direction on bus B. Thus, the virtual token travels in one direction on one bus in one round and in the opposite direction on the other bus in the following round.

In the first version, TLP-1, even the idle stations have to transmit the AS. The virtual token moves between the two extreme powered-on stations even if they are not backlogged. In the second version (TLP-2), the token movement is limited between the extreme backlogged stations. From the performance results reported in [Rodrigues et. al 1984], the delay performance of this version however, is worse than that of TLP-1.

In the above two versions, the time-out interval for the event ES is twice the end-to-end bus propagation delay τ. In the third version (TLP-3) of the protocol, the assumption that extreme stations do not change frequently is used to reduce this overhead between two consecutive rounds. Here, an extreme station, when it gets channel access in a round, starts a fresh round in the other direction. If a newly powered-on station becomes an extreme station, the initialization procedure is performed after a packet collision to establish the new station as an extreme station. Out of the three token-less protocol versions proposed, TLP-3 is the best in a high speed LAN environment, because it could achieve a maximum channel utilization of (N/(N+a)), where N is the number of stations.

1.5.1.6. Distributed Queue Dual Bus (DQDB)

DQDB (or, QPSX-Queued Packet and Synchronous Circuit Exchange, as it was known earlier) is a slotted system using the dual-unidirectional bus architecture (Figure 1.1.d). The architectural features of DQDB are described in [Budrikis et al. 1986] and the Distributed Queueing (DQ) protocol used in DQDB is described in [Newman and Hullet 1986]. [Newman et al. 1988] discusses the application of DQDB and the DQ protocol as a metropolitan area network. DQDB is being considered by the IEEE for its 802.6 MAN standard project [IEEE 1990].

A head station (called the Network Controller NC, at the upstream end of a bus) generates the frame synchronization on one bus and the end station generates the frame pattern at the same rate on the other bus. The duration of a frame is 125 μs (sampling period used in digital telephony) and a frame is subdivided into fixed length packet slots. The slots can be utilized on a circuit-switched as well as packet-switched basis. The network controller reserves some slots of a frame for synchronous use and allocates them to stations that have requested synchronous packets. Slots not reserved for synchronous circuits are available for packet switched communication. The access to these slots is determined according to the DQ protocol described below.

The access protocol for each bus is independent and identical. Therefore, the description of the DQ protocol here applies to the R-L bus, with the understanding that a similar protocol is employed on the L-R bus. The protocol uses two control bits, namely BUSY and REQ. The BUSY bit at the head of each slot indicates whether the slot is full and already being in use. When a station has a packet for transmission on the R-L bus, it sets a REQ bit on the L-R bus. This indicates to the upstream stations on the R-L bus that an additional station is awaiting access. Each station maintains a record of the number of stations queued downstream from itself by counting the REQ bits (in a RQ counter) as they pass on the L-R bus. Each time an empty slot passes on the R-L bus, the RQ counter is decremented by one. This is done because the empty slot that passes by will be used by one of the ready downstream stations.

When a station i becomes ready, it transfers the current value of the RQ counter to a second counter called the countdown counter CD, and the RQ counter is reset to zero. The CD counter thus has the number of downstream stations awaiting access at the time station i becomes ready. CD counter is then decremented for every empty slot that passes on the R-L bus. When the CD count reaches zero, station i transmits its packet in the first empty slot that passes by. This machanism ensures the FIFO order of packet transmissions on a bus.

During the time that a ready station awaits its turn to transmit, new REQs received on the L-R bus are added to the RQ counter. Therefore the RQ count will still be correct for the next transmission access.

In DQDB, priority operation can be achieved by having separate distributed queues for each level of priority. A separate REQ bit on the L-R bus for each level of priority and separate RQ counters for each priority level are used. A RQ counter operating at a particular level will count REQs at the same and higher priority levels. A CD counter operating at a particular priority level will, in addition to counting down the passing empty segments, be incremented for REQs received at higher priority levels. This allows the higher priority data packets to claim access ahead of already queued data packets.

From the above description, DQDB has obvious similarities to the Fasnet [Limb and Flores 1982]. These are the dual unidirectional bus architecture, the slotted nature of the access protocol and the use of a specific bit (namely, the BUSY bit) to indicate that a slot is in use. The use of the distributed queue does away with the need for transmission cycles.

In [Newman et al. 1988], the access delays (as a function of network loading) of DQ protocol and token ring protocols are compared. The DQ protocol is shown to be superior to the token ring in terms of access delay. At light load, the RQ counter of a station will be zero with high probability. Therefore, at light load, when a station becomes ready, it can transmit in the next slot, which would be free with high probability too. Thus, at very light load, the access delay of a station will be in the order of a slot. In a token ring, at light load, the access delay will be between zero and the total ring propagation delay. A major advantage of the DQDB protocol is that the maximum achievable channel utilization remains very high (very close to 1), irrespective of a. Therefore, it appears to be an ideal network to operate in a high speed environment. In DQDB, as in Fasnet [Limb and Flores 1982], stations require a knowledge of the relative location of other stations on the network. This makes possible two concurrent transmissions on the buses, resulting in higher network throughput. However, this will be a disadvantage if there are frequent station additions, removals or other station location changes in the network. It is also important to incorporate suitable redundancy measures at the head stations to enhance the reliability of the network.

At high a values (i.e., with a large end-to-end propagation delay and a comparatively small slot period) and high load, the DQDB protocol does not appear to be fair in its bandwidth allocation. Several papers [Filipiak 1989; Wong 1989; Hahne et al. 1990; Conti et al. 1990; Kaur and Campbell 1990] discuss how the DQDB protocol could behave in an unfair manner. [Filipiak 1989] and [Hahne et al. 1990] propose mechanisms to improve the fairness of the DQDB protocol. The mechanism proposed in [Hahne et al. 1990] has been adopted by the IEEE 802.6 standards committee as an option in the 802.6 standard for metropolitan area networks [IEEE 1989].

1.5.2. Ring Networks

A ring or a loop network can be considered as a sequence of point-to-point links closed in on itself [Penney and Baghdadi 1979a]. In ring networks, stations are generally connected to the ring using active interfaces. Signals are regenerated and repeated at each node. This allows larger separation between stations than in bus networks, but introduces at least one bit delay at each station interface. An often quoted disadvantage is the vulnerability of rings due to single node failures. Variations to the basic ring topology, such as dual counter-rotating rings, node skipping links with each node joined to its two immediate predecessors, bypassing failed nodes etc., have been considered to improve the network reliability. Generally, in ring networks a monitor station is required for ring initialization, re-establishing loss of control and handling other error conditions.

Well known medium access protocols based on ring topology are token rings, slotted rings, and register-insertion rings. Token-based protocols use a unique pattern called the control token to grant medium access rights to each node. The token is passed from one node to the adjacent node around the ring in one direction. A ready station has to first wait for a free token to come by. Once a free token is received, it is marked as busy and packet transmission is started. At the end of transmission, the transmitting station releases a new free token. Various token-passing schemes are possible depending on the time the free token is introduced and whether the transmitted packet is removed from the ring by the sending station or the receiving station. For example, in the IEEE 802.5 token ring, a new token is released once the header portion of the transmitted packet returns after one complete rotation.[2] In [Xu and Herzog 1988], a token ring protocol is described where packets are partially removed by the receiving station. An advantage of token-passing protocols is that the packet delay can be bounded, because of the absence of collisions. A major disadvantage is the initial waiting time to receive a free token even at very light loads. This initial waiting time could be appreciable, especially in large rings. As a means of reducing this initial delay, a token ring protocol using multiple tokens is proposed in [Kamal 1990a]. It is shown that multiple tokens improve the delay performance at light load, but slightly decrease the maximum achievable throughput if the number of tokens used is large. Increasing the number of tokens beyond a certain limit is shown to result in rapid performance degradation at high load, because of increased token overhead.

In slotted rings, one or more fixed-size slots circulate around the ring. A *ready* station awaits the arrival of an idle slot, marks it full, and inserts data in the data field of the slot. Several variations of the protocol are possible depending on the number of slots used and whether a used slot is marked *empty* by the source or the destination. The Cambridge Ring [Wilkes and Wheeler 1979] is an example of a slotted ring LAN in which a used slot is marked empty by the source. [Metzner 1985] and [Kamal 1990b] describe slotted ring protocols where a used slot is marked empty by the destination, while providing for acknowledgements. The protocol in [Kamal 1990b] requires a message to consist of a multiple number of packets and therefore relies on the use of a large number of small slots.

A new slotted-ring protocol is proposed in [Kamal and Hamacher 1990]. An important feature of the protocol is the reduced overhead in multi-packet message transmissions. Unlike in conventional slotted-rings, only the first packet of a multi-packet message contains the source and

[2] This is to facilitate the change of ring priority through requests.

destination addresses. In the subsequent packets, a *continuation* bit indicates that it is an intermediate packet of a multi-packet message. Another feature is the use of all empty slots received (including the slots used by the station in transmitting previous packets of the same message) until all packets of the message are transmitted. Simulation results show that, compared to conventional slotted-ring and token ring, the new protocol achieves improved performance at low and medium utilizations. However, it is noted that an implementation would be more complex and error recovery more difficult.

In register insertion rings [Hafner et al. 1974; Liu 1978], each *ready* station stores the data to be transmitted in a *transmit* shift register. When a suitable idle point on the ring is found, this register is switched in series with the medium and its contents are transmitted. During transmission, any incoming data received from preceding stations are temporarily stored in a *receive* shift register. In the register insertion scheme proposed in [Hafner et. al 1974], when the transmitted packet returns to the *receive* register after one complete rotation, the register is switched out of the ring. This way, the sender removes its transmitted data from the ring. Liu employs a variation to this technique where a packet is removed from the ring by the destination [Liu 1978]. As in slotted rings, an advantage of the register insertion rings is that several packets can be in transit simultaneously, resulting in high channel utilization. Further, in register insertion rings, the medium acquisition delay can be very small, and practically zero at very light load.

A comprehensive survey of ring networks is contained in [Penney and Baghdadi 1979a,b]. In [Saltzer et al. 1981], a comparison of a token-based ring with a contention-based broadcast bus (specifically, the Ethernet [Metcalfe and Boggs 1976]) is made. The comparison is based on a variety of technical and operational factors. It is concluded that one cannot make a clear case for either the contention-based bus or the token-based ring. A performance comparison of token, slotted and register-insertion ring protocols with other LAN protocols is done in [Liu et al. 1982] and [Stallings 1984b]. In [Davies and Ghani 1983], a comparative study of four ring protocols is presented. The performance of several slotted ring protocols is evaluated in [Zafirovic-Vukotic et al. 1988]. Approximate analytic models for slotted ring protocols (specifically, for Cambridge ring) are developed in [King and Mitrani 1987]. Kamal and Hamacher show how their approximate analysis of multiserver polling systems can be applied to obtain waiting time in a slotted ring [Kamal and Hamacher 1989].

Recently proposed ring networks designed to operate at high data rates include the Cambridge Fast Ring [Temple 1983; Hopper and Needham 1988] and the FDDI protocol [Ross 1986, 1989]. The operation and the performance of these schemes are described below.

1.5.2.1. Cambridge Fast Ring (CFR)

The Cambridge fast ring [Temple 1983; Hopper and Needham 1988] is similar in principle to the Cambridge Ring [Wilkes and Wheeler 1979]. It is designed to carry a raw data rate of 100 Mbps using a transmission medium such as coaxial or fiber optic cable.

The architecture of CFR consists of one or more rings connecting different types of nodes. These nodes could be *stations*, *monitors* and *bridges*. Devices are connected to the ring through stations. A monitor is used in each ring to initialize and maintain the slot structure and for fault handling. Rings are connected together in a back-to-back fashion using bridges. An integral number of fixed-size slots circulate in the ring.

A ready station has to wait for an empty slot to come by to transmit its data. It converts a specific bit (the *F/E* bit) of an empty slot to *Full* and transmits its packet. At the destination, the receiver may copy the information and sets the appropriate bits in the CRC field as a response to the sender. Thus, in addition to an error check, CRC is used for providing feedback information from the receiver to the sender. CRC is also used in passing control information from the sender to the receiver [Hopper and Needham 1988]. After a packet transmission, the sender waits for the used slot to return and marks it empty. This prevents hogging of ring bandwidth by busy stations.

In CFR, two types of slots, namely, *normal* and *channel*, are used. A *normal* slot has to be marked empty (as mentioned above) by the sender when it returns, even if the station has a backlog of packets for transmission. A *channel* slot however, can be used by a station for consecutive packet transmissions. Therefore, this mode could be used for the transfer of long data streams between two stations.

In CFR, approximately 16% of a slot are overhead bits. This is in contrast to about 58% overhead in the Cambridge ring [Wilkes and Wheeler 1979]. The performance of CFR is discussed in [Temple 1983] and [Hopper and Needham 1988]. In [Hopper and Needham 1988], expressions are obtained for the system bandwidth and the point-to-point bandwidth under *normal* and *channel* modes. In [Temple 1983], the maximum point-to-point bandwidth in *normal mode* and *channel mode* are shown for different ring lengths and number of stations. These figures indicate that the point-to-point bandwidth of a single ring could be very low (9.3 Mbps with four slots and supporting a maximum of 30 stations) in *normal* mode. As a way of achieving high point-to-point bandwidth, the use of small rings with many bridges is recommended, while keeping the packet transfer time of a bridge small. With this configuration, however, the performance of bridges, especially under heavy load plays an important role in the overall performance of a CFR network.

1.5.2.2. Fiber Distributed Data Interface (FDDI)

FDDI is a protocol standard proposed by the American National Standards Institute (ANSI) Committee X3T9.5 responsible for HSLANs. The protocol is an enhancement of IEEE 802.5 token-ring, to operate at a data rate of 100 Mbps using an optical fiber medium. An introduction to FDDI is given in [Ross 1986,1989].

FDDI topology consists of two independent fiber optic rings, each carrying data in opposite directions at a rate of 100 Mbps. If one ring fails, the other one is used to reconfigure the network. This improves the overall availability of the network, minimizing complete station isolation due to link failures.

A ready station has to capture a free token before it could transmit. After the transmission of one or more data packets, the transmitting station appends a new token. The sender is responsible for removing its packet from the ring when it returns after one rotation. Two timers are maintained in every station, the token rotation timer and the token holding timer. Each station measures the token rotation time (TRT), which is the time between the receipt of two consecutive free tokens. The token holding time (THT) limits the time for transmission of asynchronous packets by a station at token capture. THT is calculated as the maximum of zero and (TTRT - TRT). TTRT is the *target* token rotation time, decided during the initialization phase through a bidding process. Each station requests a TTRT value equal to one half of the maximum acceptable delay. At the end of the negotiation, the minimum TTRT requested becomes the operative TTRT for all stations [Dykeman and Bux 1988].

The priority mechanism of FDDI supports *synchronous* (high priority) and *asynchronous* (low priority) traffic modes. After gaining channel access, a station first transmits synchronous packets. Then it can transmit asynchronous packets if the token has been received within the TTRT. With asynchronous traffic, eight priority levels are possible.

Two important properties of the FDDI protocol have been proved in [Sevcik and Johnson 1987]. The first is that the *average* TRT is bounded from above by the TTRT. The second is that the *maximum* TRT is at most twice the TTRT. These properties guarantee that the protocol provides bounded delay for time-critical messages.

In [Schill and Zieher 1987], performance results of simulation experiments done on the FDDI protocol are reported. They have concluded that the throughput and realtime behaviour of the FDDI protocol cannot be optimized simultaneously, as they require opposite values of TTRT. For example, a high value of TTRT results in high throughput, as it allows a station to transmit several packets after one token capture. This minimizes the impact of token passing overhead, especially with long rings. A high TTRT however, causes a long channel access time for stations at high load. This results in a deterioration of real-time behaviour of the protocol. Therefore, by appropriate setting of TTRT, the FDDI protocol can either be tuned for optimum throughput or optimum real-time behaviour, but both cannot be optimized simultaneously.

Another paper that deals with the performance of FDDI protocol is [Dykeman and Bux 1988]. The performance results have led to a conclusion similar to that of [Schill and Zieher 1987], which is the trade-off between the throughput efficiency and the delay performance. Results have also revealed a potentially severe problem with the tuning of the protocol parameters - the relative performance of the priority levels strongly depends on the number of transmitting stations. Therefore, in order to guarantee a specified performance level for lower priority stations, it is necessary to adapt the parameters dynamically to the number of transmitting stations. This could be a very difficult task in practice.

1.5.3. Star Networks

In comparison to bus and ring LANs, less attention has been paid to the development of star LANs. The reasons could probably be the vulnerability of the network due to a central hub failure and the additional wiring costs compared to bus and ring topologies. However, star networks have the following advantages over other networks [Kamal 1987]: ready suitability for optical-fiber based implementations because of the point-to-point links from end nodes to the central hub, very high throughputs are achievable, and simpler end nodes (as most of the complexities are contained in the central hub).

Fibernet [Rawson and Metcalfe 1978], Hubnet [Lee and Boulton 1983] and Fibernet II [Schmidt et al. 1983] are examples of LAN schemes based on the star topology. Fibernet uses a passive central hub while Fibernet II uses an active hub. Both schemes use the 1-persistent CSMA/CD as the medium access protocol. Therefore, as already discussed, their performance degrades rapidly when a approaches 0.5. The architecture, operation and performance aspects of Hubnet are described below.

In [Kamal 1987], the performance of several star protocols is analyzed in depth using mathematical models. A new access protocol for star networks is proposed based on a semaphore mechanism. The performance of the new protocol is shown to outperform the token ring protocol and the star protocols considered. Another recent LAN scheme based on the star topology is described in [Mehmet-Ali et al. 1988]. The approach used is fast circuit switching using a central crosspoint switch.

1.5.3.1. Hubnet

Hubnet [Lee and Boulton 1983] is a LAN operating at a data rate of 50 Mbps, recently upgraded to work at 100 Mbps. The network architecture consists of a pair of matching rooted trees (called the *selection tree* and the *broadcast tree*), with optical fiber as the transmission medium. The two trees are rooted at a central node called the *hub*. The internal nodes in the trees are called *subhubs*, which are very similar to the root hub. Stations are connected to the corresponding leaves of the two trees through intelligent devices called *Network Access Controllers* (NACs). A NAC acts as an interface between the stations and the network. Each NAC can accomodate several stations.

Packets are transmitted on the selection tree. They are forwarded by each intermediate subhub towards the root hub. Once arrived at the selection side of the root hub, packets are sent to the broadcast side of the hub through the link connecting the two sides. Packets are broadcast from the root of the broadcast tree to reach all NACs. A packet received on the broadcast tree is ignored by all NACs except the destination and the source NACs. The destination NAC sends the packet to the addressed station if the checksum is correct. The source NAC treats the reception of its own packet (called the *echo signal*) on the broadcast tree as a kind of low level acknowledgement. If the echo signal is not received within a predetermined time (called the *retry time*, which depends on the round-trip propagation delay between the NAC and the central hub) the source NAC assumes that the packet has not been received by any NAC and retransmits it.

When a packet arrives at a subhub or the root hub which is busy (i.e., while processing another packet), it is ignored. Now, as there will not be an echo signal, the source NAC, after the retry time interval, will retransmit its packet. A new packet is not transmitted until the echo signal of the previous packet is received. Therefore, if a NAC has a long retry time (i.e., a NAC that could be far away from the central hub), then it could experience longer delays under heavy load conditions.

Results reported in [Lee and Boulton 1983] show that, with increasing number of NACs, longer message lengths cause link saturation at low packet generation rates. If the message length is short, on the other hand, the effective data rate on the network will become low as the proportion of frame overhead increases with decreasing message lengths.

Experimental results of performance of the Hubnet are reported in [Lee et al. 1988]. These results show that, at low loads a high precentage of packets are successfully delivered at the first attempt itself. As the load increases however, the number of retries required for the successful delivery of a packet increases. In fact, there is a non-zero probability that a packet suffers infinite delay due to very large number of retries that may be required. This is because, in the existing implementations, all NACs attempting to access the network have an equal probability of success at any time. As a result, there is no mechanism to avoid the possibility of an infinite series of contentions in which one particular NAC always loses.

A recent study on the behaviour of Hubnet is reported in [Hassanein and Kamal 1990]. The study has revealed that, contrary to the intuitive belief, increasing the retry time does not necessarily result in higher packet delay. For fixed packet lengths in a symmetric network, they show that packets which are integer multiples of the retry time could experience lower delays than with the employment of some lower retry times. Further, they show how this specific property can be used to implement priority classes in Hubnet.

1.5.4. Multi-channel Local Area Networks (MLANs)

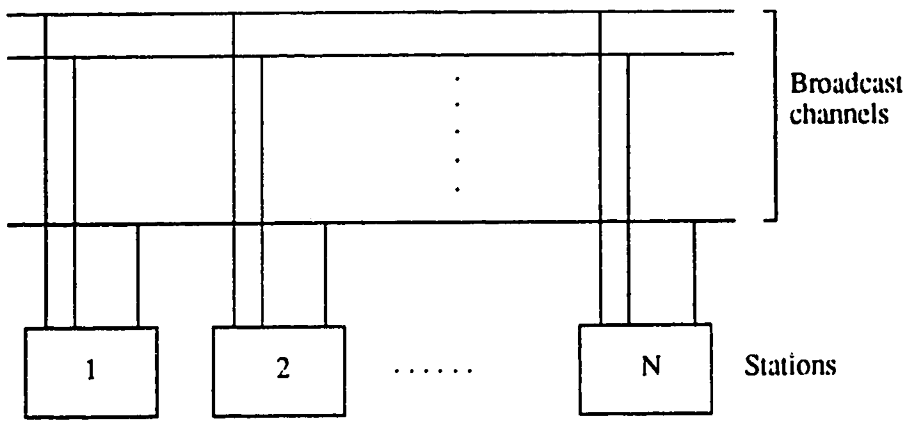


Figure 1.2. Architecture of Multi-channel LANs.

The architecture of MLANs consists of a set of parallel channels connected to all stations (Figure 1.2). The channels need not necessarily be physically separate; they can be obtained by dividing the bandwidth of a single physical connection [Marsan and Roffinella 1983]. The use of a multiple number of parallel channels could provide a very high aggregate bandwidth, while keeping the data rate on each individual channel low. Therefore, the normalized propagation delay a on each channel can be kept sufficiently low, thus avoiding any performance degradation observed with high a on a single high data rate channel. Other advantages of MLANs are [Marsan and Roffinella 1983]:

(a). The architecture is modular, allowing gradual system growth.

(b). Increased reliability and fault tolerance due to the redundant network architecture. The failure of a single channel will have a very limited impact on the network performance.

(c). Well established technologies can be used for the channel interfaces, as their individual data rates are low. Therefore, even though a higher number of channel interfaces are required, they offer an overall cost advantage over high speed channel interfaces required for high bandwidth single channels providing the same bandwidth.

An analysis of the performance of MLAN architectures is contained in [Chlamtac and Ganz 1988]. In [Marsan and Roffinella 1983], several multiple-access schemes for MLANs based on non-persistent CSMA and CSMA/CD protocols are proposed and analyzed. Two alternatives (namely, RC-Random choice, and IC-Idle channel choice) have been considered in choosing a channel by a station for its packet transmission. In RC, a *ready* station chooses a channel randomly before sensing it. In IC, a *ready* station randomly chooses a channel among those that are sensed idle. In RC approach, it is possible that packets are rescheduled even when some channels are idle. In IC, however, packets are rescheduled only when all channels are busy. It is shown that significant throughput increases with respect to the single channel case can be obtained by splitting the available bandwidth into parallel channels. The multi-channel CSMA/CD protocol with idle channel choice is shown to be the most efficient scheme, out of the protocol versions considered. In [Du et. al 1989], the idea is extended to allow a station to choose and transmit on a number of channels simultaneously, rather than just one channel. If at least one of the transmission attempts is successful, transmission is continued to end. Both the RC and IC strategies could be employed with this modified protocol. Simulation results indicate that this modified protocol is superior in performance to the original one.

1.6. Contributions and Outline of the Thesis

The main contribution in this thesis is to propose two new medium access protocols for dual-bus, fiber-optic local area networks and to analyze their performance. The two new protocols are named Z-Net and X-Net, and they are shown to perform satisfactorily in the high speed domain (i.e., even when the normalized propagation delay a exceeds 1). In addition to achieving better performance compared to some of the already proposed HSLANs, the two new schemes overcome some drawbacks in existing LAN schemes. Both protocols are completely distributed, in the sense that all stations execute the identical medium access protocol. Therefore, the vulnerability of the network to a single station failure is minimized. Stations do not require the knowledge of the relative locations of other stations in the network. Therefore, whenever a station location is changed, the information in other stations need not be updated. Analytic and simulation models have been developed to study the performance of the proposed protocols. The performance analysis shows that the channel utilization of Z-Net and X-Net does not deteriorate rapidly as a approaches 1. Both protocols exhibit bounded delay properties at all offered loads, making them suitable for supporting real-time traffic.

In the next chapter, the network architectures and the medium access protocols of Z-Net and X-Net are described. Chapter 3 contains the development of analytic models to study their performance. In Chapter 4, the performance of the proposed schemes is evaluated using the analytic and simulation results. Then the performance of Z-Net and X-Net are compared with other recently proposed bus LAN schemes. The final chapter (Chapter 5) contains concluding remarks and directions for further research. Proofs of lemmas and other detailed derivations are relegated to the appendices at the end of the respective chapters.

Chapter 2

Network Architecture and the Medium Access Protocols of Z-Net and X-Net

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2.1. Need for New Medium Access Protocols

In the previous chapter, we have discussed the need for high speed LANs and the rapid performance degradation of some of the existing LANs at high speeds (i.e., when the normalized propagation delay, a, approaches or exceeds 1). Therefore, researchers have been working on new LAN protocols that yield satisfactory performance for a wide range of a. Some of these schemes have been reviewed in the previous chapter with their merits and drawbacks.

In this chapter, we propose two new local area network schemes (named Z-Net and X-Net), that are suitable for operation at high speeds. First, the architectural features of the two networks are described. Then the salient features of the medium access protocols are described. In the last part of the chapter, features that are specific to the medium access protocols of Z-Net and X-Net are described with the aid of state diagrams. In the subsequent chapters, we model and study their performance and show that the proposed protocols indeed yield satisfactory performance even when a exceeds 1.

2.2. Architectural Features

The network architecture of Z-Net and X-Net consists of two unidirectional fiber optic buses (Figure 2.1). This architecture has several advantages: compared to the single folded bus, two independent buses can support more stations. This is because, there are fewer connection taps per bus, resulting in lower aggregate signal losses. In the case of a single bus failure, depending on the access protocol, the other bus may be used to achieve partial one-way communication among stations, or at least to detect the failure and communicate it to other stations, so that recovery action is possible. Further, depending on the access protocol again, the two buses may be used for two independent communications (as in Fasnet [Limb and Flores 1982] and DQDB [IEEE 1990]. A disadvantage of this topology over the single-folded (used in D-Net [Tseng and Chen 1983]) or double-folded bus (used in Expressnet [Tobagi et. al. 1983]) architecture is the requirement for an additional set of transmitting taps. The choice of optical fiber as the transmission medium is justified by the desirable properties they have over other media, discussed in the previous chapter.

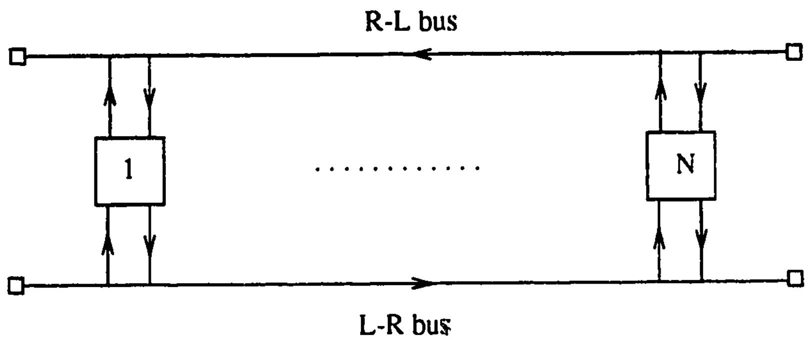


Figure 2.1. Dual-unidirectional bus architecture

In Z-Net, stations are connected to the R-L bus using passive switches (or taps), while active switches are used on the L-R bus. The X-Net architecture is similar, except that active switches are used on both buses. The operation of active switches is based on the electrooptic effect - the index of refraction of a crystal changes in proportion to an applied electric field. One possible implementation of an active switch consists of fabricating a waveguide in a crystal of a dielectric material such as $LiNbO_3$ (Lithium Niobate) [Korotky and Alferness 1988]. A detailed description of the construction and operation of active switches is contained in [Alferness 1982] and [Korotky and Alferness 1988]. In [Maxemchuk 1988], the use of active and passive switches in fiber optic LANs has been discussed. Figures 2.2 and 2.3 show the signal strengths at different points when passive and active switches are used to interface the stations to the fiber-optic buses [Maxemchuk 1988]. In these figures, X denotes the signal received from the upstream side of a fiber bus, Y is the signal injected by a station on the fiber, and α is the tap ratio. The *tap ratio* represents the proportion of the signal X that is passed to the downstream side of the fiber through the switch. In a passive switch, the tap ratio α is fixed. With α close to 1.0, a passive switch allows signals received from the upstream side of a fiber bus to pass through the switch to the downstream segment (Figure 2.2). With active switches, the tap ratio can be varied between 0 and 1 by applying a voltage V (thus, α now is denoted by $\alpha(V)$). With a high value of $\alpha(V)$, a station could sense (and copy, if necessary) the signals received from upstream section of a bus without discontinuing the signal flow. With a high value of V (thereby lowering $\alpha(V)$), a station could divert the signal flow towards point R (Figure 2.3), discontinuing the signal propagation. Thus, with the use of active switches, a station may either allow the optical signal received from the upstream side of a bus to flow towards the downstream section (i.e., from point U to D in Figure 2.3), or discontinue the signal propagation.

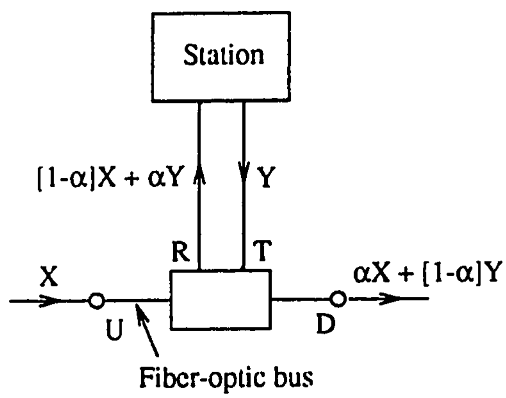


Figure 2.2. Passive switch interfacing a station to a bus

2.3. Salient Features of the Medium Access Protocols

As Z-Net and X-Net both employ hybrid-access protocols, stations can be in a *random* or *controlled* mode of operation. In X-Net, stations could additionally be in a *transient* mode. The following description applies equally to Z-Net and X-Net protocols, unless one scheme is mentioned specifically.

(a). All stations continuously monitor activities on both buses. Depending on the individual observations of each station on bus activities, each station switches from its present mode

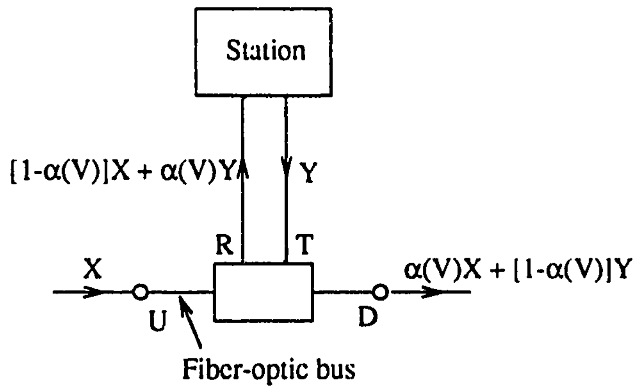


Figure 2.3. Active switch interfacing a station to a bus

to another. After initialization (or, at very light load), all stations will be in the *random* mode. In the *random* mode, if buses are sensed idle, a ready station starts its packet transmission immediately. As stations do not have to know the locations of other stations on the network, packets are transmitted on both buses simultaneously.

(b). If more than one station starts transmission in the *random* mode, packet collisions are possible. In the case of collisions, the access protocol allows one station to continue its transmission to completion, while other stations abort their transmissions. This is in contrast to schemes where all simultaneous transmissions are aborted and reattempted subsequently.

(c). Stations in the *random* mode switch to the *controlled* mode (in Z-Net) or *transient* mode (in X-Net), when they detect a signal (including signals of their own transmissions) on any bus. From the instant of leaving the *random* mode, all stations individually start counting the lengths of time intervals during which both the buses are sensed idle. We use ITC to denote this *idle time counting* operation. The total sum of the idle periods observed up to some time instant by a station i since the start of its ITC operation is called the *accumulated idle time* (AIT) of station i at that instant and is denoted by AIT_i. A station uses its AIT value to determine when to switch to the next mode.

(d). A transmission cycle in which transmission opportunities are offered to the stations in the right-to-left (left-to-right) direction is termed an R-L (L-R) cycle. In Z-Net, the cycles of transmissions are always of type R-L. In X-Net, the cycles alternate between R-L and L-R. This results in a performance improvement, as will be seen later, since this does away with an end-to-end propagation delay in detecting the end of a cycle.

(e). In the R-L (L-R) cycle, if a station becomes ready while the R-L (L-R) bus is busy, it starts transmission when the R-L (L-R) bus becomes idle. During the packet transmission, if BOC (beginning-of-carrier) is detected on the R-L (L-R) bus, transmission is aborted and the upstream transmission on the R-L (L-R) bus is allowed to progress. The AIT counter (the counter used to record the AIT value) is reset to zero and a fresh counting is started. Packet transmission is attempted again when the R-L (L-R) bus becomes idle. During transmission in the R-L (L-R) cycle, if BOC is detected on the L-R (R-L) bus, the station continues its transmission to completion. The upstream transmission on the L-R (R-L) bus is not allowed to progress any further down the L-R (R-L) bus. The station achieves this by applying a voltage V to the active switch on the L-R (R-L) bus at the beginning of its transmission and setting $\alpha(V)$ to a very low value. In the case of collisions, the use of active taps thus allows one station (which is the ready station that is most upstream on the R-L (L-R) bus in the R-L (L-R) cycle) to always complete its transmission successfully. Note that in Z-Net, as the cycles are always of type R-L, the use of active switches is necessary only on the L-R bus. In X-Net, as both types of cycles are present, active switches are used on both buses.

The following sections describe the features specific to the Z-Net and X-Net protocols with the aid of state diagrams. We start by Z-Net, since its protocol is simpler. Then, we show how the principles used in Z-Net are extended in X-Net for improving the performance.

2.4. Z-Net

The Z-Net protocol is based on two principles:

(a). A transmitting station defers to the upstream transmissions on the R-L bus. This operation can be achieved by using either passive or active taps on the R-L bus. Therefore, passive taps are used in interfacing stations to the R-L bus, since they are less expensive.

(b). When a transmitting station detects signals from the upstream side of the L-R bus, it continues its transmission to completion. The signals received from the upstream side of the L-R bus are not allowed to propagate to the downstream segment of the bus. This operation is achieved by using active taps on the L-R bus. Prior to the start of its transmission, a station applies a voltage V to the active tap on the L-R bus. This isolates the downstream segment of the L-R bus from its upstream segment, thereby discontinuing the upstream signal propagation on the L-R bus.

In Z-Net, stations could be either in the *random* or the *controlled* mode. A station in the *random* mode switches to the *controlled* mode when it detects a signal on a bus. While in the *controlled* mode, when the AIT value of a station reaches $(2\tau+\delta)$, where δ is a very small time delay that accounts for inter-packet gaps,[1] the station switches back to the *random* mode. This denotes the end of the current transmission cycle. ITC is then terminated. If the station is ready at this time, it starts transmission immediately (as it is in the *random* mode now), thereby starting a new cycle. Two consecutive transmission opportunities of a station are therefore separated by at least a time interval of $(2\tau+\delta)$.

Lemma 2.1: In the *controlled* mode of Z-Net, a station observing an AIT of $(2\tau+\delta)$ between two consecutive successful packet transmissions gives an opportunity to:

(a) all downstream stations on the R-L bus to transmit in the current cycle, and

[1] See Appendix 2.1 for an expression for δ.

(b) all upstream stations on the R-L bus to start the next cycle.

Proof: See Appendix 2.1.

When simultaneous transmissions take place, as a station defers to upstream transmissions on the R-L bus, higher numbered stations in Figure 2.1 have higher priority over lower numbered ones during a given cycle. However, the protocol does not allow unfair use of channel bandwidth by higher numbered stations because, according to Lemma 2.1 and the operation of the protocol, each station has a chance to transmit only one packet during a cycle.

The access protocol of Z-Net can be specified by a state diagram (Figure 2.4) with four states: *Idle, Ready, Txmit* and *Txmitted.* The station behaviour under each state is as follows:

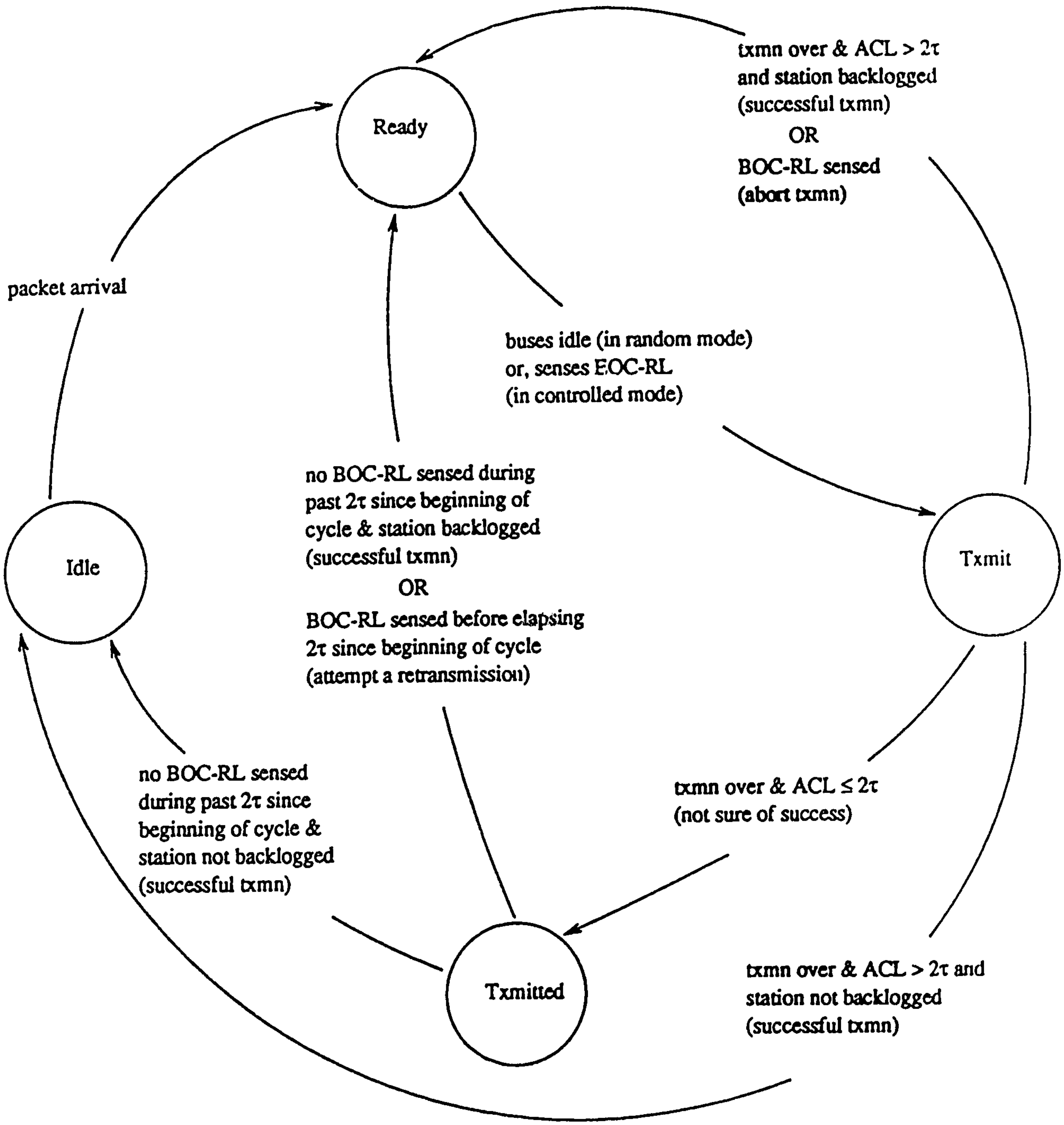
Idle : A station whose transmission buffer is empty is in this state. The station could either be in the *random* or the *controlled* mode.

Ready : A station with a non-empty transmission buffer, awaiting its turn to transmit is in this state.

Txmit : A station in this state is in the process of transmitting a packet.

Txmitted : As will be explained later, the completion of a packet transmission does not necessarily guarantee that it will be received intact by the destination. If a station is not certain about the fate of its transmitted packet (as will be explained in the next few paragraphs), at the end of a transmission it moves from the *Txmit* state to the *Txmitted* state. The station then remains in this state until such time it can determine the destiny of the transmitted packet by observing channel activities. If the channel observations indicate that the packet may not have reached the receiver, the station moves from the *Txmitted* state to the *Ready* state to attempt retransmitting the packet at the next opportunity to gain access to the channel.

When a station in the *random* mode detects BOC on a bus, it considers this event as the start of a cycle. The time elapsed from the start of a cycle until some other time instant before the end



τ : end-to-end bus propagation delay

BOC-RL : beginning of carrier on R-L bus

EOC-RL : end of carrier on R-L bus

ACL : accumulated cycle length

Idle Time Counting (ITC) is performed in all states except 'Transmit' when a station is in 'Controlled' mode of transmission.

Figure 2.4. State Diagram of Z-Net

of this cycle is defined as the *accumulated cycle length* (ACL) of the station up to that time instant. The ACL value of station i is denoted by ACL_i.

A station i in the *Idle* state moves to the *Ready* state when it receives a packet for transmission. When in the *random* mode, if both buses are sensed idle, station i moves to the *Txmit* state and starts transmission immediately. With the completion of transmission, station i moves from the *Txmit* state to either the *Idle*, *Ready* or the *Txmitted* state, depending on the accumulated cycle length, ACL_i, observed by it up to this time. Two cases are distinguished here, as stated in the following lemma:

Lemma 2.2: In Z-Net, when a station, i, completes its transmission, if ACL_i is its accumulated cycle length, then it may infer the outcome of the transmitted packet on the L-R bus as follows:

(a) If $ACL_i > 2\tau$, then it is definite that the station i's transmission on the L-R bus has not been intercepted by an upstream station on the R-L bus (which is a downstream station on the L-R bus).

(b) If $ACL_i \leq 2\tau$, the uninterrupted completion of a transmission does not necessarily imply that station i's transmission has reached all stations downstream from i on the L-R bus (because of the uncertainity that the transmission on the L-R bus has been intercepted by an upstream station j on the R-L bus).

Proof: See Appendix 2.2.

Therefore, when $ACL_i > 2\tau$, station i assumes that its transmissions on both buses have been successful. It then moves from the *Txmit* state to the *Idle* state if its transmission buffer is empty. If its transmission buffer is non-empty (indicating that there is at least one more packet awaiting transmission), it moves to the *Ready* state.

If $ACL_i \leq 2\tau$, however, as station i is uncertain of the outcome of its transmission on the L-R bus, at the end of the transmission, it moves from the *Txmit* state to the *Txmitted* state. If no BOC is sensed on the R-L bus within 2τ from the beginning of the cycle, station i is assured that no upstream station on the R-L bus has intercepted its transmission on the L-R bus. Therefore, after the elapse of 2τ from the beginning of cycle, station i moves from the *Txmitted* state to the *Ready* state if its transmission buffer is non-empty. Otherwise, it moves to the *Idle* state. The time interval of 2τ used here is an upper bound. For a pair of transmitting and receiving stations, the value that has to be actually used is the round-trip propagation delay between the two stations. However, 2τ is used (with a minor penalty in performance) as a station is not required to know the relative location of other stations on the network.

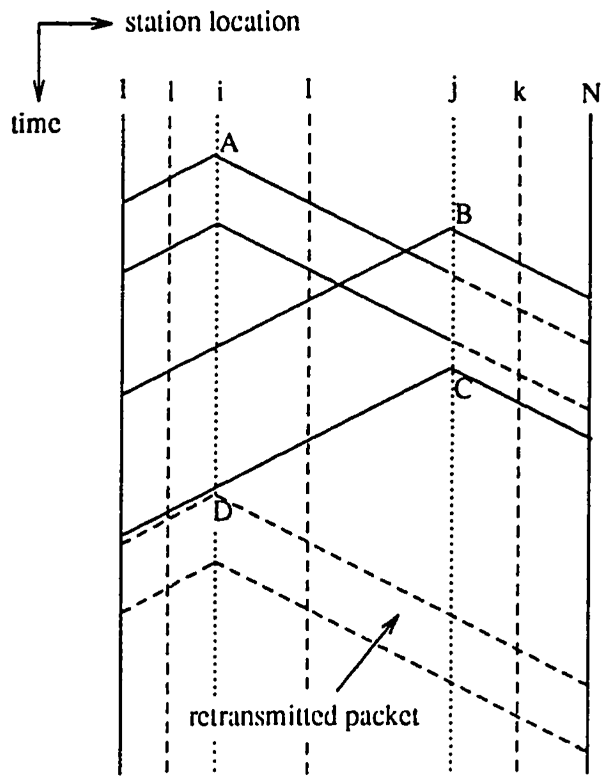


Figure 2.5. Blocking of packets on the L-R bus by station j

If a station in the *Txmitted* state detects BOC on the R-L bus within 2τ from the beginning of the cycle, its transmission could still be successful, depending on the location of the receiver on the network. For example, consider the time-space diagram in Figure 2.5. In this diagram, the vertical axis represents the time, with time values increasing in the downward direction. Locations of stations in the network are indicated in the horizontal direction, with station 1 at the left extreme and station N at the right extreme. A packet transmitted by station i propagating on the R-L bus is represented by two slanted parallel lines from station i to station 1. The same packet propagating on the L-R bus is represented by two slanted parallel lines from station i to station N. In the scenario illustrated in Figure 2.5, stations i and j are initially in the *random* mode. They start transmission as soon as they become ready (at time instants A and B in Figure 2.5). As j disconnects the L-R bus during its transmission, i's packet does not reach stations j+1,j+2,..,N. If the receiver of i's transmission is station k, ($j<k\leq N$), then i's transmission is unsuccesful. However, if the receiver is a station l such that $l<j$ (see Figure 2.5), the blocking of i's packet on the L-R bus by station j does not affect the packet reception by station l. However, as station i is not required to know the receiver's location, it pessimistically assumes that its transmission had been unsuccessful. Therefore, the packet is retransmitted as soon as the R-L bus becomes idle (at time D in Figure 2.5). If i's transmission had actually been successful (i.e., the receiver is station l and it had in fact received the complete packet) these retransmissions have to be treated as duplicates by the receiver. Using lemma 2.3, a receiver can identify such duplicates.

Lemma 2.3: In Z-Net, if a station receives two complete packets (i.e., packets which have not been aborted halfway) from the same source before its AIT reaches $(2\tau+\delta)$, then the second packet is a duplicate of the first.

Proof: See Appendix 2.3.

Thus, a packet sequence number is not explicitly required by a receiver to identify and discard duplicate packets.

2.5. X-Net

According to the Z-Net protocol, a station has to wait at least a time interval of $(2\tau+\delta)$ before it could transmit again. Therefore, the minimum time between the start of two consecutive transmission cycles in Z-Net is approximately $(T+2\tau)$.[2] The idle time counting of 2τ is used by stations in Z-Net, as the stations are not required to know their locations on the buses. If the stations know their locations in the network, we can show that this knowledge can be utilized by stations to start the next cycle in the opposite direction (i.e., in the L-R direction) at an earlier time, thereby reducing the time interval between two consecutive cycles. The X-Net protocol is based on this principle. By imposing the requirement that each station knows its location (in terms of the propagation delay to the end of the buses)[3], an R-L cycle is followed by an L-R cycle. As will be shown later, this reduces the wasted time between two consecutive cycles (averaged over a cycle) as seen by a station to τ. Therefore, X-Net achieves superior performance compared to Z-Net.

An R-L cycle in X-Net is very similar to a cycle in Z-Net (where, cycles are always R-L). Therefore, in X-Net, in the R-L cycle, a transmitting station defers to upstream transmissions on the R-L bus. During its transmission, a station blocks transmissions received from the upstream side of the L-R bus. A station does this by activating the active switch on the L-R bus when it starts transmision. The station operation in the L-R cycle is similar to that in the R-L cycle, when R-L is replaced by L-R, and vice versa. Thus, in the L-R cycle, a transmitting station defers to the upstream transmissions on the L-R bus, while upstream transmissions on the R-L bus are blocked. Therefore, in X-Net, active switches are used on the R-L bus too, as mentioned under the architectural features.

In X-Net, a station can be in one of three modes : the *Random*, the *Transient*, or the *Controlled*. A station in the *random* mode will switch to the *transient* mode when it detects BOC on any bus. In the *transient* mode, a station will assume either an R-L cycle or an L-R cycle of transmissions. How a station decides which cycle to assume in the *transient* mode will be described later. In the *transient* mode, only one cycle (either R-L or L-R) is allowed, and then the

[2] This is with just one transmission in a cycle.

[3] This can be determined by all stations during the network initialization.

next cycle of transmissions is established by transmitting a *Dummy* packet (denoted by *D-pkt*). A *D-pkt* is a very short packet of fixed length, which contains a uniquely identifiable pattern. It is transmitted on one bus only. Each station transmitting or detecting a *D-pkt* will assume the *controlled* mode. If the *D-pkt* is transmitted or detected on the R-L (L-R) bus, it denotes the start of an R-L (L-R) cycle. These *D-pkt* transmissions in X-Net are similar in concept to the *locomotive* transmissions in Expressnet [Tobagi et. al. 1983] and D-Net [Tseng and Chen 1983], and *buzz* signal in Buzznet [Gerla et. al. 1987]. Therefore, for the implementation of *D-pkts*, approaches adopted in these networks may be followed.

Under heavy load, when all or most of the stations are continuously backlogged, a station will continuously be in the *controlled* mode, alternating between the R-L and L-R cycles of transmissions. The switching from the R-L to the L-R cycle (and vice versa) in the *controlled* mode will be described later. When the network load decreases, each station will individually switch to the *random* mode, depending on the "number of consecutive *empty cycles*" that it has observed. An *empty cycle* is defined as a cycle with no transmissions, because each station happenes to be in the idle state when it gets the transmission opportunity.

2.5.1. Operation Under the Transient Mode

When more than one station start transmitting in the *random* mode simultaneously, packet collisions occur. The purpose of the *transient* mode is to resolve such collisions by establishing a cycle in the R-L or L-R direction and prepare the network to operate in a conflict-free *controlled* mode. After switching to the *transient* mode, a station assumes the R-L (or, the L-R) cycle as follows:

Before switching to the *random* mode, if the last *empty cycle* that the station had observed was an R-L (L-R) cycle, the station will assume an L-R (R-L) cycle. This principle ensures that at the start of the *transient* mode, all stations assume the same cycle (either R-L or L-R) in a consistent manner.

Suppose a station i has assumed the R-L (L-R) cycle in the *transient* mode. To switch from the *transient* mode to the *controlled* mode, and to establish the next L-R (R-L) cycle (which will be in the *controlled* mode), station i will transmit a *D-pkt* on the L-R (R-L) bus after a time interval $T_{D,i}$, measured from the start of the current cycle. If the station detects a *D-pkt* (from an upstream station on the L-R bus) before the elapse of the time interval $T_{D,i}$, it simply switches to *controlled* mode and assumes the L-R (R-L) cycle. The station does not transmit its own *D-pkt* now, because the *D-pkt* it detected would cause other downstream stations too to assume the L-R (R-L) cycle in the *controlled* mode.

In the R-L (L-R) cycle, the station i determines the time interval $T_{D,i}$ as follows:

The station first waits for an interval of 2τ from the start of the current cycle. It then starts a fresh counting of AIT from zero. When the AIT reaches $(2\tau_{i,1}+\delta)$ (or, $(2\tau_{i,N}+\delta)$ in the L-R cycle), if the station has not already detected a *D-pkt*, a *D-pkt* is transmitted on the L-R (R-L) bus. δ here is the same time delay used in Z-Net that takes into account the inter-packet gaps.[4]

Lemma 2.4: In the *transient* mode of X-Net, during the R-L (L-R) cycle, the station i ($1 \le i \le N$) waiting a time interval of $T_{D,i}$ (as determined above) to generate a *D-pkt* on the L-R (R-L) bus guarantees that:

(a) it has waited sufficient time to give opportunity to downstream stations on the R-L (L-R) bus to transmit in the current cycle,

(b) if it receives a *D-pkt* on the L-R (R-L) bus from an upstream station, it will either be:

(i) before the generation time of its *D-pkt*, or,

(ii) not later than the end of its *D-pkt* transmission.

Proof: See Appendix 2.4.

---

[4] See Appendix 2.1 for an expression for δ.

During the transmission of a *D-pkt* on the L-R (R-L) bus, stations defer to the upstream transmissions on the L-R (R-L) bus. Therefore, eventually, only the most upstream powered-on station on the L-R (R-L) bus will complete its *D-pkt* transmission. With the end of the transmission or sensing of the *D-pkt*, each station will switch to the *controlled* mode. Then, each station will assume either the L-R or the R-L cycle, depending on which bus the *D-pkt* was transmitted or detected. Figures 2.6.a and 2.6.b are time-space diagrams showing the instants of *D-pkt* transmissions, when the packet lengths are large and small, respectively. As seen from Figure 2.6.b, when packet lengths are small (such that $ACL_i<(2\tau+\delta)$), the *D-pkt* transmission does not necessarily follow the last transmission in the cycle directly. This is to give an upstream station (on the R-L (L-R) bus in the R-L (L-R) cycle) a sufficient time to start the cycle should it become ready while in the *random* mode (i.e., before detecting any BOC on a bus and switching to the *transient* mode).

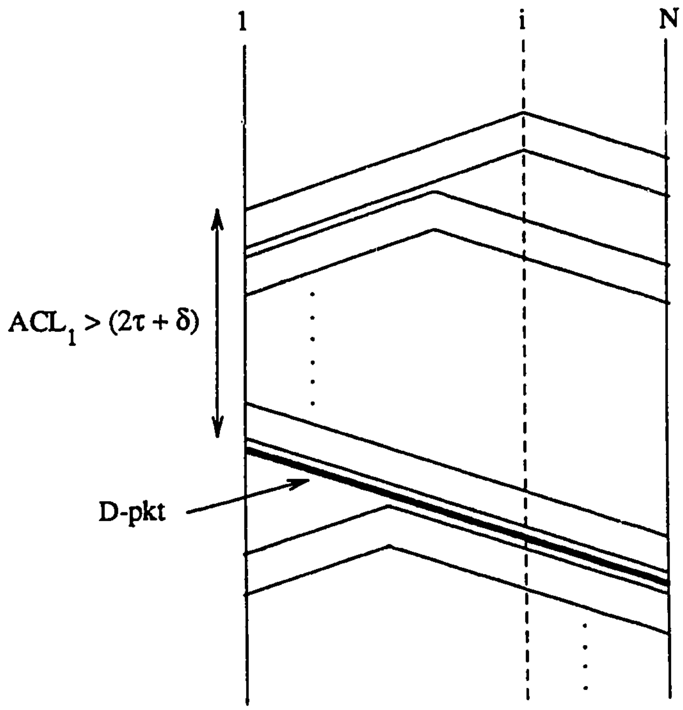
2.5.2. Operation Under the Controlled Mode

In the *controlled* mode, all stations assume the R-L (L-R) cycle, with the sensing of the end of a *D-pkt* transmission on the R-L (L-R) bus. In both cycles, all stations perform the ITC operation. Further, each station counts the number of consecutive *empty cycles* that it observes in an *Empty Cycle Counter*, ECC. ECC is reset to zero at the end of a *non-empty* cycle. Lemma 2.5 is used by a station i ($1\leq i\leq N$) to determine the end of a transmission cycle in the *controlled* mode.

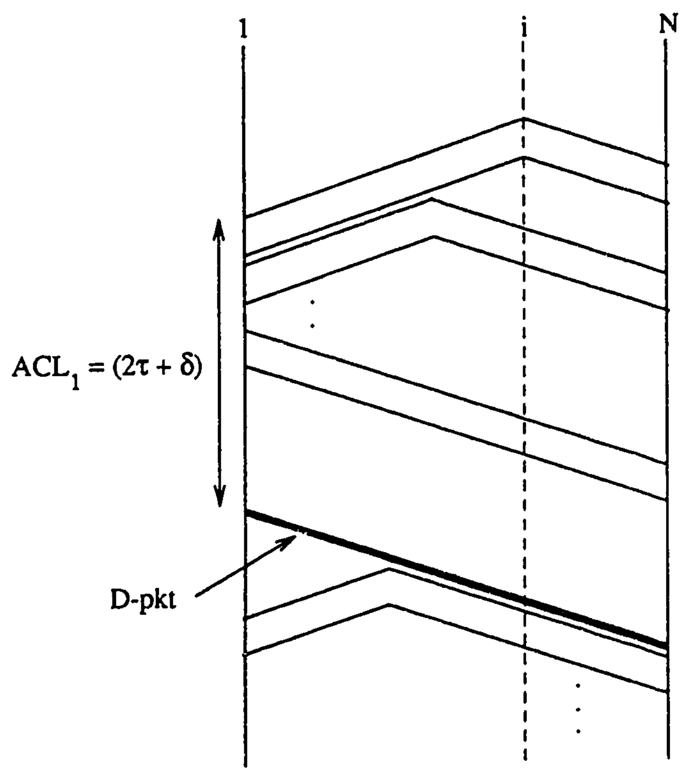
Lemma 2.5: In X-Net, during the R-L (L-R) cycle in the *controlled* mode, when the AIT value of station i exceeds $(2\tau_{i,1}+\delta)$ (or, $(2\tau_{i,N}+\delta)$ in the L-R cycle), then station i has given sufficient time for all the other downstream stations to transmit in the current cycle.

Proof: See Appendix 2.5.

At the end of a cycle in the *controlled* mode, a station executes the following procedure either to start the next cycle, or to switch to the *random* mode:



(a). when packet lengths are long.



(b). when packet lengths are short.

Figure 2.6. *D-pkt* transmissions in the *transient* mode of X-Net

```
if the cycle had been empty, then
     begin
          ECC := ECC + 1;
          if ECC < EC then
               begin
                    transmit a D-pkt on the L-R (R-L) bus;
               end
          else
               begin
                    ECC := 0;
                    switch to the random mode;
               end
     end
else
     begin
          ECC := 0;
          transmit a D-pkt on the L-R (R-L) bus;
     end
```

According to the above description, transmissions in the *controlled* mode will consist of alternating R-L and L-R cycles, separated by *D-pkt* transmissions. In the above procedure, EC (denoting *Empty Cycles*) is an integer parameter set in all stations to the same value at the network initialization time, such that EC $\geq$ 1. If EC=1, after observing one *empty cycle*, stations switch from the *controlled* mode to the *random* mode. When EC=3, for example, a station has to observe three consecutive *empty cycles* in the *controlled* mode before switching to the *random* mode.

The access protocol of X-Net can be represented by the state diagram shown in Figure 2.7. It consists of five states, four of which are *Idle*, *Ready*, *Txmit* and *Txmitted* as in the Z-Net state diagram. The station operation in these states is very similar to that in the Z-Net. The additional state in the X-Net protocol is the *D-Txmit* state. In this state, a station transmits a *D-pkt* on the R-L or the L-R bus, defering to upstream transmissions on that bus. As described later, the *Txmitted* state is applicable only when a station is in the *transient* mode and the *D-Txmit* state is applicable only when a station is in the *transient* or the *controlled* mode.

In X-Net, a station in the *Idle* state can be in any of the three modes, *random*, *transient* or *controlled*. With the arrival of a packet, an idle station moves from the *Idle* to the *Ready* state. If the station is in the *random* mode, it moves immediately from the *Ready* to the *Txmit* state, then switches to the *transient* mode, and assumes either R-L or L-R cycle (determined as described previously) to begin its packet transmission. At the completion of packet transmission, the transition from the *Txmit* state to the next st. : is influenced by whether the station is in the *transient* or the *controlled* mode. The transition from the *Txmit* state under each of these modes is described separately in the following paragraphs.

Case 1: Transition from the *Txmit* state in the *Transient* mode

In this case, at the completion of transmission, the station infers the outcome of its transmission using lemma 2.6. Based on this, the station moves from the *Txmit* to either the *Idle*, *Ready* or the *Txmitted* state.

Lemma 2.6: In the *transient* mode of X-Net, when a station, i, completes its transmission in the R-L (L-R) cycle, if ACL_i is its accumulated cycle length, then it may infer the outcome of the transmitted packet on the L-R (R-L) bus as follows:

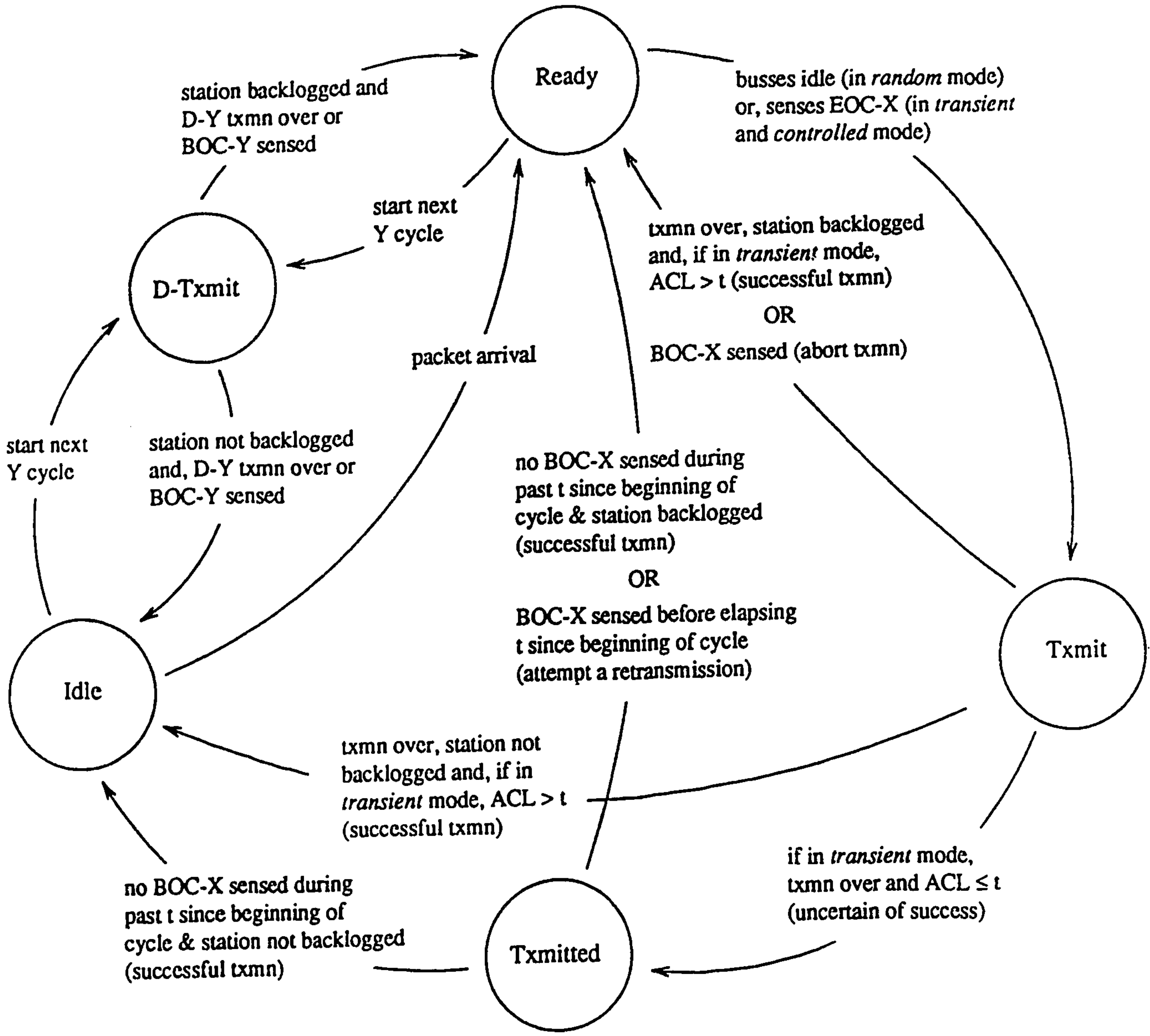
(a) If $ACL_i > 2\tau_{i,N}$ ($2\tau_{i,1}$), then it is definite that station i's transmission on the L-R (R-L) bus has not been intercepted by an upstream station on the R-L (L-R) bus.

(b) If $ACL_i \leq 2\tau_{i,N}$ ($2\tau_{i,1}$), the uninterrupted completion of a transmission does not necessarily imply that station i's transmission has reached all stations downstream from i on the L-R (R-L) bus (because of the uncertainity that the transmission on the L-R (R-L) bus has been intercepted by an upstream station j on the R-L (L-R) bus).

Proof: See Appendix 2.6.

Assume that station i had transmitted in a R-L cycle in the *transient* mode.[5] When $ACL_i > 2\tau_{i,N}$, according to lemma 6, station i assumes that its transmission on both buses is successful.

[5] Should station i had transmitted in an L-R cycle, $\tau_{i,1}$ has to be used in the following discussion, instead of $\tau_{i,N}$. Also, R-L has to be used in lieu of L-R.



Notes:

a. It is assumed that, wherever applicable, the station is currently in cycle X, where X = R-L or L-R

b. When X = R-L, Y will be L-R and vice versa

c. Time interval $t = 2\tau_{i,N}$ for station i when it is in R-L cycle.
$t = 2\tau_{i,1}$ for station i when it is in L-R cycle.

Abbreviations:

D-Y - D-pkt transmission on bus Y (Y = R-L or L-R)

BOC-X - beginning of carrier on bus X (X = R-L or L-R)

EOC-X - end of carrier on bus X

Figure 2.7. State Diagram of X-Net

Therefore, the station moves to the *Ready* state if its transmission buffer is non-empty. Otherwise, it moves to the *Idle* state. With $ACL_i \leq 2\tau_{i,N}$ at the end of the transmission, because of the uncertainity of success on one bus (L-R bus in the R-L cycle, and vice versa), the station moves from the *Txmit* state to the *Txmitted* state. If no BOC is sensed on the R-L bus within $2\tau_{i,N}$ from the beginning of the cycle, it ensures that no upstream station on the R-L bus has intercepted its transmission on the L-R bus. Therefore, after the elapse of $2\tau_{i,N}$ from the beginning of cycle, the station moves from the *Txmitted* state to the *Ready* state if it is backlogged. Otherwise, it moves to the *Idle* state.

If a station in the *Txmitted* state detects BOC on the R-L bus within $2\tau_{i,N}$ of the beginning of the cycle, the station assumes that its transmission had been unsuccessful. Therefore, the packet is retransmitted as soon as the R-L bus becomes idle. If its transmission had actually been successful (ie. the packet was received successfully and without interruption) these retransmissions have to be treated as duplicates by the receiver. As in Z-Net, the identification of a duplicate packet can be done easily: if a station receives two packets from the same source during the *transient* mode, the second packet is definitely a duplicate and is discarded.

Case 2: Transition from the *Txmit* state in the *Controlled* mode

When a station completes its packet transmission in the *controlled* mode, it moves from the *Txmit* state to the *Idle* state, if its transmission buffer is empty. If the transmission buffer is non-empty, the station moves to the *Ready* state. Note that a station does not have to move to the *Txmitted* state now because a completion of packet transmission in the *controlled* mode guarantees its success. The reason for this is that, in the *controlled* mode, a higher priority station (i.e., an upstream station on the R-L (L-R) bus in the R-L (L-R) cycle) which becomes ready after missing its turn in a cycle has to wait its turn in the next cycle. Therefore, the *Txmitted* state in the state diagram is applicable only when a station is in the *transient* mode.

In *D-Txmit* state, a station transmits a *D-pkt* on the R-L (or L-R) bus, defering to upstream transmissions on the R-L (L-R) bus. As the purpose of the *D-pkt* is to denote the start of the next cycle in the *controlled* mode, no *D-pkts* are transmitted in the *random* mode. Therefore, transitions to the *D-Txmit* state are applicable only when a station is in the *transient* or *controlled* mode.

Figure 2.8 is a time-space diagram illustrating transitions between different modes in X-Net. Initially, all stations are in the *random* mode. Stations i and j start transmissions as soon as they become ready. They switch to the *transient* mode at the start of their transmissions. Assuming that the last *empty cycle* observed by the stations (prior to switching to the *random* mode) had been of type R-L, stations i and j assume an L-R cycle. All other stations detecting either i's or j's BOC will switch to the *transient* mode and similarly assume an L-R cycle. Even though station j completes its transmission in the *transient* mode uninterrupted, it detects i's BOC before ACL_j reaches $2\tau_{j,1}$. Therefore, it retransmits the packet at time A. As explained before, the receiver will identify this packet as a duplicate and discard it. Assuming that the packet lengths are short, station N transmits a *D-pkt* at time B. Then it switches to the *controlled* mode and assumes an R-L cycle. Other stations which are in the process of transmitting *D-pkts* abort their transmissions when they detect the BOC of station N's *D-pkt*. At the end of *D-pkt* reception on the R-L bus, all stations switch to the *controlled* mode and assume the R-L cycle. Assume that the parameter EC is set to 1 in all stations. As long as the cycles are non-empty, stations are continuously in the *controlled* mode and transmission cycles alternate between types R-L and L-R. Finally, a *D-pkt* is followed by an *empty cycle* (of type R-L, for example). Stations then switch to the *random* mode individually, as shown in Figure 2.8. The next transmission cycle (not shown in Figure 2.8) following the *random* mode will be of type L-R, as the last *empty cycle* that the stations had observed was of type R-L. As seen from Figure 2.8, in X-Net, the transitions among the different modes of a station are in the cyclic order, *random* to *transient*, then to *controlled* and back to *random*. In Figure 2.8, note that there is only one cycle in the *transient* mode, while there can be several consecutive alternating R-L and L-R cycles in the *controlled* mode.

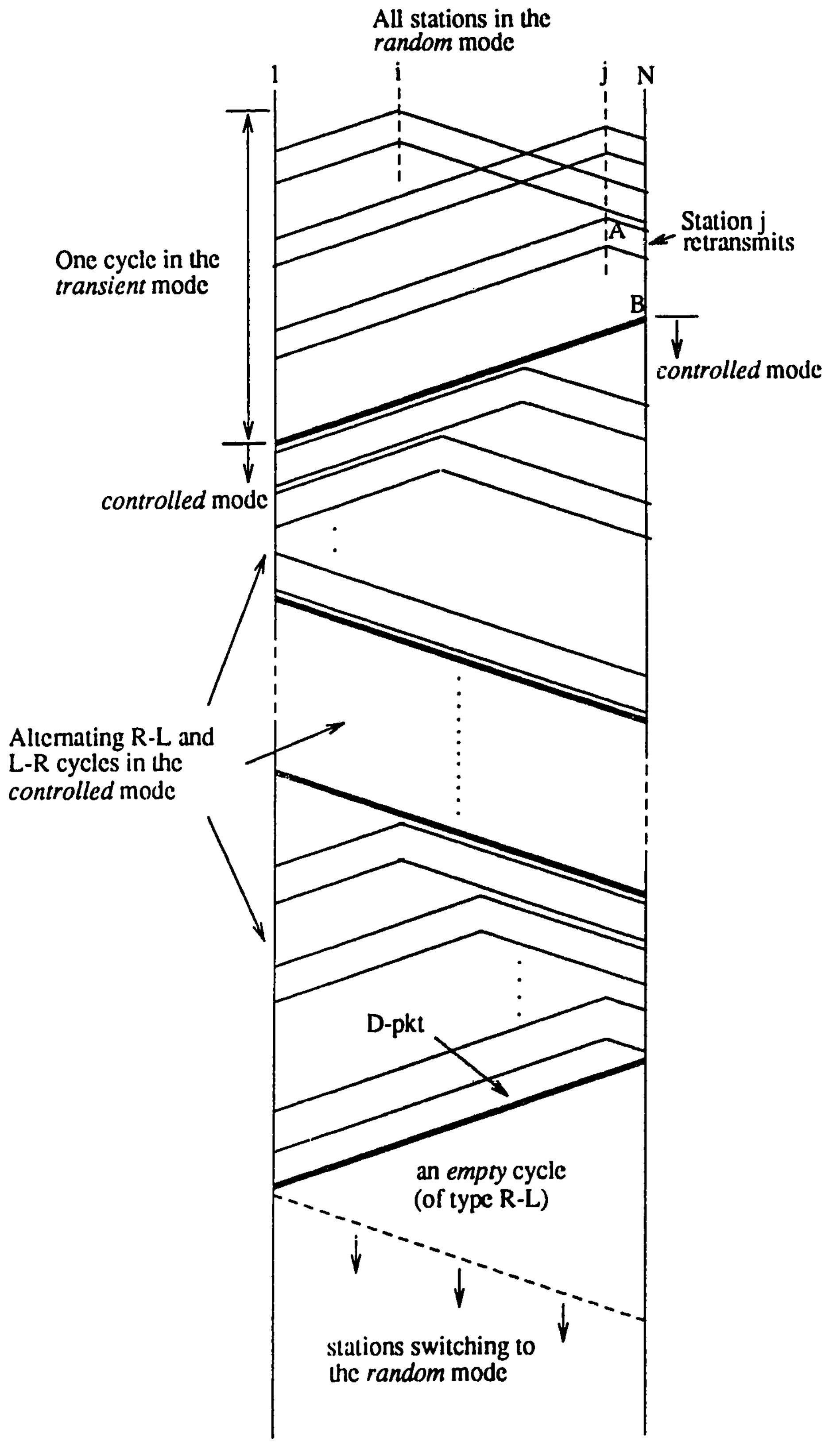


Figure 2.8. Transitions between modes in X-Net

Appendix 2.1: Proof of Lemma 2.1

(a). Figure 2.9 shows a cycle of transmissions started by station i. Consider a station j and let n be the number of stations downstream from j on the R-L bus that transmit in the cycle. Let these stations be numbered $k_1, k_2, \ldots, k_n$ from right to left. Let:

N = total number of stations

τ = end-to-end bus propagation delay

$\tau_{i,j}$ = bus propagation delay between stations i and j

θ = time taken by a station to detect the beginning-of-carrier (BOC) on a bus

β = time taken by a station to detect the end-of-carrier (EOC) on a bus

γ = time taken by a station to start transmission

During the cycle, station j will observe the bus idle intervals of $t_1, t_2, \ldots, t_{n-1}, t_n$ (see Figure 2.9), given by:

$$t_1 = 2\tau_{j,k_1} + (\gamma + \theta) + \beta \quad \text{and,}$$

$$t_x = 2\tau_{k_x,k_{x-1}} + (\gamma + \theta)\,; \quad 2 \le x \le n.$$

If station 1 transmits in the cycle (which will then be the last transmission), station j will detect its BOC at time A. At time A, AIT_j is given by:

$$AIT_j = t_1 + t_2 + \ldots\ldots + t_n + t_{n+1}$$

where,

$$t_{n+1} = 2\tau_{k_n,1} + (\gamma + \theta)$$

Therefore,

$$AIT_j = 2\tau_{j,1} + (n+1)(\gamma+\theta) + \beta \qquad \text{(A2.1.1)}$$

By waiting until AIT_j exceeds the time interval in expression (A2.1.1) above, station j gives opportunity for n downstream stations to transmit in the cycle. Should all (j-1) downstream stations transmit; it is sufficient for station j to wait until its AIT exceeds $[2\tau_{j,1} + (j-1)(\gamma+\theta) + \beta]$. The

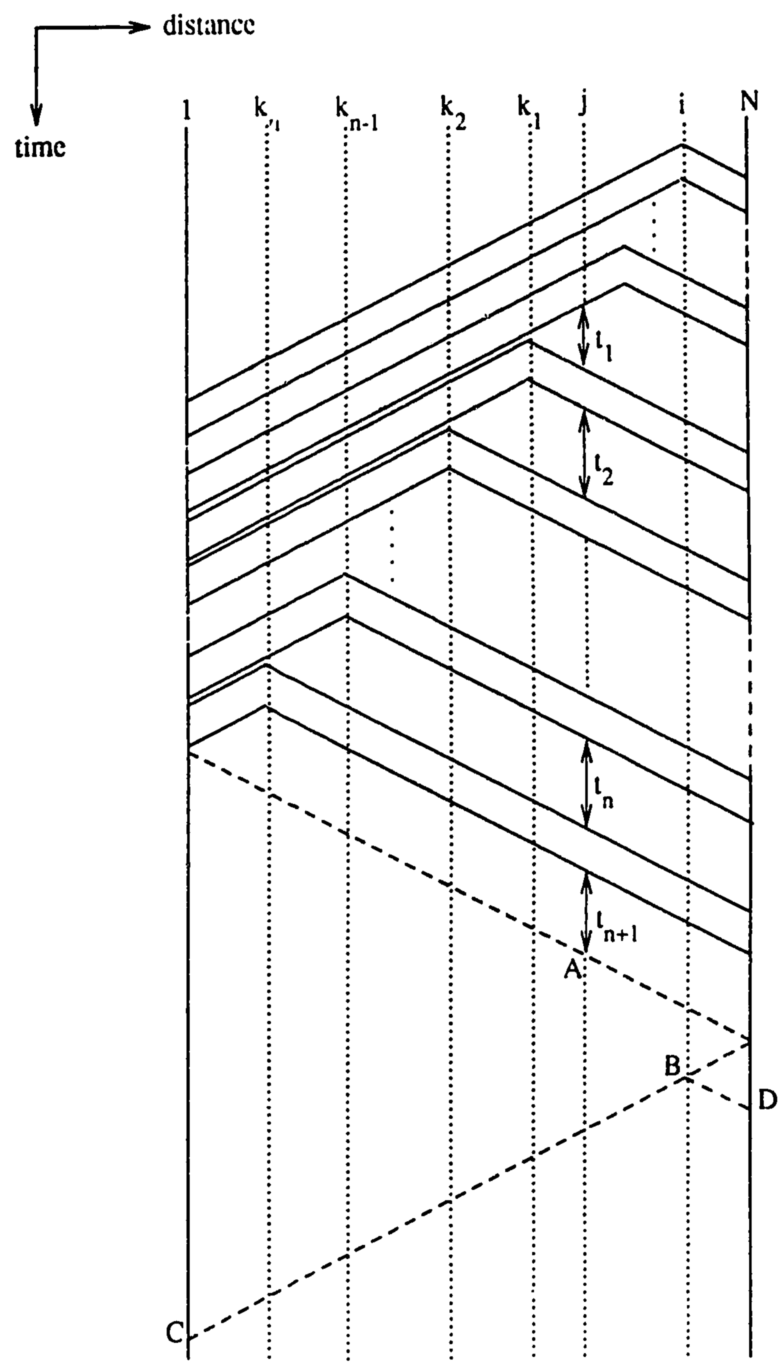


Figure 2.9. Idle time intervals seen by station j

highest AIT will be when j=N and all stations transmit in the cycle. Under this condition, AIT_N is given by:

$$AIT_N = 2\tau_{N,1} + (N-1)(\gamma+\theta) + \beta = 2\tau + (N-1)(\gamma+\theta) + \beta$$

Therefore, station N observing an AIT of $(2\tau + \delta)$, where $\delta = (N-1)(\gamma+\theta) + \beta$ in between two consecutive tra[illegible] allows all other stations to transmit during a cycle. In Z-Net, as stations are not require[illegible] locations, they all observe the same AIT of $(2\tau+\delta)$ in between two consecutive pac[illegible]sions. □

(b). In Figure 2.9, the lines BC and BD are the loci of time instances the stations 1,2,...,i and i,i+1,...,N complete the AIT of $(2\tau+\delta)$. Thus, they indicate the earliest opportunity at which a station could transmit, following the current cycle. Consider two stations p and q such that $p<q$ (i.e., q is located upstream from p on the R-L bus) and $q \le N$, which are ready and await transmission in the next cycle. When q detects p's BOC on the L-R bus, it had already started its transmission. Therefore, considering the stations p and q, q will be the potential starter of the next cycle. Thus, by observing an AIT of $(2\tau+\delta)$, a station also waits sufficient time for upstream stations on the R-L bus to start the next cycle.[6] □

Note that Figure 2.9 does not show the effect of packet collisions, which are possible at the beginning of the cycle, when more than one station start transmission in the *random* mode. In this case, the completion of AIT of $(2\tau+\delta)$ by each station will be at or after the time instances indicated by lines BC and BD. Thus, lemma 2.1 will still hold for this case.

Appendix 2.2: Proof of Lemma 2.2

(a) : When $ACL_i > 2\tau$

By the time station i finishes transmission, if $ACL_i > 2\tau$ (see Figure 2.10), it guarantees that all the stations upstream from i on the R-L bus have had an opportunity to transmit in the current cycle. Therefore, the uninterrupted completion of i's transmission implies that its packet has

[6] Eventually, the starter of the next cycle will be the most upstream station on the R-L bus which awaits its turn in the next cycle.

reached all stations downstream from i on the L-R bus without pre-emption. □

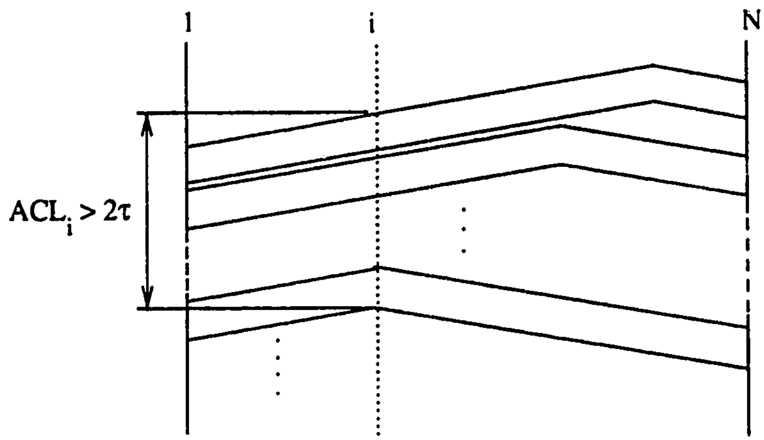


Figure 2.10. Time-space diagram when $ACL_i > 2\tau$

(b) : When $ACL_i \leq 2\tau$

As seen from Figure 2.11, stations i, j, and other stations in between i and j have completed their transmissions. However, an upstream station k has intercepted j's transmission on the L-R bus. Depending on the duration of k's and other transmissions, station k may partially or completely intercept one or more packets on the L-R bus, or, it may not intercept any transmission at all. Therefore, when $ACL_i \leq 2\tau$, station i is uncertain of the outcome of its transmission on the L-R bus. □

Appendix 2.3: Proof of Lemma 2.3

Under any load condition, the normal protocol operation does not allow a station to transmit two complete packets spaced within an AIT of $(2\tau+\delta)$ (ie., within the same cycle of transmission in the *controlled* mode, according to lemma 2.1). The only exception to this is the case where a station transmits a packet and subsequently a duplicate of it in the same cycle. Therefore, if a station receives two complete packets from the same source before its AIT reaches $(2\tau+\delta)$, the second packet should be a duplicate. □

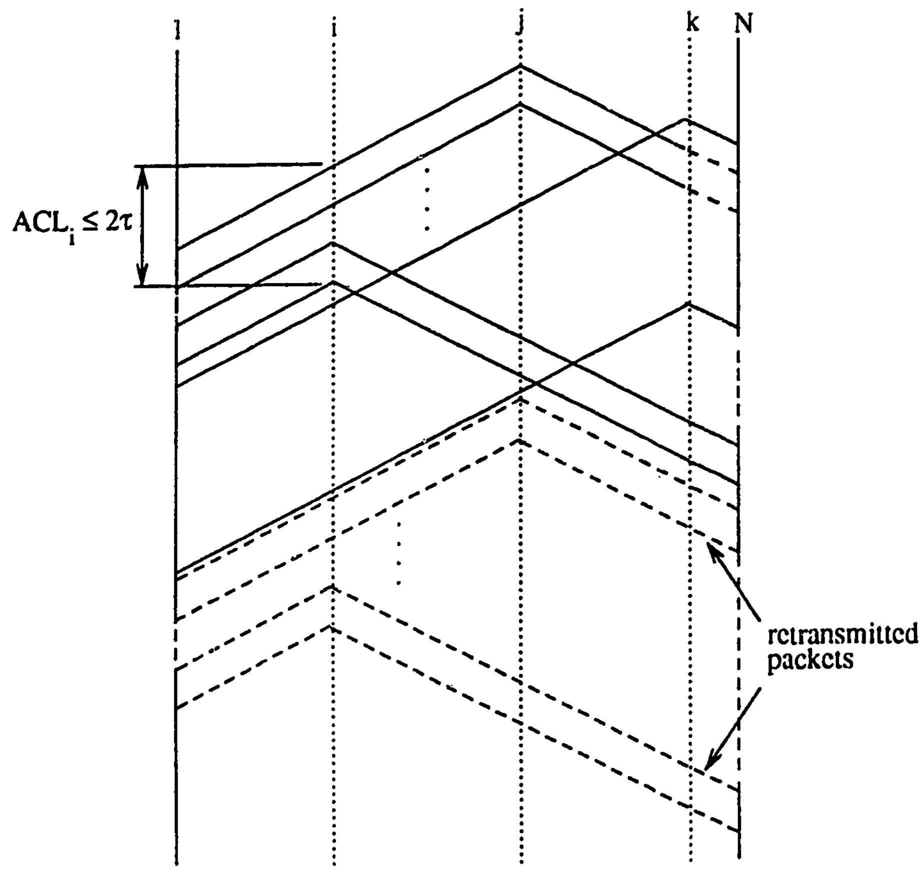


Figure 2.11. Time-space diagram when $ACL_i \leq 2\tau$

Appendix 2.4: Proof of Lemma 2.4

In the proof, we consider an R-L cycle in the *transient* mode. Same line of arguments can then be applied to prove the lemma for an L-R cycle.

(a). Station i waiting until $AIT_i > [2\tau_{i,1}+(i-1)(\gamma+\theta)]$ gives sufficient time for other downstream stations on the R-L bus to transmit in the current R-L cycle (see proof of Lemma 2.1 in Appendix 2.1). If $D_{gen,i}$ is the generation time of a *D-pkt* by station i, then:

$$D_{gen,i} = 2\tau \text{ from the start of cycle} + AIT_i \text{ of } (2\tau_{i,1}+\delta); \quad \text{where, } \delta = (N-1)(\gamma+\theta)+\beta.$$

$$\text{Thus, } D_{gen,i} > AIT_i \text{ of } [2\tau_{i,1}+(i-1)(\gamma+\theta)].$$

Therefore, by the time station i generates a *D-pkt*, all stations have transmitted in the R-L cycle. □

(b). For a station x, $1 \le x \le N$, let:

$t_{st,x}$ = the start time of an R-L cycle

$D_{gen,x}$ = scheduled generation time of a *D-pkt*

$D_{end,x}$ = expected completion time of *D-pkt* transmission

$D_{rec,x}$ = time at which station x receives a *D-pkt* from an upstream station

B_x = channel busy time seen by station x, since ACL_x reaches 2τ, until AIT_x exceeds $2\tau_{x,1}$.

Let l_D be the length of a *D-pkt*. For a pair of stations i and j, such that i>j, we have:

$$D_{gen,i} = t_{st,i} + 2\tau + B_i + 2\tau_{i,1} + \delta$$

$$D_{end,i} = D_{gen,i} + l_D$$

$$D_{gen,j} = t_{st,j} + 2\tau + B_j + 2\tau_{j,1} + \delta$$

Station i will receive station j's *D-pkt* at time $D_{rec,i}$, given by:

$$D_{rec,i} = D_{gen,j} + \tau_{j,i}$$

$$= D_{end,i} - (t_{st,i} - t_{st,j} + \tau_{i,j} + B_i - B_j + l_D) \qquad \text{(A2.4.1)}$$

To prove part (b) of lemma 2.4, we show that $D_{rec,i} \le D_{end,i}$.

In Figure 2.12, ACL of stations i and j reach 2τ at time instants A and B, respectively. They start idle time counting from these time instants. Before the AIT_i and AIT_j exceed $2\tau_{i,1}$ and $2\tau_{j,1}$, respectively, B_j can be larger than B_i by $|\tau_{i,j}-(t_{st,j}-t_{st,i})|$ at most, because B_j could include channel busy times which are not counted in B_i as a result of overlap in transmissions. This is true irrespective of whether $t_{st,i} \ge$ or $< t_{st,j}$; ($|t_{st,j}-t_{st,i}| \le \tau_{i,j}$). Therefore,

$$B_j \le B_i + (t_{st,i}-t_{st,j}+\tau_{i,j}) \quad \text{and,}$$
$$B_i - B_j + t_{st,i} - t_{st,j} + \tau_{i,j} \ge 0$$

As $l_D>0$, from expression (A2.4.1), $D_{rec,i} \le D_{end,i}$. □

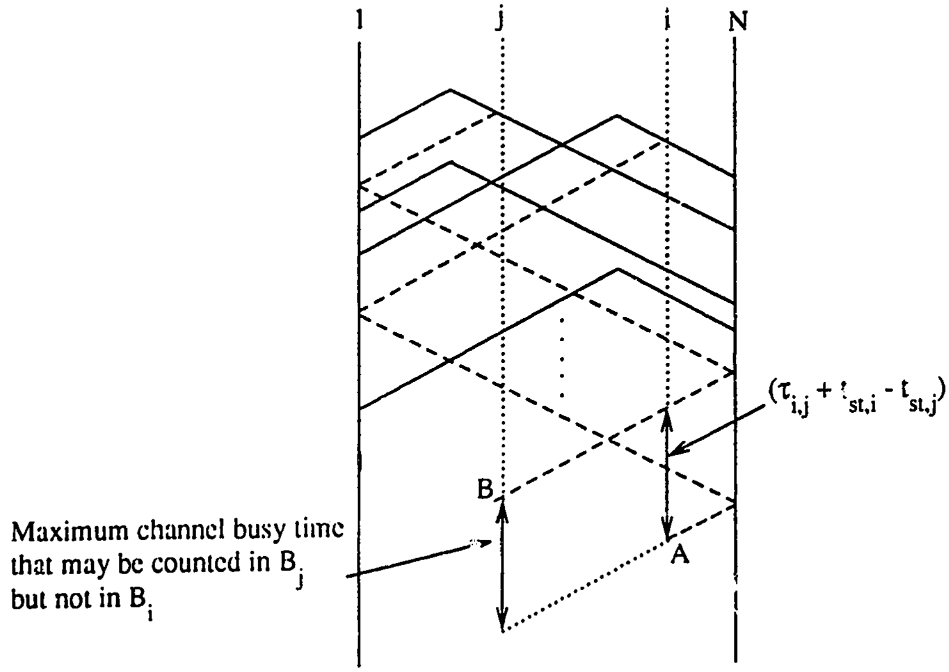


Figure 2.12. Channel busy times seen by stations i and j

Appendix 2.5: Proof of Lemma 2.5

The proof is very similar to the proof of Lemma 2.1, when an R-L cycle is considered. In Z-Net, stations have to use the value of 2τ in the AIT value, as the stations do not know their location in the network. In X-Net, in the R-L cycle, a station i, $(1 \le i \le N)$, can use $2\tau_{i,1}$ instead of 2τ, as stations know their locations. The same reasoning can be applied to an L-R cycle, with a station i using an AIT value of $2\tau_{i,N}$. □

Appendix 2.6: Proof of Lemma 2.6

Considering the R-L cycle, the proof is identical to the proof of Lemma 2.2 of Z-Net. In Z-Net, ACL_i is compared with 2τ, as the stations do not know their location in the network. In X-Net, as station i know its location on the buses, $2\tau_{i,N}$ can be used instead of 2τ. Same reasoning can be applied to prove the lemma for an L-R cycle. □

Chapter 3

Analytic Performance Modeling of Z-Net and X-Net

This chapter describes the development of approximate analytic models to study the performance of Z-Net and X-Net. First, the general properties of the models are described. Next, an analytic model for Z-Net is developed. Then, the remaining part of the chapter shows how the Z-Net model can be extended to model the behaviour of the X-Net protocol.

3.1. The General Model

A system of N stations is considered. Each station has a single buffer. Without loss of generality, stations are numbered from left to right from 1 to N, as shown in Figure 2.1. A station could either be in the *idle* or *ready* (i.e., backlogged) states. The packet inter-arrival time at each station is exponentially distributed with mean $1/\lambda$. The end-to-end propagation delay of the network is denoted by τ. The packet length is assumed to be fixed and the packet transmission time is T seconds. Each packet is preceded by a short, fixed-length preamble (for receiver synchronization) and, a packet contains some fixed number of overhead bits (for addressing, error-detection purposes, etc.). The preamble and overhead bits in a packet are assumed to be negligible compared to the information bits in the packet. It is assumed that $T \geq 2\tau$. This assumption is made so that the uninterrupted completion of a packet transmission implies its success. A station in the ready state does not generate packets and assumes the idle state when the packet transmission is completed.

For a given station, the right opportunity to start a packet transmission (as governed by the medium access protocol), is referred to as the *server entering* that station. The end of transmission opportunity is referred to as the *server departing* the station. Thus, for a station which had been in the ready state when the *server entered* it, the time instant of *server departure* is the end of that station's packet transmission. For an idle station, the time instants of *server entering* and *server departing* the station coincide.

The time interval from the start of a transmission cycle until the start of the next cycle is called the *cycle time*. The number of transmissions in a cycle is referred to as the *cycle length*. The following notations are used in the remaining sections of this chapter:

$t_{st,y}^{x}$ - time instant of the start of x^{th} transmission cycle by station y

$i(j)$ - identity of the station that started the r^{th} $((r+1)^{th})$ transmission cycle

$k(l)$ - number of stations in ready state at time $t_{st,i}^{r}$ $(t_{st,j}^{r+1})$, which are located downstream from i (j) on the R-L bus

n - cycle length of r^{th} cycle

e - identity of the station that transmitted last in the r^{th} cycle

t_c^r - cycle time of r^{th} cycle (i.e., $t_c^r = t_{st,j}^{r+1} - t_{st,i}^{r}$)

$t_{ent,x}$ - time instant of the server entering station x

$t_{dep,x}$ - time instant of the server departing station x

$\tau_{x,y}$ - propagation delay from station x to station y

For simplicity, we drop the superscript indicating the cycle number from $t_{st,i}^{r}$, $t_{st,j}^{r+1}$ and t_c^r.

3.2. Performance Modeling of Z-Net

In Z-Net, the station j (the station that started the $(r+1)^{th}$ cycle) and the number of ready stations l (downstream from j at the start of the cycle) depend only on the packet arrivals between the time instants $t_{st,i}^{r}$ and $t_{st,j}^{r+1}$. Therefore, the system can be modeled as an embedded Markovian process with *the start of a transmission cycle* taken as the embedding point. The state descriptor is taken as: *the identity of the station that started a transmission cycle* and *the number* *of stations in the ready state at the start of the cycle, that are downstream from the cycle starter on the R-L bus.*

Consider the two embedding points, the start of r^{th} and $(r+1)^{th}$ transmission cycles, with system states (i,k) and (j,l), respectively. The transition probability from state (i,k) to state (j,l) is given by:[1]

$$p^Z_{(i,k);(j,l)} = \sum_n \sum_e S_Z(j,l \mid n,e,i,k) \cdot R_Z(n,e \mid i,k) \tag{3.1}$$

where,

$S_Z(j,l \mid n,e,i,k)$ = prob($(r+1)^{th}$ cycle is started by station j, with l ready stations downstream from it at the start of cycle; given n,e,i,k), and,

$R_Z(n,e \mid i,k)$ = prob(length of r^{th} cycle is n and the last transmission in the cycle is by station e; given i,k)

In the following sections, the derivations of $S_Z(j,l \mid n,e,i,k)$ and $R_Z(n,e \mid i,k)$ are described. The terms *downstream* and *upstream* are used in relation to the R-L bus, unless the L-R bus is mentioned specifically. For example, the stations downstream from i refer to stations $i-1, i-2, .., 2, 1$ which are on the downstream segment of the R-L bus from station i (see Figure 2.1).

Derivation of R_Z (n,e | i,k) :

Define the function $D(s,g,h \mid n,i,k)$ as:

$D(s,g,h \mid n,i,k)$ = Prob(s^{th} transmission in the cycle is by station g, with h ready stations downstream from it at the start of the s^{th} transmission; given n,i,k)

To derive $D(.|.)$, assume that the $(s-1)^{th}$ transmission was by station g', which is upstream from station g. Also, assume that there were h' stations in the ready state at time $t_{ent,g'}$, where

[1] The superscript Z in equation (3.1) denotes Z-Net.

$t_{ent,g'}$ denotes the start of the $(s-1)^{th}$ transmission by the station g'. Then, for $(k+1)<s\le n$, $D(s,g,h \mid n,i,k)$ can be computed in terms of $D(s-1,g',h' \mid n,i,k)$ as follows:

$$D(s,g,h \mid n,i,k) = \sum_{g'=g+1}^{i+2-s} \sum_{h'=h'_{min}}^{h'_{max}} p_1 \cdot p_2 \cdot p_3 \cdot D(s-1,g',h' \mid n,i,k) \tag{3.2}$$

where:

$$h'_{\min} = \begin{cases} max(k+2-s, max(h-1,0)), & \text{if } g'<i \\ k, & \text{if } g'=i \end{cases}$$

$$h'_{\max} = \begin{cases} h+1, & \text{if } g'<i \\ k, & \text{if } g'=i \end{cases}$$

p_1 = prob(stations $g'-1, g'-2, \ldots, g+1$ are not ready at time $t_{ent,g'}$)

p_2 = prob(each station $x; x=g'-1, g'-2, \ldots, g+1$ does not become ready during $T+\tau_{g'x}$)

and,

p_3 = prob(station g is ready at time $t_{ent,g'} \mid$ stations $g'-1, \ldots, g+2, g+1$ were not ready at $t_{ent,g'}$) $\cdot$

prob($max(h-(h'-1),0)$ arrivals at $(g-1)-(h'-1)$ idle stations in the set of stations $\{1,2,\ldots,g-1\}$ during time $(T+\tau_{g'g})$)

+

prob(station g is not ready at time $t_{ent,g'} \mid$ stations $g'-1, \ldots, g+2, g+1$ were not ready at $t_{ent,g'}$) $\cdot$ prob(g becomes ready during $(T+\tau_{g'g})$) $\cdot$ prob($h-h'$ arrivals at $(g-1)-h'$ idle stations in the set $\{1,2,\ldots,g-1\}$ during time $(T+\tau_{g'g})$)

To derive an expression for the probability p_1, we assume that each of the stations $g'-1, g'-2, \ldots, e+1, e$, could be in the ready state at time $t_{ent,g'}$ with equal probability. Then,

prob(station g'-1 is in the idle state at time $t_{ent,g'}$) = $\left[1 - \frac{h'}{g'-e}\right]$, and

prob(station g'-2 is in the idle state at time $t_{ent,g'} \mid$ station g'-1 is idle at time $t_{ent,g'}$)

$$= \left[1 - \frac{h'}{g'-e-1}\right]$$

Following the same line of reasoning, we can obtain,

$$p_1 = \prod_{x=0}^{g'-g-2} \left[1 - \frac{h'}{g'-e-x} \right]$$

It is easy to see that p_2 is given by:

$$p_2 = \prod_{x=g+1}^{g'-1} e^{-\lambda(T'+\tau_{g',x})}$$

To derive an expression for p_3, we use,

$$\text{prob}(x \text{ arrivals at } y \text{ idle stations during time } t) = \binom{y}{x} (1-e^{-\lambda t})^x (e^{-\lambda t})^{y-x}$$

p_3 can then be expressed as:

$$p_3 = \phi \binom{g-h'}{h-(h'-1)} \left[1-e^{-\lambda t} \right]^{h-h'+1} \left[e^{-\lambda t} \right]^{g-h-1} +$$

$$(1-\phi)(1-e^{-\lambda t}) \binom{(g-1)-h'}{h-h'} \left[1-e^{-\lambda t} \right]^{h-h'} \left[e^{-\lambda t} \right]^{g-1-h}$$

where,

ϕ = prob(g is ready at time $t_{ent,g'}$ | h' stations in ready state at time $t_{ent,g'}$ and stations g'-1,...,g+2,g+1 are not ready at $t_{ent,g'}$)

$= \frac{h'}{g}$, and

$t = (T' + \tau_{g',g})$.

The model is an approximate one since the exact locations of the k ready stations at the start of the r^{th} cycle are not known. As a simplifying assumption, we assume that these k stations are the stations $i-1, i-2, ..., i-k$ (i.e., k adjacent stations downstream from i). Another assumption considered was that, these k stations could be any combination from the stations $\{1,2,...,i-1\}$ with equal probability. However, intuitively, the former assumption is more realistic for a set of stations with equal arrival rates. This is because, at high offered loads, during the previous cycle (i.e., $r-1^{th}$ cycle), the server would have departed a higher numbered station (such as $i-1$) much earlier than a lower numbered station (like 1 or 2). Therefore, a higher numbered station has a higher probability of being in the ready state at time $t_{st,i}$ than a lower numbered station. At light loads, k would most likely be 0. Therefore, the assumption about the location of the k ready stations is relevant only at higher offered loads.

With $s=n$, $g=e$ and $h=0$ in equation (3.2), the function $D(n,e,0 \mid n,i,k)$, which is the prob(n^{th} transmission in the cycle is by station e with 0 ready stations downstream from it when the server enters e, given n,i and k) can be recursively computed, with the following boundary conditions:

When k=0:

$$D(1,g,h \mid n,i,k) = \begin{cases} 1, & \text{if } g=i \text{ and } h=0 \\ 0, & \text{otherwise} \end{cases}$$

because the first transmission in the cycle is by station i.

When $k > 0$:

$D(k+1,i-k,h' \mid n,i,k) =$ prob(h' arrivals at $(i-k-1)$ idle stations during the time $(k\text{T}+\tau_{i,i-k})$)

$$= \binom{i-k-1}{h'} \left[1-e^{-\lambda t}\right]^{h'} \left[e^{-\lambda t}\right]^{i-k-1-h'}$$

where, $t = (kT+\tau_{i,i-k})$.

This is because the $(k+1)^{th}$ transmission is by the station $(i-k)$ and h' number of stations have become ready from the set $\{1,2,\ldots,i-k-1\}$ during the time interval between the server entering the stations i and $(i-k)$.

The probability $R_Z(n,e \mid i,k)$ can now be readily computed as follows:

$R_Z(n,e \mid i,k) = D(n,e,0 \mid n,i,k) \cdot$ prob(each station x, $x=e-1,e-2,\ldots,1$, in the idle state during the time interval $(\text{T}+\tau_{e,x})$)

$$= D(n,e,0 \mid n,i,k) \cdot \prod_{x=1}^{e-1} e^{-\lambda(T+\tau_{e,x})} \tag{3.3}$$

Derivation of $S_Z(j,l \mid n,e,i,k)$:

Depending on the location of station j with respect to station i, two cases are possible; Case 1: $j>i$, and Case 2: $j \leq i$.

Case 1: j>i

Under this case, three subcases 1.1, 1.2 and 1.3 are considered, depending on the length of the cycle time t_c. The reason for these different subcases is that the evaluation of $S_Z(j,l \mid n,e,i,k)$ could be somewhat different depending on the length of t_c. The conditional state transition probability $S_Z(j,l \mid n,e,i,k)$ for each of these subcases is denoted by $s_{j,l \mid n,e,i,k}^{1,x}$, where $x=1$, 2 or 3.

Subcase 1.1 : $t_c = t_j+2\tau_{j,i}$ where, $t_j = \tau_{i,1}+nT+\tau_{1,N}+\tau_{N,j}$ (see Figure 3.1)

According to the medium access protocol, station j cannot start the new cycle until t_c is at least $(t_j+2\tau_{j,i})$, even if j is ready before this time. This is the reason why we consider the smallest value of t_c to be $(t_j+2\tau_{j,i})$.

Subcase 1.2 : $(t_j+2\tau_{j,m+1}) < t_c \leq (t_j+2\tau_{j,m})$; m=1,2,...,(i-1) (see Figure 3.2)

In this case, $(i-1)$ adjacent subintervals of t_c from $(t_j+2\tau_{j,i})$ to $(t_j+2\tau_{j,1})$ are considered, with the range of each subinterval determined by m. For a given value of m, where $1 \leq m \leq (i-1)$, suppose the length of the cycle time, t_c, is such that $(t_j+2\tau_{j,m+1}) < t_c \leq (t_j+2\tau_{j,m})$. For some arbitrary value of t_c in this time interval, let $s_{j,l \mid n,e,i,k}^{1,2}(m,t_c)$ be the conditional state transition probability from state (i,k) to state (j,l), given m, n and e.

Integrating $s_{j,l \mid n,e,i,k}^{1,2}$ (m,t_c) over the time interval from $(t_j+2\tau_{j,m+1})$ to $(t_j+2\tau_{j,m})$, and summing up for all values of m, the conditional state transition probability $s_{j,l \mid n,e,i,k}^{1,2}$ is expressed as:

$$s_{j,l \mid n,e,i,k}^{1,2} = \sum_{m=1}^{i-1} \left[\int_{t_j+2\tau_{j,m+1}}^{t_j+2\tau_{j,m}} s_{j,l \mid n,e,i,k}^{1,2} \ (m,t_c) \ dt \right]$$

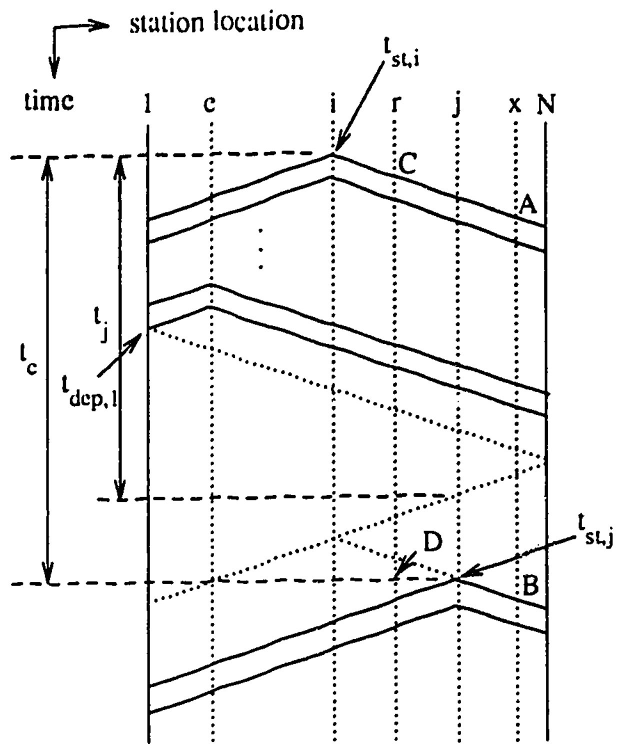


Figure 3.1. Time-space diagram for Subcase 1.1

Subcase 1.3 : $t_c > (t_j+2\tau_{j,1})$ (see Figure 3.3)

Given n, e and some arbitrary value of t_c such that $t_c > (t_j+2\tau_{j,1})$, let $s_{j,l\,|\,n,e,i,k}^{1,3}(t_c)$ be the conditional state transition probability from state (i,k) to state (j,l).

Integrating $s_{j,l\,|\,n,e,i,k}^{1,3}(t_c)$ from $t_j+2\tau_{j,1}$ to ∞, $s_{j,l\,|\,n,e,i,k}^{1,3}$ for subcase 1.3 is given by:

$$s_{j,l\,|\,n,e,i,k}^{1,3} = \int_{t_j+2\tau_{j,1}}^{\infty} s_{j,l\,|\,n,e,i,k}^{1,3}(t_c)\, dt$$

The probability $S_Z(j,l \mid n,e,i,k)$ for Case 1 is therefore given by:

$$S_Z(j,l \mid n,e,i,k) = s_{j,l\,|\,n,e,i,k}^{1,1} + s_{j,l\,|\,n,e,i,k}^{1,2} + s_{j,l\,|\,n,e,i,k}^{1,3} \tag{3.4}$$

$s_{j,l\,|\,n,e,i,k}^{1,x}$ for each subcase $x=1$, 2 and 3, is evalauated using:

$$s_{j,l\,|\,n,e,i,k}^{1,x} = p_{nr}^{1,x} \cdot p_j^{1,x} \cdot p_l^{1,x} \ ; \ x=1,2,3$$

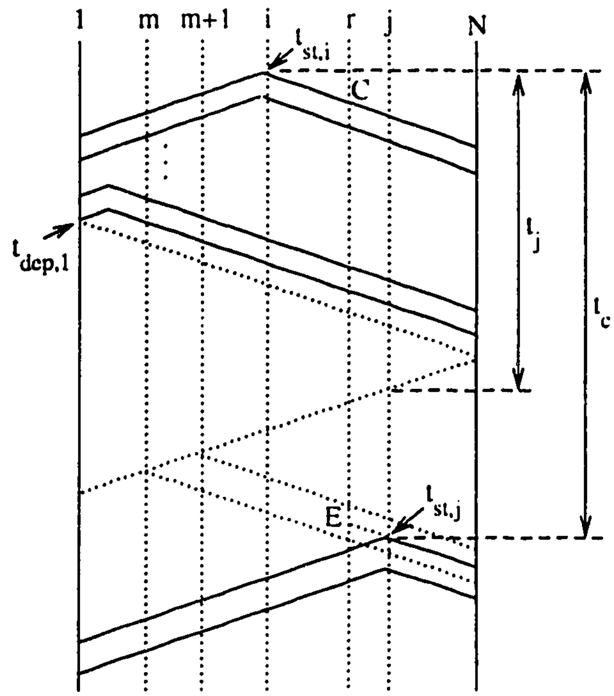


Figure 3.2. Time-space diagram for Subcase 1.2

where,

$p_{nr}^{1,x}$ = prob(stations r; $r=j+1, j+2,...,N$; remaining idle during the time interval AB in Figure 3.1),

$p_j^{1,x}$ = prob(station j in ready state at time $t_{st,j}$), and

$p_l^{1,x}$ = prob(l stations from the set $\{1,2,...,j-1\}$ in ready state at time $t_{st,j}$).

For all three subcases, $p_{nr}^{1,x}$ is given by:

$$p_{nr}^{1,x} = \left[e^{-\lambda(t_c-\tau_{i,j})}\right]^{N-j} \quad ; \; x=1,2,3$$

$p_j^{1,x}$ is given by:

$$p_j^{1,x} = \begin{cases} 1 - e^{-\lambda(t_c-\tau_{i,j})}, & \text{for } x = 1 \\ \lambda e^{-\lambda(t_c-\tau_{i,j})}, & \text{for } x = 2,3 \end{cases}$$

In evaluating $p_l^{1,x}$, note that l could be any combination of l stations from station $(j-1)$ to 1. We consider the stations from $j-1$ to $i+1$ and the stations from i to 1 as two separate sets. Let

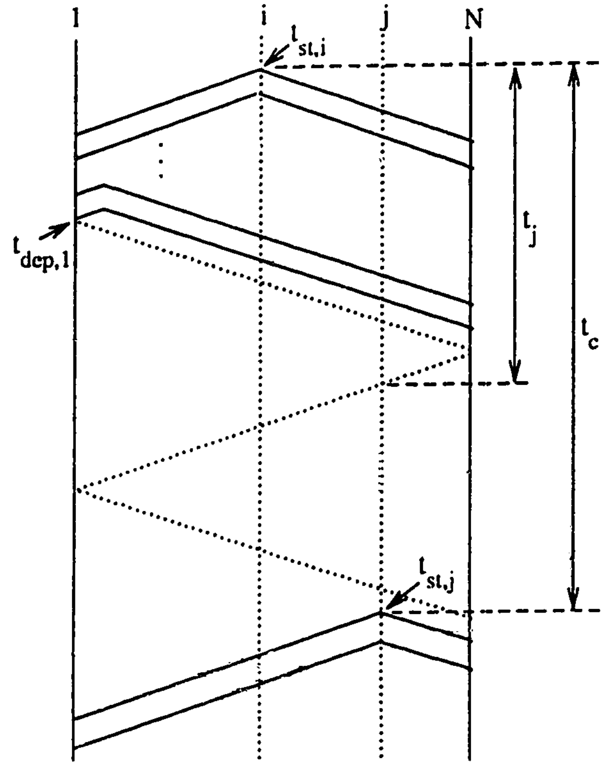


Figure 3.3. Time-space diagram for Subcase 1.3

$l = x + y,$

where,

x = number of ready stations among $\{i+1, i+2, \ldots, j-1\}$; $\max(0, l-i) \leq x \leq (j-1)-i$, and,

y = number of ready stations among $\{1, 2, \ldots, i\}$; $y = (l-x)$.

Let the probability of x be p_x and the probability of y, given x be $p_{y|x}$. $p_l^{1.3}$ is then given by:

$$p_l^{1.3} = \sum_{x=\max(0,l-i)}^{j-1-i} p_x \cdot p_{y|x} \text{ for } y = l - x$$

The derivations of p_x and $p_{y|x}$ for each subcase follow the same procedure, but the intermediate steps have some variations. For subcase 1.1, the evaluation procedure for p_x and $p_{y|x}$ are described in Appendix 3.1. Appendices 3.2 and 3.3 then outline the variations from this for subcases 1.2 and 1.3, respectively.

The derivation of $S_Z(j,l \mid n,e,i,k)$ for Case 2 (i.e., j≤i) follows a similar procedure (with three subcases, again) with minor variations from Case 1, and is outlined in Appendix 3.8.

As both probabilities in the R.H.S. of equation (3.1) can now be evaluated, the state transition probabilities $p^Z_{(i,k):(j,l)}$ for $1 \le i,j \le N$ and $0 \le k \le (i-1)$, $0 \le l \le (j-1)$ can be found. Then, using standard techniques, the steady state probabilities $\pi_{i,k}$ can be obtained.

Next, we describe how the above model can be extended for modeling the behaviour of X-Net. Then, using the steady state probabilities, expressions are derived for the mean channel utilization of both Z-Net and X-Net. Using the mean channel utilization, a common expression is then obtained to compute the mean packet transfer delay of Z-Net and X-Net.

3.3. Performance Modeling of X-Net

In X-Net, an *R-L* cycle of transmissions is very similar to a transmission cycle in Z-Net (where all transmission cycles are of type *R-L*). Therefore, the principles used in the Z-Net model can also be applied to model the behaviour of the X-Net protocol. However, the state descriptor must now be enhanced to capture the additional properties of the X-Net protocol. Firstly, in X-Net, a transmission cycle could be either of type *R-L* or *L-R*. This must be reflected in the state descriptor. Secondly, the operation mode of the cycle leader (the station that starts a new cycle) just prior to the start of the cycle could be either *random* or *controlled* (whereas this is always *random* in Z-Net). The state descriptor should also represent this *mode* information. To make these extensions to the state descriptor, we additionally introduce the following notations:

$a(b)$ - type of r^{th} ($(r+1)^{th}$) transmission cycle (*R-L* or *L-R*)

$c(d)$ - mode of station i (j) just before the start of r^{th} ($(r+1)^{th}$) cycle (*random* or *controlled*)

The rest of the notations used are the same as those used in the Z-Net model (defined in Section 3.1). In the derivations, we assume the parameter EC is 1 and the length of *D-pkts* is negligible compared to the length of a cycle.

Considering the time instant $t_{st,j}$, the station j (station that starts the $(r+1)^{th}$ cycle), the number of ready stations l (downstream from j at the start of the cycle), the type of cycle b and the mode d of station j depend only on the packet arrivals between the time instants $t_{st,i}$ and $t_{st,j}$. Therefore, the behaviour of X-Net can be modeled as an embedded Markovian process with the state descriptor as (a,c,i,k) and the start of a transmission cycle as the embedding point.

Consider the two embedding points, the start of r^{th} cycle with system state (a,c,i,k), and the start of $(r+1)^{th}$ cycle with system state (b,d,j,l). The transition probability from the state (a,c,i,k) to the state (b,d,j,l) can be expressed as:[2]

$$p^X_{(a,c,i,k);(b,d,j,l)} = \sum_n \sum_e S_X(b,d,j,l \mid n,e,a,c,i,k) \cdot R_X(n,e \mid a,c,i,k) \qquad (3.5)$$

where,

$S_X(b,d,j,l \mid n,e,a,c,i,k)$ = prob($(r+1)^{th}$ cycle is of type b and is started by station j in mode d, with l ready stations downstream from it at the start of cycle; given n,e,a,c,i,k), and,

$R_X(n,e \mid a,c,i,k)$ = prob(length of r^{th} cycle is n and the last transmission in the cycle is by station e; given a,c,i,k)

The following observations simplify the task of computing the state transition probabilities:

(a). Only the transition probabilities $p^X_{(R-L,c,i,k);(b,d,j,l)}$ need to be computed. Because of the similarity of behaviour of $R-L$ and $L-R$ cycles, $p^X_{(L-R,c,i,k);(b,d,j,l)})$ can then be obtained using the following symmetry property:

$$p^X_{(L-R,c,i,k);(b,d,j,l)}) = p^X_{(R-L,c,i',k);(b',d,j',l)})$$

[2] The superscript X in equation (3.5) denotes X-Net.

where,

$$x' = N + 1 - x \;;\; x = i, j$$

$$b' = \begin{cases} R-L, & \text{if } b = L-R \\ L-R, & \text{if } b = R-L \end{cases}$$

(b). The probability $R_X(n, e \mid a, c, i, k)$ does not depend on the mode c of the station i that started the r^{th} cycle. Therefore, when the cycle type a is R-L, $R_X(n, e \mid R-L, ., i, k)$ is identical to the probability $R_Z(n, e \mid i, k)$ of Z-Net (see equations 3.1 and 3.3).

In the light of these observations, only the derivation of the probability $S_X(b, d, j, l \mid n, e, R-L, c, i, k)$ need to be described. In the following sections, during the R-L cycle, the terms *downstream* and *upstream* are used in relation to the R-L bus. In the L-R cycle, these terms apply to the L-R bus.

Derivation of $S_X(b, d, j, l \mid n, e, R-L, c, i, k)$ **:**

As in the Z-Net model, two cases are possible; Case 1: $j > i$, and Case 2: $j \le i$.

Case 1: j>i

Four subcases 1.1, 1.2, 1.3 and 1.4 are considered, depending on the length of the cycle time t_c. The conditional probability $S_X(b, d, j, l \mid n, e, R-L, c, i, k)$ for each subcase is denoted by $s^{1x}_{b,d,j,l \mid n,e,R-L,c,i,k}$, where x=1, 2, 3 or 4.

Subcase 1.1 : $t_c = \tau_{i,1}$+nT+$\tau_{1,j}$ (see Figure 3.4)

According to the X-Net medium access protocol, the earliest time that station j can start the new cycle is after a time interval of $(\tau_{i,1}+nT+\tau_{1,j})$ from the start of the previous cycle. Therefore, the smallest possible value of t_c is $(\tau_{i,1}+nT+\tau_{1,j})$. As shown in Figure 3.4, the type of the new cycle in this case is L-R.

Subcase 1.2 : $t_c = \tau_{i,1}$+nT+ τ + $\tau_{N,j}$ (see Figure 3.5)

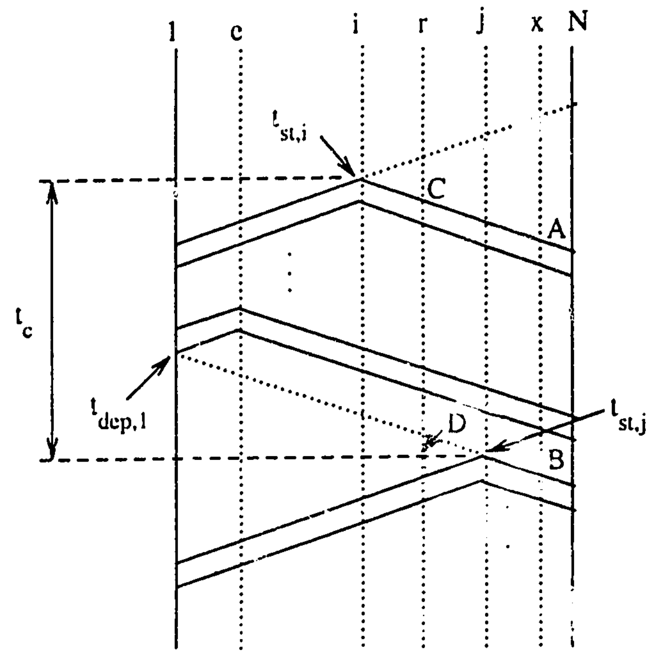


Figure 3.4. Time-space diagram for Subcase 1.1

If station j was not ready to start the new cycle under subcase 1.1, then according to the medium access protocol, the next earliest time that j can start the cycle is when t_c reaches $(\tau_{i,1}+nT+\tau+\tau_{N,j})$ as shown in Figure 3.5. Note that the cycle type in this case (and the remaining subcases 1.3 and 1.4) is R-L.

Subcase 1.3 : $(t_j+2\tau_{j,m+1}) < t_c \leq (t_j+2\tau_{j,m})$; m=1,2,...,(j-1) (see Figure 3.6)

This subcase is somewhat similar to the subcase 1.2 of the Z-Net model. Here, $(j-1)$ adjacent subintervals of t_c from $(t_j+2\tau_{j,i})$ to $(t_j+2\tau_{j,1})$ are considered, with the range of each sub interval being determined by m. For a given value of m, where $1 \leq m \leq (j-1)$, suppose the length of the cycle time, t_c, is such that $(t_j+2\tau_{j,m+1}) < t_c \leq (t_j+2\tau_{j,m})$. For some arbitrary value of t_c in this time interval, let $s^{1.3}_{b,d,j,l \mid n,e,a,c,i,k}(m,t_c)$ be the conditional state transition probability from state (a,c,i,k) to state (b,d,j,l), given m, n and e.

The conditional state transition probability $s^{1.3}_{b,d,j,l \mid n,e,a,c,i,k}$ is then expressed as:

$$s^{1.3}_{b,d,j,l \mid n,e,a,c,i,k} = \sum_{m=1}^{j-1} \left[\int_{t_j+2\tau_{j,m+1}}^{t_j+2\tau_{j,m}} s^{1.3}_{b,d,j,l \mid n,e,a,c,i,k}(m,t_c)\, dt \right]$$

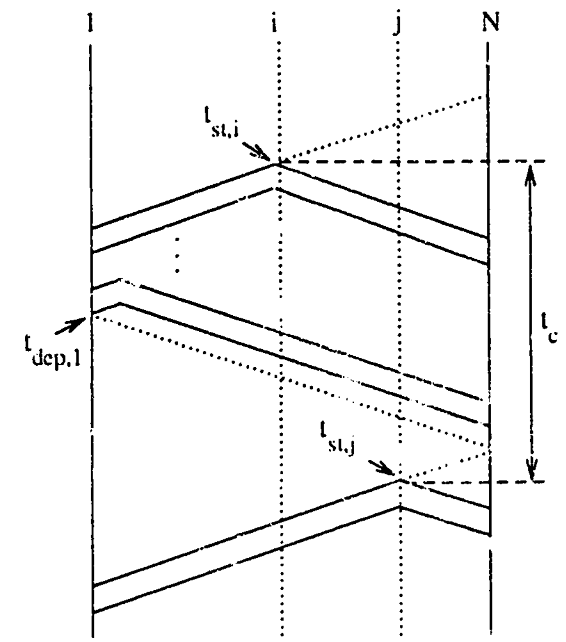


Figure 3.5. Time-space diagram for Subcase 1.2

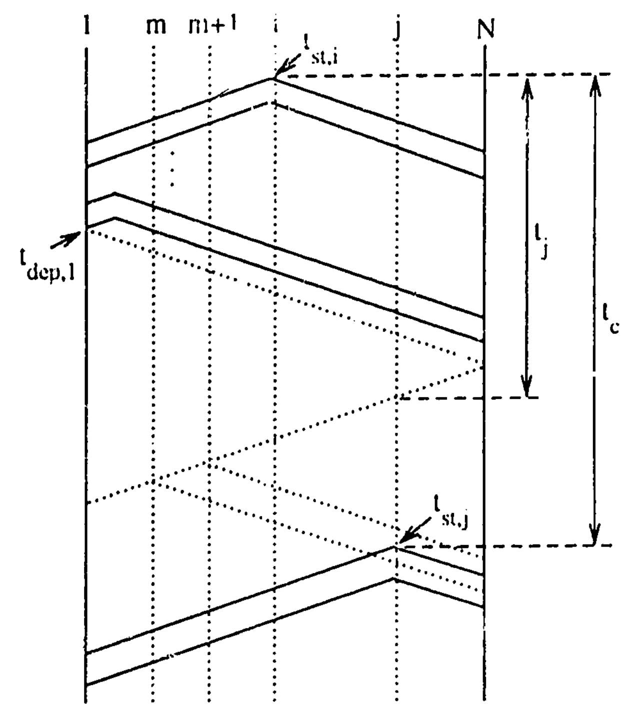


Figure 3.6. Time-space diagram for Subcase 1.3

Subcase 1.4 : $t_c > (t_j + 2\tau_{j,1})$ (see Figure 3.7)

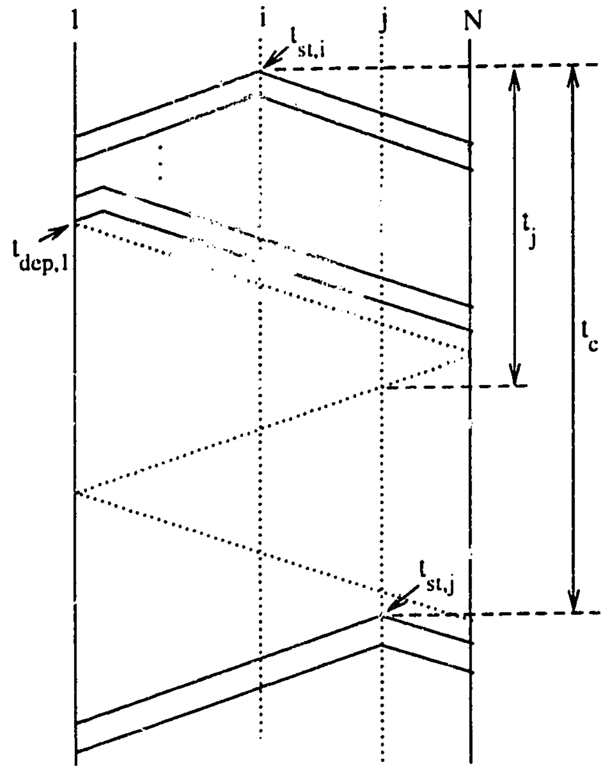


Figure 3.7. Time-space diagram for Subcase 1.4

Given n, e and some arbitrary value of t_c such that $t_c > (t_j + 2\tau_{j,1})$, let $s^{1.4}_{b,d,j,l\,|\,n,e,a,c,i,k}(t_c)$ be the conditional state transition probability from state (a,c,i,k) to state (b,d,j,l).

Then $s^{1.4}_{b,d,j,l\,|\,n,e,a,c,i,k}$ for subcase 1.4 is given by:

$$s^{1.4}_{b,d,j,l\,|\,n,e,a,c,i,k} = \int_{t_j+2\tau_{j,1}}^{\infty} s^{1.4}_{j,l\,|\,n,e,a,c,i,k}(t_c)\, dt$$

Subcase 1.1 is the only one in which station j can start a cycle of type *L-R*. In this case, the mode d of station j just before the start of the cycle is *controlled*. In the remaining cases 1.2, 1.3 and 1.4, the type of cycle started by station j is of type *R-L*. In these cases, the mode d of station j is ***random***. With these observations, the conditional probability $S(b,d,j,l\,|\,n,e,R-L,c,i,k)$ for Case 1 can then be evaluated as follows:

$$S_X(L-R, random, j, l \mid n, e, R-L, c, i, k) = 0$$

$$S_X(L-R, controlled, j, l \mid n, e, R-L, c, i, k) = s^{1.1}_{L-R, controlled, j, l \mid n, e, R-L, c, i, k}$$

$$S_X(R-L, random, j, l \mid n, e, R-L, c, i, k) = s^{1.2}_{R-L, random, j, l \mid n, e, R-L, c, i, k} + s^{1.3}_{R-L, random, j, l \mid n, e, R-L, c, i, k} + s^{1.4}_{R-L, random, j, l \mid n, e, R-L, c, i, k}$$

$$S_X(R-L, controlled, j, l \mid n, e, R-L, c, i, k) = 0$$

$s^{1.x}_{b,d,j,l \mid n,e,R-L,c,i,k}$ for each subcase x, where $x=1, 2, 3$ and 4, is evalauated using:

$$s^{1.x}_{b,d,j,l \mid n,e,R-L,c,i,k} = p_{nr}^{1.x} \cdot p_j^{1.x} \cdot p_l^{1.x} \; ; \; x=1,2,3,4$$

where,

$p_{nr}^{1.x}$ = prob(when the *server enters*, each station upstream from j is not ready)[3]

$p_j^{1.x}$ = prob(station j in the ready state at time $t_{st,j}$), and

$p_l^{1.x}$ = prob(l stations downstream from j in the ready state at time $t_{st,j}$).

For subcase 1.1,

$p_{nr}^{1.1}$ = prob(each station y, y = $1,2,\ldots,j-1$, in the idle state until time $(t_{dep,1}+\tau_{1,y})$)

For the other subcases, with x = 2,3,4,

$p_{nr}^{1.x}$ = prob(each station y, y = $j+1, j+2, \ldots, N$, in the idle state until time $(t_{st,j}+\tau_{j,y})$)

The evaluation of $p_{nr}^{1.x}$, x = 1,2,3,4, is described in Appendix 3.6.

$p_j^{1.x}$ is given by the following:

When $c = random$,

$$p_j^{1.x} = \begin{cases} 1 - e^{-\lambda(t_c - \tau_{1,j})}, & \text{for } x = 1 \\ e^{-\lambda(t_c - \tau_{1,j} - 2\tau_{j,N})} \cdot (1 - e^{-\lambda 2\tau_{j,N}}) & \text{for } x = 2 \\ \lambda\, e^{-\lambda(t_c - \tau_{1,j})}, & \text{for } x = 3 \text{ and } 4 \end{cases}$$

[3] Therefore, these upstream stations do not start the new cycle.

When c = *controlled*,

$$p_j^{1,x} = \begin{cases} 1 - e^{-\lambda(t_c + \tau_{i,j})}, & \text{for } x = 1 \\ e^{-\lambda(t_c + \tau_{i,j} - 2\tau_{j,N})} \cdot (1 - e^{-\lambda \cdot 2\tau_{j,N}}) & \text{for } x = 2 \\ \lambda\, e^{-\lambda(t_c + \tau_{i,j})}, & \text{for } x = 3 \text{ and } 4 \end{cases}$$

In evaluating $p_l^{1.1}$ (i.e., for subcase 1.1), l could be any combination of l stations from station $(j+1)$ to N. For the other three subcases (i.e., 1.2, 1.3 and 1.4), l could be any combination from station 1 to $(j-1)$. p_l^{1x}, x = 1,2,3,4 are then given by:

$p_l^{1.1}$ = prob(l packet arrivals at stations y; $y = j+1, j+2, \ldots, N$; until the time instant $t_{st,j}$)

$p_l^{1.2}$ = prob(each station y; $y = 1,2,\ldots,j-1$ in idle state until $(t_{dep,1}+\tau_{1,y})$) $\cdot$ prob(l packet arrivals at stations y; $y = 1,2,\ldots,j-1$; during the time interval from $(t_{dep,1}+\tau_{1,y})$ to $t_{st,j}$)

For a given station m, where $1 \le m \le (j-1)$,

$p_l^{1.3}(m)$ = prob(each station y, $y = 1,2,\ldots,m$, in the idle state until $(t_{dep,1}+\tau_{1,y})$) $\cdot$ prob(each station y, $y = m+1,\ldots,j-1$, in the idle state until time $(t_{st,j}-\tau_{j,y})$) $\cdot$ prob(packet arrivals at l stations out of $1,\ldots,j-1$)[4]

$p_l^{1.4}$ = prob(each station y, $y = 1,\ldots,j-1$, in the idle state until time $(t_{st,j}-\tau_{j,y})$) $\cdot$ prob(packet arrivals at l stations out of $1,\ldots,j-1$ during the interval $\tau_{j,y}$ preceding $t_{st,j}$)

In computing $p_l^{1.1}$, each station y, $y = j+1, j+2, \ldots, N$, could be in the ready state at time $t_{st,j}$ with the probability:

$$\begin{cases} 1 - e^{-\lambda(t_c - \tau_{i,y})}, & \text{when } c = \textit{random}, \\ 1 - e^{-\lambda(t_c + \tau_{i,y})}, & \text{when } c = \textit{controlled} \end{cases}$$

[4] Packets may arrive at each station y, $y=1,2,\ldots,m$, from the time instant $(t_{dep,1}+\tau_{1,y})$ until the time instant $t_{st,j}$. For each station y, $y=m+1,\ldots,j-1$, packets may arrive during an interval $\tau_{j,y}$ preceding the time instant $t_{st,j}$.

$$\text{where}, \ t_c = (\tau_{i,1} + nT + \tau_{1_j}).$$

As these individual probabilities are known, $p_l^{1,1}$ can be computed using the recursive algorithm described in Appendix 3.4.

Considering the stations $\{1,2,...,i\}$ and $\{i+1,i+2,...,j-1\}$ as two separate sets, the evaluation procedure for $p_l^{1,x}$, x = 2,3,4, is similar to that of $p_l^{1,x}$, x = 1,2,3, of the Z-Net model. Hence, $S_X(b,d,j,l \mid n,e,R-L,c,i,k)$ for Case 1 can be computed. Using similar principles, $S_X(b,d,j,l \mid n,e,R-L,c,i,k)$ for Case 2 (i.e., j≤i) can be computed, as outlined in Appendix 3.9.

The state transition probabilities $p^X_{(a,c,i,k),(b,d,j,l)}$ for $a,b = R-L, L-R$, $c,d = random, controlled$, $1 \le i,j \le N$, and $0 \le k \le (i-1)$, $0 \le l \le (j-1)$, can therefore be found. Standard techniques are then used to obtain the steady state probabilities $\pi_{a,c,i,k}$.

The remaining sections of the chapter describe how the mean channel utilization of Z-Net and X-Net can be obtained using the steady state probabilities.

3.4. Mean Channel Utilization

Mean channel utilization for a given station (*stationwise channel utilization*) is defined as the proportion of time the channel is utilized by that station for successful packet transmissions. Mean channel utilization for the network (denoted simply by *channel utilization*) is calculated by adding the individual channel utilizations of all stations. Let $\bar{n}$ be the mean number of transmissions per cycle (i.e., mean cycle length), and $\bar{c}$ be the mean time between the start of two consecutive cycles (i.e., mean cycle time), both averaged over all cycles. Then, the mean channel utilization $\bar{U}$ is given by:

$$\bar{U} = \frac{\bar{n} \cdot T}{\bar{c}} \tag{3.6}$$

Mean Channel Utilization for Z-Net :

For Z-Net, $\bar{n}$ and $\bar{c}$ can be found as follows:

$$\bar{n} = \sum_n \left[n \sum_{i,k} prob(n \mid i,k) \cdot \pi_{i,k} \right] ,$$

where,

$$prob(n \mid i,k) = \sum_e prob(n,e \mid i,k) \text{, and}$$

$$\bar{c} = \sum_n \sum_{i,k} E(c \mid n,i,k) \cdot prob(n \mid i,k) \cdot \pi_{i,k} ,$$

with,

$$E(c \mid n,i,k) = \sum_{j,l} \bar{c}_{j,l \mid n,i,k} .$$

$\bar{c}_{j,l \mid n,i,k}$ is defined as the mean time for transition from state (i,k) to state (j,l), when the cycle length is n.

Considering the three different subcases of Case 1, we compute $\bar{c}_{j,l \mid n,i,k}$ as follows:

$$\bar{c}_{j,l \mid n,i,k} = (t_j + 2\tau_{j,i}) \cdot p_{j,l \mid n,e,i,k}^{1,1} + \sum_{m=i-1}^{1} \int_{t=t_j+2\tau_{j,m+1}}^{t_j+2\tau_{j,m}} \left[t \cdot p_{j,l \mid n,e,i,k}^{1,2} \right] dt + \int_{t=t_j+2\tau_{j,1}}^{\infty} \left[t \cdot p_{j,l \mid n,e,i,k}^{1,3} \right] dt$$

where,

$$t_j = (\tau_{i,1} + nT + \tau_{1,N} + \tau_{N,j}).$$

Note that in the above expression, $p_{j,l \mid n,e,i,k}^{1,x}$, x=1,2,3, is independent of e, as this probability does not depend on the identity of the station that transmitted last in the cycle. For Case 2 (i.e., j≤i), $\bar{c}_{j,l \mid n,i,k}$ can be found in an identical manner.

Mean Channel Utilization for X-Net :

Similar to that of the Z-Net, $\bar{n}$ and $\bar{c}$ for X-Net are given by:

$$\bar{n} = \sum_n \left[n \sum_{a,c,i,k} prob(n \mid a,c,i,k) \cdot \pi_{a,c,i,k} \right] ,$$

where,

$$prob(n \mid a,c,i,k) = \sum_{e} prob(n,e \mid a,c,i,k)\text{, and}$$

$$\bar{c} = \sum_{n}\sum_{a,c,i,k} E(c \mid n,a,c,i,k) \cdot prob(n \mid a,c,i,k) \cdot \pi_{a,c,i,k} \ .$$

with,

$$E(c \mid n,a,c,i,k) = \sum_{b,d,j,l} \bar{c}_{b,d,j,l \mid n,a,c,i,k} \ .$$

$\bar{c}_{b,d,j,l \mid n,a,c,i,k}$ is defined as the mean time for transition from state (a,c,i,k) to state (b,d,j,l), when the cycle length is n.

Considering the four different subcases of Case 1, we compute $\bar{c}_{b,d,j,l \mid n,a,c,i,k}$ as follows:

$$\bar{c}_{b,d,j,l \mid n,a,c,i,k} = (\tau_{i,1}+nT+\tau_{1,j}) \cdot p^{1.1}_{b,d,j,l \mid n,e,a,c,i,k} + t_j \cdot p^{1.2}_{b,d,j,l \mid n,e,a,c,i,k} + \sum_{m=i-1}^{1} \int_{t=t_j+2\tau_{j,m+1}}^{t_j+2\tau_{j,m}} \left[t \cdot p^{1.3}_{b,d,j,l \mid n,e,a,c,i,k} \right] dt + \int_{t=t_j+2\tau_{j,1}}^{\infty} \left[t \cdot p^{1.4}_{b,d,j,l \mid n,e,a,c,i,k} \right] dt$$

For Case 2 (i.e., j≤i), $\bar{c}_{b,d,j,l \mid n,a,c,i,k}$ can be found in an identical manner.

3.5. Mean Packet Delay

Packet delay is measured in terms of two quantities, the *insertion delay*, and the *transfer delay*. The insertion delay, denoted by d_{ins}, is defined as the time interval between a packet moving to the head of the transmitting queue and the beginning of its successful transmission [Gerla et. al. 1985]. Transfer delay d_{tfr} is defined as the time interval between the arrival of a packet at a station and the successful reception of it by the receiving station. Consider two consecutive packet arrivals at a station x, $(1 \le x \le N)$, as shown in Figure 3.8. The station starts its packet transmission at time A, after a mean delay of $\bar{d}_{ins,x}$, during which time the packet waits in the transmitting queue. At the completion of packet transmission at time B, the packet generation process is enabled.

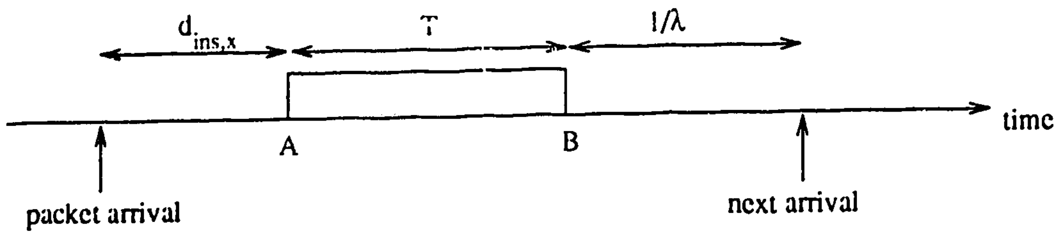


Figure 3.8. Two consecutive packet arrivals at a station x

The mean channel utilization of station x, denoted by $\bar{u}_x$, is then given by:

$$\bar{u}_x = \frac{T}{\bar{d}_{ins,x} + T + (1/\lambda)}$$

Therefore, the mean insertion delay for station x, $\bar{d}_{ins,x}$, is given by:

$$\bar{d}_{ins,x} = T \left[\frac{1}{\bar{u}_x} - 1 \right] - \frac{1}{\lambda} \tag{3.7}$$

The mean insertion delay, $\bar{d}_{ins}$, calculated over all stations, can then be computed as follows:

$$\bar{d}_{ins} = \frac{\sum_{x=1}^{N} \bar{d}_{ins,x}\, \bar{u}_x}{\bar{U}} \tag{3.8}$$

where $\bar{U}$ is given by:

$$\bar{U} = \sum_{x=1}^{N} \bar{u}_x$$

Substituting the expression (3.7) for $\bar{d}_{ins,x}$ in equation (3.8), and then simplifying, the following expression can be obtained for $\bar{d}_{ins}$:

$$\bar{d}_{ins} = T \left[\frac{N}{\bar{U}} - 1 \right] - \frac{1}{\lambda} \tag{3.9}$$

The mean transfer delay $\bar{d}_{tfr}$ is then given by:

$$\bar{d}_{tfr} = \bar{d}_{ins} + T + \bar{d}_p \tag{3.10}$$

where,

$\bar{d}_p$ = mean propagation delay between a pair of transmitting and receiving stations

$$= \frac{\tau(N+1)}{3(N-1)}$$

Appendix 3.1: Evaluation of p_x and $p_{y|x}$ in Subcase 1.1 of the Z-Net Model

For evaluating p_x for Subcase 1.1, consider a station r in the set $\{i+1, i+2, ..., j-1\}$. It could be in ready state at time $t_{st,j}$ with probability p_r where,

$$p_r = \text{prob(an arrival at } r \text{ during time interval CD in Figure 3.1)} = 1 - e^{-\lambda(t_j + 2\tau_{j,i} - \tau_{i,r})}$$

Each p_r can be found for $r = i+1, i+2, ..., j-1$. Then, using the recursive algorithm given in Appendix 3.4, p_x can be computed. For evaluating $p_{y|x}$, consider the time intervals from $t_{st,i}$ to $t_{dep,1}$ and $t_{dep,1}$ to $t_{st,j}$ separately (see Figure 3.1). Let $y = y_1 + y_2$, where,

$y_1 =$ number of ready stations in the set $\{1,2,...,i\}$ at time $t_{dep,1}$; $0 \le y_1 \le i$ and

$y_2 =$ number of stations in the set $\{1,2,...,i\}$ that become ready during the time from $t_{dep,1}$ to $t_{st,j}$

Let p_{y_1} be the probability of y_1 and, $p_{y_2|y_1}$ be the probability of y_2, given y_1. Then,

$$p_{y|x} = \sum_{y_1=0}^{min(y,i-1)} p_{y_1} \cdot p_{y_2|y_1} \; ; \quad y_2 = y - y_1 \qquad \text{(A.3.1.1)}$$

$$\text{and}, \; p_{y_2|y_1} = \begin{bmatrix} i-y_1 \\ y-y_1 \end{bmatrix} \left[1-e^{-\lambda t}\right]^{y-y_1} \left[e^{-\lambda t}\right]^{i-y} \qquad \text{(A.3.1.2)}$$

where, t = time interval from $t_{dep,1}$ to $t_{st,j} = \tau_{1,N} + \tau_{N,i} + \tau_{i,j}$ (see Figure 3.1)

To compute the probability p_{y_1}, we use the function P_r(y | i,k,e,n) defined as follows:

P_r(y | i,k,e,n) = prob(when the server leaves station 1 (i.e., at time $t_{dep,1}$), there are y stations upstream from station 1 in ready state among stations $\{2,3,...,r\}$ AND all stations $\{r+1, r+2, ..., i\}$ remain in idle state; given i, k, e, n); $1 \le r \le i$.

This function is also used in the other subcases 1.2 and 1.3 and its derivation is given in Appendix 3.5. The use of parameter r will be apparent in subcase 1.2. With r=i and y=y_1,

$P_i(y_1$ | i,k,e,n) = prob(when the server leaves station 1 there are y_1 stations upstream from station 1 in ready state among stations $\{2,...,i-1,i\}$; given i, k, e, n)

Clearly, $p_{y_1} = P_i(y_1 \mid i,k,c,n)$. □

Appendix 3.2: Evaluation of p_x and $p_{y \mid x}$ in Subcase 1.2 of the Z-Net Model

For evaluating p_x for Subcase 1.2, consider a station r in the set $\{i+1, i+2, ..., j-1\}$. It should not become ready during the time interval CE in Figure 3.2. This is because, should it become ready before the time instant E, its BOC (beginning of carrier) will reach station j before j becomes ready at time $t_{st\,j}$. Then, station r would be the starter of new cycle instead of j. However, r could become ready during the last $\tau_{r,j}$ interval from $t_{st\,j}$, because, by the time r's BOC reaches j, j had already started the new cycle. p_x is given by:

$p_x =$ prob(each station r, $r = i+1, i+2, ..., j-1$, remaining idle during $(t_c - \tau_{i,r} - \tau_{r,j})$) $\cdot$ prob(x stations out of stations $r = i+1, i+2, ..., j-1$, becoming ready, each during the last interval $\tau_{r,j}$ from $t_{st\,j}$)

The first probability on the R.H.S. of the above expression is given by:

$$\sum_{r=i+1}^{j-1} e^{-\lambda(t_c - \tau_{i,r} - \tau_{r,j})} \qquad (A.3.2.1)$$

The second probability can be computed by considering the following: each station r, $r = i+1, i+2, ..., j-1$, may become ready during the interval $\tau_{r,j}$, each with probability $(1 - e^{-\lambda\tau_{r,j}})$. Using the recursive algorithm in Appendix 3.4, the prob(x stations out of $i+1, i+2, ..., j-1$, in ready state at time $t_{st\,j}$) can then be computed.

To evaluate $p_{y \mid x}$, the stations $1, 2, ..., m$ and the stations $m+1, m+2, ..., i$ are considered separately under (i) and (ii) below:

(i). Each station r, $r = m+1, m+2, ..., i$, should remain in the idle state from the time the server leaving them until the time instant $(t_{st\,j} - \tau_{r,j})$. Let this probability be p_{idle}. Each of these stations may become ready during the last interval $\tau_{r,j}$ from $t_{st\,j}$.

To evaluate p_{idle}, first consider the time interval from the instant the server leaving a station

r, $r=m+1,...,i$, until time instant $t_{dep,1}$, and then, the time interval from $t_{dep,1}$ until the time instant $(t_{st\,j}-\tau_{r\,j})$. p_{idle} is then given by:

p_{idle} = prob(all stations $m+1, m+2,..,i$ remaining in idle state until time $t_{dep,1}$) ·

prob(each station r, $r=m+1, m+2,...,i$ remaining idle from $t_{dep,1}$ until time $(t_{st\,j}-\tau_{r\,j})$)

$$= p_{idle}\ (\text{until } t_{dep,1}) \cdot \prod_{r=m+1}^{i} e^{-\lambda(t-\tau_{r,j})} \qquad \text{(A. 3.2.2)}$$

where, p_{idle} (until $t_{dep,1}$) = prob(stations $m+1, m+2,...,i$ in idle state until time $t_{dep,1}$),

and, t = time interval from $t_{dep,1}$ to $t_{st\,j}$ = $t_c-\tau_{i,1}-nT$

The probability p_{idle} (until $t_{dep,1}$) will be included in the evaluation of one of the remaining terms of p_{j}, as seen later under section (ii) below.

Let y_1 be the number of stations in ready state at time $t_{st,j}$ in the set $\{m+1, m+2,...,i\}$, and p_{y_1} be the probability of y_1. As already mentioned above, each station r, $r=m+1, m+2,...,i$, can become ready only during the last interval $\tau_{r,j}$ from $t_{st\,j}$.

p_{y_1} can be computed using the recursive algorithm in Appendix 3.4 again, since the individual probabilities of each station r, $r=m+1, m+2,...,i$, in the ready state at time $t_{st,j}$ is given by $(1-e^{-\lambda\tau_{r,j}})$.

(ii). Each station r, $r=1,2,...,m$ may become ready from the time the server leaving them until $t_{st\,j}$.

Let y_2 be the number of stations in the set $\{1,2,...,m\}$ in ready state at time $t_{st\,j}$. Let $p_{y_2|y_1}$ be the probability of y_2 given y_1. Calculation of $p_{y_2|y_1}$ is then very similar to the calculation of $p_{y|x}$ in Appendix 3.1 for Subcase 1.1 (see equation (A.3.1.1)), with the time intervals from $t_{st,i}$ to $t_{dep,1}$ and $t_{dep,1}$ to $t_{st\,j}$ considered separately. Let $y_2=y_3+y_4$, where,

y_3 = number of ready stations in the set $\{1,2,...,m\}$ at time $t_{dep,1}$; $0\le y_3\le y_2$, and,

y_4 = number of stations in the set $\{1,2,...,m\}$ that becomes ready during the time interval from $t_{dep,1}$ to $t_{st,j}$.

Let p_{y_3} be the probability of y_3, and $p_{y_4|y_3}$ be the probability of y_4 given y_3. Then,

$$p_{y_2|y_1} = \sum_{y_3=0}^{min(y_2,m-1)} p_{y_3} \cdot p_{y_4|y_3} \; ; \quad y_4=y_2-y_3 \tag{A.3.2.3}$$

$$\text{and} \quad p_{y_4|y_3} = \begin{bmatrix} m-y_3 \\ y_2-y_3 \end{bmatrix} \left[1-e^{-\lambda t}\right]^{y_2-y_3} \left[e^{-\lambda t}\right]^{m-y_2} \tag{A.3.2.4}$$

where, t = time interval from $t_{dep,1}$ to $t_{st,j} = t_c - \tau_{i,1} - nT$ (see Figure 3.2)

To find p_{y_3}, the function $P_r(y \mid i,k,e,n)$ is used (which is defined previously). With r=m and $y=y_3$ in this function,

$P_m(y_3 \mid i,k,e,n)$ = prob(when the server leaves station 1 there are y_3 stations upstream from station 1 in ready state among stations $\{2,...,m-1,m\}$ AND, all stations $\{m+1,m+2,...,i\}$ remain in idle state at time $t_{dep,1} \mid i,k,e,n$)

$$= p_{y_3} \cdot \text{prob}(\text{stations } m+1,m+2,...,i, \text{ in idle state at time } t_{dep,1})$$

$$= p_{y_3} \cdot p_{idle}(\text{until } t_{dep,1}) \tag{A.3.2.5}$$

Note that, $P_m(y_3 \mid i,k,e,n)$ includes not only p_{y_3} but also the probability $p_{idle}(\text{until } t_{dep,1})$, which is required in evaluating p_{idle}. Equations (A.3.2.3) and (A.3.2.5) yield:

$$p_{y_2|y_1} \cdot p_{idle}(until\ t_{dep,1}) = \sum_{y_3=0}^{y_2} P_m(y_3 \mid i,k,e,n) \cdot p_{y_4|y_3} \; ; \quad y_4=y_2-y_3 \tag{A.3.2.6}$$

$$\text{Hence,} \quad p_{y|x} = p_{idle} \cdot \sum_{y_1=max(0,y-m)}^{min(y,i-m)} p_{y_1} \cdot p_{y_2|y_1} \tag{A.3.2.7}$$

Appendix 3.3: Evaluation of p_x and $p_{y|x}$ in Subcase 1.3 of the Z-Net Model

The evaluation of p_x is identical to the evaluation of p_x in Appendix 3.2 for Subcase 1.2. In evaluating $p_{y|x}$, the following should be taken into account:

(a). Each station r, r=1,2,..,i, should remain in the idle state from the time the server leaving station r until the time instant $(t_{st,j} - \tau_{r,j})$.

(b). Each of these stations r may become ready during the last interval $\tau_{r,j}$ from $t_{st,j}$.

$p_{y|x}$ can then be evaluated following the same line of arguments in Appendices 3.1 and 3.2. □

Appendix 3.4: Recursive Algorithm for evaluating the function f(m,n)

Suppose there are m stations, numbered from 1 to m, which could be in ready state at a particular time instant t with individual probabilities $p_1, p_2, ..., p_m$, respectively. p_i, $i = 1,..,m$, is given by: $p_i = 1 - e^{-\lambda t_i}$

where, λ = arrival rate, and

t_i = time interval up to the time instant t, during which an arrival could take place at station i.

The function f(m,n) is defined as:

f(m,n) = prob(out of m given stations, n are in ready state at time instant t)

f(m,n) is computed recursively using the following equations:

$$f(m,n) = f(m-1,n-1)\,p_m + f(m-1,n)(1-p_m)$$

$$f(x,y) = f(x-1,y-1)\,p_x + f(x-1,y)(1-p_x); \quad 1 \le x \le m; \; 1 \le y \le n.$$

$$f(x,0) = \sum_{i=1}^{x} (1-p_i); \quad 1 \le x \le m.$$

$$f(1,1) = p_1. \; \square$$

Appendix 3.5: Evaluation of the function $P_r(y \mid i,k,e,n)$

For a given r such that $1 \le r \le i$, the function P_r(y | i,k,e,n) is defined as:

$P_r(y \mid i,k,e,n)$ = prob(when the server leaves station 1, there are y stations in ready state in the set $\{2,..,r-1,r\}$ AND all the stations $\{r+1,r+2,..,i\}$ remain in idle state | i,k,e,n,r)

To evaluate this probability, a slightly different function is defined as follows:

$U_r(s,g,h \mid i,k,e,n,y)$ = prob(s^{th} transmission in the cycle is by station g, with h stations in ready state in the set $\{g+1,g+2,..,r\}$ at the end of s^{th} transmission AND all the stations $\{r+1,r+2,...,i\}$ remain in idle state at the end of s^{th} transmission | i,k,e,n,r,y)

$U_r(s,g,h \mid i,k,e,n,y)$ can then be computed in terms of $U_r(s-1,g',h' \mid i,k,e,n,y)$ as shown below. Depending on the location of r, two cases, Case (a). $e \le r \le i$, and Case (b). $1 \le r < e$, are possible. Under Case (a), three subcases are possible, depending on the location of station g. These are: Case (a.1). $r \le g \le i$; Case (a.2). $e \le g < r$; and Case (a.3). $1 \le g < e$. Under Case (b), only the case where $1 \le g < e$ is relevant.

Assuming that $e \le r \le i$ and $e \le g < r$, derivation of $P_r(s,g,h \mid i,k,e,n,u)$ under case (a.2) is decribed below. The procedure for other cases use the same reasoning with minor variations.

$U_r(s,g,h \mid i,k,e,n,y)$ is given by:

$$U_r(s,g,h \mid i,k,e,n,y) = \sum_{g'=min(r,g+1)}^{i+2-s} \sum_{h'=0}^{h'_{max}} p_1 \cdot \alpha \cdot p_2 \cdot p_3 \cdot U_r(s-1,g',h' \mid i,k,e,n,y) \quad (A.3.5.1)$$

where:

$$h'_{max} = \begin{cases} 0, & \text{if } g' > r \\ min(h,(r-g')), & \text{else} \end{cases}$$

p_1 = prob(stations $g'-1,g'-2,..,g+1$ did not transmit | $s-1^{th}$ transmission is by station g')

$$= \prod_{x=1}^{g'-g-1} \left[1 - \frac{n-s}{g'-e-x} \right]$$

α = prob(station g transmits | $s-1^{th}$ transmission is by g' and stations $g'-1,..,g+2,g+1$ did not transmit)

$$= \frac{n - s}{g - e}$$

$p_2 =$ prob($(h-h')$ new arrivals at $(r-g-h')$ idle stations in the set $\{g+1, g+2, .., r\}$), and,

$p_3 =$ prob(no arrivals at stations $\{r+1, r+2, .., i\}$ during the time interval between server leaving g' and g)

Evaluation of p_2 above depends on the location of station g' with respect to r.

When $r > g'$:

Stations $g+1, g+2, .., r$ are divided into two sets $\{g+1, g+2, ..., g'-1\}$ and $\{g', g'+1, ..., r\}$.

(a). Idle stations in the set $\{g', g'+1, ..., r\}$ may become ready during ($\tau_{g',g}$ + T), each with the same probability $(1-e^{-\lambda(\tau_{g',g}+T)})$.

(b). Each of the stations $x, x=g+1, g+2, ..., g'-1$; may become ready during ($\tau_{x,g}$ + T), with individual probabilities $(1-e^{-\lambda(\tau_{x,g}+T)})$.

With the above, the individual probabilities of any station in the set $\{g+1, g+2, .., r\}$ becoming ready are known. Therefore, p_2 can be evaluated using the recursive algorithm in Appendix 3.4.

When $r \leq g'$:

Now, each of the stations $x, x=g+1, g+2, .., r$; may become ready during ($\tau_{x,g}$ + T), with individual probabilities $(1-e^{-\lambda(\tau_{x,g}+T)})$. Therefore, p_2 can be computed again using the recursive algorithm in Appendix 3.4.

$U_r(s, g, h \mid i, k, e, n, y)$ can now be computed recursively, using the following conditions:

$U_r(., r, x \mid i, k, e, n, y) = 0$, if $x \neq 0$

≥ 0, if $x = 0$, and this could be found using the case (a.1) where $r \leq g \leq i$.

To evaluate $P_r(y \mid i, k, e, n)$, we consider the time instants the server leaving station e (i.e., $t_{dep,e}$), and the server leaving station 1 (i.e., time $t_{dep,1}$). Let $y = y_1 + y_2$, where,

y_1 = number of ready stations in the set $\{e+1, e+2, ..., r\}$ at time $t_{dep,e}$, $0 \le y_1 \le y$; and,

y_2 = number of stations in the set $\{2,3,...,r\}$ that becomes ready during the time from $t_{dep,e}$ to $t_{dep,1}$

Let p_{y_1} be the probability of y_1 and $p_{y_2|y_1}$ be the probability of y_2, given y_1. $P_r(y \mid i,k,e,n)$ is then given by:

$$P_r(y \mid i,k,e,n) = \sum_{y_1=0}^{min(r-e,y)} p_{y_1} \cdot p_{y_2|y_1} \cdot p_{idle} \;\; ; \;\; y_2 = y - y_1 \qquad \text{(A.3.5.2)}$$

where, p_{idle} = prob(stations $r+1, r+2, ..., i$ in idle state at time $t_{dep,1}$)

= prob(stations $r+1,...,i$ idle at time $t_{dep,e}$) $\cdot$

prob(stations $r+1,...,i$ remaining idle from time $t_{dep,e}$ until $t_{dep,1}$)

$$= \text{prob(stations } r+1,...,i \text{ idle at time } t_{dep,e}) \cdot \left[e^{-\lambda \tau_{e,1}}\right]^{i-r} \qquad \text{(A. 3.5.3)}$$

To find p_{y_1}, the function $U_r(s,g,h \mid i,k,e,n,y)$ is used with $s=n$, $g=e$, $h=y_1$ and $y=y_1$.

$U_r(n,e,y_1 \mid i,k,e,n,y_1)$ = prob(n^{th} transmission is by station e with y_1 ready stations at time $t_{dep,e}$ AND stations $r+1, r+2, ..., i$ remaining idle at time $t_{dep,e} \mid i,k,e,n,y_1$)

$$= p_{y_1} \cdot \text{prob(stations } r+1,...,i \text{ in idle state at time } t_{dep,e}) \qquad \text{(A.3.5.4)}$$

$p_{y_2|y_1}$ can be computed using the recursive algorithm in Appendix 3.4, considering the following:

During the time the server leaving stations e and 1 (i.e., time interval $\tau_{e,1}$), y_2 arrivals have occured at the $(r-1-y_1)$ idle stations in the set $\{2,3,...,r\}$. To find this probability (which is $p_{y_2|y_1}$), consider the stations $\{2,3,...,e\}$ and $\{e+1, e+2, ..., r\}$ separately. A station x in $\{2,3,...,e\}$ may become ready during the time $\tau_{x,1}$ with probability $(1-e^{-\lambda\tau_{x,1}})$. Any idle station in $\{e+1,...,r\}$ may become ready during the time $\tau_{e,1}$ with the same probability $(1-e^{-\lambda\tau_{e,1}})$. Now, the recursive algorithm in Appendix 3.4 can be used to compute $p_{y_2|y_1}$.

Equations A.3.5.2, A.3.5.3 and A.3.5.4 yield:

$$P_r(y \mid i,k,e,n) = \sum_{y_1=0}^{\min(r-e,y)} U_r(n,e,y_1 \mid i,k,e,n,y_1) \cdot p_{y_2|y_1} \cdot \left[e^{-\lambda\tau_{r,1}}\right]^{i-r} \qquad \text{(A.3.5.5)}$$

Appendix 3.6: Evaluation of $p_{nr}^{1,x}$, $x=1,2,3,4$, of the X-Net Model

Evaluation of $p_{nr}^{1,1}$:

$p_{nr}^{1,1}$ is given by:

$$p_{nr}^{1,1} = p_1 \cdot p_2 \cdot p_3$$

where:

$p_1 =$ prob(Stations {2,3,..,i} in idle state at time $t_{dep,1}$)

$p_2 =$ prob(Each station y; y = 2,3,..,i; in idle state from time $t_{dep,1}$ to $(t_{dep,1}+\tau_{1,y})$)

$$= \prod_{y=2}^{i} e^{-\lambda \cdot \tau_{1,y}}$$

$p_3 =$ prob(Each station y; $y = i+1, i+2,..,j-1$; in idle state until $(t_{dep,1}+\tau_{1,y})$)

$$= \begin{cases} \left[e^{-\lambda(t_c-\tau_{1,j})}\right]^{(j-1-i)}, & \text{for } c = \textit{random} \\ \prod_{y=i+1}^{j-1} e^{-\lambda(t_c+\tau_{1,y})}, & \text{for } c = \textit{controlled} \end{cases}$$

To evaluate p_1, we define a function Q_r(y | i,k,c,n) as follows:

Q_r(y | i,k,c,n) = prob(At time $t_{dep,1}$, there are y stations among $\{r+1, r+2,..,i\}$ in ready state AND all stations $\{1,2...,r-1\}$ (if any) remain in idle state; given i,k,e,n); $1 \le r \le i$.

The derivation of Q_r(y | i,k,c,n) is given in Appendix 3.7. With r=1 and y=0,

Q_1(0|i,k,c,n) = prob(At time $t_{dep,1}$, there are no stations in ready state among stations $\{2,..,i-1,i\}$; given i,k,e,n)

Therefore, $p_1 = Q_1(0 \mid i,k,c,n)$.

$p_{nr}^{1,x}$; x = 2,3,4; are given by the following expressions :

$$p_{nr}^{1,x} = \begin{cases} \left[e^{-\lambda(t_c-\tau_{i,j})}\right]^{(N-j)} , & \text{for } c = random \\ \prod_{y=j+1}^{N} e^{-\lambda(t_c+\tau_{i,y}+\tau_{j,y})} , & \text{for } c = controlled \end{cases}$$

where, $t_c = (\tau_{i,1}+nT+\tau+\tau_{N,j})$. □

Appendix 3.7: Evaluation of the function $Q_r(y \mid i,k,e,n)$

For a given r such that $1 \le r \le i$, the function $Q_r(y \mid i,k,c,n)$ is defined as:

$Q_r(y \mid i,k,e,n) =$ prob(when the server leaves station 1, there are y stations in ready state in the set $\{r+1,r+2,...,i\}$ AND all the stations $\{2,3,..,r-1\}$ remain in the idle state $\mid i,k,e,n,r)$

In evaluating this probability for a given r, $1 \le r \le i$, the function $W_r(s,g,h \mid i,k,e,n,y)$ is defined as follows:

$W_r(s,g,h \mid i,k,e,n,y) =$ prob(s^{th} transmission in the cycle is by station g, with h stations in ready state in the set $\{x+1,x+2,...,i\}$ at the end of s^{th} transmission AND if $g<(r-1)$, then all the stations $\{g+1,g+2,...,r-1\}$ remain in idle state at the end of s^{th} transmission $\mid i,k,e,n,r,y)$, where, x = $\max(r,g)$.

$W_r(s,g,h \mid i,k,e,n,y)$ can then be computed in terms of $W_r(s-1,g',h' \mid i,k,e,n,y)$. Depending on the location of r, three cases, Case (a). $(i-k) \le r \le i$, Case (b). $e \le r < (i-k)$, and Case (c). $1 \le r < e$ are possible.

Assuming that $(i-k) \le r \le i$, derivation of $W_r(s,g,h \mid i,k,e,n,y)$ for Case (a) is decribed below. Other cases use the same reasoning with minor variations in the derivations.

$W_r(s,g,h \mid i,k,e,n,y)$ can be computed in a recursive manner as follows:

$$W_r(s,g,h \mid i,k,e,n,y) = \sum_{g'=min(r,g+1)}^{i+2-s} \sum_{h'=0}^{h'_{max}} p_1 \cdot \alpha \cdot p_2 \cdot p_3 \cdot p_4 \cdot W_r(s-1,g',h' \mid i,k,e,n,y)$$

where:

$h'_{max} = \min(y, \min(h, i-r))$

p_1 = prob(stations $g'-1, g'-2, \ldots, g+1$ did not transmit | $s-1^{th}$ transmission is by station g')

$$= \prod_{x=1}^{g'-g-1} \left[1 - \frac{n-s}{g'-e-x} \right]$$

α = prob(station g transmits | $s-1^{th}$ transmission is by g' and stations $g'-1, \ldots, g+2, g+1$ did not transmit)

$$= \frac{n-s}{g-e}$$

p_2 = prob($(h-h')$ new arrivals at $(i-r-h')$ idle stations in the set $\{r+1, r+2, \ldots, i\}$ during the interval $(T+\tau_{g',g})$)

$$= \binom{i-r-h'}{h-h'} \left[1-e^{-\lambda t}\right]^{h-h'} \left[e^{-\lambda t}\right]^{i-r-h} \quad \text{where}, \quad t = (T+\tau_{g',g}).$$

p_3 = prob(no arrivals at each station x; $x = g+1, g+2, \ldots, g'$ during the interval $(T+\tau_{x,g})$)

$$= \prod_{x=g+1}^{g'} e^{-\lambda(T+\tau_{x,g})}$$

and,

p_4 = prob(no arrivals at stations $\{g'+1, g'+2, \ldots, r-1\}$ during the interval $(T+\tau_{g',g})$)

$$= \left[e^{-\lambda(T+\tau_{g',g})}\right]^{(r-1)-(g'+1)}$$

The function $W_r(s,g,h \mid i,k,e,n,y)$ can then be computed using the following boundary conditions:

$$W_r(k+1,x,\cdot \mid i,k,e,n,y) = 0 \text{ if } x \neq (i-k)$$ [5]

$$W_r(k+1,i-k,x \mid i,k,e,n,y) = p_a \cdot p_b$$

where,

p_a = prob(x stations out of $\{r+1,r+2,..,i\}$ ready at time $t_{dep,i-k}$), and

p_b = prob(each station q; $q = i-k+1,i-k+2,..,r-1$; remaining in idle state during the interval $(T+\tau_{q,i-k})$)

To compute p_a: each station q; $q = r+1,r+2,..,i$; may become ready during the interval t_q, with probability $(1-e^{-\lambda t_q})$; where, $t_q = ((q-i+k)T+\tau_{q,i-k})$. Therefore, the recursive algorithm described in Appendix 3.4, can be used to compute p_a.

p_b is given by:

$$p_b = \prod_{q=i-k+1}^{r-1} e^{-\lambda(T+\tau_{q,i-k})}$$

$Q_r(y \mid i,k,e,n)$ is then given by:

$$Q_r(y \mid i,k,e,n) = \sum_{x=0}^{min(y,i-r)} \Big[\text{prob}(x \text{ stations in the set } r+1,r+2,..,i \text{ ready at time } t_{dep,e}) \cdot$$
$$\text{prob}((y-x) \text{ arrivals at the } (i-r-x) \text{ stations in the set } r+1,r+2,...,i \text{ during the interval } \tau_{e,1})\Big] \cdot$$

prob(stations $e+1,e+2,...,r-1$ remaining in idle state during the interval $\tau_{e,1}$) .

prob(each of the stations q; $q = e,e-1,..,2$ remaining in idle state during the interval $\tau_{q,1}$) .

$$= \sum_{x=0}^{min(y,i-r)} \left[W_r(n,e,x \mid i,k,e,n,y) \cdot \binom{i-r-x}{y-x} \left[1-e^{-\lambda\tau_{e,1}}\right]^{y-x} \left[e^{-\lambda\tau_{e,1}}\right]^{i-x-y} \right]$$

[5] This is because the (k+1)th transmission should be by station (i-k).

$$\cdot \left[e^{-\lambda\tau_{e,1}} \right]^{(r-1-e)} \cdot \prod_{q=2}^{e} e^{-\lambda\tau_{q,1}} \quad \square$$

Appendix 3.8 : Derivation of $S_Z(j,l \mid n,e,i,k)$ when $j \le i$ (Case 2)

As in Case 1, three subcases 2.1, 2.2 and 2.3 are considered, depending on the length of the cycle time t_c. The conditional state transition probability $S_Z(j,l \mid n,e,i,k)$ for each of these subcases is denoted by $s_{j,l \mid n,e,i,k}^{2,x}$, where $x = 1$, 2 or 3.

Subcase 2.1 : $t_c = t_j$ where, $t_j = \tau_{i,1} + nT + \tau_{1,N} + \tau_{N,j}$ (see Figure 3.9)

Note that, according to the medium access protocol, station j cannot start the new cycle until t_c is at least t_j, even if j is ready before this time. Therefore, the smallest possible value of t_c is t_j.

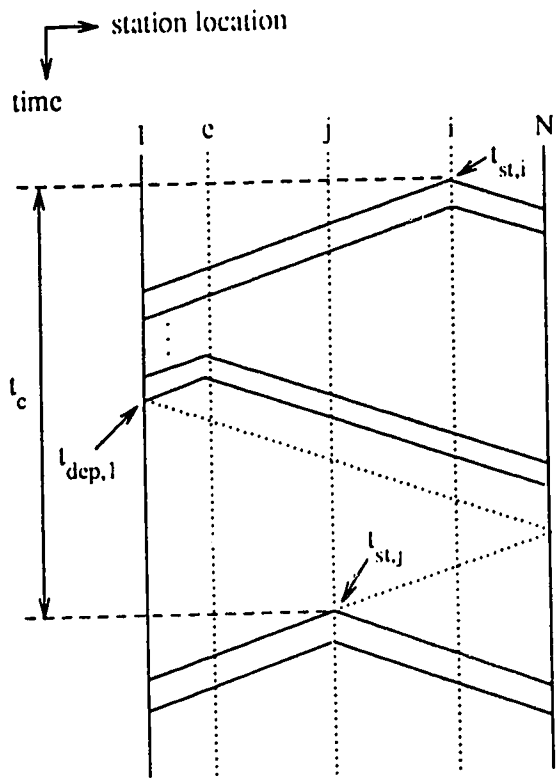


Figure 3.9. Time-space diagram for Subcase 2.1

Subcase 2.2 : $(t_j+2\tau_{j,m+1}) < t_c \leq (t_j+2\tau_{j,m})$; m=1,2,...,(j-1) (see Figure 3.10)

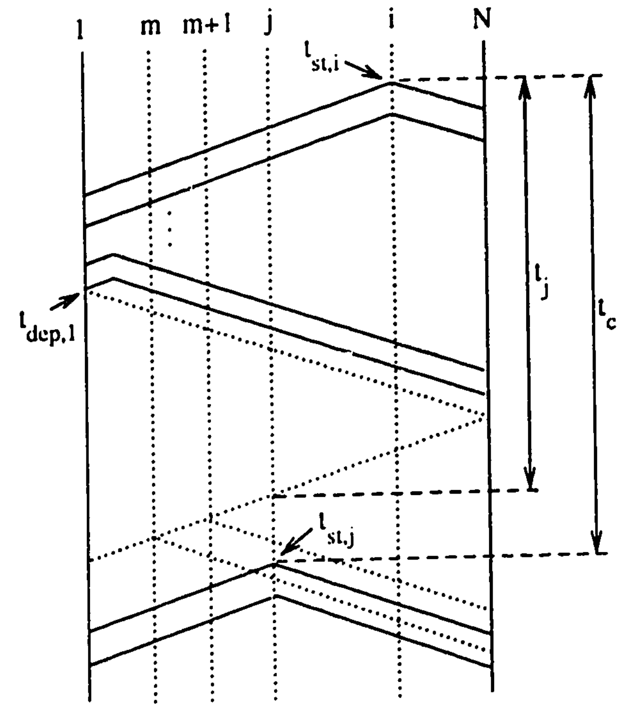


Figure 3.10. Time-space diagram for Subcase 2.2

This case is very similar in principle to subcase 1.2. However, only $(j-1)$ adjacent subintervals of t_c from t_j to $(t_j+2\tau_{j,1})$ are considered (instead of $(i-1)$ subintervals in subcase 1.2), with the range of each subinterval determined by m. For a given value of m, where $1 \leq m \leq (j-1)$, suppose the length of the cycle time, t_c, is such that $(t_j+2\tau_{j,m+1}) < t_c \leq (t_j+2\tau_{j,m})$. For some arbitrary value of t_c in this time interval, let $s_{j,l|n,e,i,k}^{2,2}(m,t_c)$ be the conditional state transition probability from state (i,k) to state (j,l), given m, n and e.

Integrating $s_{j,l|n,e,i,k}^{2,2}$ (m,t_c) over the time interval from $(t_j+2\tau_{j,m+1})$ to $(t_j+2\tau_{j,m})$, and summing up for all values of m, the conditional state transition probability $s_{j,l|n,e,i,k}^{2,2}$ is given by:

$$s_{j,l|n,e,i,k}^{2,2} = \sum_{m=1}^{j-1} \left[\int_{t_j+2\tau_{j,m+1}}^{t_j+2\tau_{j,m}} s_{j,l|n,e,i,k}^{2,2} \ (m,t_c) \ dt \right]$$

Subcase 2.3 : $t_c > (t_j + 2\tau_{j,1})$ (see Figure 3.11)

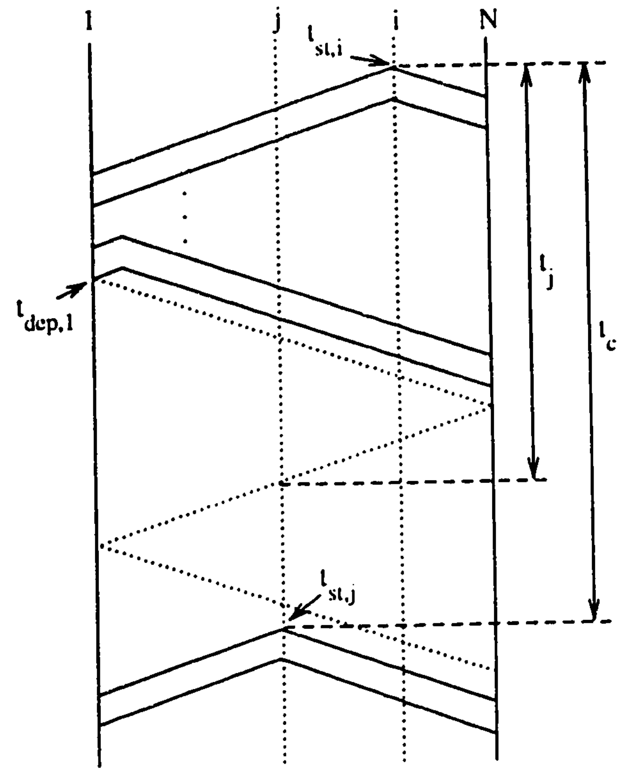


Figure 3.11. Time-space diagram for Subcase 2.3

Given n, e and some arbitrary value of t_c such that $t_c > (t_j + 2\tau_{j,1})$, let $s^{2,3}_{j,l \mid n,e,i,k}(t_c)$ be the conditional state transition probability from state (i,k) to state (j,l).

Integrating $s^{2,3}_{j,l \mid n,e,i,k}(t_c)$ from $(t_j + 2\tau_{j,1})$ to ∞, $s^{2,3}_{j,l \mid n,e,i,k}$ for subcase 2.3 is given by:

$$s^{2,3}_{j,l \mid n,e,i,k} = \int_{t_j + 2\tau_{j,1}}^{\infty} s^{2,3}_{j,l \mid n,e,i,k}(t_c)\, dt$$

The probability $S_Z(j,l \mid n,e,i,k)$ for Case 2 is therefore given by:

$$S_Z(j,l \mid n,e,i,k) = s^{2,1}_{j,l \mid n,e,i,k} + s^{2,2}_{j,l \mid n,e,i,k} + s^{2,3}_{j,l \mid n,e,i,k} \qquad \text{(A.3.8.1)}$$

The procedure for evaluating $s^{2,x}_{j,l \mid n,e,i,k}$ for each subcase $x=1$, 2 and 3, is similar in principle to Case 1. $s^{2,x}_{j,l \mid n,e,i,k}$ is given by:

$$s^{2,x}_{j,l \mid n,e,i,k} = p^{2,x}_{nr} \cdot p^{2,x}_{j} \cdot p^{2,x}_{l} \ , \quad x=1,2,3$$

where,

$p_{nr}^{2.x}$ = prob(each station $r, r=j+1, j+2, ... N$, remaining in the idle state until the time instant $(t_{st\,j}+\tau_{j,r})$)

$p_j^{2.x}$ = prob(station j in the ready state at time $t_{st\,j}$), and

$p_l^{2.x}$ = prob(l stations from the set $\{1,2,...,j-1\}$ in the ready state at time $t_{st\,j}$)

Expressions for $p_{nr}^{2.x}$, $p_j^{2.x}$ and $p_l^{2.x}$ can be derived using principles similar to that of Case 1, with some variations in the intermediate steps. $S_Z(j,l \mid n,e,i,k)$ for Case 2 can therefore be evaluated using equation (A.3.8.1). □

Appendix 3.9: Derivation of $S_X(b,d,j,l \mid n,e,R-L,c,i,k)$ when $j \le i$ (Case 2)

As in the Case 1, four subcases 2.1, 2.2, 2.3 and 2.4 are considered, depending on the length of the cycle time t_c. The conditional probability $S_X(b,d,j,l \mid n,e,R-L,c,i,k)$ for each subcase is denoted by $s_{b,d,j,l \mid n,e,R-L,c,i,k}^{2.x}$, where $x = 1, 2, 3$ or 4.

Subcase 2.1 : $t_c = \tau_{i,1}+nT+\tau_{1,j}$ (see Figure 3.12)

According to the X-Net medium access protocol, the earliest time that station j can start the new cycle is after a time interval of $(\tau_{i,1}+nT+\tau_{1,j})$ from the start of the previous cycle. The cycle type in this case is *L-R*.

Subcase 2.2 : $t_c = \tau_{i,1}+nT+\tau+\tau_{N,j}$ (see Figure 3.13)

If station j was not ready to start the new cycle under subcase 2.1, according to the medium access protocol, the next earliest time that j can start the cycle is when t_c reaches $(\tau_{i,1}+nT+\tau+\tau_{N,j})$ as shown in Figure 3.13. Note that the cycle type in this case (and the remaining cases 2.3 and 2.4) is L-R.

Subcase 2.3 : $(t_j+2\tau_{j,m+1}) < t_c \le (t_j+2\tau_{j,m})$; m=1,2,...,(j-1) (see Figure 3.14)

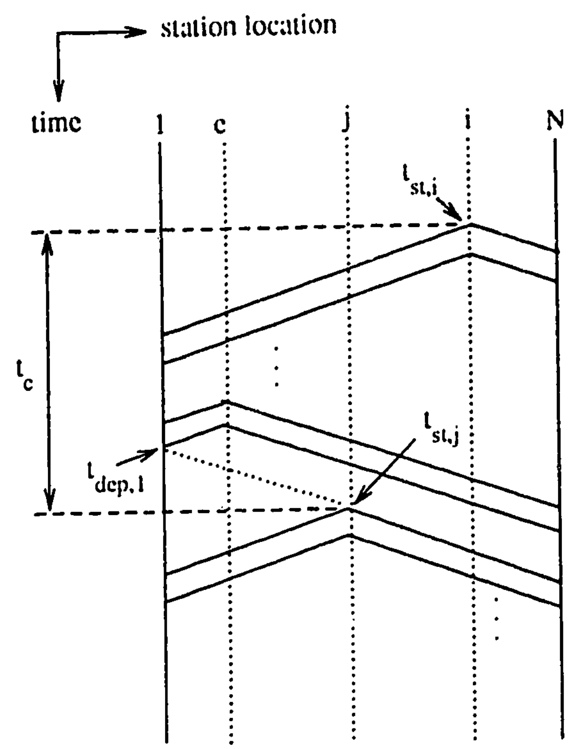


Figure 3.12 . Time-space diagram for Subcase 2.1

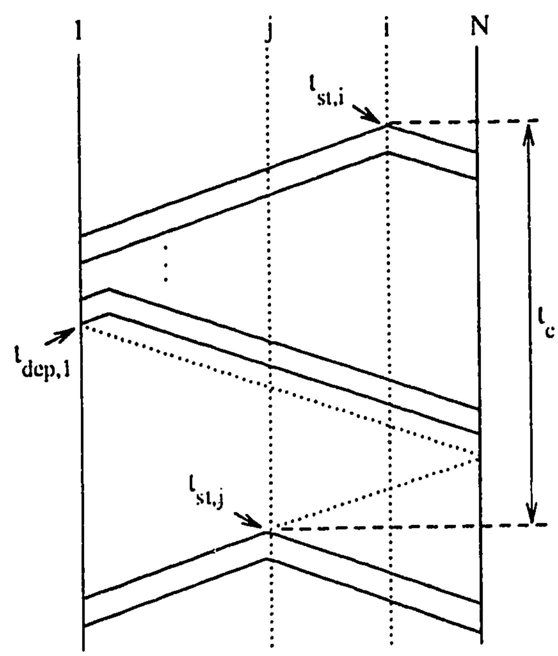


Figure 3.13. Time-space diagram for Subcase 2.2

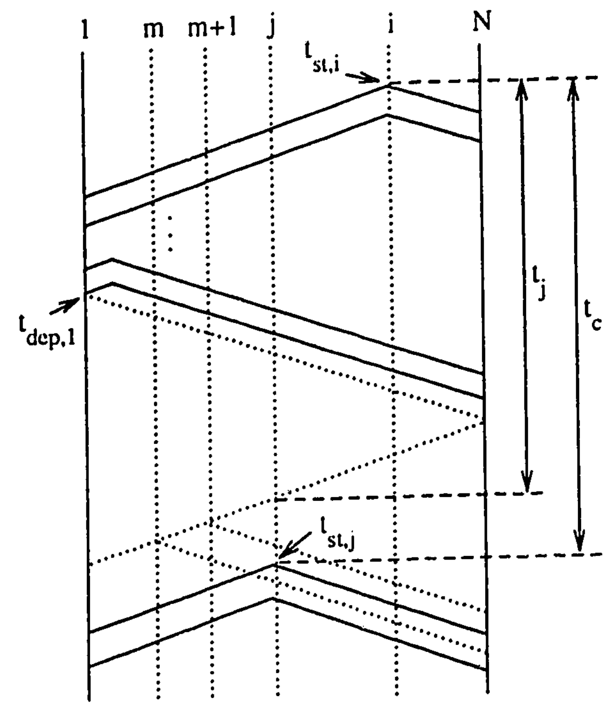


Figure 3.14. Time-space diagram for Subcase 2.3

As in subcase 1.3, $(j-1)$ adjacent subintervals of t_c from t_j to $(t_j+2\tau_{j,1})$ are considered (where, $t_j = (\tau_{i,1}+nT+\tau+\tau_{N,j})$), with the range of each sub interval being determined by m. For a given value of m, where $1 \le m \le (j-1)$, suppose the length of the cycle time, t_c, is such that $(t_j+2\tau_{j,m+1}) < t_c \le (t_j+2\tau_{j,m})$. For some arbitrary value of t_c in this time interval, let $s^{2.3}_{b,d,j,l\,|\,n,e,a,c,i,k}(m,t_c)$ be the conditional state transition probability from state (a,c,i,k) to state (b,d,j,l), given m, n and e.

The conditional state transition probability $s^{2.3}_{b,d,j,l\,|\,n,e,a,c,i,k}$ is then expressed as:

$$s^{2.3}_{b,d,j,l\,|\,n,e,a,c,i,k} = \sum_{m=1}^{j-1} \left[\int_{t_j+2\tau_{j,m+1}}^{t_j+2\tau_{j,m}} s^{2.3}_{b,d,j,l\,|\,n,e,a,c,i,k}\ (m,t_c)\ dt \right]$$

Subcase 2.4 : $t_c > (t_j+2\tau_{j,1})$ (see Figure 3.15)

Given n, e and some arbitrary value of t_c such that $t_c > (t_j+2\tau_{j,1})$, let $s^{2.4}_{b,d,j,l\,|\,n,e,a,c,i,k}(t_c)$ be the conditional state transition probability from state (a,c,i,k) to state (b,d,j,l).

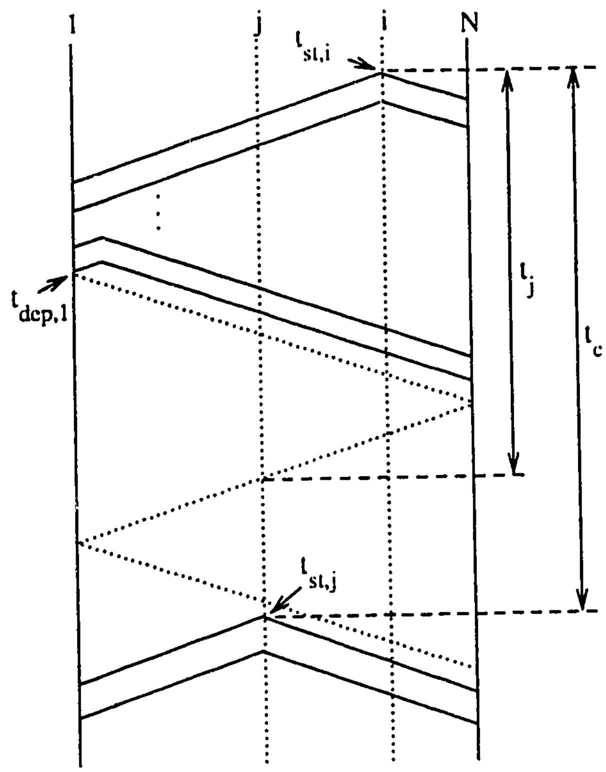


Figure 3.15. Time-space diagram for Subcase 2.4

Then $s^{2.4}_{b,d,j,l \mid n,e,a,c,i,k}$ for subcase 2.4 is given by:

$$s^{2.4}_{b,d,j,l \mid n,e,a,c,i,k} = \int_{t_j+2\tau_{j,i}}^{\infty} s^{2.4}_{j,l \mid n,e,a,c,i,k}(t_c)\, dt$$

Subcase 2.1 is the only one in which station j can start a cycle of type L-R. In this case, the mode d of station j just before the start of the cycle is *controlled*. In the remaining cases 2.2, 2.3 and 2.4, the type of cycle started by station j is of type R-L. In these cases, the mode d of station j is *random*. With these observations, the conditional probability $S(b,d,j,l \mid n,e,R-L,c,i,k)$ for Case 2 can then be evaluated as follows:

$$S_X(L-R, random, j, l \mid n, e, R-L, c, i, k) = 0$$

$$S_X(L-R, controlled, j, l \mid n, e, R-L, c, i, k) = s^{2.1}_{L-R, controlled, j, l \mid n, e, R-L, c, i, k}$$

$$S_X(R-L, random, j, l \mid n, e, R-L, c, i, k) = s^{2.2}_{R-L, random, j, l \mid n, e, R-L, c, i, k} +$$

$$s^{2.3}_{R-L,random,j,l \mid n,e,R-L,c,i,k} + s^{2.4}_{R-L,random,j,l \mid n,e,R-L,c,i,k}$$

$$S_X(R-L, controlled, j, l \mid n, e, R-L, c, i, k) = 0$$

$s^{2.x}_{b,d,j,l \mid n,e,R-L,c,i,k}$ for each subcase x, where $x=1, 2, 3$ and 4, is evalauated using:

$$s^{2.x}_{b,d,j,l \mid n,e,R-L,c,i,k} = p^{2.x}_{nr} \cdot p^{2.x}_{j} \cdot p^{2.x}_{l} \ ; \ x=1,2,3,4$$

where,

$p^{2.x}_{nr}$ = prob(when the *server enters*, each station upstream from j is not ready) [6]

$p^{2.x}_{j}$ = prob(station j in the ready state at time $t_{st,j}$), and

$p^{2.x}_{l}$ = prob(l stations downstream from j in the ready state at time $t_{st,j}$).

The expressions for above probabilities can be derived following similar principles used in Case 1, with some variations in the intermediate steps. Thus, $S_X(b, d, j, l \mid n, e, R-L, c, i, k)$ for Case 2 (i.e., j≤i) can be computed. □

[6] Therefore, these upstream stations do not start the new cycle.

Chapter 4

Performance Evaluation of Z-Net and X-Net

The performance of Z-Net and X-Net is evaluated in this chapter. First, the analytic models developed in Chapter 3 are used to study the channel utilization and the delay characteristics of the protocols. These results are validated by comparing them with results obtained using simulation models. The simulation models are based on discrete-event simulation. The single-run method is used with the run time divided into 61 contiguous batch intervals. The mean values of channel utilization and delay of each batch are considered as independent samples and these sample values are used to compute the mean throughput and delay at a particular value of the offered load. Stations with single-capacity and infinite-capacity buffers have been incorporated in the simulations. Performance of the proposed schemes is compared with other recently proposed bus LAN schemes using the maximum achievable channel utilization and the insertion delay as performance measures. Towards the end of the chapter, possible performance improvements in Z-Net are discussed. The final section contains a discussion of the effect of the parameter EC[1] on the performance of X-Net.

4.1. Channel Utilization

Figures 4.1 and 4.2 show the variations of channel utilization with offered load for different values of **a** in Z-Net and X-Net, respectively. These results are obtained using the analytic models developed in Chapter 3. Ten equally spaced, identical, single-buffered stations have been considered. Packet lengths are assumed to be fixed with the packet interarrival times being exponentially distributed. As seen from the figures, for a given value of the offered load, channel utilization decreases with increasing a. This is especially apparent at higher offered loads, which is

[1] As mentioned in Chapter 2, the parameter EC specifies the number of consecutive *empty cycles* necessary to switch from the *controlled* mode to the *random* mode in X-Net.

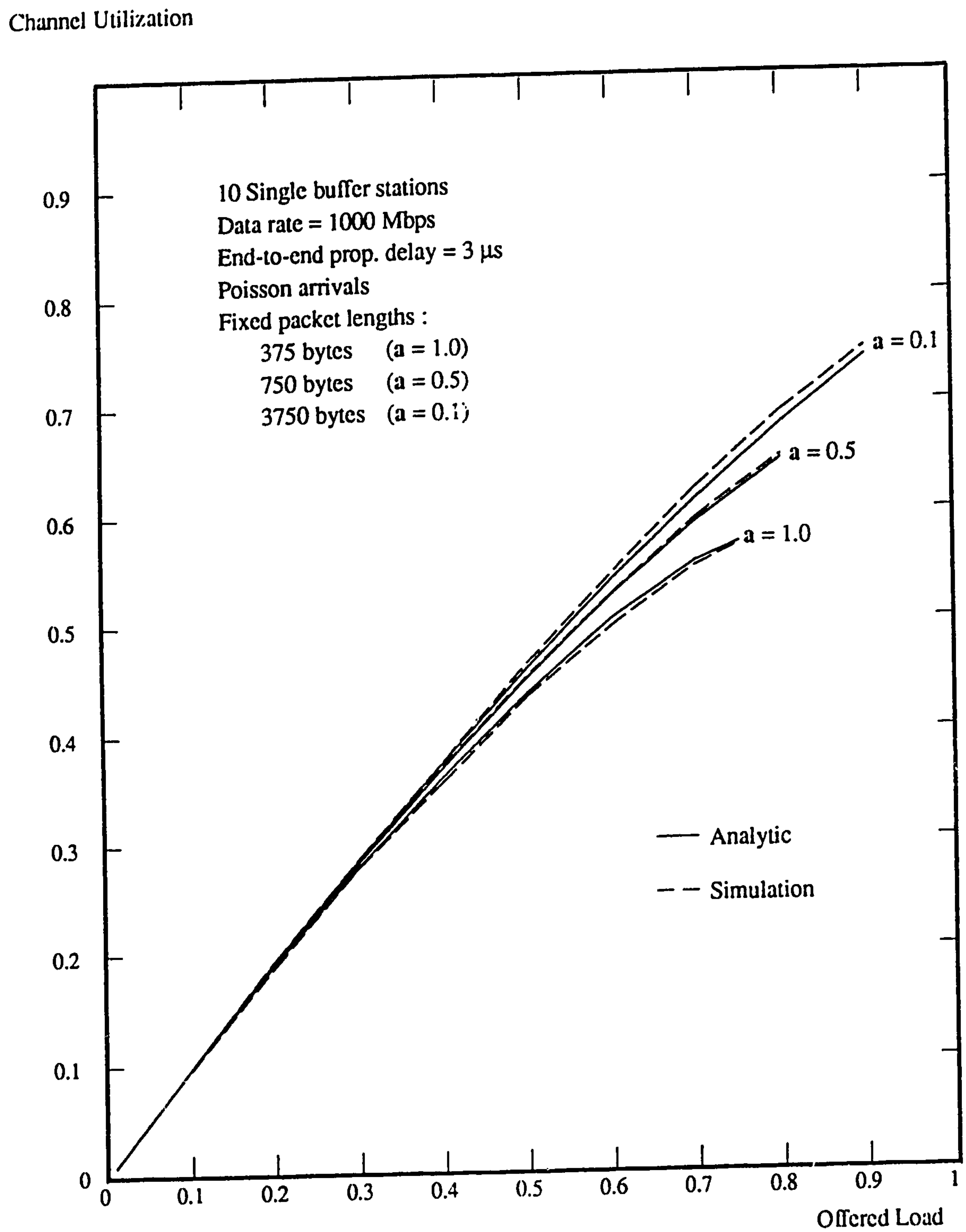


Figure 4.1. Channel Utilization vs Offered Load for Z-Net

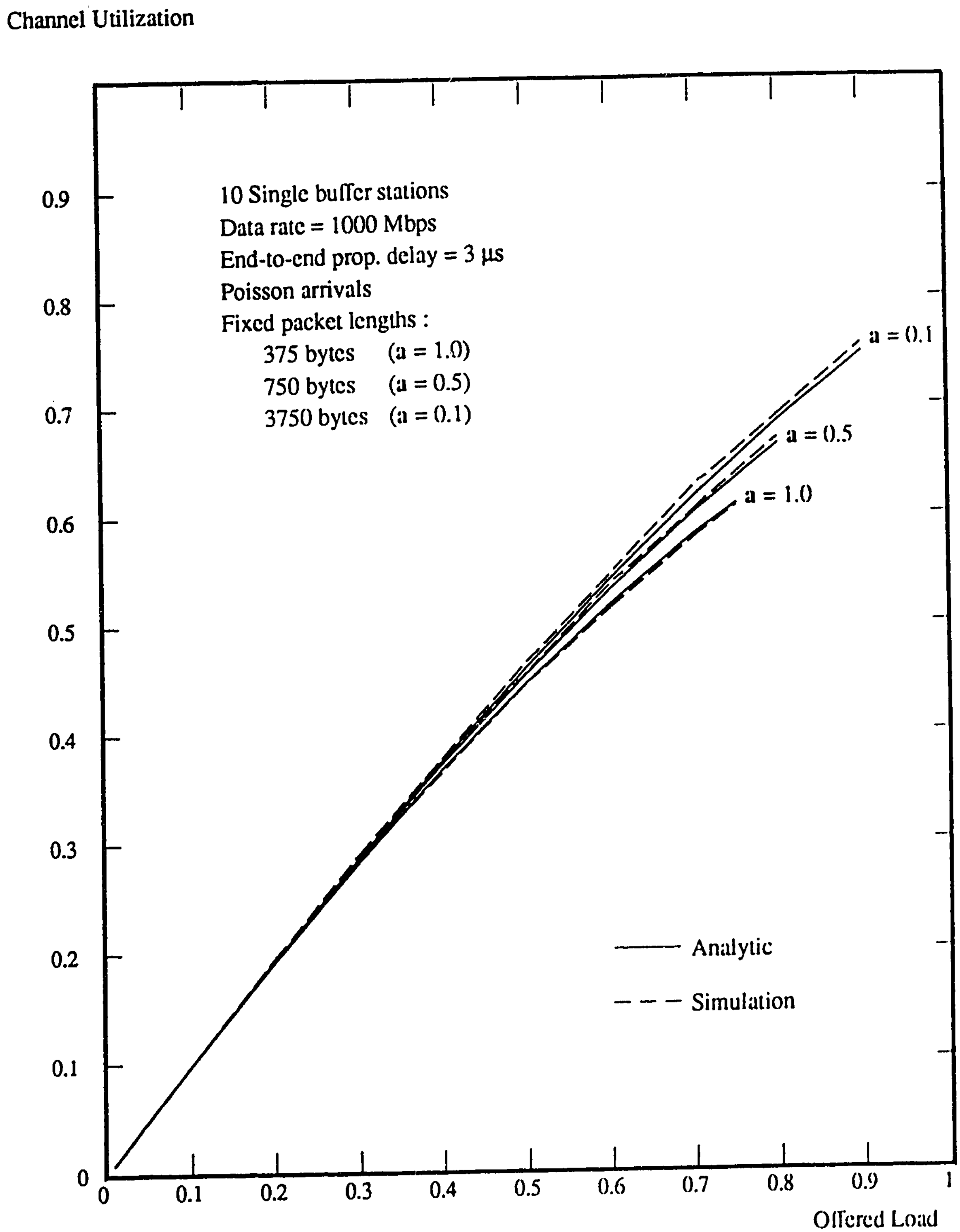


Figure 4.2. Channel Utilization vs Offered Load for X-Net

because of the increasing channel idle time, with increasing a, compared to the time during which the channel is utilized for packet transmissions. We have developed simulation models of Z-Net and X-Net to validate the results obtained from the analytic models. In Figures 4.1 and 4.2, the simulation results are indicated with dashed curves (using the same assumptions and parameters as in the analytic models). As seen from the figures, the analytic and simulation results are in very close agreement, in both Z-Net and X-Net.

Channel utilizations of Z-Net and X-Net are close to each other at low values of a. Figure 4.3 shows the channel utilization of the Z-Net and X-Net protocols at higher values of a. Simulation models have been used to obtain these results. At high values of a, the superior channel utilization of X-Net is clearly visible from the figure. This is expected, because in X-Net, the overhead time between two consecutive transmission cycles is less than that of the Z-Net.

4.2. Packet Transfer Delay

Packet transfer delays in Z-Net and X-Net are estimated using the equation (3.8) derived in Chapter 3. Channel utilization values computed from the analytic models are substituted in this expression to obtain the packet transfer delays. These results are presented in Figures 4.4 and 4.5 for Z-Net and X-Net, respectively. At very low values of utilization (i.e., at light loads) the insertion delays in both Z-Net and X-Net are zero. This is because, at light load, an idle station will be in the *random* mode most of the time, and therefore, can start a packet transmission as soon as it becomes ready. Therefore, at light loads, the packet transfer delay in both schemes consists of mainly the channel propagation delay. As utilization increases, delay increases gradually. Packet transfer delays obtained using the simulation models are also indicated in Figures 4.4 and 4.5 by dashed curves. The 95% confidence intervals at the simulation points are also shown. In equation (3.7), the mean insertion delay $\bar{d}_{ins}$ is very sensitive to the mean channel utilization $\bar{U}$, especially at low values of $\bar{U}$. Therefore, even though the results of analytic and simulation models for channel utilization match very closely, the results of the analytic and simulation models for packet

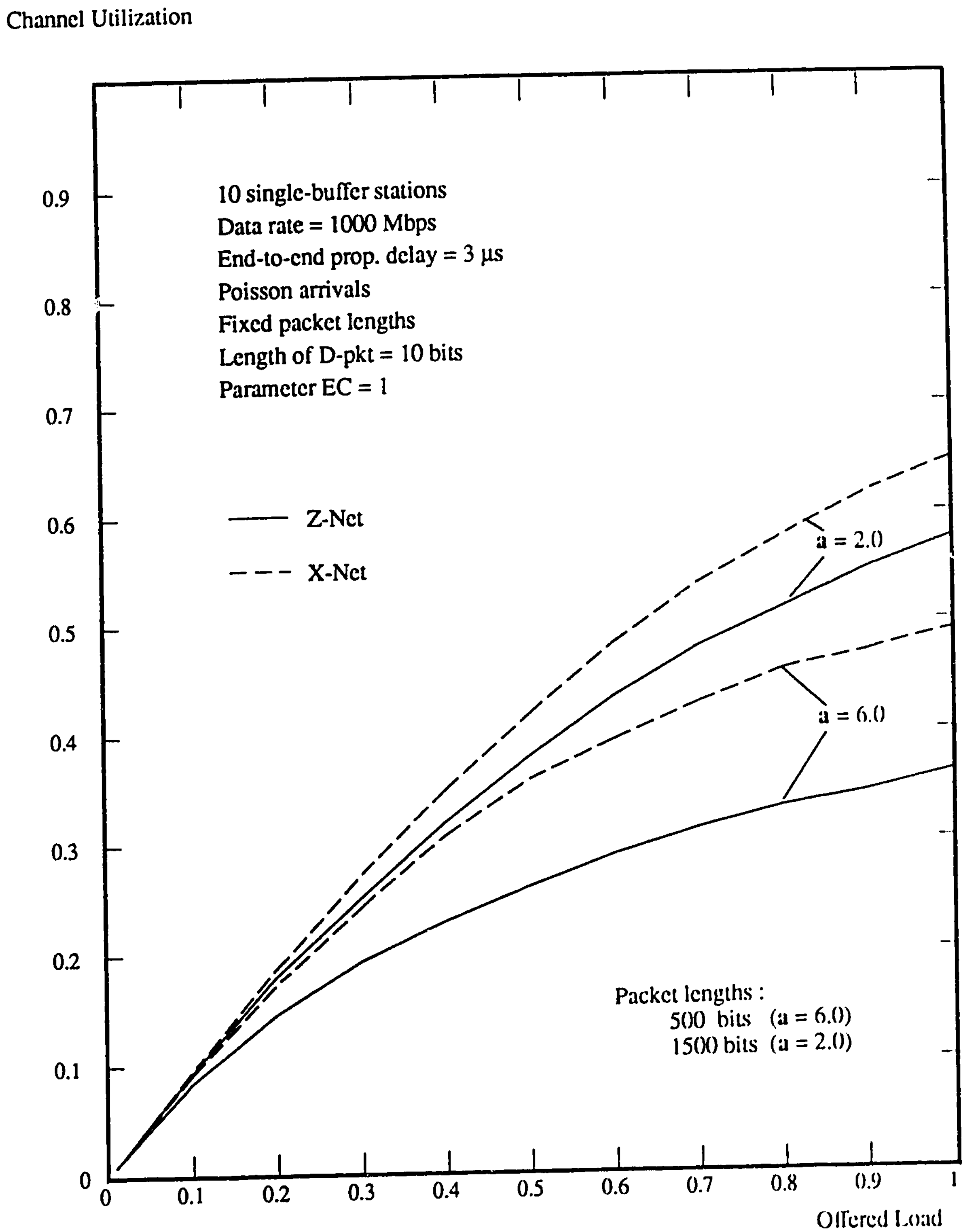


Figure 4.3. Channel Utilization vs Offered Load for Z-Net and X-Net at high a

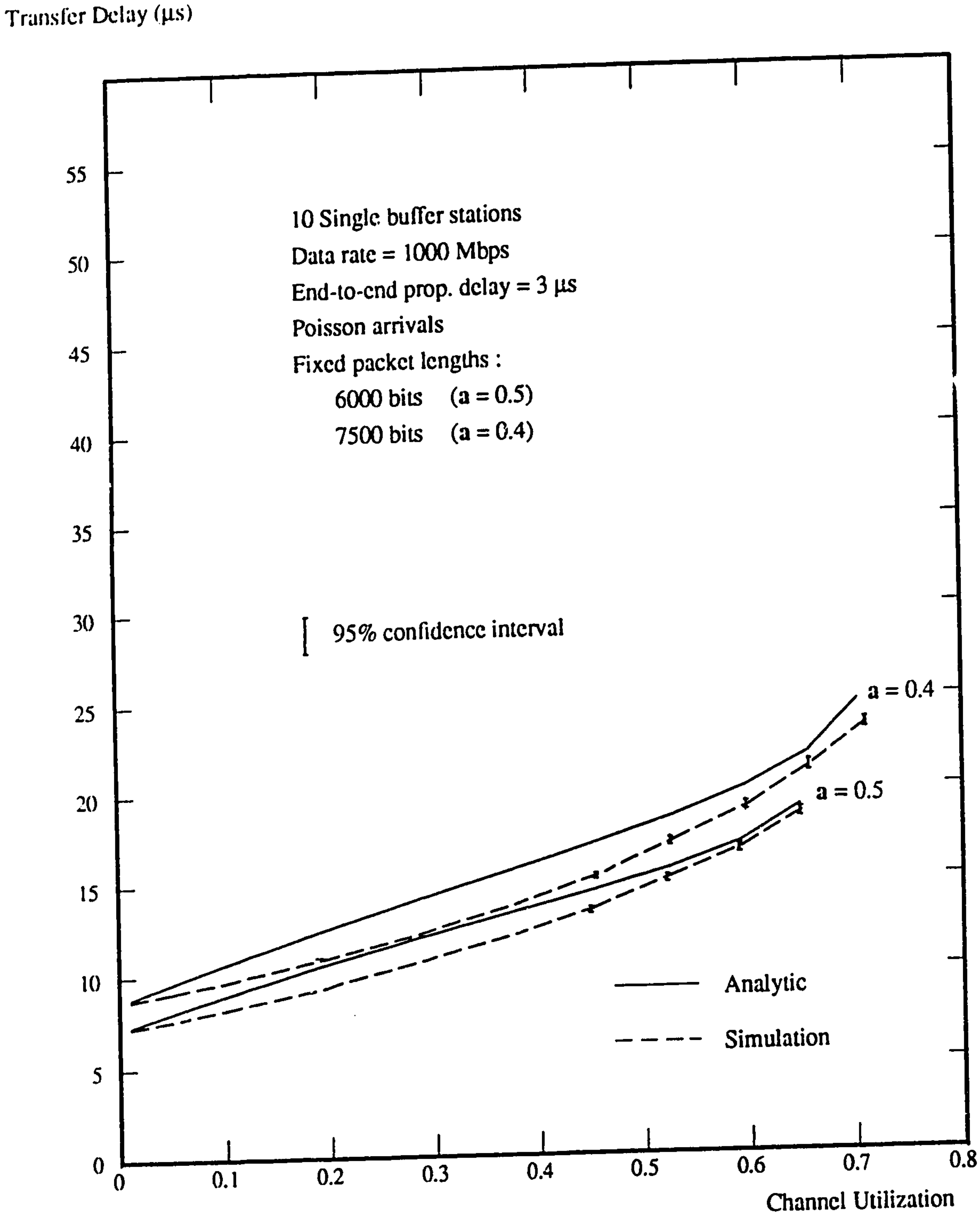


Figure 4.4. Transfer Delay vs Channel Utilization for Z-Net

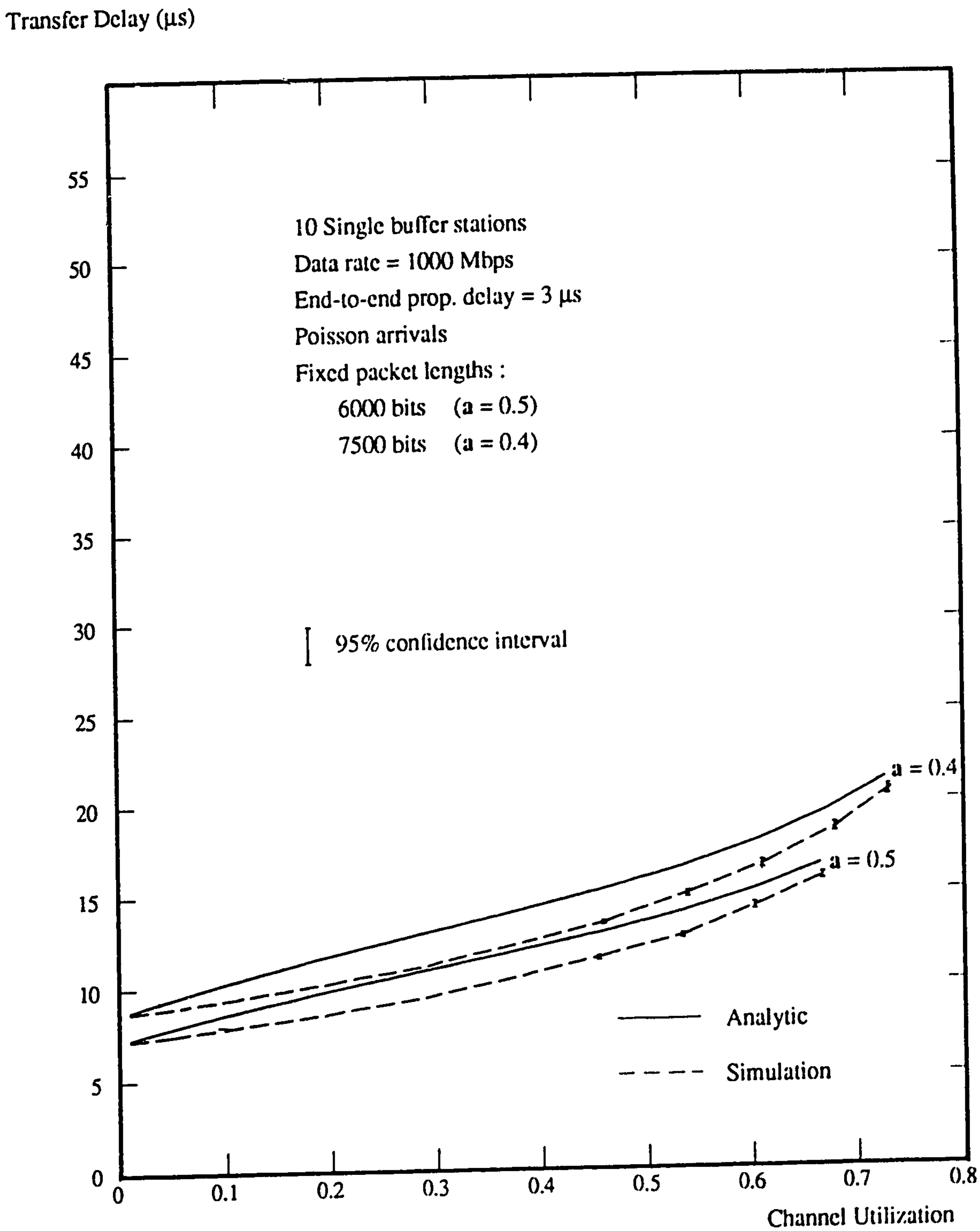


Figure 4.5. Transfer Delay vs Channel Utilization for X-Net

transfer delay show a wider difference at the lower and intermediate range of utilization.

4.3. Comparison of Performance with other Bus Schemes

Maximum achievable channel utilization is used as one of the measures to compare the performance of Z-Net and X-Net with other recently proposed bus schemes. Expressions for the maximum channel utilization of Z-Net and X-Net can be derived as follows:

Let x stations out of a total of N be continuously backlogged. During the time interval between two consecutive cycles, x packets are transmitted. Therefore, the channel utilization $U_{N,x}$ under this condition is given by:

Z-Net:

$$U_{N,x} = \frac{x \times \text{Mean packet length (bits)}}{\text{Mean time between two consecutive cycles (bits)}}$$

With x transmissions in a cycle, neglecting the inter-packet gaps, the time interval between two consecutive cycles in Z-Net is approximately given by $(xT + 2\tau)$. Therefore,

$$U_{N,x} = \frac{xT}{xT + 2\tau} = \frac{1}{1 + \frac{2a}{x}} ,$$

where $a = \frac{\tau}{T}$

X-Net:

Consider two consecutive cycles of transmissions in the *controlled* mode. During the time interval between the start of the first cycle and the end of the second cycle, $2x$ packets are transmitted. Therefore, $U_{N,x}$ is given by:

$$U_{N,x} = \frac{2x \times \text{Mean packet length (bits)}}{\text{Length of R-L and L-R cycles (bits)}}$$

Ignoring the inter-packet gaps and the length of the *D-pkts*, the channel utilization is approximately given by:

$$U_{N,x} = \frac{2xT}{2xT + 2\tau} = \frac{1}{1 + \frac{a}{x}}$$

The above expressions yield the maximum utilization, U_{max}, when $x = N$, i.e., when all the stations are continuously backlogged. The maximum achievable channel utilization for several bus network schemes is shown in Table 4.1. These are plotted as a function of a in Figure 4.6 for two different values of N.

Table 4.1: Maximum channel utilization for different bus networks

Network	Max. Channel Utilization
Buzznet	$\frac{1}{1 + 6a/N}$
D-Net, Expressnet, Z-Net	$\frac{1}{1 + 2a/N}$
DQDB	1
Fasnet	$\frac{1}{1 + (\lceil 2a \rceil + 1)/N}$
TLP-1, 2	$\frac{1}{1 + 3a/N}$
TLP-3, X-Net	$\frac{1}{1 + a/N}$

As seen from Figure 4.6, DQDB achieves the highest utilization, which is almost insensitive to a. However, in DQDB, stations are required to know the relative locations of other stations in the network. The channel utilizations of X-Net and TLP-3 at heavy load are similar, and are better than that of Buzznet, Expressnet, Fasnet, TLP-1,2 and Z-Net. Even though the U_{max} of these networks decrease with increasing a, with large number of backlogged stations, high utilizations can be achieved for a wide range of a values.

We use insertion delay as a means of comparing the delay performance of Z-Net and X-Net with other bus schemes. Figure 4.7 shows the variation of the insertion delay with the utilization for Z-Net, X-Net and several other network schemes. Except for Z-Net and X-Net results, other results in this figure are based on the simulation results presented in [Gerla et. al 1985]. As seen

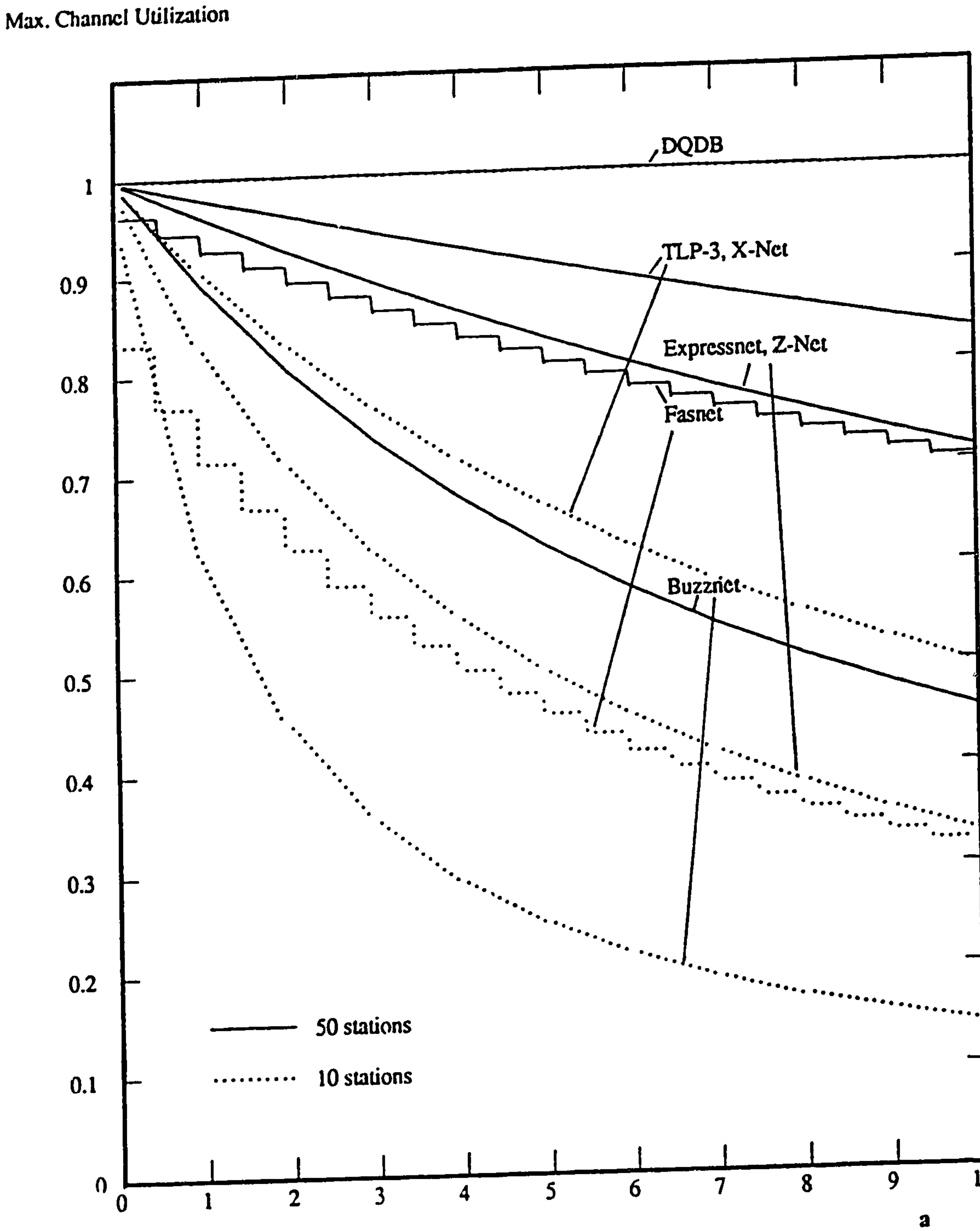


Figure 4.6. Maximum Channel Utilization vs a for several bus networks

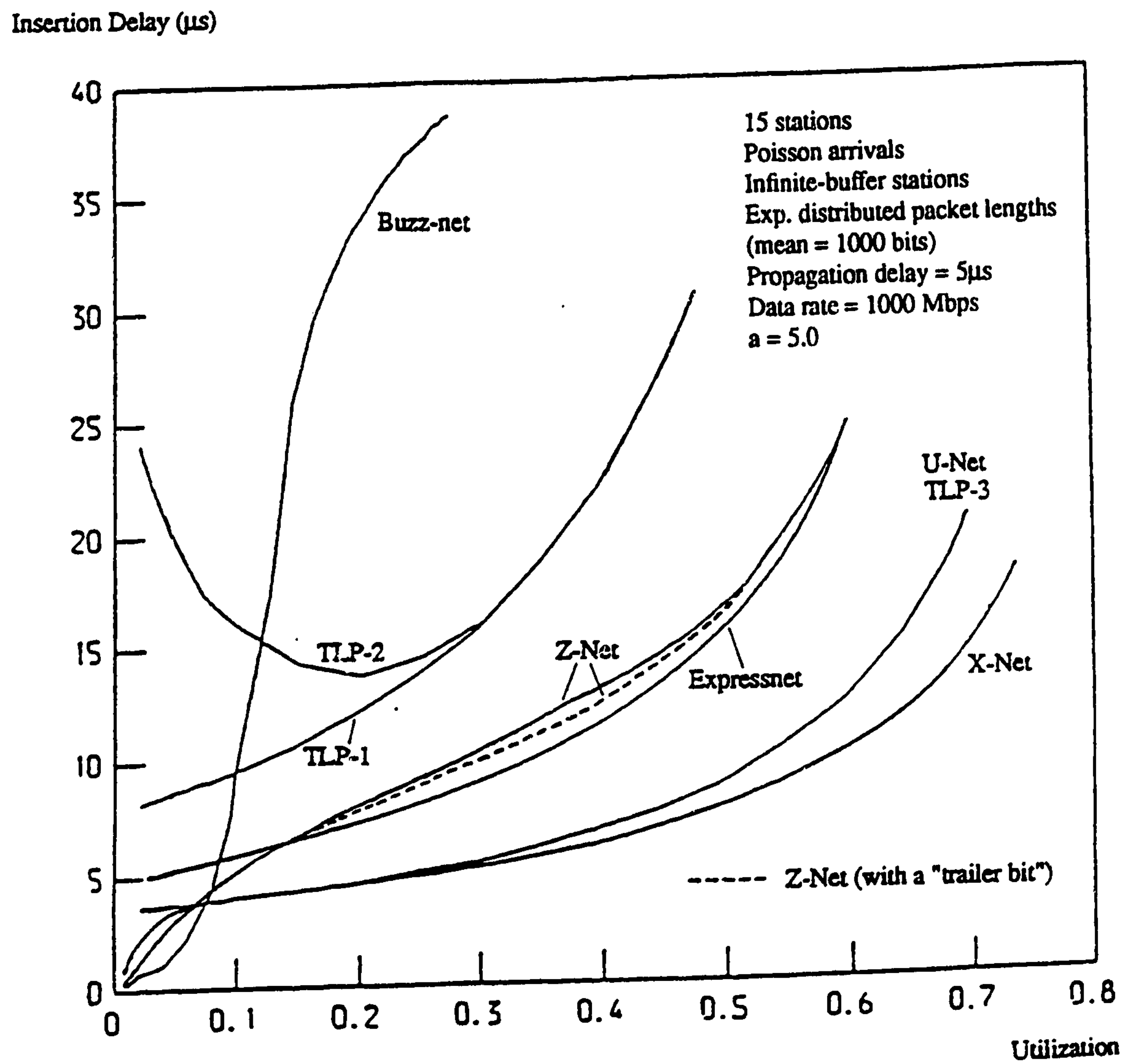


Figure 4.7. Insertion Delay vs Channel Utilization for several bus networks

from the figure, the insertion delay of Z-Net is better than that of TLP-1 and TLP-2 at all loads. Compared to Buzznet, Z-Net achieves superior delay performance, except at light load. At intermediate loads, Expressnet outperforms Z-Net. X-Net performance at light load is slightly worse than that of Buzznet and Z-Net, but superior to Expressnet, U-Net and Tokenless protocols. At all other loads,[2] the delay performance of X-Net is better than all the other schemes considered.

4.4. Performance Improvements in Z-Net

In Z-Net, as discussed in Chapter 2, a station i that has completed transmission retransmits its packet if it senses beginning-of-carrier (BOC) on the R-L bus, before ACL_i reaches 2τ. If the upstream station on the R-L bus which originated the second transmission has not blocked any transmissions on the L-R bus, channel bandwidth is wasted by station i transmitting a duplicate. Some of these unnecessary retransmissions can be eliminated by attaching an additional bit (called the *trailer* bit) to the end of a packet. A station j in the *Txmit* state, detecting any BOC on the L-R bus during its transmission will set the *trailer* bit to 1. On the other hand, if station j has not blocked any transmissions on the L-R bus, it sets the *trailer* bit to zero. A downstream station i (on the R-L bus) in the *Txmitted* state will wait for the *trailer* bit when it senses BOC on the R-L bus. A retransmission is effected only if the *trailer* bit is 1, which indicates that an upstream station (on the R-L bus) has blocked a transmission (not necessarily station i's) on the L-R bus. This technique minimizes unnecessary retransmissions, but it cannot completely eliminate them. The dashed curve in Figure 4.7 shows the simulation results of insertion delay when a *trailer* bit is used in Z-Net. As seen from the figure, the performance improves slightly, especially in the intermediate range of utilizations.

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[2] At heavy load, we expected the delay performance of U-Net and TLP-3 to be similar to that of X-Net. For some reason not known to us, the insertion delay of U-Net and TLP-3 appear to exceed that of X-Net at heavy load.

4.5. Effect of the parameter EC on the Performance of X-Net

In X-Net, more transitions are made from the *controlled* to the *random* mode when EC=1, compared to other values of EC. This is because, with higher values of EC, a greater number of consecutive empty cycles are required for switching to the *random* mode. Figure 4.8 shows the simulation results of the insertion delay against channel utilization with different values of EC for a = 5. As the load increases, the probability of a cycle being empty decreases. Therefore, at high loads, even with EC=1, a station is in the *controlled* mode most of the time. Thus, at high loads, the parameter EC will have no effect on the performance, as seen from Figure 4.8. At low loads, however, the probability of several consecutive empty cycles is higher (compared to heavy load). Therefore, at low loads, the number of transitions from the *controlled* mode to the *random* mode can be controlled by varying EC. With low values of EC, the delay performance is better at light loads. This is because, the bulk of the transmissions is started from the *random* mode with zero delay. Only very few transmissions are made in the following *transient* and *controlled* modes. At intermediate loads however, a transmission in the *random* mode is usually followed by several other transmissions in the *transient* and *controlled* modes. When the mean packet length is small compared to τ (i.e., with high a), even after all transmissions in the *transient* mode are over, there could be an additional waiting time (as seen from Figure 2.6.b) before switching to the *controlled* mode to start the next cycle. Therefore, with high a values, at low loads, better performance can be achieved by allowing less transitions to the *random* mode. The reason being, even though the delay is zero for the station that starts transmission in the *random* mode, the other stations that have to wait their turn in the *controlled* mode now have to wait for a longer period. At high a values therefore, there is a trade-off in performance with different EC values. At light loads, performance is better with low EC, but this yields slightly higher delay at intermediate loads. With increasing EC, delay increases slightly at light load, but decreases at intermediate loads. This effect is clearly visible from Figure 4.8. Therefore, with an intermediate value of EC, a compromise between these two extremes can be achieved. When EC is made extremely large, all

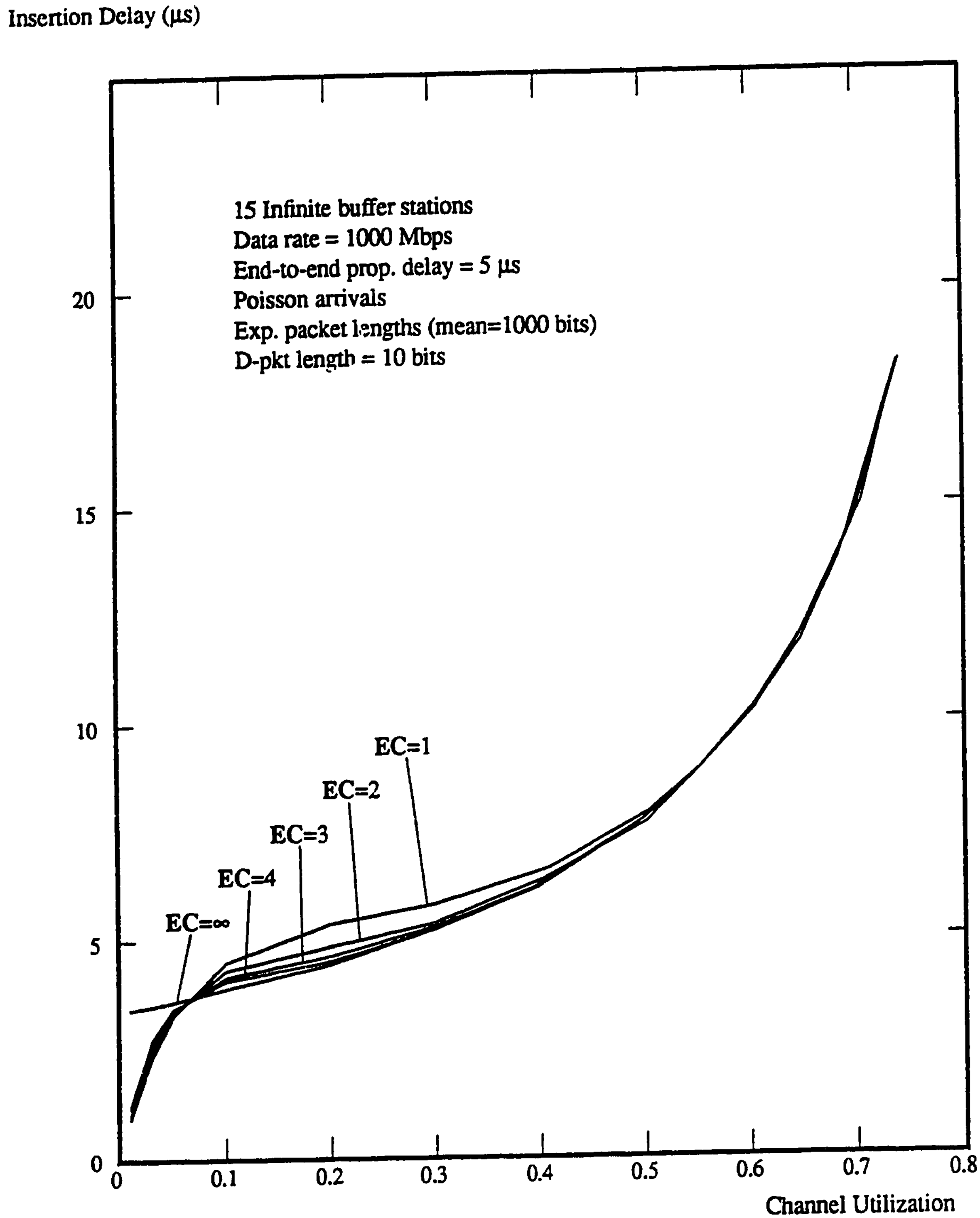

Figure 4.8. Variation of Insertion Delay with parameter EC in X-Net (a = 5.0)

transmissions (at any load) take place only in the *controlled* mode, as no transitions are made to the *random* mode. The operation of X-Net under this situation is similar to that of U-Net and TLP-3. The trade-off in performance discussed above is not present at lower a values. This is because, there is no additional waiting time involved in switching from the *transient* to the *controlled* mode (see Figure 2.6.a) in this case. Once the transmissions in the *transient* mode are over, *D-pkt* transmission can immediately be started to switch to the *controlled* mode and start the next cycle. The simulation results presented in Figure 4.9 for a = 0.2 with different values of EC show the insensitivity of performance to the parameter EC when a is small.

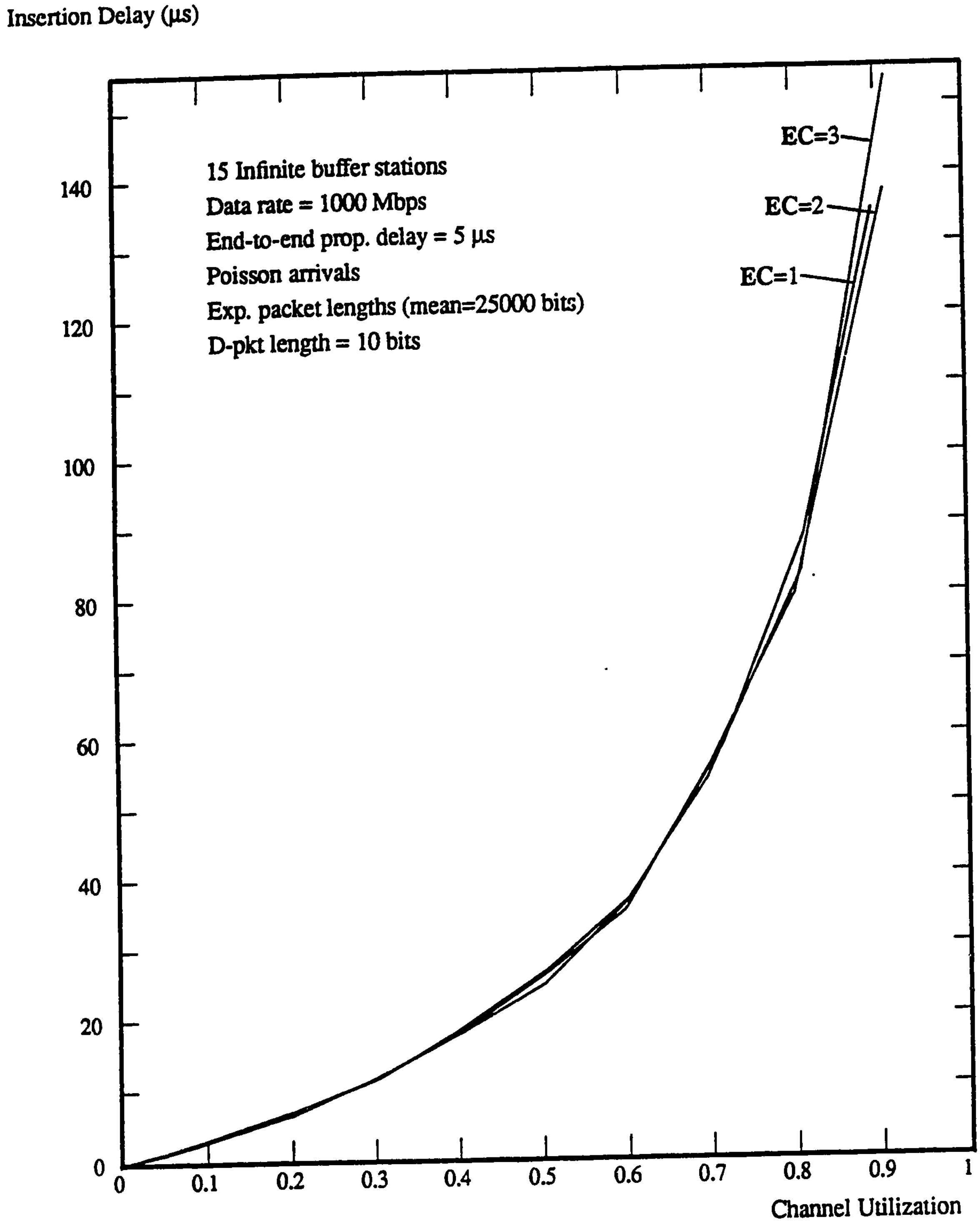

Figure 4.9. Variation of Insertion Delay with parameter EC in X-Net (a = 0.2)

Chapter 5

Conclusions and Directions for Further Research

In this thesis, we have proposed two new local area network protocols named Z-Net and X-Net, and evaluated their performance using analytic and simulation models. The network architecture of both Z-Net and X-Net consists of two, unidirectional fiber-optic channels. The medium access protocols of the proposed schemes fall into the category of *hybrid access* protocols. This is because they behave as *random access* schemes at light load and as *controlled access* schemes (with implicit token-passing) at higher loads.

A novel feature of the network architecture is the use of active switches in interfacing a station to a fiber-optic bus, to reduce the waste of bandwidth due to packet collisions. This is achieved by allowing one transmission to continue to completion, while other participants in the collision abort their transmissions. Z-Net uses active switches on the L-R bus and passive switches on the R-L bus. In Z-Net, a transmitting station defers to upstream transmissions on the R-L bus. At the start of a packet transmission, a station applies a voltage to the active switch, thereby altering the tap ratio to a very low value. This allows the station to intercept transmissions that it may detect from the upstream side of the L-R bus during its packet transmission. A station continues its transmission, regardless of any transmissions detected on the L-R bus. However, these upstream transmissions on the L-R bus are intercepted and not allowed to propagate any further down the L-R bus. With this arrangement, when packet collisions occur, the transmitting station that is most upstream on the R-L bus continues its transmission to completion, while the other transmissions are aborted. A cycle of transmissions consists of packets from the most upstream ready station on the R-L bus towards the downstream ready stations (i.e., in the right to left direction, and therefore called an R-L cycle). During a cycle, all ready stations get an opportunity to transmit. At the end of a cycle, a new cycle is started again by the most upstream ready station on the R-L bus.

In X-Net, active switches are used on both buses. The operation of a transmission cycle in the right-to-left direction (i.e., an R-L cycle) is identical to a transmission cycle in Z-Net. Additionally, because of the use of active switches on the R-L bus, it is possible to have transmission cycles in the left-to-right direction (which are called L-R cycles) too. An R-L cycle is followed by an L-R cycle, in contrast to the (only) R-L cycles in Z-Net. The start of consecutive transmission cycles by the most upstream ready stations on the two buses in opposite and alternating directions reduces the overhead time between adjacent cycles. Therefore, as revealed from the performance results, X-Net exhibits superior throughput and delay performance compared to Z-Net.

The combination of random and controlled access protocol features of Z-Net and X-Net result in zero access delay at light load, while the delay is always bounded. The use of implicit token passing reaps the benefits of controlled access schemes while avoiding the problems of token loss, multiple tokens etc., in the explicit token schemes. Both medium access protocols are distributed, with all stations executing identical functions. This makes the network less vulnerable to single station failures. Both protocols need not know the location of other stations in the network. Therefore, if there are frequent changes in the station locations, much less maintenance effort is required.

The results obtained from the analytic and simulation models show that the performance of the proposed protocols does not degrade rapidly with increasing values of a. Therefore, they are suitable for operation in a high speed environment, where the packet transmission time could become comparable to, or even less than the end-to-end medium propagation delay. With the bounded delay property at all loads, the proposed networks are also suitable for supporting real-time traffic. Compared to Z-Net, the throughput and delay performance of X-Net is superior. However, in X-Net, stations are required to know their own locations in the network. In Z-Net, this knowledge is not required. Further, a physical implementation of X-Net would be more expensive than that of Z-Net, because of the use of active taps on both buses. Therefore, there is a trade-off of cost and performance between the two schemes. In applications where the performance is critical, X-Net may be used at a higher cost, while Z-Net may be used in other applications at a lower cost.

As a direction for further research, the extension of Z-Net and X-Net protocols for supporting different services (such as voice, data and video) could be identified. For this purpose, the protocols should be capable of supporting different levels of service priorities. For real-time traffic such as voice, the protocols should guarantee a certain channel bandwidth at periodic intervals. For traffic of non-time-critical nature, the remaining bandwidth should be allocated in a fair manner.

In the thesis, the effect of varying the parameter EC in X-Net has been discussed. In the proposed version of X-Net, a fixed value of EC is used. At high speeds (i.e., at high a), the effect of EC on performance has been demonstrated using simulation results. Further research may be undertaken to extend the X-Net protocol to dynamically select optimum values of EC with varying values of the offered load. Another area for future research is the fairness issues. Allocation of channel bandwidth equally among all stations (of the same priority) is important to ensure the fairness of the medium access protocol. In Z-Net and X-Net, the transmissions are arranged into cycles and each station gets an opportunity to transmit in a cycle. Therefore, the bandwidth allocation among stations in Z-Net and X-Net should be expected to be fair. An in-depth analysis of fairness of the Z-Net and X-Net protocols under different loading conditions and buffer capacities is an area that needs further study.

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