1987

Fail-Safe Local Area Networking Using Channel Redundancy.

Ibibia Karisemie Dabipi
Louisiana State University and Agricultural & Mechanical College

Order Number 8719857

Fail-safe local area networking using channel redundancy

Dabipi, Ibibia Karisemie, Ph.D.

The Louisiana State University and Agricultural and Mechanical Col., 1987

U·M·I
300 N. Zeeb Rd.
Ann Arbor, MI 48106

PLEASE NOTE:

In all cases this material has been filmed in the best possible way from the available copy. Problems encountered with this document have been identified here with a check mark ✓.

1. Glossy photographs or pages ______
2. Colored illustrations, paper or print ______
3. Photographs with dark background ______
4. Illustrations are poor copy ______
5. Pages with black marks, not original copy ______
6. Print shows through as there is text on both sides of page ______
7. Indistinct, broken or small print on several pages ✓
8. Print exceeds margin requirements ______
9. Tightly bound copy with print lost in spine ______
10. Computer printout pages with indistinct print ______
11. Page(s) ______ lacking when material received, and not available from school or author.
12. Page(s) ______ seem to be missing in numbering only as text follows.
13. Two pages numbered ______. Text follows.
14. Curling and wrinkled pages ______
15. Dissertation contains pages with print at a slant, filmed as received ______
16. Other ______

FAIL-SAFE LOCAL AREA NETWORKING USING CHANNEL REDUNDANCY

A Dissertation

Submitted to the Graduate Faculty of the
Louisiana State University and
Agricultural and Mechanical College
in partial fulfillment of the
requirements for the degree of
Doctor of Philosophy

in

The Department of Electrical and Computer Engineering

by
Ibibia Karisemie Dabipi
M.S.EE, Louisiana State University, 1981
B.S.EE, B.S. Physics/Mathematics,
Texas A & I University, 1979

May 1987

DEDICATED TO MY PARENTS

Sydney Tamunotonye Dabipi

and

Naomi Inibusealotomari Dabipi

WITH ALL MY LOVE.

ACKNOWLEDGEMENTS

The author wishes to express deep appreciation to Dr. Subhash C. Kak who has directed and guided him through this dissertation. His sincere gratitude to Dr. Andres Albanese of Bell Communications Research for his suggestions and encouragement. To his friends Tamunokuro Opuaji, Tamunoibuomi Abarabiye, Mr. and Mrs. Daniel Asechemie, Mr. and Mrs. Emmanuel E. Bassey, and those not mentioned, he offers his thanks for their encouragement and love throughout this period. The author also expresses his gratitude to Joeseph Meyinsse, Dennis Agboh for their help and encouragement. To his relations whose love and encouragement have sustained him through this period; he offers his thanks. To his wife Jene, and his brother Dixon, he express his love and thanks for being with him through these trying times. To his Professors Dr. Magd Zohdi, Dr. Olan Kruse and S. Burgin Dunn who have influenced his academic career he expresses his sincere gratitude. To his parents who made it all possible, it is impossible to express how grateful he is to them. He only hopes that he has made them proud.

Finally, to Mrs. Deborah Ann August whose patience and cooperation in typing this dissertation has made it all worthwhile, the author offers his appreciation.

TABLE OF CONTENTS

LIST OF FIGURES

FIGURE PAGE

FIGURE		PAGE

DABIPI, IBIBIA KARISEMIE, B.S.E.E, BS Physics/Mathematics
Texas A&I University, Kingsville, 1979
M.S.E.E., Louisiana State University, Baton Rouge, 1981
Doctor of Philosophy, Spring Commencement, 1987
Major: Electrical Engineering; Minor: Industrial Engineering

Fail-Safe Local Area Networking Using Channel Redundancy

Dissertation directed by Professor S.C. Kak
Pages in Dissertation, 140. Words in Abstract, 353.

ABSTRACT

Local area networks represent a major development in data communications. The ring topology is the most popular of the local area networks. Its inherent problem, however, is the breakdown of the entire network in the event of a station failure resulting in total shut-down until repairs can be made. This dissertation is a study of designs for fail-safe local area networks.

Two methods for maintaining the operation of a ring network are analyzed and compared. The first method uses a double-ring where the inner loop is optically coupled to the outer ring whenever a node failure occurs. The analysis performed here in terms of the scantime reveals that, for a breakdown in the network, the additional delay is at most equal to the single channel scantime delay. When "a" (defined as the propagation time divided by the transmission time) is less than 1, the throughput for the double-ring is comparable to that for the single-ring for all protocols studied. However, for "a" greater than 1, the throughput is lower. For the token-passing ring protocol, the throughput drops to a value of half the single-ring value.

The second method uses spokes in the ring that are activated to make a logical ring network connection when a node fails. This design's performance is superior to that of the double-ring in scantime, efficiency under heavy load, and throughput performance. The analysis is based on the assumption that the activation time for the spoked-ring and the double-ring reconfiguration time are negligible.

The redundancy in double-ring and spoked-ring networks that allows network operation during station breakdown is not put to any use during normal conditions. This dissertation presents a new design of ring operation where normal operation does use the 'redundant' hardware. A network called dual access bi-ring network (DABNET) is proposed where the stations are divided into two sets (rings) so that most of the communication is ordinarily between stations of each set. A protocol is designed so that communication across rings is allowed. This new concept of two rings that allows communication across them in the event of ring breakdown or when stations across the rings wish to transfer information could be generalized to more than two rings, though the protocols governing such rings would be quite complex.

CHAPTER 0

INTRODUCTION

0.1 MOTIVATION AND PERSPECTIVE

Recent advances in computer and communication technologies have made the interconnection of geographically isolated computing systems an attractive method to share computing resources and transmit data. Local area networks (LANs) are a class of computer networks that provide interconnection to a variety of devices over a small area. The transmission media in use range from twisted pair to optical fiber.

LANs are generally characterized by their topologies. There are three topologies in use.

1. The bus topology, a special form of the tree topology, uses a multipoint medium approach as in FASNET [28].
2. The star topology uses a central switching element to establish dedicated point to point connections between a pair of stations wishing to communicate.
3. The ring topology is made up of a closed loop with nodes attached to repeating elements. Data circulates the ring in a serial fashion.

The choice of topology depends on media access protocols and other user specifications [30].

Providing fail-safe operation under conditions of cable or node failure is a fundamental issue for the LAN designer. The reliability characteristics of the three topologies are very different.

For the star topology, the entire network is disabled whenever the central node fails. This is one of the reasons why this topology is not widely used.

The bus network, an example of which is the well-known Ethernet suffers from a variety of problems. One such problem is that of signal balancing: when two devices exchange data over a link, the signal strength of the transmitter must be adjusted such that it is strong enough to maintain adequate signal to noise ratio, but not too strong to overload the circuitry of the transmitter. This problem places physical limitations on the spacing between stations. Since the bus topology uses passive taps to attach stations on the bus, the length of the bus is limited by the signal attenuation for a given medium. This attenuation as well as the receiver threshold determine the physical distance over which the signal can be transmitted. It is often difficult to isolate faults on the cable. Furthermore a break in the cable can disable the network, in part or entirely. There exists also the problem of finding ways to determine if data is being received wrongly by an unauthorized station, due to its having been assigned a duplicate or incorrect address.

The ring network has been viewed as an alternative to the bus network. First proposed by Farmer and Newhall [63] in 1969, the token ring allows a station to transmit an arbitrary length message when it is in possession of the token. A variation of the token ring is the slotted ring where the ring is divided into fixed sizes. This was proposed by Pierce in 1972 [64]. The cambridge ring is a working example of the slotted ring. A second variation to the ring is the

register-insertion ring by Hafner, Nendal and Tschantz [65] proposed in 1974.

Since the ring network uses point to point communication, transmitted signals are regenerated at each node. This method has several advantages over the bus topology.

1. Transmission errors are reduced due to the regeneration of signals at each node.
2. Greater distances can be reached without loss of data integrity which is a limitation for the bus topology.
3. The ring can accommodate optical fiber links which provide very high data rates and excellent electromagnetic interference (EMI) characteristics.
4. Fault isolation and recovery are simpler.
5. Duplicate address problem is easily solved. The first station with an address match that is encountered by a packet can modify a bit in the packet to acknowledge reception [30]. Subsequent stations with the same address will recognize the problem.

These advantages suggest that the ring topology might, in future, supercede the bus network. The interest in the ring network is evidenced by the large number of research papers on its implementation that have been written in the recent years [53-57]. Network projects have been undertaken by several universities, IBM, AT&T, and the ring network has been the subject of research.

Reliability in the ring network has been provided by the introduction of some form of redundancy. Zafiropulo in 1974 [67] proposed the double ring and the recent advances in lightwave technology

have accelerated the implementation of the double-ring concept. The spoked-ring [26] is also a ring network which introduces redundancy in the form of spokes. This network has not been analyzed and, so far, the analysis done on the double-ring networks is limited to the question of availability of the channel under node or channel failures.

The motivation for the research in this dissertation is to fill the above gap in the literature. The performance of the double ring networks has been computed for different data rates and node failure combinations and spoked networks have been analyzed for the first time. Similar conditions have been considered for the spoked-ring and the double-ring so that their performance can be compared. The carrier sense multiple access with collision detection (CSMA/CD) and the token passing protocols [68] have been used to study this performance.

0.2 CONTRIBUTION OF THIS WORK

The double-ring and proposed spoked-ring networks have been analyzed and compared under single failure conditions. While the double-ring network sustains serial node failures, it is not clear how the network will perform in the event of arbitrary two node failures. The scantime and efficiency analysis for the double-ring are compared to the single-ring network performance.

Similar analysis is performed for the spoked-ring network. The length of the spokes for the spoked-ring, which is based on the number of stations on the ring and the number of consecutive failures the network is allowed to sustain to operate normally has been computed. The analysis shows that the spoked-ring network out performs the double-ring network.

The double-ring and the spoked-ring networks are inherently wasteful of bandwidth. The double-ring comes to use only when there is an expansion of the number of stations on the ring or when a station fails. Therefore the period of time when no such events occur, the inner-ring bandwidth is being wasted. For the low frequency of such occurrences, the added reliability may be too costly. Besides, the idea of a loop back to form a network does exclude every other transmission media except the fiber optical medium. This implies that every existing network modification cannot be implemented. Double-ring application in other media would be in the form of a redundant ring which is switched to in the event of a failure. This will also mean a wastage of bandwidth in the absence of a failure or expansion of the network.

The spoked-ring primarily has to be rearranged everytime an expansion is needed or expensive equipment would be placed on expected expansion locations in order to avoid the constant rebuilding of the network. These major problems have been addressed by the proposed dual access bi-ring network (DABNET). This network introduces two independent rings that are connected by gateways. These networks communicate with each other during failure or no failure situations. Thus, the inner-ring or dual access bi-ring does not suffer from the same bandwidth waste and reconfiguration problems associated with the double-ring and the spoked-ring networks respectively. A protocol for the operation of DABNET has been proposed.

0.3 ORGANIZATION OF THIS WORK

This dissertation is organized as follows: Chapter One provides an overview of computer networks and their general characteristics. The concept of fail-safe networking is also introduced. Chapter Two deals

with the idea of supervision as it relates to single and double channel supervision. In Chapter Three, the double-ring network analysis is performed and several results with respect to the network efficiency, scantime and protocol performance are derived. These results are compared to the single-ring network results under same conditions. Chapter Four introduces the spoked-ring and studies its performance. The spoke activation time and the network connectivity are also examined. These results are compared to those of the single-ring. Chapter Five introduces the proposed DABNET. The merits of the network over the double-ring and spoked ring networks are examined. A protocol for the DABNET is developed and examined using an example.

Chapter Six includes comparison of the performance of the double-ring, the spoked-ring and DABNET. Suggestions for future work are also made.

CHAPTER ONE

COMPUTER NETWORKS

1.1 AN OVERVIEW OF COMPUTER COMMUNICATION NETWORKS

Computer and communications technologies have advanced rapidly over the past twenty years. Advances in VLSI have led to the development of faster and smaller devices resulting in more powerful and compact computers. Satellite and optical fiber communications have opened up new possibilities in communications. These new advances, create new possibilities in the use of computers, especially in the sharing of resources.

When independent major computing systems ("hosts") communicate with one another to share resources, the resulting system is termed a computer network. Advances in communications technology have also led to a reduction of network costs. The availability of low priced computers suitable for carrying out the functions required to operate a network has made resource sharing very attractive. In a corporate structure, for example, departments which were previously isolated from the organization's main system may be connected to it. These islands (departmental systems) can eliminate job duplication and produce savings with little change in system configuration.

Computer networks are often robust. By interconnecting independent computing systems, job orders on any failed or crashed computer in the network can be redirected to other computers in the network where such task could be performed. It is also possible to

shorten delays in the system by routing jobs to less busy systems.

Communication networks are of two types:

1. Circuit switched, in which case a dedicated path between the source and destination has to be established prior to the beginning of communication.

2. Packet switched, where messages are broken into packets each with all the necessary information for proper delivery at the destination and then transmitted randomly without the existence of a dedicated path between source and destination. A packet is a collection of data and control bits usually of variable length.

The control bits are used to provide source, destination and error control information which is necessary for a proper placement of data in the packet. A packet switched network is usually made up of nodes that are geographically dispersed and connected by dedicated high speed data links as in the ARPANET and DECNET. The nodes are stored program computers whose internal data links are connected to other nodes and external data links connected to local computers and terminals. The basic network components as shown in Figures 1.1 and 1.2 are the following:

1. The USER TERMINAL, which could be either a teletype keyboard and printer, or a local network joining multiple user facilities in a building or campus area.

2. The HOST COMPUTER, which functions primarily as a medium providing services and translations necessary to get data out and back over the network to associated terminals.

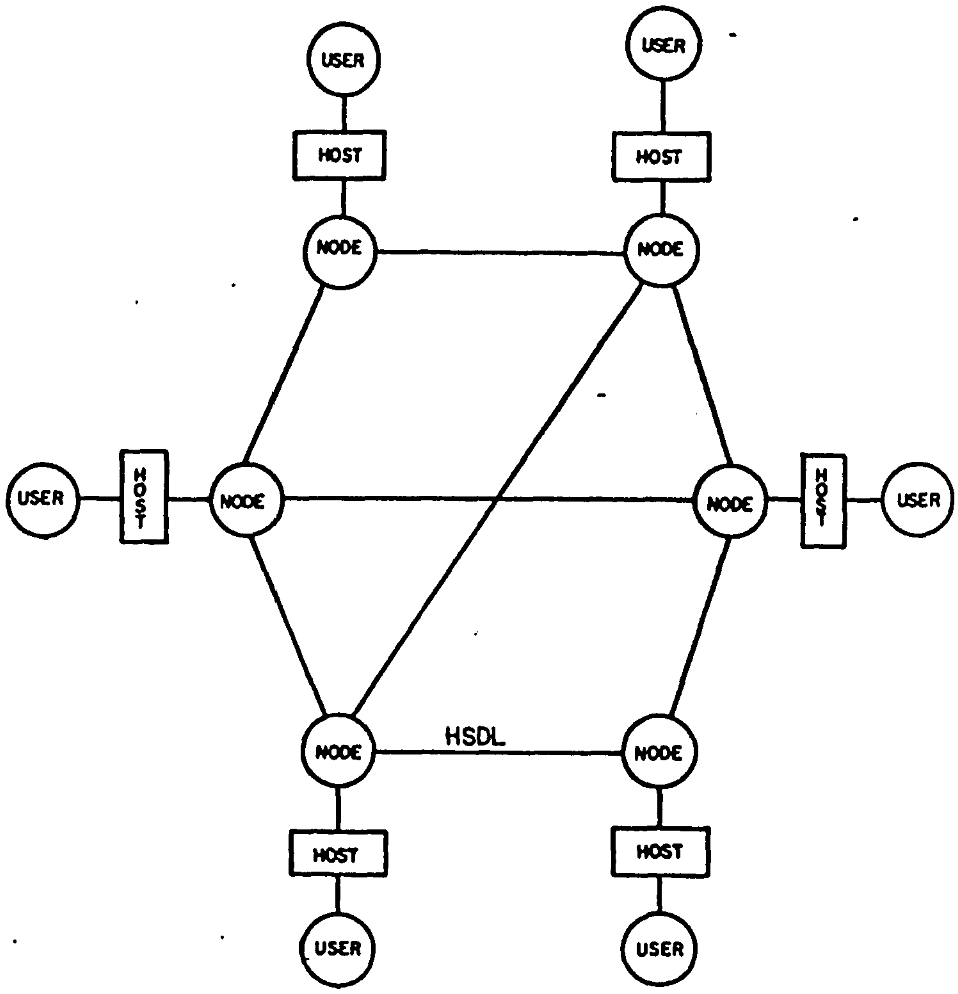

FIGURE 1.1 BASIC COMPUTER NETWORK STRUCTURE

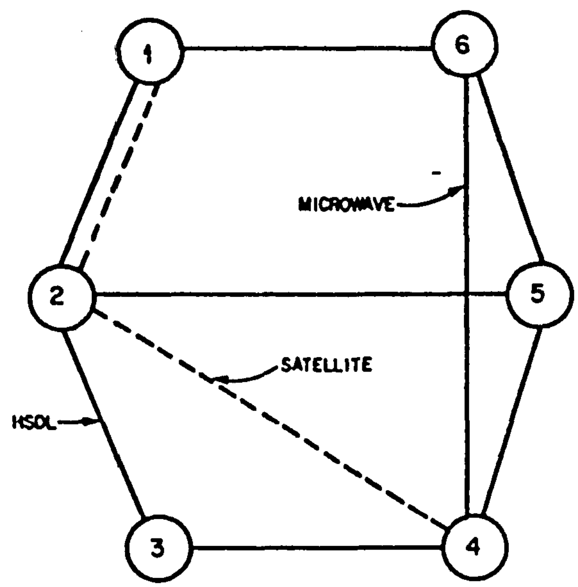

FIGURE 1.2 COMPUTER NETWORK WITH SOME REDUNDANCY

3. The NODES, whose primary function is the routing of the packets through the network. Each packet received at an intermediate node is typically stored in a buffer memory until processing capacity is available to decide which of the many queues it has to be placed in to continue its journey toward its destination. This is a store and forward process. To keep the queuing delays short to permit interactive transactions, the packet size is chosen to be small.

4. The HIGH-SPEED DATA LINKS, that have a bit rate capacity that is high compared to that of the user terminal.

Local area networks (LANs) are a class of the general purpose networks that interconnect devices within a small area. LANs are characterized as networks with high data rates transmitted over short distances. Typical data rates are between 0.1 Mbps to 100 Mbps. With the introduction of fiber optical links, it is conceivable to have data rates in the gigabit per sec range. The error rates associated with LANs are very low (10^{-8} - 10^{-11}).

1.2 COMPARISON BETWEEN CIRCUIT AND PACKET SWITCHING

Circuit switching (as in telephone or voice switching) is currently widely used. In circuit switching, a dedicated communication path or circuit is established at the beginning of the communication between the source and the destination. This communication circuit is broken at the end of the communication. The major problem associated with circuit switching is in the establishment of the communication path. Once this task is achieved, all messages transmitted through this established link are guaranteed arrival to the destination in the sequence in which they are

sent. No special headers are required for identification of messages. In packet switching no such dedicated paths exist. Messages are fragmented into packets which are then transmitted through the network using a dynamically allocated transmission capacity scheme. The message segmentation is done by the computer. Once a message is segmented, the resulting message capsules or packets are transmitted independently through the network. At the receiver, another computer reassembles the data to provide a meaningful message which is the actual replica of the transmitted message. To achieve this recovery, each packet carries enough information necessary to allow efficient reassembly. This additional information is the overhead associated with the message.

1.3 PACKET PROTOCOLS

Protocols are rules that govern the necessary packet switching transactions within the system. The use of protocols ensures that messages transmitted in the network will be received and interpreted correctly. Protocols establish:

1. Standard data elements, which may include characters, files, jobs and graphic displays.
2. Conventions, which may include those on packet size and format.
3. Standard communication paths, for addressing, priority sequence and error control.

Network efficiency is a result, in part, of the choice of the protocols. Several factors need to be taken into consideration in making this choice. For example, are the packet lengths variable or

fixed and whether acknowledgement is required for transmitted packets or not? The channel error characteristics and the nature of the channel (whether it is half-duplex or full-duplex) are also important factors. Should all packets not received after transmission "hold time" be retransmitted or can the message be recovered from partial retransmission are questions to which answers must be provided at the outset. How these issues are resolved determines the channel utilization. Several protocols have been researched [51,52]. The most commonly used protocols are the Carrier Sense Multiple Access with collision detection (CSMA/CD) and the token-passing protocol. For the CSMA/CD protocol, a station wishing to transmit first senses the medium and transmits only if it is idle. The station continues to listen and ceases transmission when a collision occurs. With the token-ring protocol, a token circulates around the ring. A station wishing to transmit may do so by seizing the token, inserting a packet onto the ring and then retransmitting the token. Another ring protocol is the slotted ring protocol where the ring is divided into slots which may be designated empty or full. A station wishing to transmit may fill an empty slot with a packet as the slot passes by. The register insertion protocol uses the existence of registers at the stations to temporarily hold a circulating packet when the station is transmitting its packets.

1.4 FAIL-SAFE NETWORKING

When a system works well everytime it is called upon to perform its function, it is said to be reliable. System reliability may be viewed from three standpoints: how failure will affect device safety, whether mission performance will be viable and what are the unscheduled

maintenance requirements. In a telephone system environment, customers expect and usually obtain a good quality service anytime under varying conditions. Here, an important consideration in system design is the level of reliability that should be designed into individual elements like the customer terminals or local switching systems. The network reliability depends primarily on performance under component failures. Component failure is generally specified by the meantime between failures (MTBF). It is assumed that the "up" time for a component is exponentially distributed:

$$P_r(T < t) = 1 - e^{-\lambda t}$$

where $e^{-\lambda t}$ is the probability that a component will function for at least a time t and $1/\lambda$ is the mean time between failures. These specifications are toughest for safety-related failures and least exacting for those that merely require maintenance.

Improvement in the system reliability usually comes from the introduction of redundancy. These component redundancies can take the form of a standby, parallel or series-parallel combination. In a series dependence, each component must function in order for the entire system to function. In parallel or redundant connection, only parts of the connections have to function in order for the entire system to function.

Standby redundancy requires that the external unit that makes the decision to switch to the standby be more reliable than the redundant circuit. However, the standby unit is subject to much less stress until it is put to use. Parallel systems work well when the fault to be guarded against is straight-forward, like an open circuit. For

logical failures, more complicated scheme may be required since a faulty circuit would still operate but give the wrong output. Bimodal parallel-series or series-parallel redundancy is often used when the commonly expected failures are shorts or opens in the network. For higher reliability, voting systems are employed as in space hardware. A simple majority voting is used to determine system performance. However this approach fails when more than half the components have failed. This problem is avoided by using adaptive majority voting where a comparator identifies an element when it first "goes out of step" and disables it by taking a majority vote of the remaining elements. The above approaches are concerned with hardware or fail-safe networking with component failures. They may result in more elements or components in the system leading to increased maintenance frequency. However, these redundancies coupled with repair implies that the system can operate continuously while being repaired, thus reducing overall failures significantly. For example, a power utility is expected to serve its customers continuously, even while maintaining its equipment and fixing or replacing failed or failing components. In this context, reliability takes a different meaning, where functional reliability rather than component reliability is being stressed. Such reliability can be provided by digital processors with large memories which perform diagnostic tests and provide network reconfiguration with software controls. This yields functional redundancy and/or a fault-tolerant capability. It is also clear that software reliability becomes an extremely important issue. Network control, management and fault isolation are usually software oriented. This approach provides for the tailoring of the network to specific applications while using a

general purpose hardware.

In local area networks both software and hardware redundancy is employed. While component duplication increases the probability of fail-safe operation, software redundancy provides for the network reconfiguration after a given node/nodes failure. One may also talk of a fail-safe node that allows the network to reconfigure in the event of a network break. For an optical fiber ring, such a node will consist of a light guide receiver and a light guide transmitter electrically connected by a regenerator and optically connected by a directional coupler. The optical coupler usually provides continuity when a power failure occurs at a node [17].

With channel redundancy, the network will be considered fail-safe if in the event of a break on the main channel or some other failure, the network still operates. A double-ring fiber optical network is an example of a channel redundancy network. When a break on the outer ring occurs, the inner ring becomes optically coupled to the outer ring resulting in an operational ring network (See Chapter three). Depending on the number of failures one may end up with a number of fragmented ring networks. Such network bypasses may result in a partial operation of the original network. The network performance can be measured using the network's responsiveness and throughput criteria.

In Chapter Four, another example of a channel redundancy is shown where a number of spokes in the ring network define the introduced redundancy. Chapter Five introduces the dual access bi-ring where redundancy is obtained through the establishment of gateways.

While it is true that the introduction of component redundancy

enhances the reliability of a network, the proper operation of the network depends also on the reliability of the software, which controls hardware operation. As shown earlier, hardware reliability is associated with component failure due to the physical deterioration and probability of failure associated with each component. Software reliability, on the other hand, is not so simple to determine. Software reliability is formally defined as the probability that a software fault that causes deviation from required output by more than specified tolerances in a specified environment does not occur during a specified exposure period. In most applications, the exposure period is chosen as the run corresponding to the selection point from the input domain of a program. The exposure period is expected to be independent of factors like machine execution time, programming environment and so on. Software failure could result from faulty software design such as errors in compiler, operating system and microcode. Sometimes faulty hardware design can result in software failure. So far, there are no generally accepted methods of determining how reliable a software is [27]. A variety of software reliability models have been developed and characterized under the following:

1. Estimation models, which are used in an environment in which, once an error is found, it is corrected before testing is continued. These models require data such as the number of errors found in a fixed period, the interval between two detected errors and the time to correct errors in addition to failure rate.

2. Measurement models, which measure software failure in an operational environment while assuming that the software is not modified during the measurement period even if an error is found.

3. Prediction models, which unlike the estimation and measurement models, use the internal structure (like the number of statements of each type or the number of variables) of the software to establish reliability.

Common to these models are measures such as mean time to failures (MTTF) and the assumption that the compiler is correct when a program is written in a higher-level language. For a large and complex system design, it becomes almost impossible to deal with software reliability. Fail-safe networks use software that is compatible with hardware reliability requirements.

This dissertation does not deal with software reliability issues. Hardware redundancy as provided for by an additional ring and through spokes will be used to increase the reliability of the ring network. The additional ring will be used in the traditional single access or the proposed dual access modes.

CHAPTER TWO

NETWORKS WITH SUPERVISION

2.1 THE IDEA OF SUPERVISION

Supervision to ensure integrity and proper use is a major concern of the network designer. Efficient management of the network depends on how well the network control structure is maintained. Adequate allocation of available resources requires scheduling and automatic control of peripherals. With the growth of the network, it becomes increasingly important to protect and maintain network reliability by monitoring resources for proper use. Some nodes or devices, depending on their use, may require prioritization. Some form of supervision has been introduced in networks relative to the nature of data the network carries. This data could be fairly steady as in voice, telemetry and bulk file transfer or bursty as in the interaction between terminal and host traffic. In both cases, reservation strategies such as TDMA, S-ALOHA, priority demand assignment (PODA) and random access technique like CSMA/CD and contention are employed. The success of the approach employed usually lies on whether the control is centralized or distributed. Complete supervision takes into account flow and error control if the proper operation of the network is to be maintained.

As user demands change, the network may expand with the change and become more complex or adapt in a manner that accommodate the change. Most often, previously effective methods become inadequate and new measures have to be introduced. This basic problem is examplified by the different types of local area networks existing today. With the

emergence of new technologies, a fundamental concern is how to upgrade or include the new technology in the existing system. Equipment compatibility and the question of flexibility so that one is not locked into the offerings of a specific vendor become real. In Britain, the British Telecommunication's recent Integrated Digital Access (IDA) service is an example where communication bandwidth is being increased from the normal 3 kHz used for standard voice signals and 9.6 kbits/sec used for high quality data transmission to an expected 144 kbits/sec service. Value added networks (VANS) have added more demand for greater bandwidth. In Japan and the U.S., much work has been done on the proposed Integrated Services Digital Network (ISDN) that, when implemented, will perform all voice, data and video transmission services. How such a network might interact with the existing networks is being dealt with by regulatory organizations like the CCITT, and ISO. Supervision may thus be viewed at the individual network level as well as in the regulation of data flow across network and national boundaries.

2.2 SINGLE CHANNEL SUPERVISION

In packet communication systems that go to define computer networks, data transmission and supervisory functions are designed to share the same channel. Usually, the supervisory functions are considered as overhead costs for such transmissions. This approach works well with low data rate. However, at heavy traffic, network efficiency drops considerably due to backlog and the necessary control measures used to decongest traffic. This compares to a traffic network where lights and ramps (for highways) are used as control measures to

reduce traffic intensity. At low traffic periods (usually off rush hours), these measures are adequate to control queue length at stop lights. During peak hours these measures do not work well introducing long delays and alternate routes have to be found to avoid traffic jams. Many measures such as buffer reservation, token-passing and choke packets to name a few have been introduced via centralized or distributed routing algorithms in packet communications networks with various degrees of success.

One of the major problems with single channel supervision is the potential loss of data when a link is abruptly broken. Assuming the existence of a minimum connectivity between nodes for a single branch failure, the network will eventually establish another path to complete the ongoing transmission. Its supervisory controls will have to take care of the associated problems like packet duplication and so on. A new path established under these conditions may be sub-optimal relative to the original message path. It is assumed here that the network through its algorithm establishes an optimum path when messages are sent. This path could be associated to cost or delay depending on the network type (private or public). Reliability of such networks with respect to prioritized messages will then depend on the message hold time versus time taken by network to establish a new path.

2.3 DOUBLE CHANNEL SUPERVISION

One way to avoid the problems faced by single communication channel systems is to introduce a second channel. With such a system it is possible to provide supervisory services with one channel while

transmitting data with the other channel. The supervisory channel can carry control packets such as call request, call accepted, clear request, interrupt, and acknowledgement if there is no reverse traffic. The question arises: Could it be a potential data carrier also? As a potential data carrier, the supervisory channel capacity relative to the other channel becomes the key parameter when being compared to a single channel system. At low data rate or low system failure rate, it is possible that the introduction of a second channel might be wasteful. If a channel in a two-channel computer network is used solely for supervision, then its channel capacity must be a small fraction of the data transmission channel capacity. In such a case, the cost analysis may prove that the introduced efficiency does not justify cost. However, if it does have a data transmission channel capacity, not only has network reliability been improved since both channels can transmit data at peak periods while maintaining the same network optimal path, network efficiency is automatically greater as congestion is minimized. This is exemplified in the traffic network by the average traffic congestion profiles of single lane roads versus double or triple lane roads. If a one lane road is bad, traffic has to be diverted through an alternate route. However, for roads with more than one lane, it is easy to block the bad lane while allowing traffic through the good lanes. With two channel computer networks the worst case analysis for a single channel breakdown will be as good as a single channel throughput analysis. The second channel may be defined by means of additional bandwidth allocation in the same physical medium, or it may be a new physical link. When considered as a new

link for the ring network one would thus have a double-ring. In such a case, both the rings carry data and control packets.

The situation where the additional channel carries special supervisory information can lead to interesting possibilities. These possibilities are best considered in relation to the design of specific networks.

CHAPTER THREE

DOUBLE-RING NETWORKS

3.1 INTRODUCTION

While the ring network is the most popular local area network, its inherent problem of total breakdown when a node or station fails has resulted in various approaches being developed to enhance its efficiency. The FASNET [28] which is basically two undirectional transmission lines with stations making attachments to both the upper and lower lines uses passive coupling to solve a station breakdown problem as shown in Figure 3.1.

While FASNET is not a ring network, it uses the double link approach. FIPNET [22], however, is a ring network that exhibits the problem that affects all single-ring networks. A breakdown simply stops the ring from operating with its repair time as the minimum delay required to get the ring operational again. The double-ring network which introduces a redundancy by optically coupling the inner ring when there exists a failure is shown in Figures 3.2 and 3.3. The optical coupling (fiber optics application) is done at the node interface and the structure can tolerate as many failures or expansions the ring is designed for. The obvious advantage of this network structure lies in its ability to minimize the network breakdown service time.

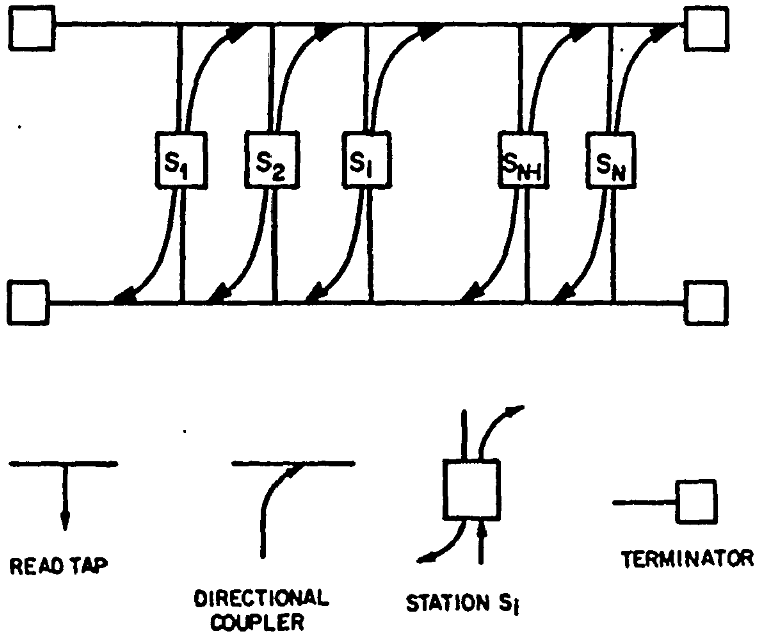

FIGURE 3.1 PHYSICAL CONFIGURATION OF FASNET

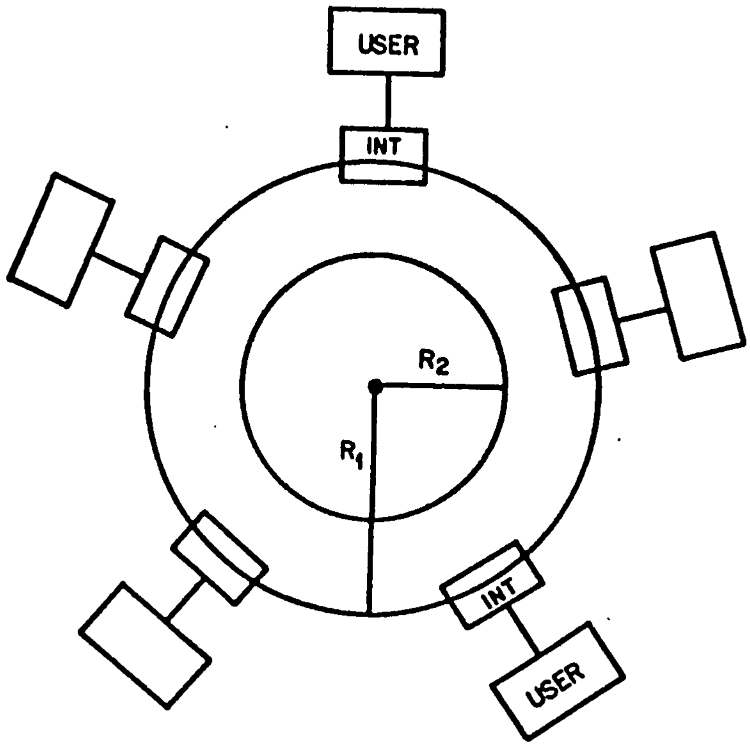

FIGURE 3.2 DOUBLE RING STRUCTURE

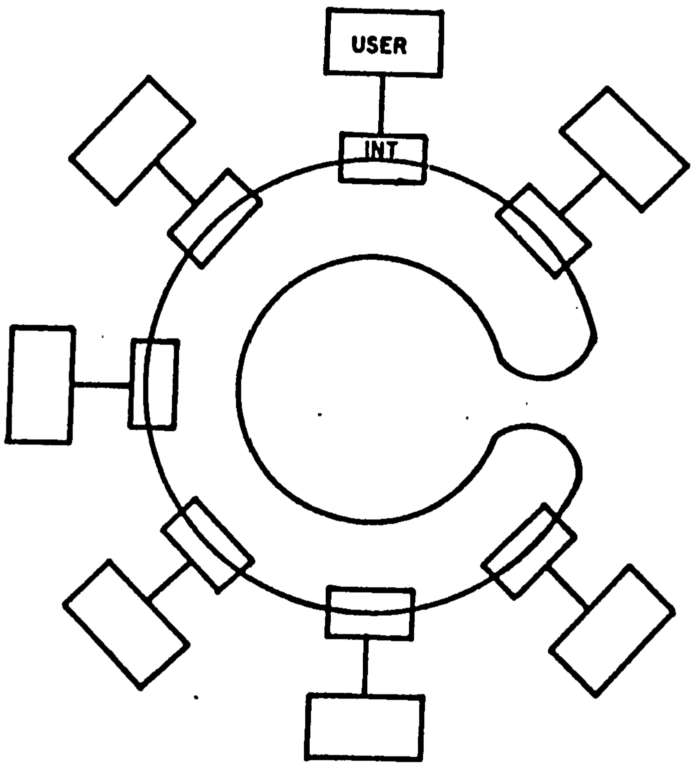

FIGURE 3.3 DOUBLE RING STRUCTURE AFTER A STATION BREAKDOWN

3.2 SCANTIME ANALYSIS

Let the number of stations on the ring be N. Let us assume that packets are generated according to a Poisson process with each station allowed to empty all queued packets. The queue length is q and the arrival rate of all the N stations is λ, each station contributing λ/N. Assume also, a service rate μ (number of packets a station can transmit). If the walk time is w (time it takes a packet to go around the ring when the ring is in an idle state), the scantime for a single-ring is given by [12]

$$S = \frac{w}{(1-\rho)} \qquad 3.1$$

where $$\rho = \frac{\lambda}{\mu}\ ,\ \rho<1.$$

If the length of the ring is L and the propagation rate is B m/μsec then

$$w = \frac{L}{B}$$

$$= \frac{2\pi R}{B} \qquad 3.2$$

For a double-ring network, the waiting time when a station breaks down assuming equal spacing between stations on the main ring is given by

$$w = \frac{(L_1 - L_1/N) + L_2}{B} \qquad 3.3$$

where $L_1 = 2\pi R_1$ and $L_2 = 2\pi R_2$.

Substituting for L_1 and L_2 yields

$$w = \frac{2\pi[(1 - 1/N)R_1 + R_2]}{B}$$

$$= \frac{2\pi(R_1 + R_2)}{B} \qquad 3.4$$

for large N.

It is assumed here that $R_1=R_2$ such that the coupling length is negligible. If R_1 is not equal to R_2, there will be spokes linking both rings which will introduce a new component to equation 3.3.

$$W = \frac{(L_1 - L_{1/N}) + L_2 + (R_1 - R_2)}{B}$$

where (R_1-R_2) is the additional length introduced due to the change in the radius of the inner ring.

$$W = \frac{2\pi[(1 - 1/N + 1/2\pi)R_1 + R_2(1 - 1/2\pi)]}{B}$$

For $R_1=R_2$, this reduces to equation 3.4

$$W = \frac{2\pi(R_1 + R_2)}{B} = \frac{4\pi R_1}{B}$$

We shall consider the case where $R_1=R_2$ in all discussions from now on and R_1 and R_2 can be used interchangeably.

For a single-ring, the scantime is given by

$$S = \frac{W}{(1-\rho)}$$

$$= \frac{2\pi R}{B(1-\rho)} \qquad 3.5$$

For the double-ring, the scantime becomes

$$S_D = \frac{W}{(1-\rho)}$$

$$= \frac{2\pi\ [(1 - 1/N)^{R_1} + {}^{R_2}]}{B(1-\rho)}$$

$$= \frac{2\pi({}^{R_1} + {}^{R_2})}{B(1-\rho)} \qquad 3.6$$

The amount of additional delay introduced due to the new ring length is given by

$$\Delta s = \frac{2\pi R_2}{B(1-\rho)}$$

This is the penalty suffered for the redundancy introduced which in the worst case is equal to

$$\Delta s = \frac{2\pi R_1}{B(1-\rho)}$$

This in most cases will be less than the service time required to make the single-ring operational again when a failure occurs.

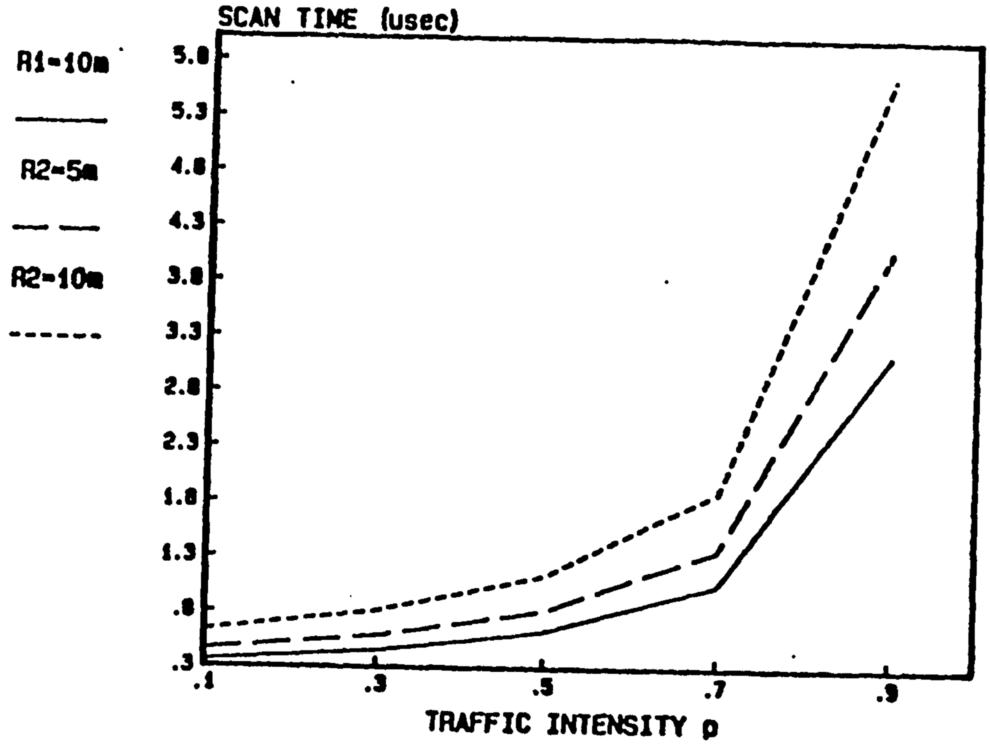

FIGURE 3.4 SCAN TIME AS A FUNCTION OF TRAFFIC INTENSITY FOR THE DOUBLE-RING.

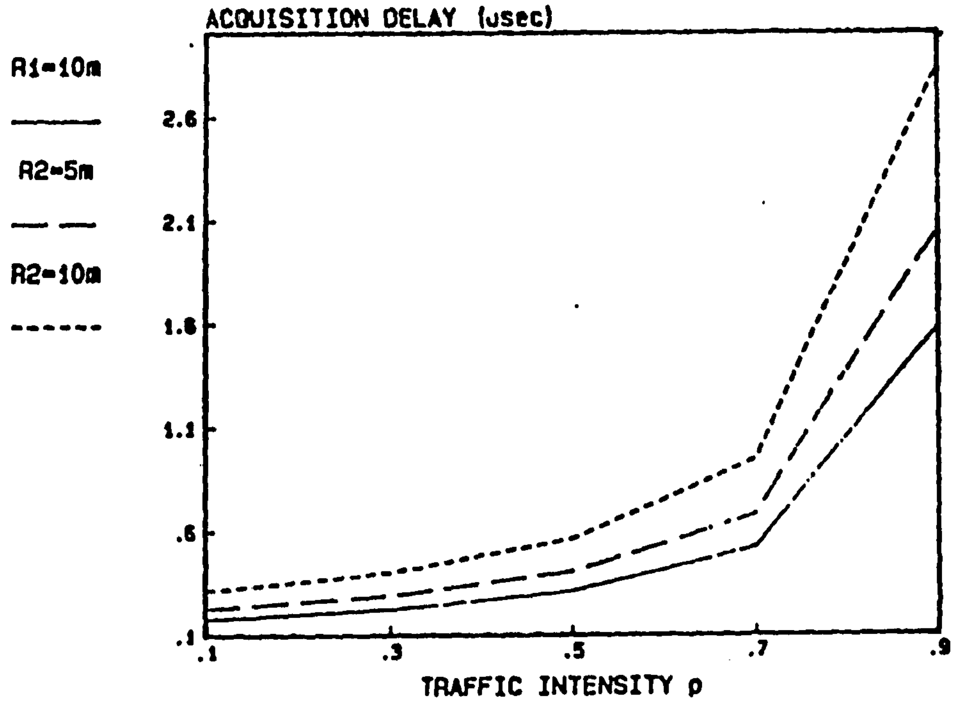

FIGURE 3.5 CHANNEL ACQUISITION DELAY AS A FUNCTION OF TRAFFIC INTENSITY FOR THE DOUBLE-RING

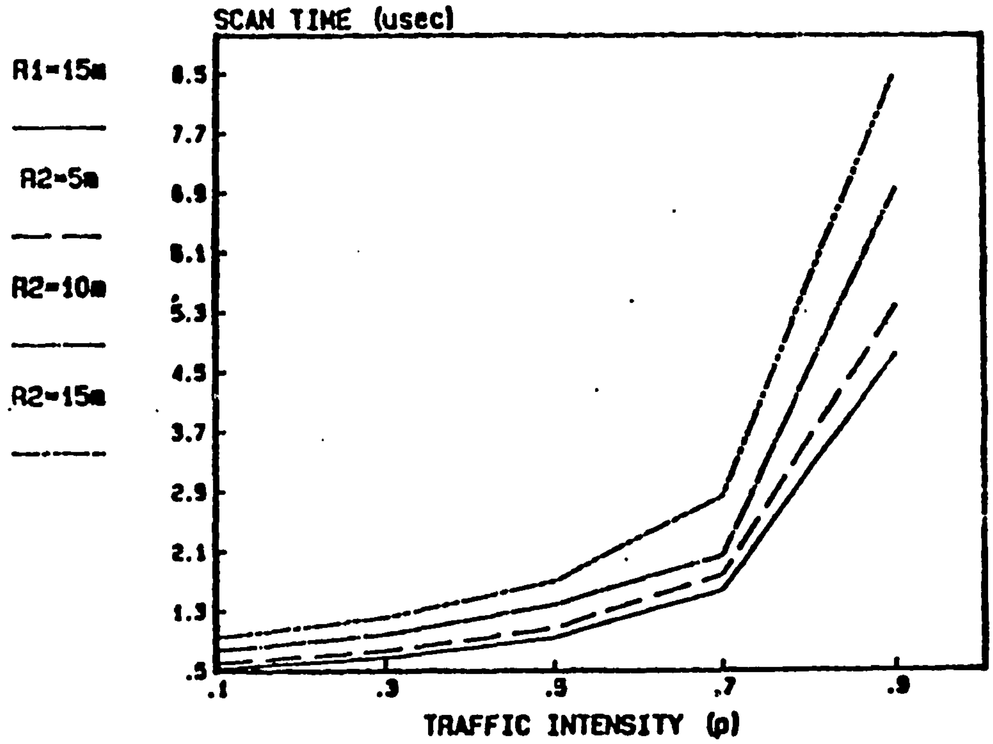

FIGURE 3.6 SCANTIME AS A FUNCTION OF TRAFFIC INTENSITY FOR THE DOUBLE-RING

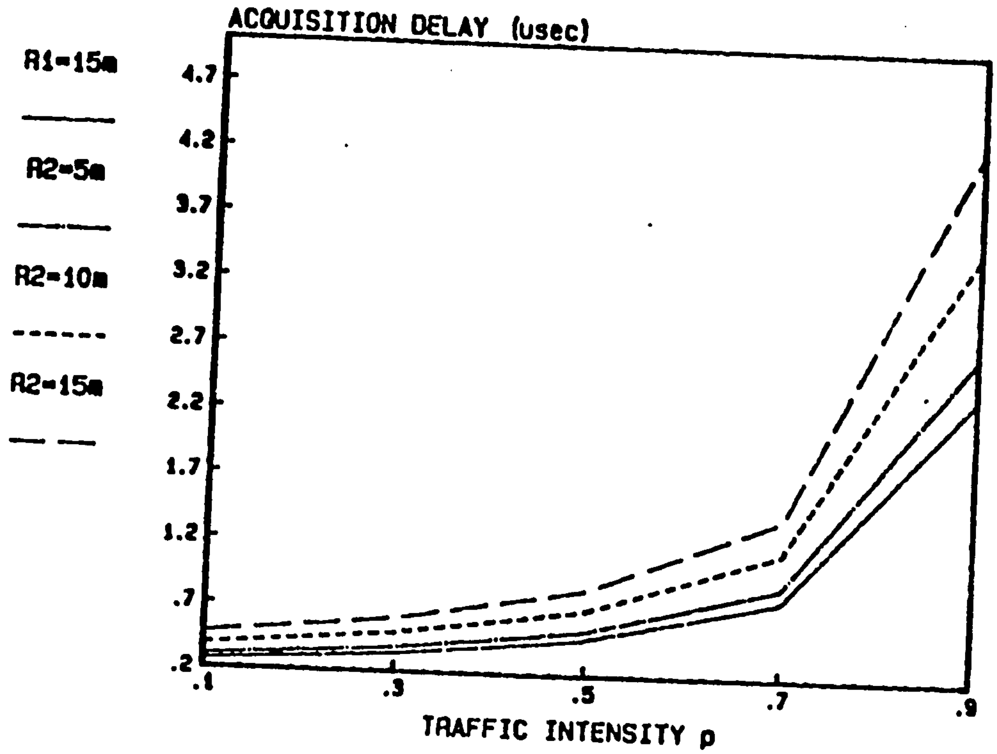

FIGURE 3.7 CHANNEL ACQUISITION DELAY AS A FUNCTION OF TRAFFIC INTENSITY FOR THE DOUBLE-RING.

3.3 NETWORK EFFICIENCY ANALYSIS

The network efficiency analysis will be governed by two factors: the channel-acquisition delay and channel efficiency under heavy load.

3.3.1 CHANNEL-ACQUISITION DELAY

The channel-acquisition delay is usually about half the scantime [12].

Thus,
$$\frac{1}{2}S = CD = \frac{L}{2B(1-\rho)} \tag{3.7}$$

where CD is the acquisition delay.

Thus, for a regular ring network this value becomes

$$CD = \frac{\pi R}{B(1-\rho)} \tag{3.8}$$

For the double-ring network, we have the channel-acquisition delay as

$$CD = \frac{\pi\,[(1 - 1/N)R_1 + R_2]}{B(1-\rho)}$$

$$\cong \frac{\pi[R_1 + R_2]}{B(1-\rho)}$$

for large N.

3.3.2 CHANNEL EFFICIENCY UNDER HEAVY LOAD

A heavy load situation occurs when all stations have packets queued for transmission such that a round robin protocol is followed. This means that for a token ring, the token is passed right to left resulting in equal access to the ring after a full rotation of token. Thus, the only overhead associated with the network is w/N; (the walk

time between stations). With an average station transmission time of q/μ and defining the channel efficiency as the ratio of the transmission time to the overhead (walk time between stations), we have the channel efficiency E as

$$E = \frac{q/\mu}{W/N}$$

$$= \frac{qN}{\mu W} \qquad 3.10$$

For the single-ring network

$$E = \frac{qNB}{L\mu}$$

$$= \frac{qNB}{2\pi R\mu} \qquad 3.11$$

For the double-ring, the channel efficiency is given by

$$E_D = \frac{qNB}{2\mu\pi[(1 - 1/N)R_1 + R_2]}$$

$$\approx \frac{qNB}{2\mu\pi(R_1 + R_2)} \qquad 3.12$$

3.4 MEDIUM ACCESS PROTOCOLS EVALUATION

In evaluating the network performance, we have considered it proper to compare the double-ring network to the single-ring using the results of some of the protocols which are currently being used in LANs. The measures of performance that are commonly used for LANs are

1) D: the delay that occurs between the time a packet is ready for transmission from a node and the completion of

successful transmission.

2). ST: the throughput of the local network.

3). U: the utilization of the local network medium.

We shall also define a parameter "a" which has been determined in [30], as

$$a = \frac{\text{propagation time}}{\text{transmission time}}$$

$$= \frac{\text{Data Rate x distance}}{\text{propagation speed x packet length}}$$

Using this parameter "a", the network throughput is defined as

$$\text{Throughput} = \frac{\text{packet length/(propagation + transmission time)}}{\text{Data rate}}$$

The network utilization then is given as Throughput/Data rate

$$U = \frac{1}{1 + a}$$

3.4.1 TOKEN-PASSING PROTOCOL

a. THROUGHPUT

The throughput of the token-passing protocol is given in [30] as

$$ST = \begin{cases} \dfrac{1}{1 + a/N} & a < 1 \\ \dfrac{1}{a(1 + 1/N)} & a > 1 \end{cases} \qquad 3.13$$

Where N is the number of stations on the ring. To compare the performance of the rings using this expression would require that the value of "a" be known. We can easily see that when a ring breaks and reconfigures to the form shown in Figure 3.3, the number of stations on the ring does not change. However, an additional length has been added

to the original ring length. The value of "a" is determined from the expression

$$a = \frac{\text{propagation time}}{\text{transmission time}}$$

let us assume a data rate of D_r Mbps and a packet length of L_p bits. With a propagation speed of B m/μsec; "a" for the single-ring will be given by

$$a_1 = \frac{D_r \times L}{B \times L_p}$$

$$a_1 = \frac{D_r \times 2\pi R_1}{B \times L_p} \qquad 3.14$$

For the double-ring, we have

$$a_D = \frac{D_r \times 2\pi[R_1 + R_2]}{B \times L_p} \qquad 3.15$$

The ratio

$$\frac{a_D}{a_1} = \frac{D_r \times 2\pi[R_1 + R_2](B \times L_p)}{D_r \times 2\pi R_1(B \times Lp)}$$

which reduces to

$$\frac{a_D}{a_1} = [R_2/R_1 + 1]$$

or

$$a_D = a_1 [1 + R_2/R_1] \qquad 3.16$$

Thus the throughput for the single-ring will be

$$ST_1 = \begin{cases} \dfrac{1}{(1 + a_1/N)} & a_1 < 1 \\[2ex] \dfrac{1}{a_1(1 + 1/N)} & a_1 > 1 \end{cases} \qquad 3.17$$

and for the double-ring

$$ST_D = \begin{cases} \dfrac{1}{(1 + a_D/N)} & a_D < 1 \\[2ex] \dfrac{1}{a_D(1 + 1/N)} & a_D > 1 \end{cases} \qquad 3.18$$

b. CHANNEL UTILIZATION

The channel utilization is defined as

$$U = S/\text{Data Rate}$$

For the single-ring, we have

$$U_1 = \begin{cases} \dfrac{1}{D_r[1 + a_1/N]} & a_1 < 1 \\[2ex] \dfrac{1}{D_r a_1[1 + a_1/N]} & a_1 > 1 \end{cases} \qquad 3.19$$

The double-ring has

$$U_D = \begin{cases} \dfrac{1}{D_r[1 + a_D/N]} & a_D < 1 \\[2ex] \dfrac{1}{D_r\ a_D[1 + 1/N]} & a_D > 1 \end{cases} \qquad 3.20$$

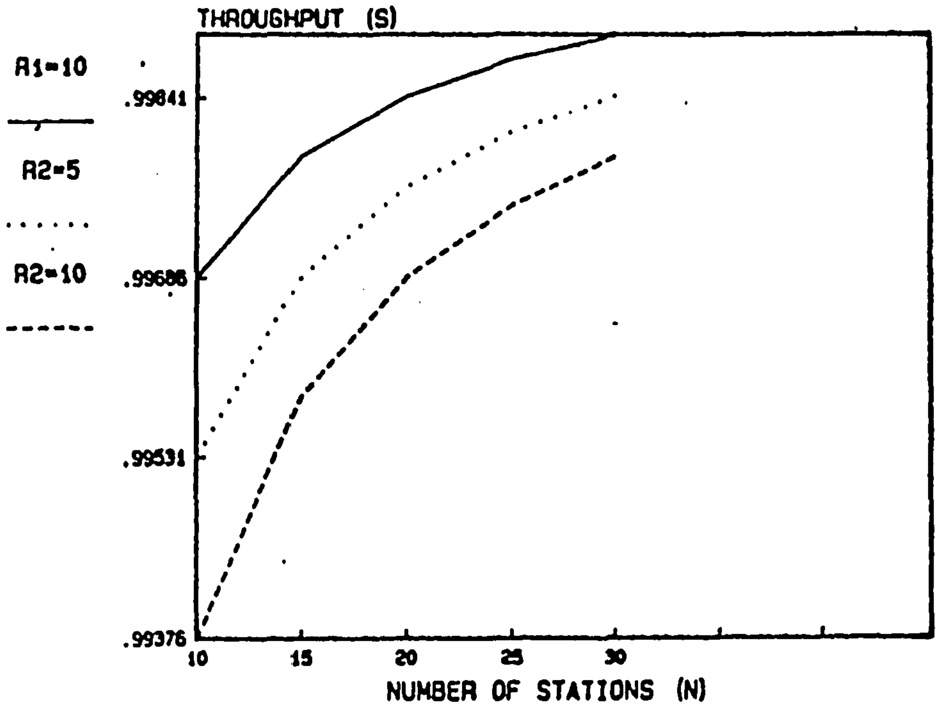

FIGURE 3.8 THROUGHPUT AS A FUNCTION OF N FOR THE TOKEN-PASSING PROTOCOL DOUBLE-RING
Lp = 500 bits DR = 50 Mbps

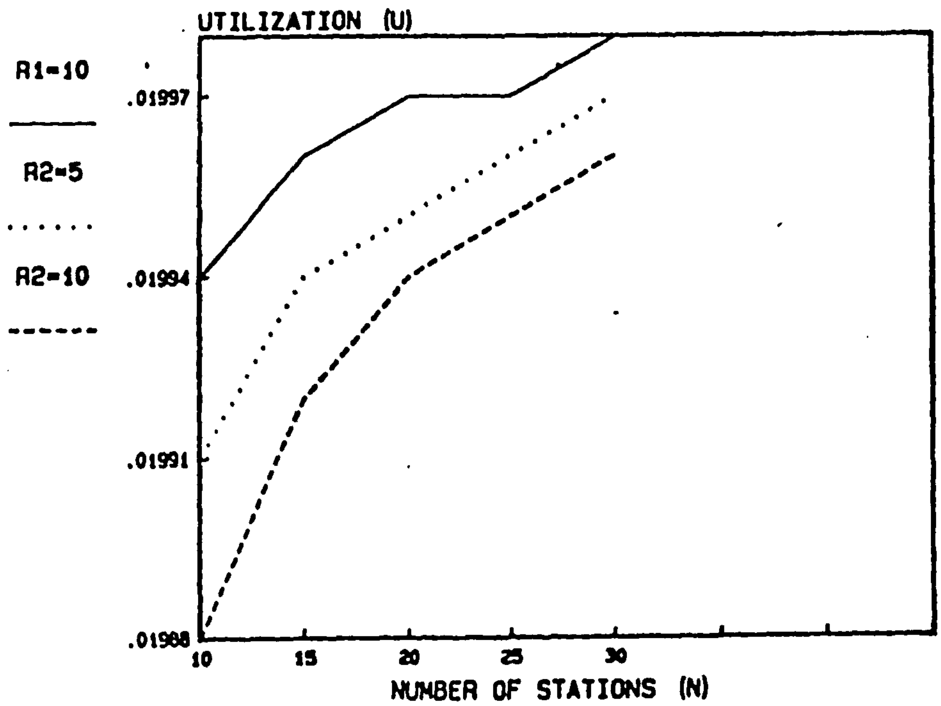

FIGURE 3.9 UTILIZATION AS A FUNCTION OF N
FOR THE TOKEN-PASSING DOUBLE RING
Lp = 500 bits DR = 50 Mbps

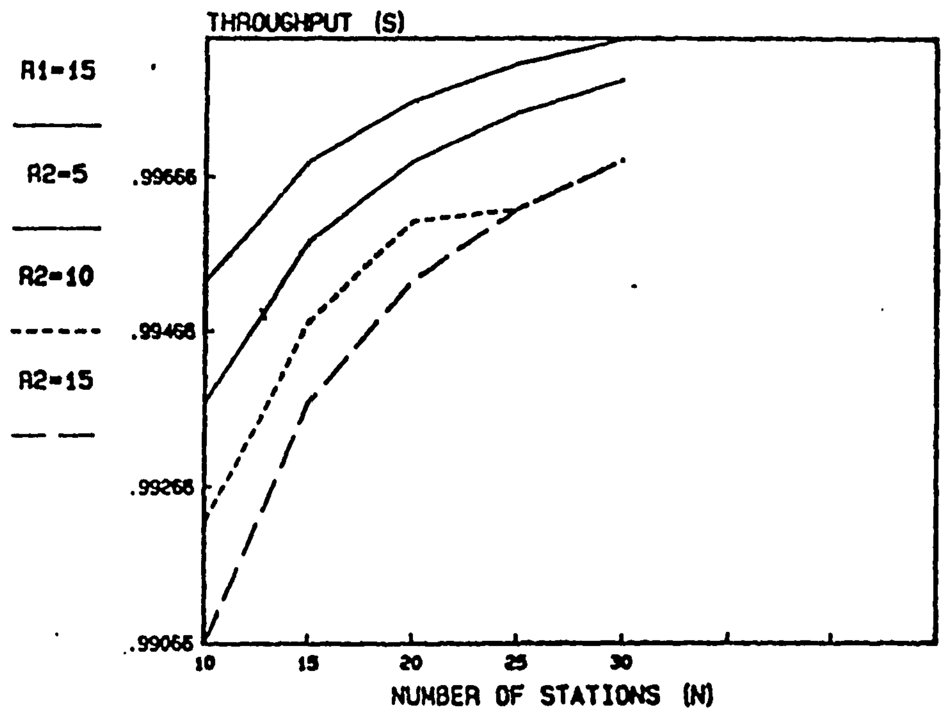

FIGURE 3.10 THROUGHPUT AS A FUNCTION OF N FOR THE TOKEN-PASSING PROTOCOL DOUBLE-RING Lp = 500 bits DR = 50 Mbps

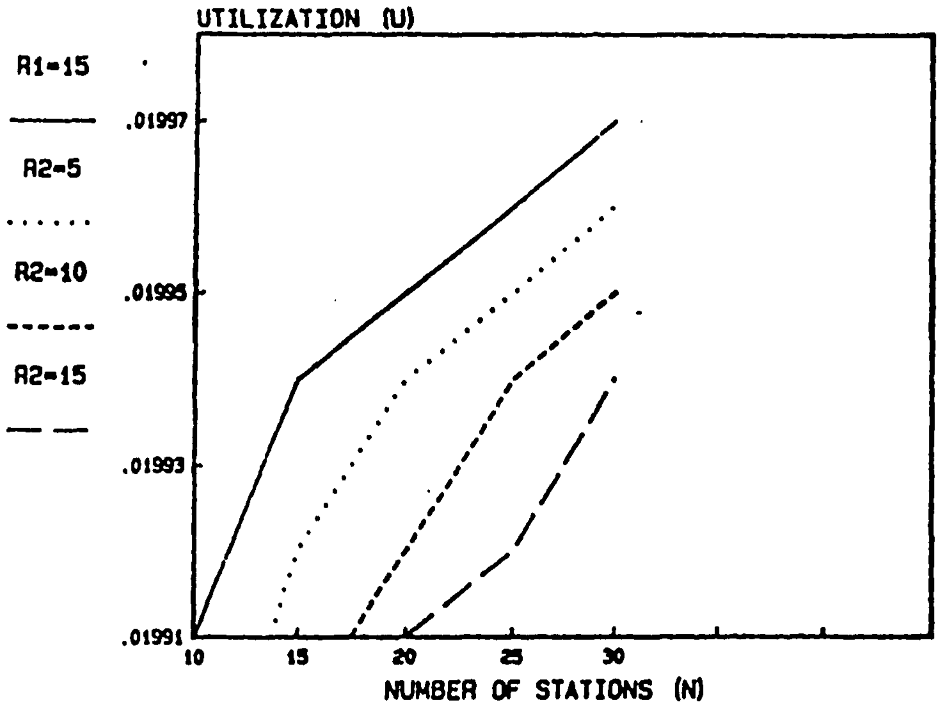

FIGURE 3.11 UTILIZATION AS A FUNCTION OF N FOR THE TOKEN-PASSING DOUBLE-RING
Lp = 500 bits DR = 50 Mbps

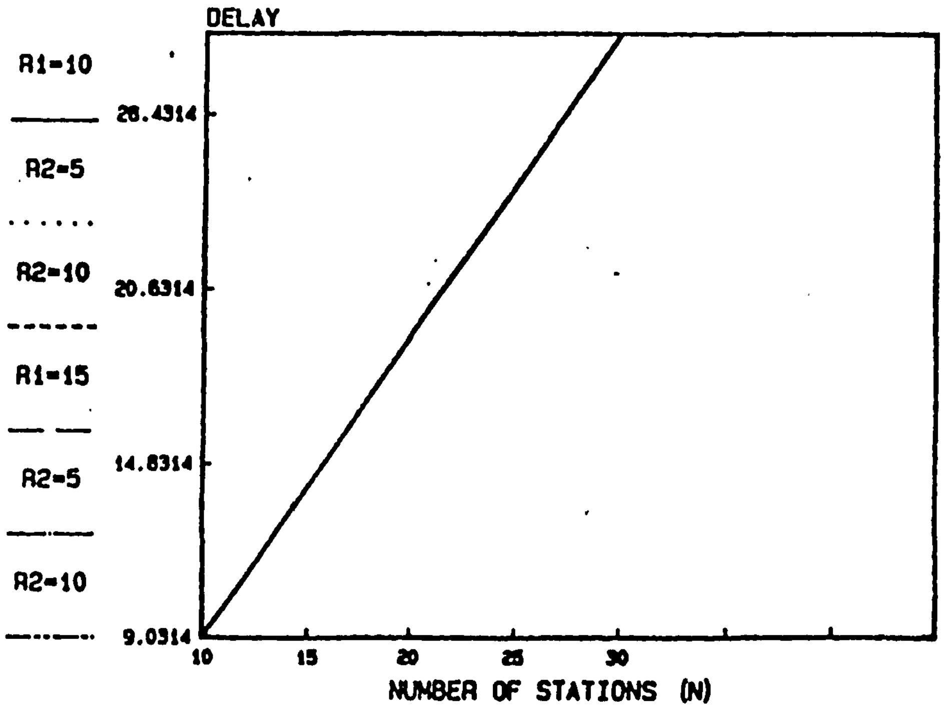

FIGURE 3.12 DELAY AS A FUNCTION OF N FOR THE TOKEN-PASSING DOUBLE-RING
Lp = 500 bits DR = 50 Mbps

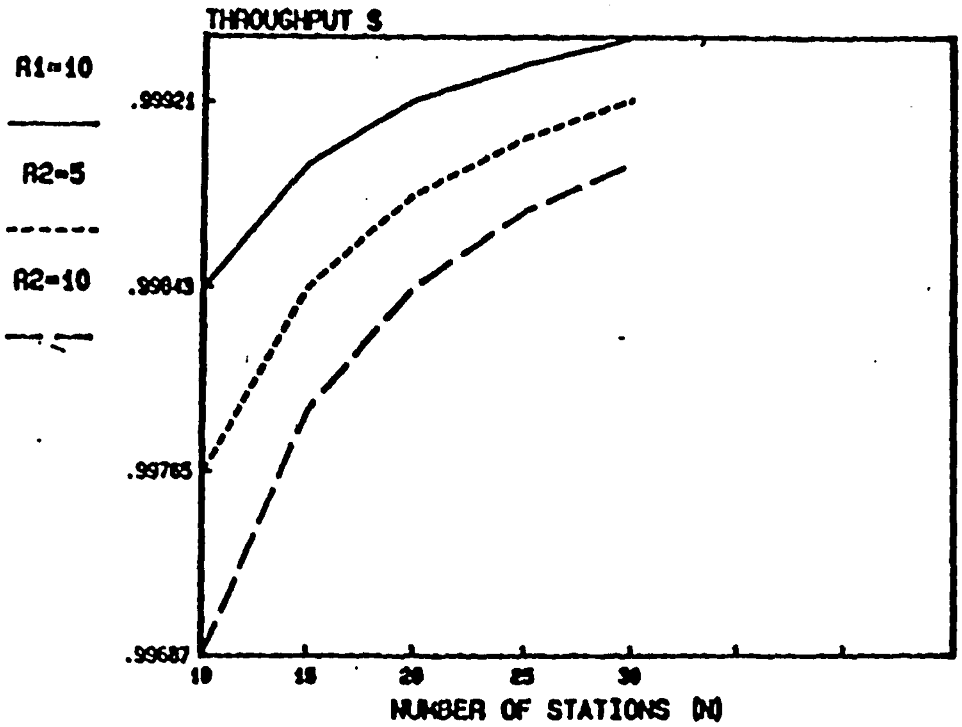

FIGURE 3.13 THROUGHPUT AS A FUNCTION OF N FOR THE TOKEN-PASSING PROTOCOL DOUBLE-RING
Lp = 1000 bits DR = 50 Mbps

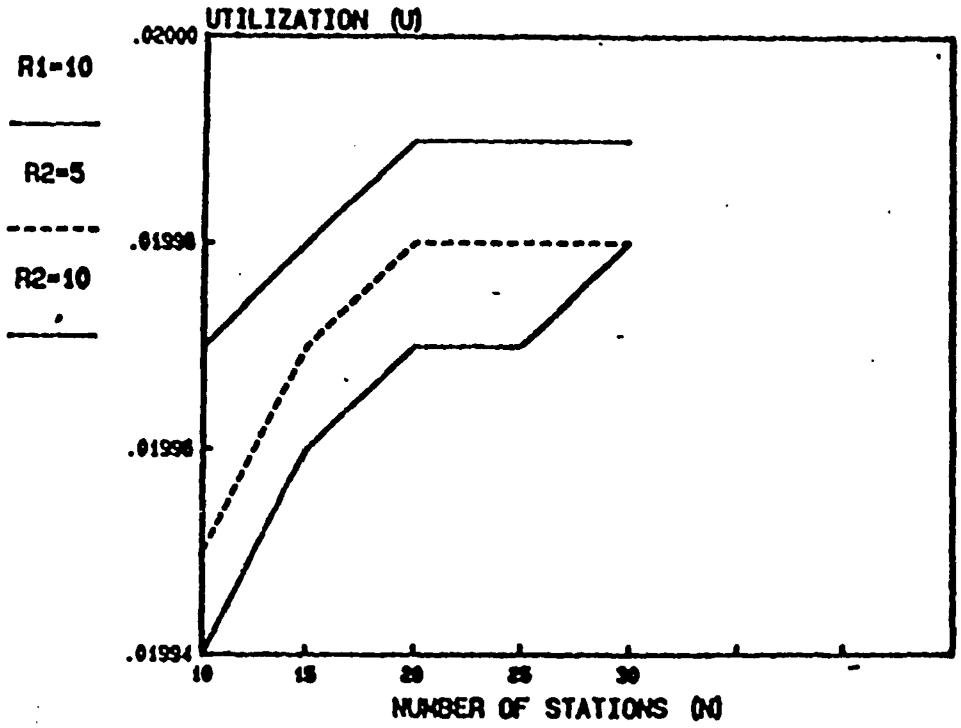

FIGURE 3.14 UTILIZATION AS A FUNCTION OF N FOR THE TOKEN-PASSING PROTOCOL DOUBLE-RING
Lp = 1000 bits DR = 50 Mbps

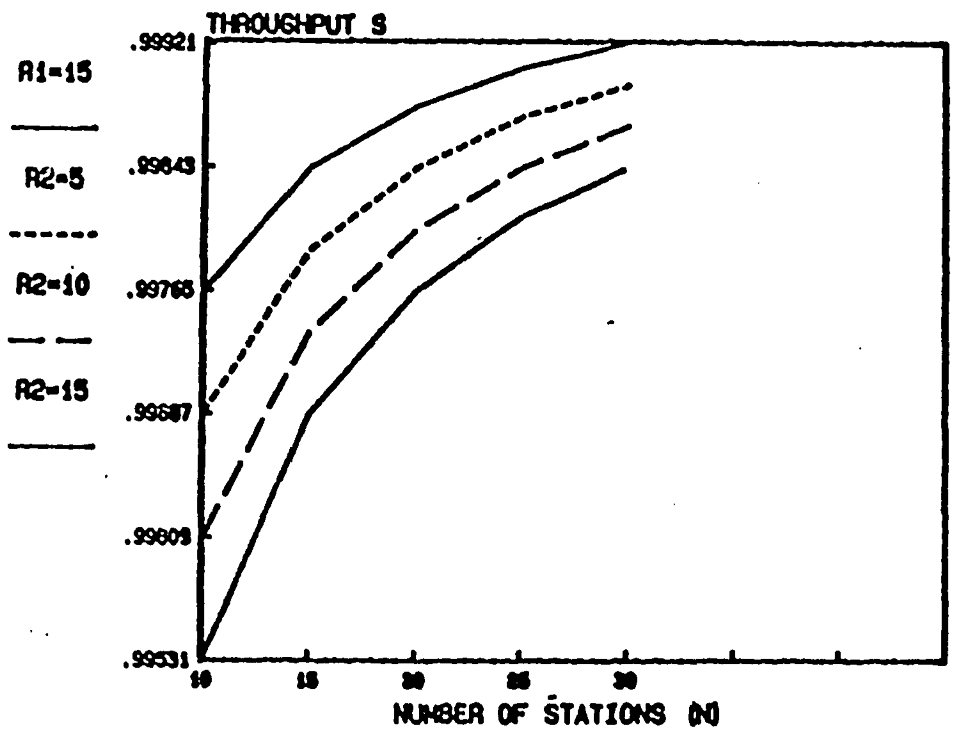

FIGURE 3.15 THROUGHPUT AS A FUNCTION OF N FOR THE TOKEN-PASSING PROTOCOL DOUBLE-RING
L_p = 1000 bits DR = 50 Mbps

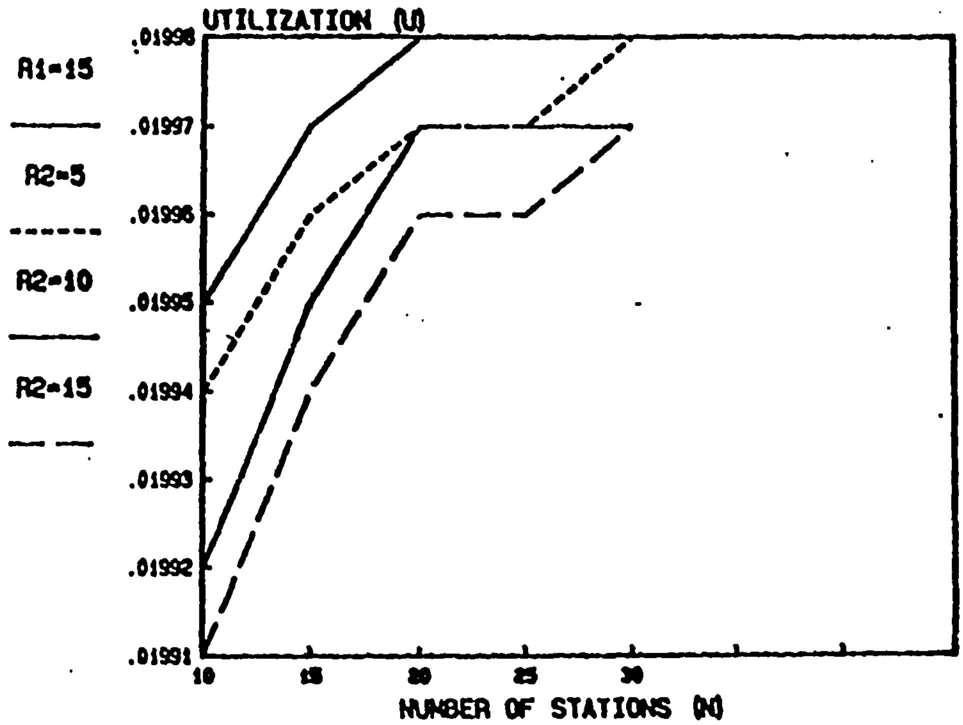

FIGURE 3.16 UTILIZATION AS A FUNCTION OF N FOR THE TOKEN-PASSING PROTOCOL DOUBLE-RING
Lp = 1000 bits DR = 50 Mbps

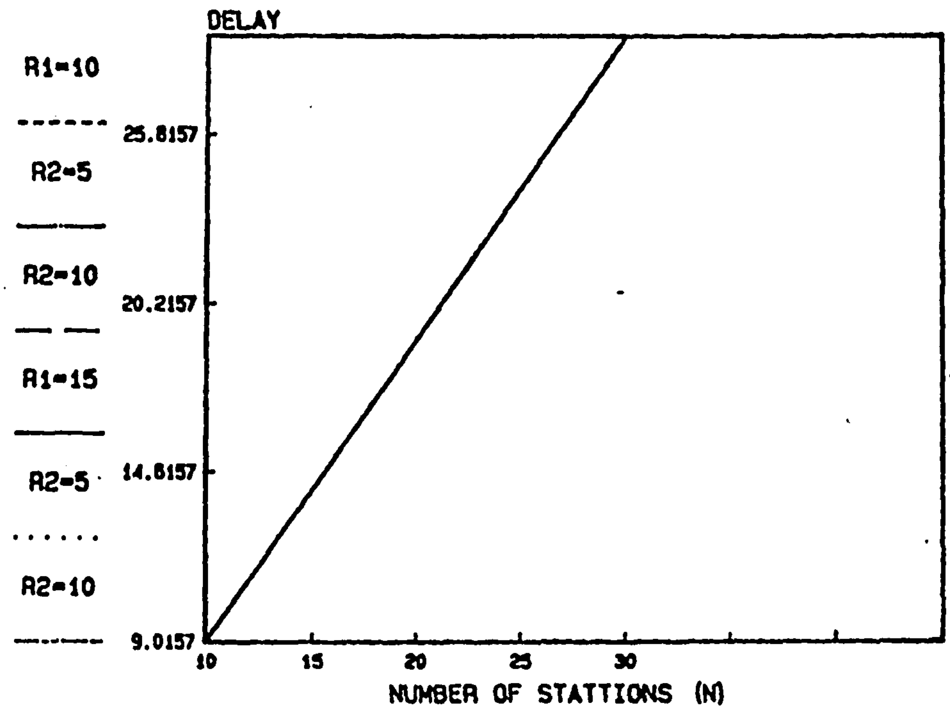

FIGURE 3.17 DELAY AS A FUNCTION OF N FOR THE TOKEN-PASSING PROTOCOL DOUBLE-RING Lp = 1000 bits DR = 50 Mbps

c. DELAY

The delay in the token-passing protocol consists of (N-1) cycles plus a/N, the token-passing time. This result is given as

$$D = \begin{cases} N + a - 1 & a < 1 \\ aN & a > 1 \end{cases} \qquad 3.21$$

So this implies that for the single-ring

$$D_1 = \begin{cases} N + a_1 - 1 & a_1 < 1 \\ a_1 N & a_1 > 1 \end{cases} \qquad 3.22$$

and for the double-ring we have

$$DD = \begin{cases} N + a_D - 1 & a_D < 1 \\ a_D N & a_D > 1 \end{cases} \qquad 3.23$$

where

$$a_D = a_1(1 + R_2/R_1)$$

3.4.2 CSMA/CD PROTOCOL

a. THROUGHPUT

From [30], the throughput for the CSMA/CD is given as

$$ST = \frac{1/2a}{1/2a + \frac{1-A}{A}} \qquad 3.24$$

where $A = (1 - 1/N)^{N-1}$

N = number of active stations.

For the single-ring network this transforms to

$$ST_1 = \frac{1}{[1 + 2a_1(\frac{1-A}{A})]} \qquad 3.25$$

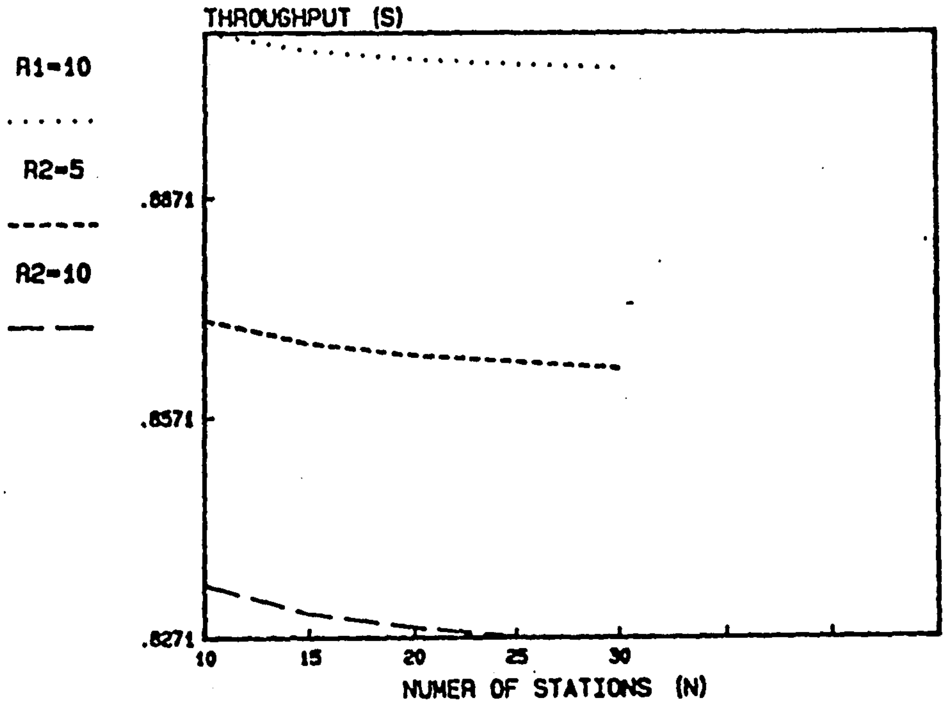

FIGURE 3.18 THROUGHPUT AS A FUNCTION OF N FOR THE CSMA/CD PROTOCOL DOUBLE-RING
Lp = 100 bits DR = 10 Mbps

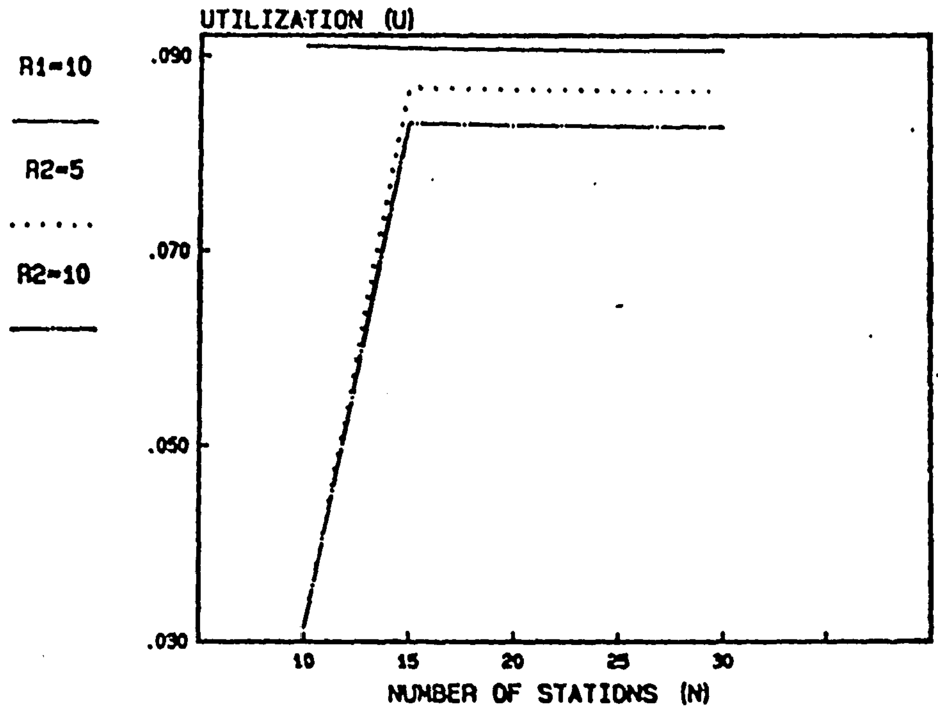

FIGURE 3.19 UTILIZATION AS A FUNCTION OF N FOR THE
CSMA/CD PROTOCOL DOUBLE-RING
Lp = 100 bits DR = Mbps

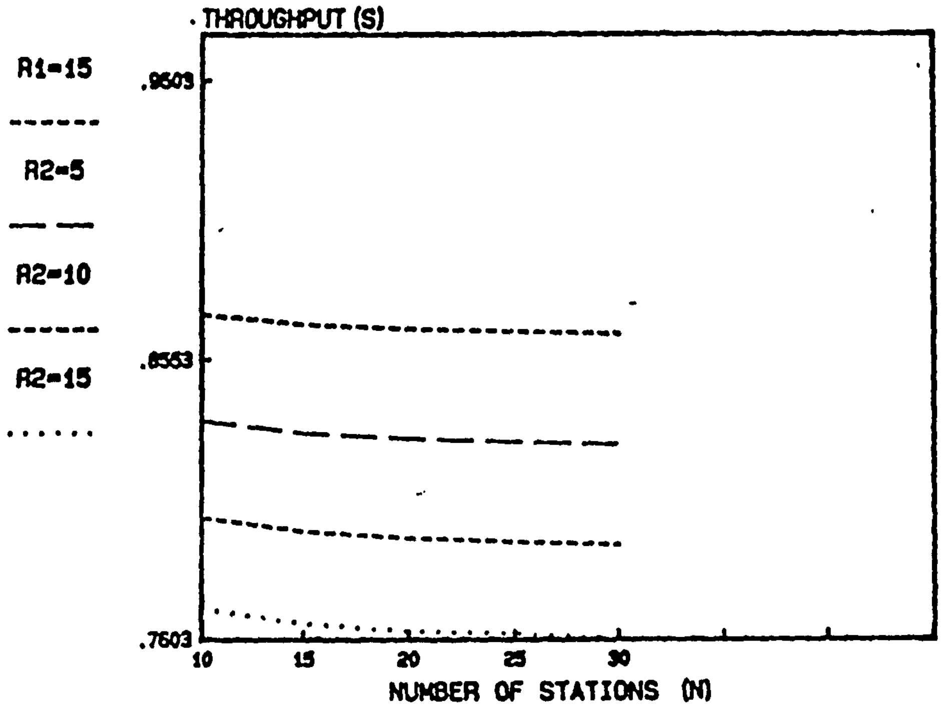

FIGURE 3.20 THROUGHPUT AS A FUNCTION OF N FOR THE CSMA/CD PROTOCOL DOUBLE-RING
Lp = 100 bits DR = 10 Mbps

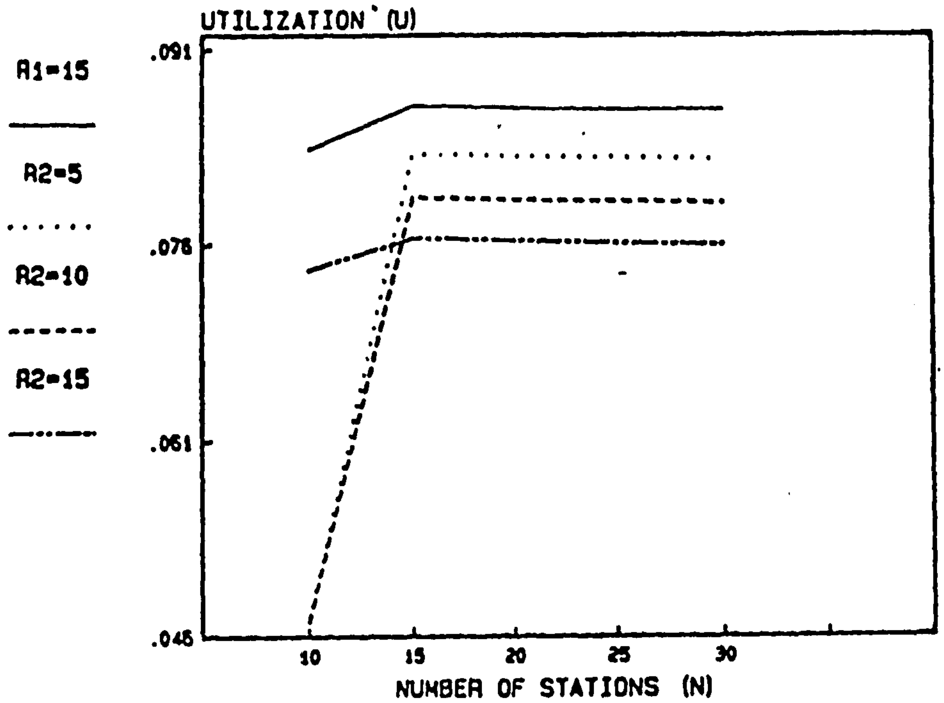

FIGURE 3.21 UTILIZTION AS A FUNCTION OF N FOR THE CSMA/CD PROTOCOL DOUBLE-RING
Lp = 100 bits DR = 10 Mbps

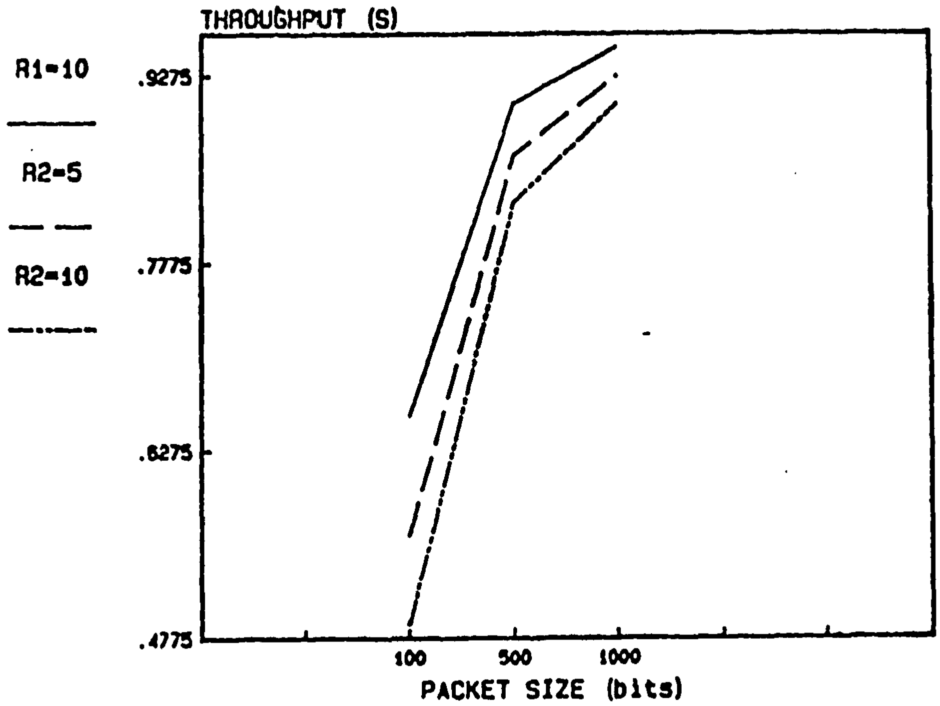

FIGURE 3.22 THROUGHPUT AS A FUNCTION OF PACKET LENGTH Lp FOR THE CSMA/CD PROTOCOL DOUBLE-RING N = 30 STATIONS DR = 50 Mbps

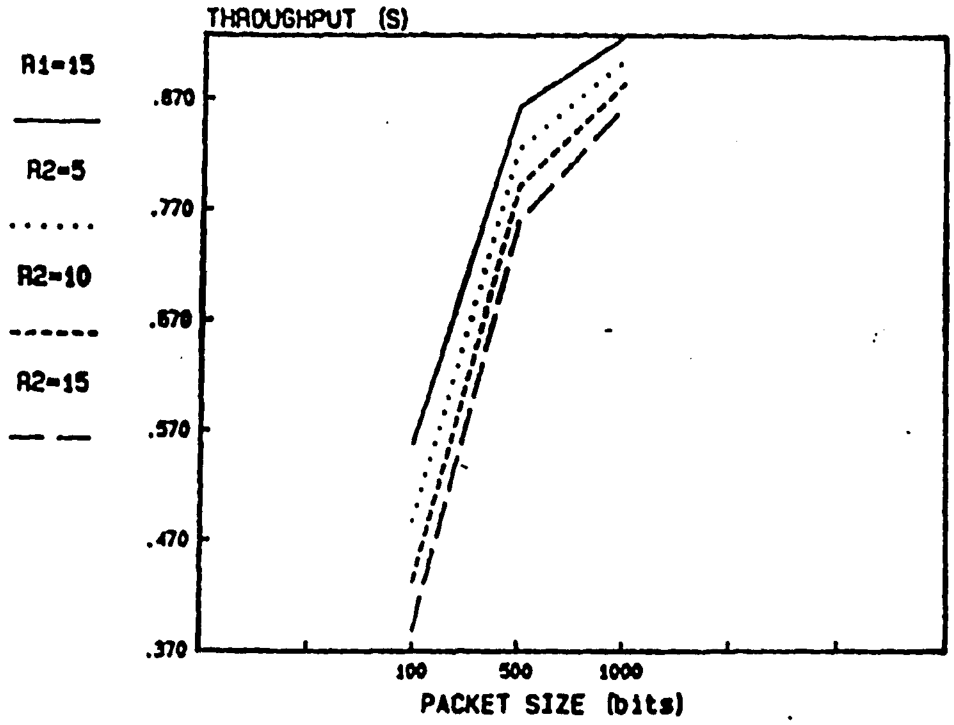

FIGURE 3.23 THROUGHPUT AS A FUNCTION OF PACKET LENGTH Lp FOR THE CSMA/CD PROTOCOL DOUBLE-RING N = 30 STATIONS DR = 50 Mbps

and

$$ST_D = \frac{1}{[1 + 2a_1(1 + R_2/R_1)[\frac{1-A}{A}]]} \qquad 3.26$$

for the double-ring network.

b. UTILIZATION

The network utilization for the single-ring becomes

$$U_1 = S_1/\text{Data rate}$$

$$= \frac{1}{D_r[1 + 2a_1(\frac{1-A}{A})]} \qquad 3.27$$

and for the double-ring

$$U_D = \frac{1}{D_r[1 + 2a_1(1 + R_2/R_1)(\frac{1-A}{A})]} \qquad 3.28$$

3.4.3 SLOTTED-RING PROTOCOL

The slotted-ring delay analysis was performed by W•Bux [52] in which the following result was obtained.

$$D = \left(\frac{2}{1-\rho}\right)\left[\frac{L_H + L_D}{L_D}\right] E\,\{x\} + \tau/2 \qquad 3.29$$

where L_H is the length of the header field, L_D the data field length, $E\,\{x\}$, the average packet service time, τ the propagation delay time and ρ the channel utilization defined as $\lambda E\{x\}$. Using the same criteria as in [51] we immediately see that the main change in delay between the single-ring and the double-ring comes from τ.

$$\tau = \frac{\text{ring length}}{\text{propagation speed}}$$

which for the single-ring becomes

$$\tau_1 = \frac{2\pi R_1}{B} \qquad 3.30$$

In the case of the double-ring, τ_2 becomes

$$\tau_D = \frac{2\pi[R_1 + R_2]}{B}$$

$$= \frac{2\pi}{} \frac{R_1[1 + R_2/R_1]}{B}$$

$$= \tau_1 [1 + R_2/R_1] \qquad 3.31$$

Thus the corresponding delays are for the single-ring

$$D_1 = \left(\frac{2}{1-\rho}\right)\left(\frac{LH + LD}{LD}\right) E\{x\} + \frac{\pi R_1}{B} \qquad 3.32$$

and for the double-ring

$$D_2 = \left(\frac{2}{1-\rho}\right)\left(\frac{LH + LD}{LD}\right) E\{x\} + \frac{\tau_1}{2} (1 + R_2/R_1) \qquad 3.33$$

If the header length L_H is small compared to the data length, then

$$\frac{L_H + L_D}{L_D} \cong 1.$$

3.5 ANALYSIS OF RESULTS

3.5.1 SCANTIME ANALYSIS

As Shown in equations 3.5 and 3.6, the additional delay or penalty introduced with the addition of a second channel is given by

$$\Delta S = S_D - S$$

$$= \frac{2\pi R_1}{B(1-\rho)} [(1 - 1/N) + R_2/R_1] - \frac{2\pi R_1}{B(1-\rho)}$$

$$= S[(1 - 1/N) + R_2/R_1 - 1]$$

which reduces to S

for large N and $R_1 = R_2$

Since the scantime depends on both the radius of the ring R and the traffic intensity ρ, a comparison between the double-ring scantime and the single-ring scantime for R_1=10 meters and R_1=15 meters has been made. While it is true that the ring may be a few kilometers long rather than meters as used in this analysis, we can easily see that the change from meters to kilometers only changes the obtained results for the scantime from μsecs to msec. Figures 3.4 and 3.6 show the variation of the double-ring scantime for changing values of the inner ring length R_2. Since the channel-acquisition delay is one half the scantime, its variation to R is same as that of the scantime. This variation is shown in Figures 3.5 and 3.7.

3.5.2 NETWORK EFFICIENCY UNDER HEAVY LOAD ANALYSIS

In comparing the efficiency of the single-ring and double-ring networks, one must evaluate this comparison in terms of N the number of stations and the radius R. The single channel efficiency is given as

$$E = \frac{qNB}{2\pi R_1 \mu}$$

while the double-ring network efficiency is

$$E_D = \frac{qNB}{2\pi\mu[(1 - 1/N)R_1 + R_2]}$$

The change in efficiency is given as

$$E - E_D = \Delta E$$

$$= \frac{qNB}{2\pi\mu R_1} \left[1 - \frac{1}{[(1 - 1/N) + R_2/R_1]}\right]$$

$$\Delta E = E\left[1 - \frac{1}{[(1 - 1/N) + R_2/R_1]}\right] \qquad 3.34$$

For large N and $R_1 = R_2$, we have

$$\Delta E = 1/2\ E$$

This change approaches zero as R_1 increases. We can easily see that the change in efficiency for large N will not be as significant as the change in scantime.

3.5.3 PROTOCOLS PERFORMANCE EVALUATION

a. TOKEN-PASSING PROTOCOL

As shown in Figures 3.8 through 3.17, the channel throughput, utilization and delay analysis have been performed for both the single-ring and the double-ring. These results detail the variation of the throughput, utilization, and delay with respect to number of stations, data rate and packet length for various lengths of R_1 and R_2. The values of throughput obtained are solely dependent on "a". In this

analysis, a was always less than 1. As expected, the throughput as a function of the number of stations for a constant data rate increases with increase in packet length. The interesting result here, however, is the fact that there is no significant change in both channel throughput and utilization when R_2 is equal to R_1. Furthermore, the delay in this case is also acceptable when compared with the delay of the single-ring network. Since

$$a_D = a_1[1 + R_2/R_1],$$

the double-ring performance will drop significantly. This we can easily show by looking at the throughput as an example. The single-ring throughput for a > 1 is

$$ST = \frac{1}{a_1(1 + 1/N)}$$

For the double-ring, the throughput is given as

$$ST_D = \frac{1}{a_D(1 + 1/N)}$$

$$= \frac{1}{a_1(1 + R_2/R_1)[1 + 1/N]} \qquad 3.35$$

The change in throughput is found by

$$\Delta ST = ST - S_{TD}$$

$$\Delta ST = \frac{1}{a_1(1 + 1/N)} - \frac{1}{a_1(1 + R_2/R_1)(1 + 1/N)}$$

$$= \frac{R_2/R_1 \quad ST}{(1 + R_2/R_1)} \qquad 3.36$$

For $R_1 = R_2$, we have

$$\Delta ST = \frac{1}{2} ST$$

This would be the worst case analysis. This analysis applies to the channel utilization. Thus, the change in channel utilization will be given as

$$\Delta U = \Delta S/D_r$$
$$= 1/2\ U.$$

The token-ring delay for $a > 1$ is given as

$$D = aN$$

which for the single-ring is given as

$$D = a_1 N$$

For the double-ring, we have

$$DD = a_D N$$
$$= a_1(1 + R_2/R_1)N \qquad 3.37$$

when $R_1 = R_2$, $DD = 2D$ which is similar to the scantime delay results.

b. CSMA/CD PROTOCOL

From Figures 3.18 to 3.23 we can see that there is a slight change in performance. The token-passing ring protocol performed better than the CSMA/CD protocol. This comparison is based on the throughput and channel utilization analysis. As in the token-passing protocol, the

obtained results are based on "a" less than 1. For a greater than 1, this protocol performance will drop significantly. The change in throughput is given here as

$$\Delta S = S - S_D$$

$$= \frac{1}{1 + 2a_1\frac{(1-A)}{A}} - \frac{1}{1 + 2a_1(1 + R_2/R_1)\frac{(1-A)}{A}}$$

$$\Delta S = \frac{R_2/R_1}{\left(\frac{A}{2a_1(1-A)} + 1\right)\left(\frac{A}{2a_1(1-A)} + (1 + R_2/R_1)\right)}$$

For $R_1 = R_2$, this reduces to

$$\Delta S = \frac{1}{\left(\frac{A}{2a_1(1-A)} + 1\right)\left(\frac{A}{2a_1(1-A)} + (1 + R_2/R_1)\right)} \qquad 3.38$$

c. SLOTTED-RING PROTOCOL

With this protocol, a simple comparison in delay was performed. The change in delay as in the previous protocols was found to be twice the original delay when $R_1=R_2$. This delay analysis is carefully monitored in the sense that it was arrived at by looking at the propagation delays and noting that the rest of the expression on expression 3.29 remains unchanged. In the true sense of the word delay as applied in this protocol analysis will be given as

$$D_1 = K_1 + \tau_1$$

$$= K_1 + \frac{2\pi R_1}{B}$$

while for the double-ring we have

$$D_D = K_1 + \tau_D$$

$$= K_1 = \tau_1 (1 + R_2/R_1)$$

$$\Delta D = DD - D_1$$

$$= \tau_1(1 + R_2/R_1) - \tau_1$$

$$= \tau_1(1 + R_2/R_1 - 1)$$

For $R_1 = R_2$, $DD = \tau_1$

or $DD = 2D_1$.

This conforms with the previously obtained results like the scantime delay.

CHAPTER FOUR

SPOKED-RING NETWORKS

4.1 INTRODUCTION

As with the double-ring network, the spoked-ring network [26] is shown in Figure 4.1

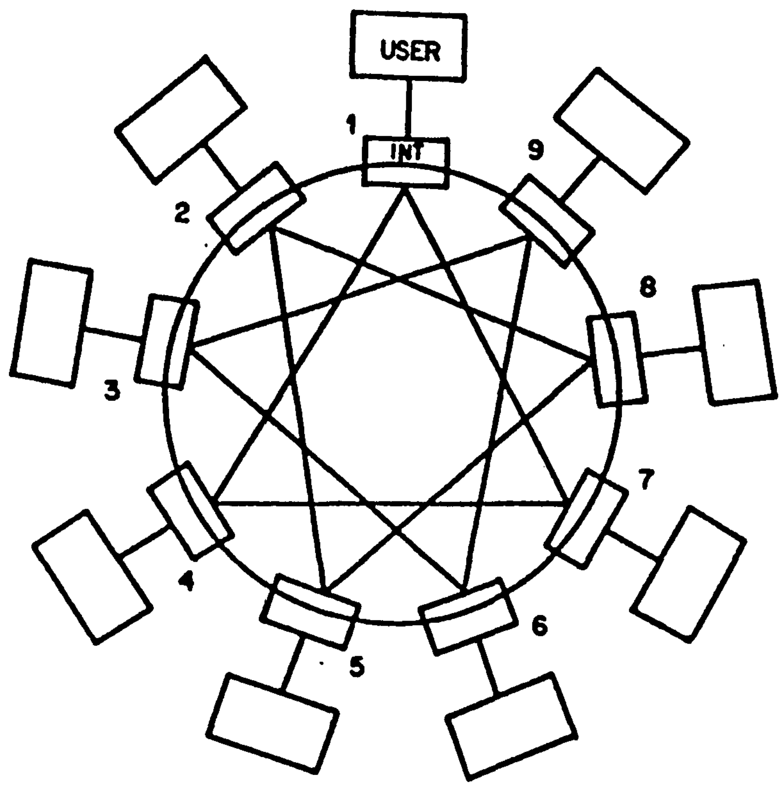

FIGURE 4.1 A SPOKED-RING WITH 9 USERS

It consists of a ring where the interfaces are also connected independently by means of spokes. The pattern of connection of the spokes is entirely arbitrary but regular pattern connection examples will be used to simplify performance analysis.

4.2 NETWORK OPERATIONS

While the spoked-ring network could be operated in either contention or CSMA/CD, we consider the network first as a token-passing spoked-ring where each ring interface transmits the incoming message $\alpha + \beta$ in two phases: one, along the ring with undiminished magnitude, and two, along the outgoing spoke with magnitude that is a fraction of r of this value as shown in Figures 4.2 and 4.3.

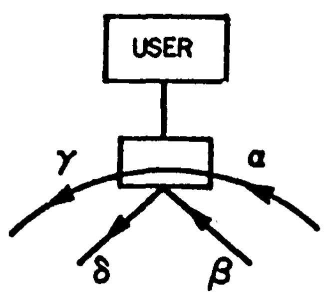

FIGURE 4.2 OPERATION OF RING INTERFACE IN LISTENING MODE

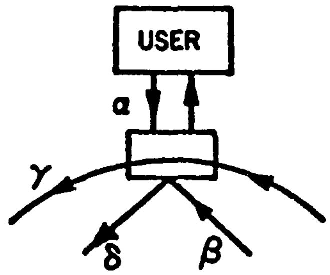

FIGURE 4.3 RING INTERFACE IN TRANSMIT MODE

$\lambda = \alpha$

$\delta = \frac{\alpha}{r}$, $r >> 1$

The value of r is chosen such that the input $\alpha + \frac{\alpha}{r}$ to a listening interface is generated as α because $\frac{\alpha}{r}$ is below the threshold value of the detector in the interface. This implies that during normal operation of the ring, transmission of message is through the ring itself and not through the spokes. When a break-down occurs, either for additional node attachments or due to node failure, the affected spoke is amplified by a value of r after a defined time (wait time) which results in message transmission to the other nodes in the network. The penalty here is the wait time required for spoke activation, while a regular ring simply must stop all transmissions resulting in a wait time equal to the time it takes to add or make repairs on the ring.

4.3 NETWORK CONNECTIVITY

In order for the spoked-ring network to operate properly in the face of breakdowns and/or repairs, it must maintain connectivity to all nodes except the broken nodes. Thus, from Figures 4.2 and 4.3, we can see that the minimum degree requirement for the spoked-ring network is four.

One of the problems associated with the spoked-ring is that of establishing how many spokes are required to achieve connectivity. For complete connectivity one requires N(N-1)/2 spokes where N is the number of stations or nodes to be connected. This provides enough redundancy for the network to remain operational for several kinds of node failures. The price paid in terms of additional links may be too high, however, should one not be able to afford complete connection, the question of how to link nodes for efficient working comes up. Too few spoke connections result in a loss of network total connectivity when there is a station failure.

A combination of alternate and two-station connection is proposed for the spoked-ring. This introduces at most N connections for an N-station or node spoked-ring. As shown in Figures 4.4 and 4.5, we see that for even number of stations, the alternate connection works well. However, with a constraint that no two consecutive stations be directly connected except through the ring itself, the alternate station connection introduces at most N connections. Such connectivity ensures that no station is disconnected from the network due to a failure of an adjacent station. Our spoked-rings will thus have N spokes. If the spoke lengths add up to the length of the original ring, one can compare the situation to introducing an additional ring path as in the

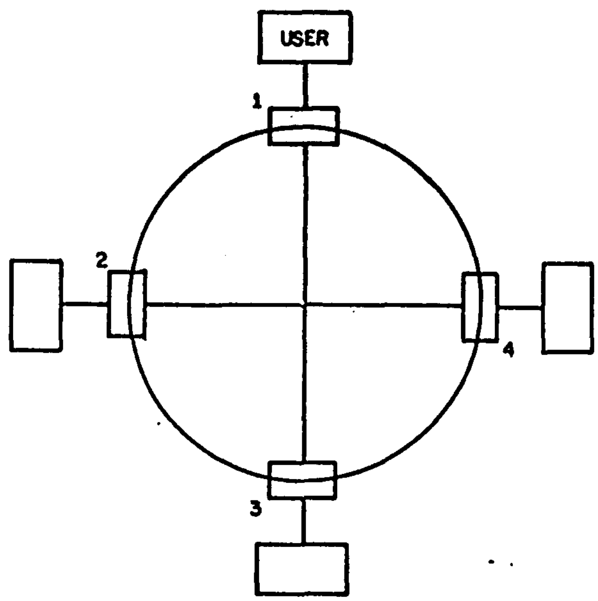

FIGURE 4.4a 4 STATIONS, 2 SPOKES

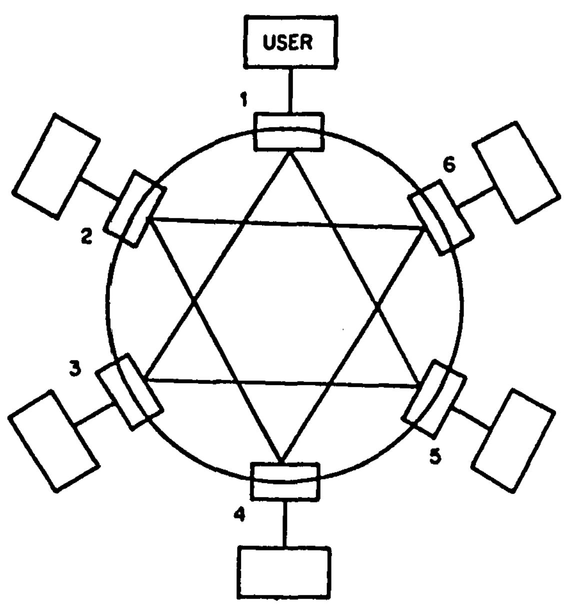

FIGURE 4.4b 6 STATIONS, 6 SPOKES

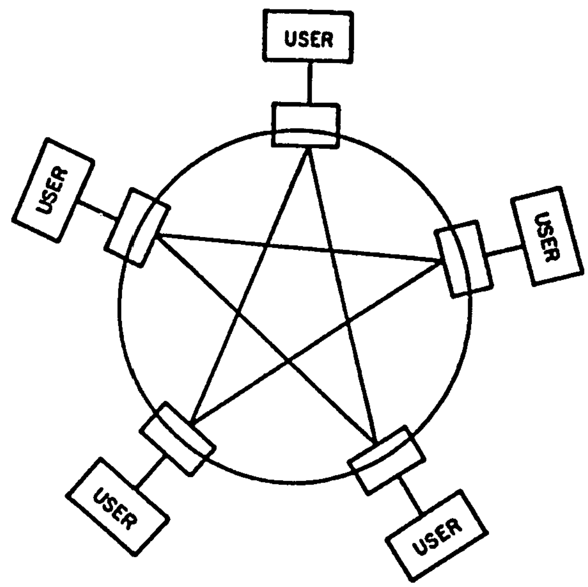

FIGURE 4.5a 5 STATIONS, 5 SPOKES

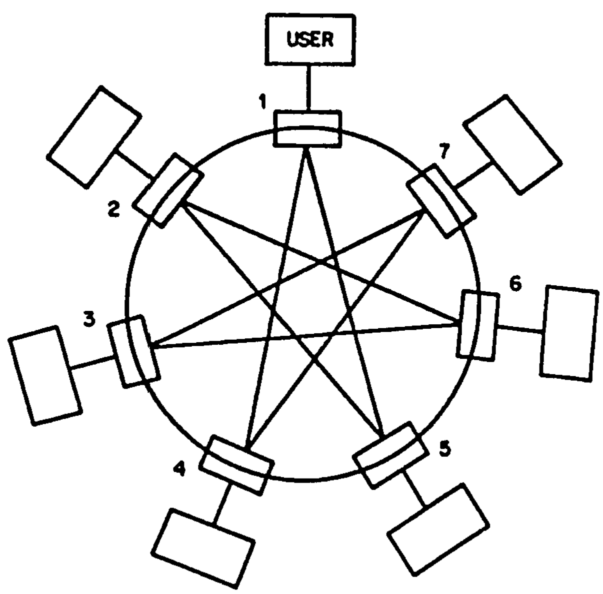

FIGURE 4.5b 7 STATIONS, 7 SPOKES

double-ring network.

4.4 SPOKE AMPLIFIER ACTIVATING TIME ANALYSIS

Let us assume the length of the ring to be L, it is further taken that the N stations attached to the ring are equally spaced such that the distance between any two stations is L/N. Further assume that the spoke length between any two stations is d. Consider the case where a station is down and the objective is that the spoked-ring network should be operational. This means that a path between the transmitting station and all other stations must exist while avoiding the broken station. In the context of Figure 4.6, take station one to be transmitting some message through the ring and that station two is down or broken, one requires at least $\frac{3L}{N}$ length time for station one to realize the existence of a break down and activate the spoke amplifier. Once the amplifier is activated, it will take d length delay to travel to station four and another d delay to travel from station nine to station three. Thus, a total of 2d delay is required to have the spoked-ring operational after a break down. It follows then that it will require at least $(K+1)\frac{L}{N}$ length time for a message to travel from the transmitting station to its spoke-connected station; where K is the number of stations between the spoke-connected nodes or stations. For alternate connections, K=1 and for a two-station skip connection K=2.

Assuming an average bit delay of length $\frac{L}{N}$ in the ring network then the minimum requirement for the spoke amplifier to be activated will be given by

$$T_{SA} = (K + 1)\frac{L}{N} + \frac{L}{2N} \quad \text{length time.} \qquad 4.1(a)$$

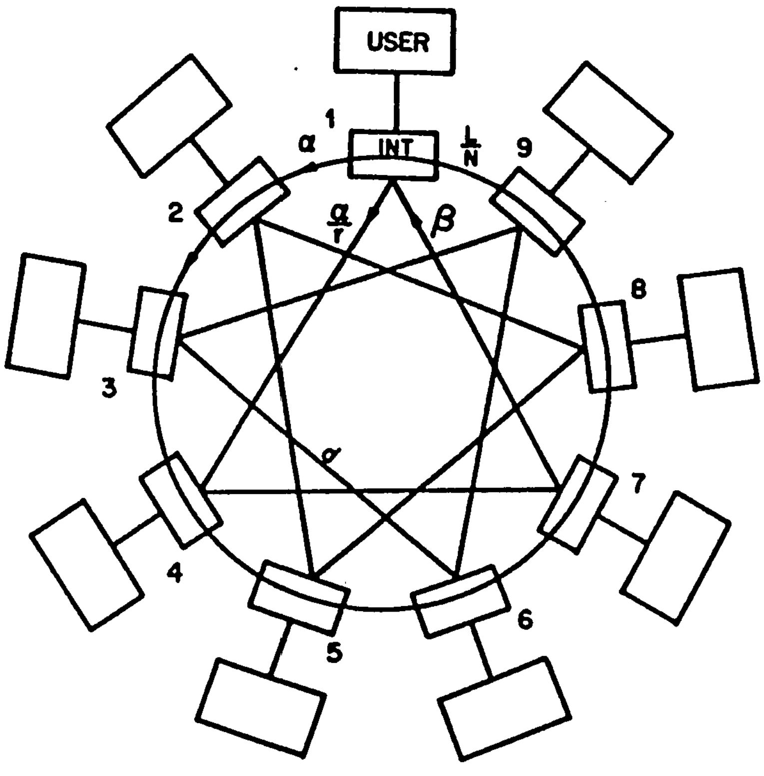

FIGURE 4.6 9 STATION SPOKED-RING NETWORK

Thus, for a propagation speed of B meters per microsecond (B m/μs)

$$T_{SA} = \frac{L[2k + 3]}{2NB} \text{ bits time} \qquad 4.1(b)$$

where L is in meters which is same as

$$T_{SA} = \frac{L}{2NB} \; [2k+3] \text{ sec.} \qquad 4.1(c)$$

An upper bound for the spoke distance d is $(K+1)\frac{L}{N}$ which is the maximum distance travelled by a message between spoke-connected stations.

Thus, $$d < (k+1)\frac{L}{N} \qquad 4.2$$

4.5 EFFICIENCY ANALYSIS

Let us assume that packets are generated according to a Poisson process, and that when a station receives the permission to send, it empties itself of all packets with mean queue length of q. Let the rate of arrival for all the N stations be λ packet/sec such that each station arrival rate is λ/N. Let the service rate be μ (the number of packets a station can transmit.) S, the scantime, is given by

$$S = w + \frac{Nq}{\mu} \text{ ; for a regular ring.}$$

For the spoked-ring with one failure, we have the following:

$$w = \frac{\text{ring length}}{\text{propagation rate}}$$

$$= \frac{L + 2d - \frac{KL}{N}}{B/\mu\text{sec}}$$

$$= \frac{L\left[1 - \frac{k}{N}\right] + 2d}{B} \qquad 4.3$$

which approximates to

$$W = \frac{L + 2d}{B} \qquad 4.4$$

for large N.

Thus, the scantime becomes

$$S = \frac{w}{1-\rho}$$

$$= \frac{L + 2d}{B[1-\rho]} \qquad 4.5$$

where $$\rho = \frac{\lambda}{\mu}, \ \rho<1.$$

This introduces a penalty of $\frac{2d}{B(1-\rho)}$

For $$d = \frac{KL}{N} + \frac{L}{2N} \ < (K+1)\frac{L}{N}$$

S becomes

$$Sp = \frac{L[N+2K+1]}{BN[1-\rho]} \tag{4.6}$$

4.5.2 CHANNEL-ACQUISITION DELAY

The channel-acquisition delay is about half the scantime [12]. For a regular ring, this delay is about

$$\frac{1}{2}S = CD = \frac{L}{2B(1-\rho)} \tag{4.7}$$

For the spoked-ring, we have

$$\frac{1}{2}S = CD = \frac{L[N+2k+1]}{2NB(1-\rho)} \tag{4.8}$$

when K=1,

$$CD_p = \frac{L\ [N+3]}{2NB[1-\rho]} \tag{4.9}$$

when K=2,

$$CD_p = \frac{L\ [N+5]}{2NB[1-\rho]} \tag{4.10}$$

4.5.3 CHANNEL EFFICIENCY UNDER HEAVY LOAD

Under heavy load condition, the only overhead associated with the network is w/N (the walk time between stations.) Since every station has something to transmit, the average transmission time per station is q/μ. If we define channel efficiency as the ratio of the station transmission time to the overhead (walk time between station) then the channel efficiency is given by

$$E = \frac{q/\mu}{w/N}$$

$$= \frac{qN}{N\mu} \qquad 4.11$$

For a ring network, $w = \frac{L}{B}$

resulting in

$$E = \frac{qNB}{L\mu} \qquad 4.12$$

For the spoked-ring,

$$w = \frac{L[1-\frac{K}{N}] + 2d}{B}$$

Let $d = \frac{KL}{N} + \frac{L}{2N}$

Then, $$E_p = \frac{qN^2B}{\mu L[N+k+1]} \qquad 4.13$$

For large N, E_p reduces to E as in equation 4.12.

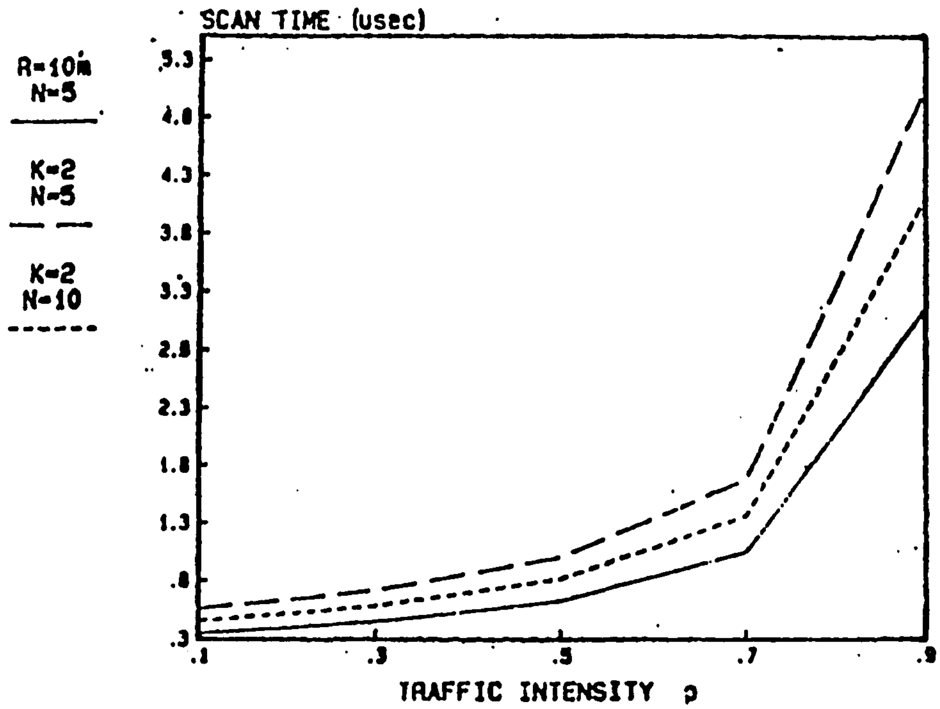

FIGURE 4.7 SCANTIME AS A FUNCTION OF TRAFFIC INTENSITY FOR THE SPOKE-RING

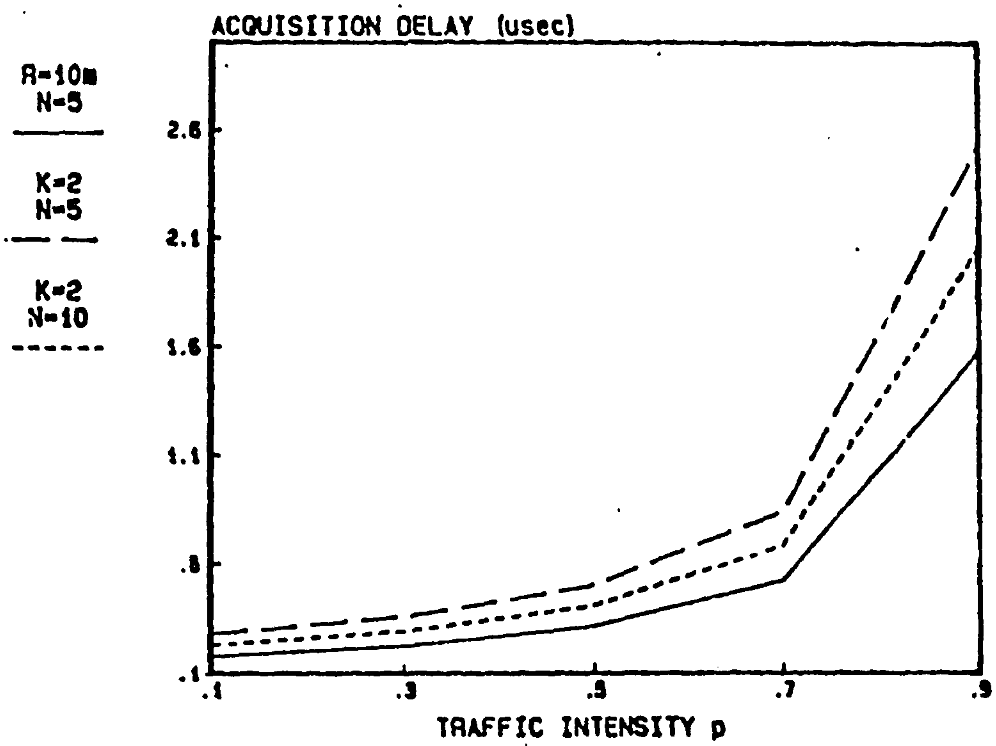

FIGURE 4.8 CHANNEL ACQUISTION DELAY AS A FUNCTION TRAFFIC INTENSITY FOR THE SPOKE-RING

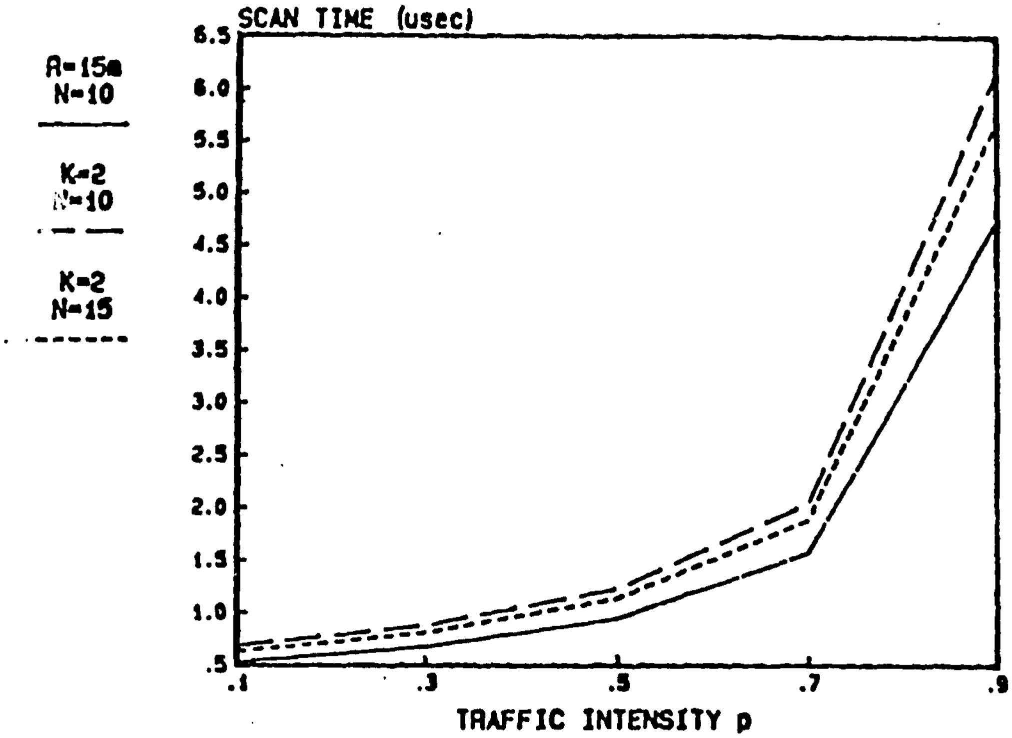

FIGURE 4.9 SCANTIME AS A FUNCTION OF TRAFFIC INTENSITY FOR THE SPOKE-RING

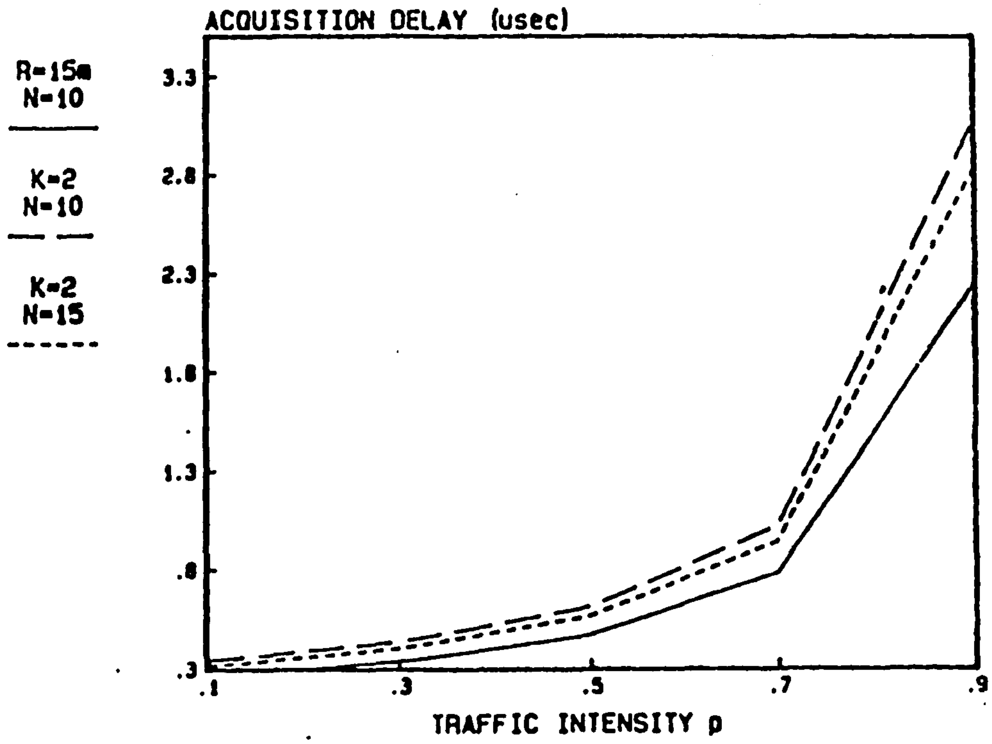

FIGURE 4.10 CHANNEL ACQUISITION DELAY AS A FUNCTION OF TRAFFIC INTENSITY FOR SPOKED- RING

4.6 MEDIUM ACCESS PROTOCOL EVALUATION

As in the previous chapter, we will compare the performance of the spoked-ring to that of the single-ring. Note that "a" is given

$$a = \frac{2\pi R_1 D_r}{BxL_p}$$

4.6.1 TOKEN-PASSING PROTOCOL

a. THROUGHPUT

The throughput for the ring is

$$S_1 = \begin{cases} \dfrac{1}{(1 + a_1/N)} & a_1 < 1 \\[2ex] \dfrac{1}{a_1(1 + 1/N)} & a_1 > 1 \end{cases}$$

while for the spoked-ring we have

$$S_p = \begin{cases} \dfrac{1}{(1 + a_p/N)} & a_p < 1 \\[2ex] \dfrac{1}{a_p\ (1 + 1/N)} & a_p > 1 \end{cases}$$

Knowing that the length of the spoked-ring is [L + 2d - KL/N], the worst case analysis yields

$$a_p = \frac{D_r(L + 2d - KL/N)}{B \times L_p}$$

$$a_p = \frac{2\pi R_1 D_r}{BxL_p} \left[1 + \frac{d}{\pi R_1} - \frac{K}{N} \right]$$

$$a_p = a_1\left[1 + \frac{d}{\pi R_1} - \frac{K}{N} \right] \qquad 4.14$$

where $1 \leq K \leq 2$

and N is the number of stations on the ring.

For $d = \frac{KL}{N} + \frac{L}{2N}$

$$a_p = a_1[1 + (K + 1)/N] \qquad 4.15$$

b. CHANNEL UTILIZATION

The channel utilization is given by

$$U = S/\text{Data rate}$$

which for the spoked-ring becomes

$$U = S_p/D_r$$

$$U_p = \begin{cases} \dfrac{1}{D_r[1 + a_p/N]} & a_p < 1 \\ \dfrac{1}{D_r\, a_p(1 + 1/N)} & a_p > 1 \end{cases} \qquad 4.16$$

This utilization is compared to equation 3.19.

c. DELAY

As in section 3.5.1 the delay for the token-passing ring in the spoked-ring will be compared to equation 3.21

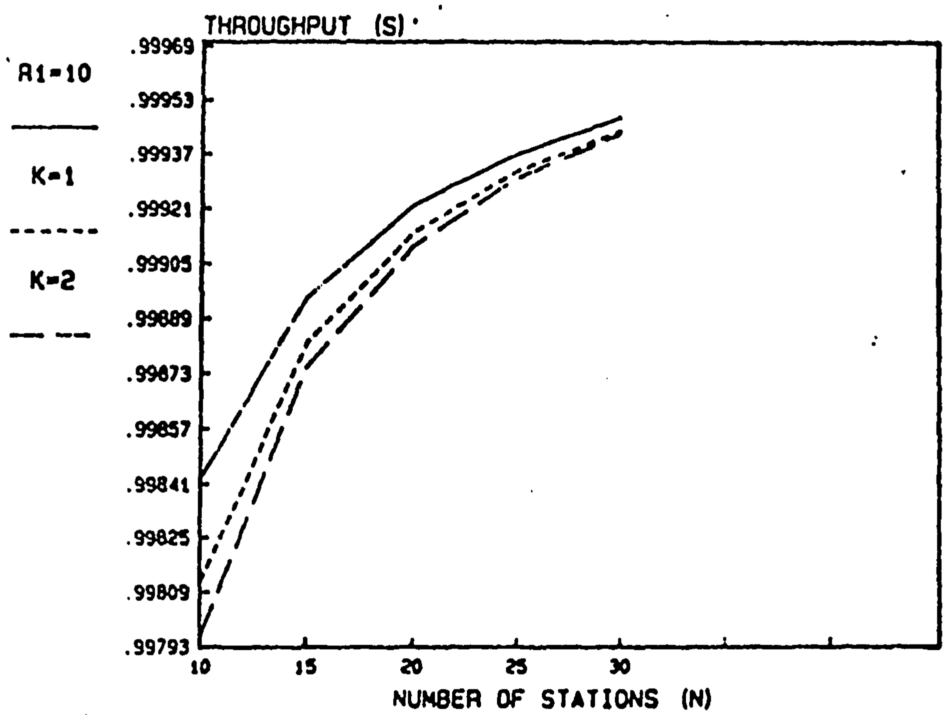

FIGURE 4.11 THROUGHPUT AS A FUNCTION OF N FOR THE TOKEN-PASSING PROTOCOL SPOKED-RING
Lp = 500 bits DR = 50 Mbps

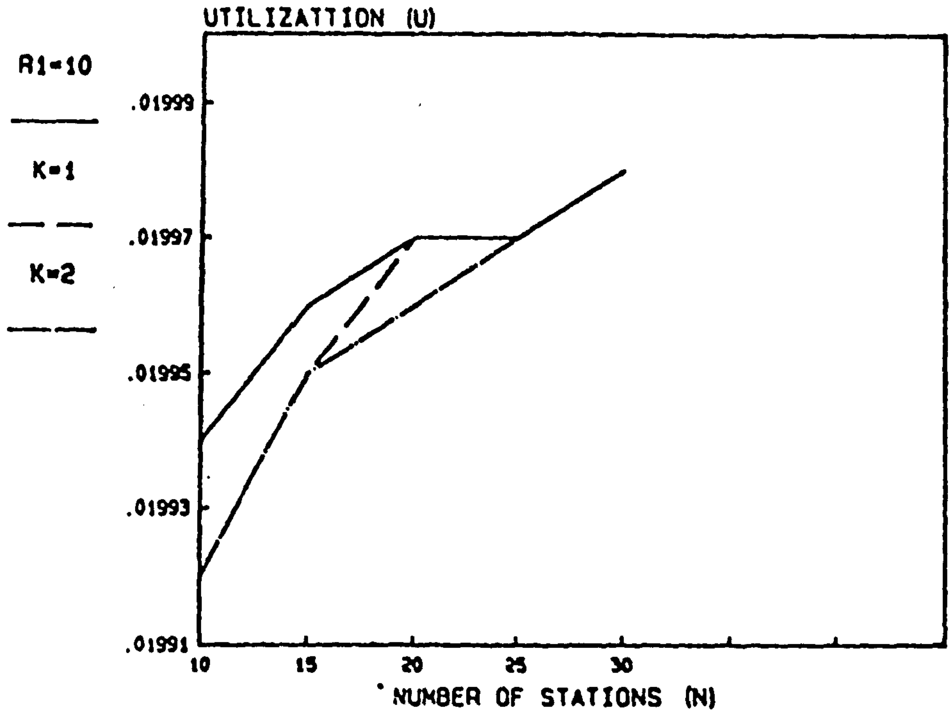

FIGURE 4.12 UTILIZATION AS A FUNCTION OF N FOR THE TOKEN-PASSING SPOKED-RING
Lp = 500 bits DR = 50 Mbps

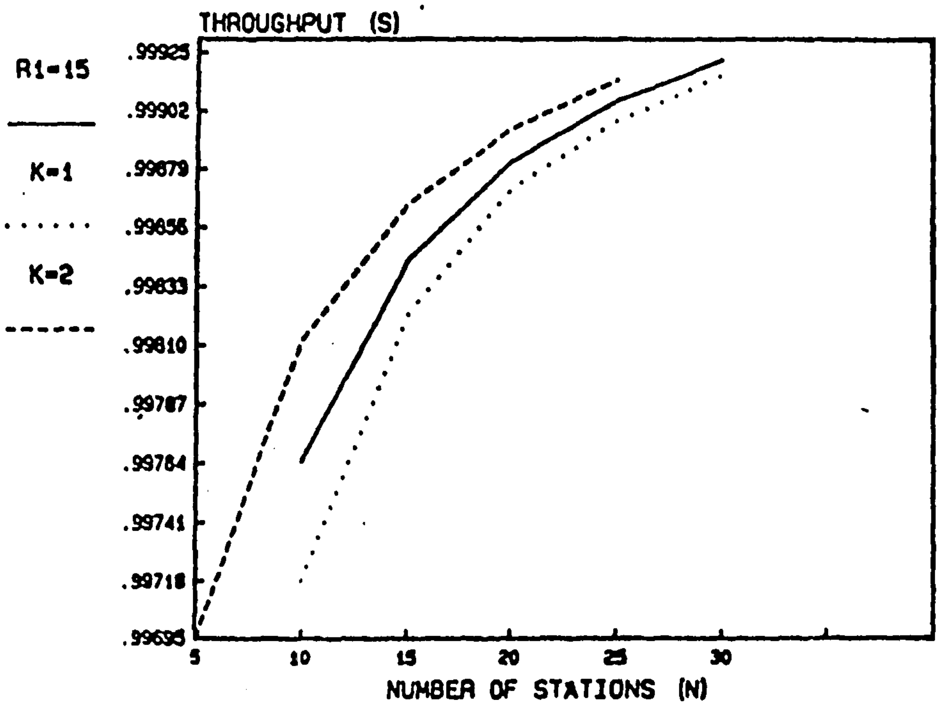

FIGURE 4.13 THROUGHPUT AS A FUNCTION OF N FOR THE TOKEN-PASSING PROTOCOL SPOKED-RING
Lp = 500 bits DR = 50 Mbps

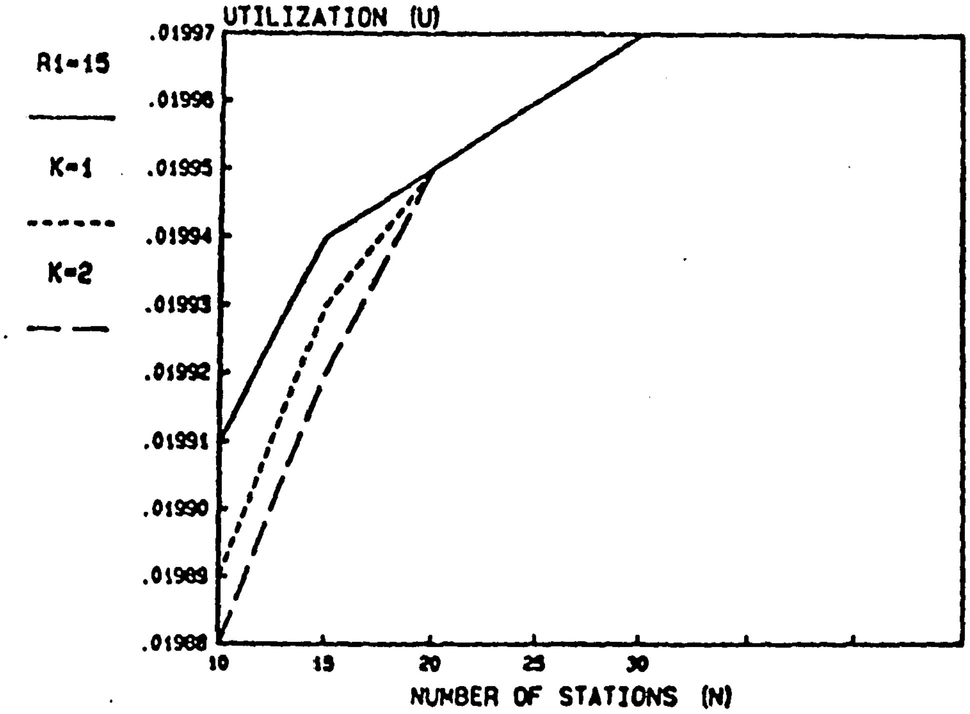

FIGURE 4.14 UTILIZATION AS A FUNCTION OF N FOR THE TOKEN-PASSING SPOKED-RING
L_p = 500 bits DR = 50 Mbps

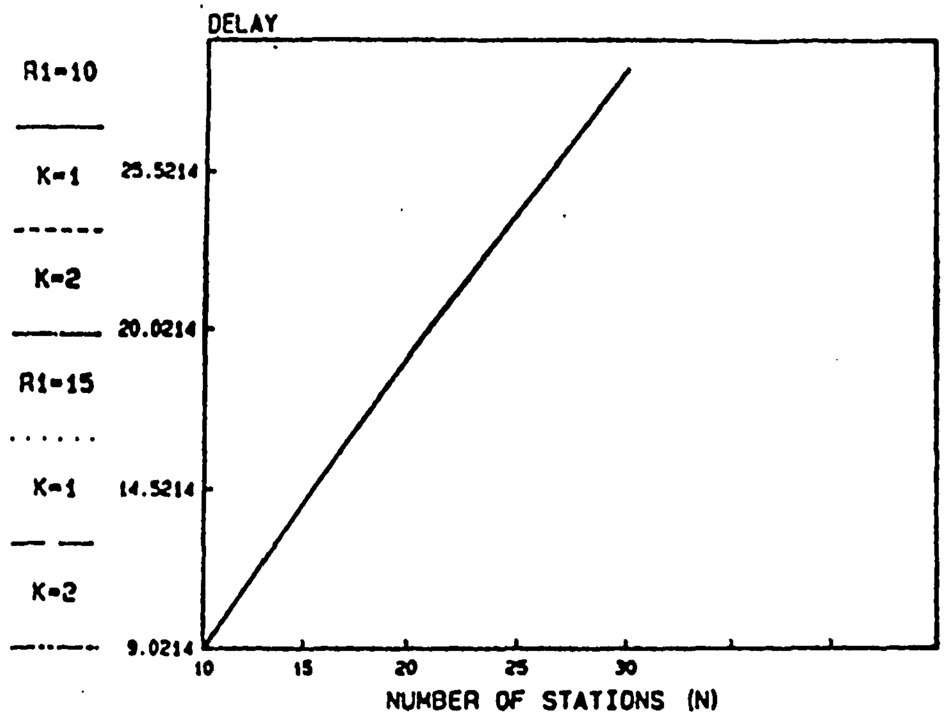

FIGURE 4.15 DELAY AS A FUNCTION OF N FOR TOKEN-PASSING PROTOCOL SPOKED-RING. Lp = 500 bits DR = 50 Mbps

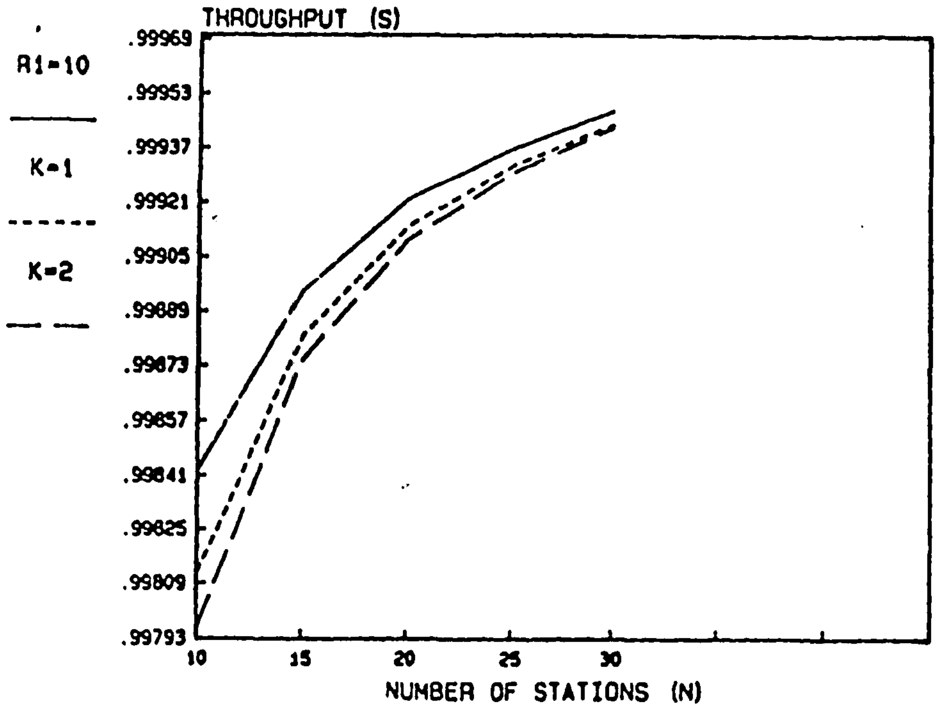

FIGURE 4.16 THROUGHPUT AS A FUNCTION OF N FOR THE TOKEN-PASSING PROTOCOL SPOKE-RING
Lp = 1000 bits DR = 50 Mbps

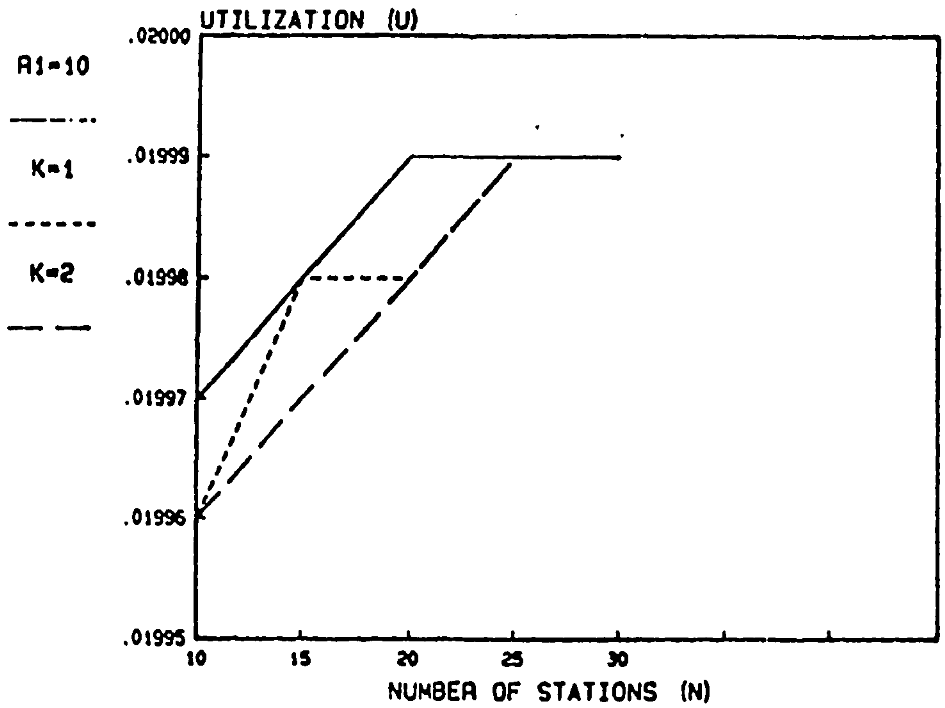

FIGURE 4.17 UTILIZATION AS A FUNCTION OF N FOR THE TOKEN-PASSING PROTOCOL SPOKE-RING
Lp = 1000 bits DR = 50 Mbps

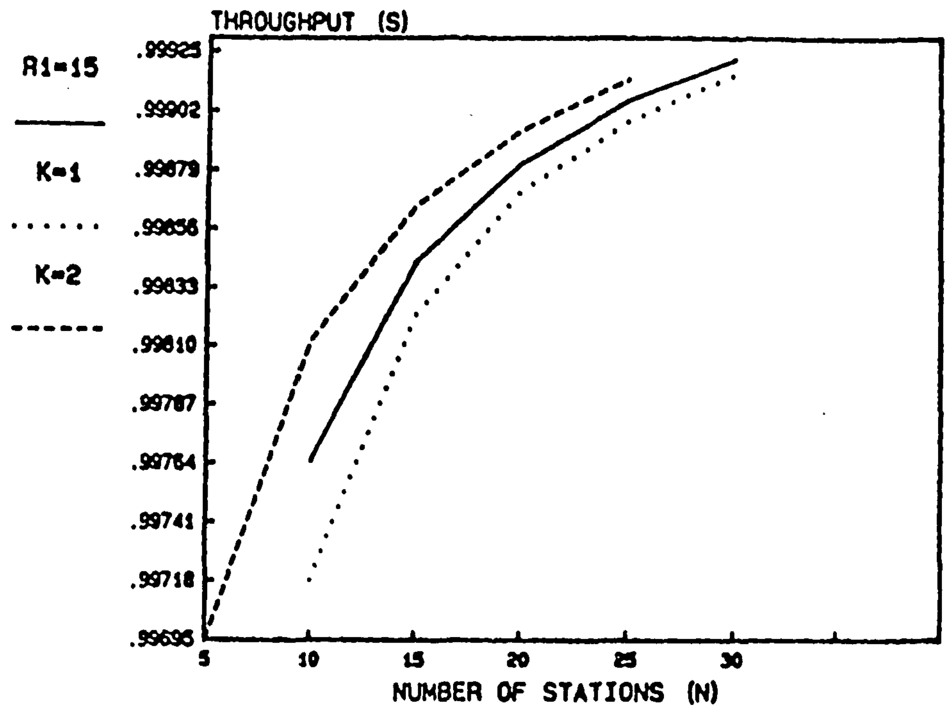

FIGURE 4.18 THROUGHPUT AS A FUNCTION OF N FOR THE TOKEN-PASSING PROTOCOL SPOKE-RING
L_p = 1000 bits DR = 50 Mbps

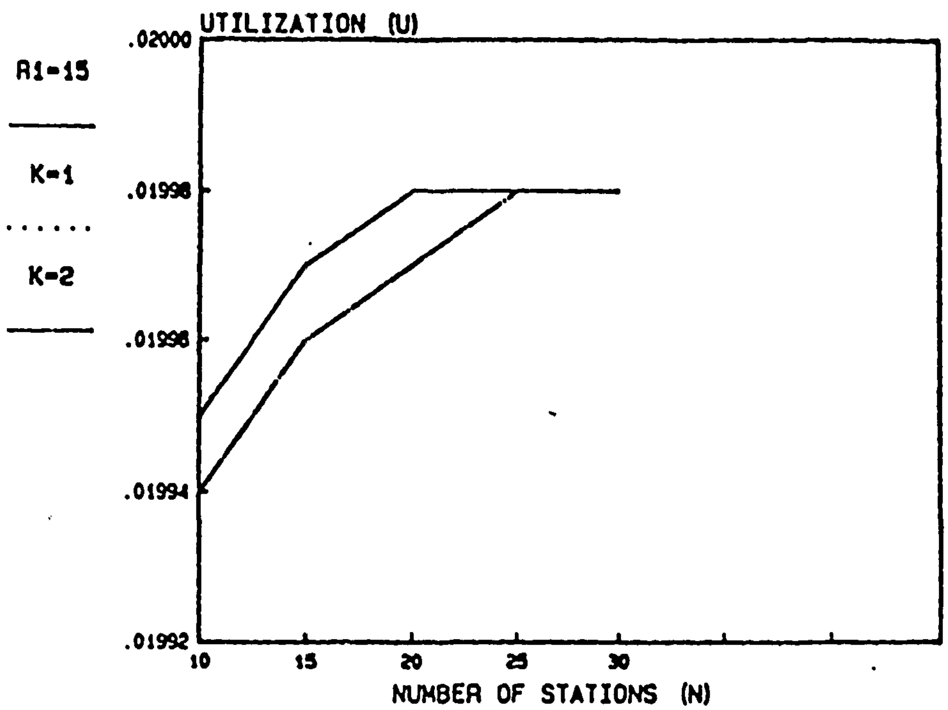

FIGURE 4.19 UTILIZATION AS A FUNCTION OF N FOR THE TOKEN-PASSING PROTOCOL SPOKE-RING
Lp = 1000 bits DR = 50 Mbps

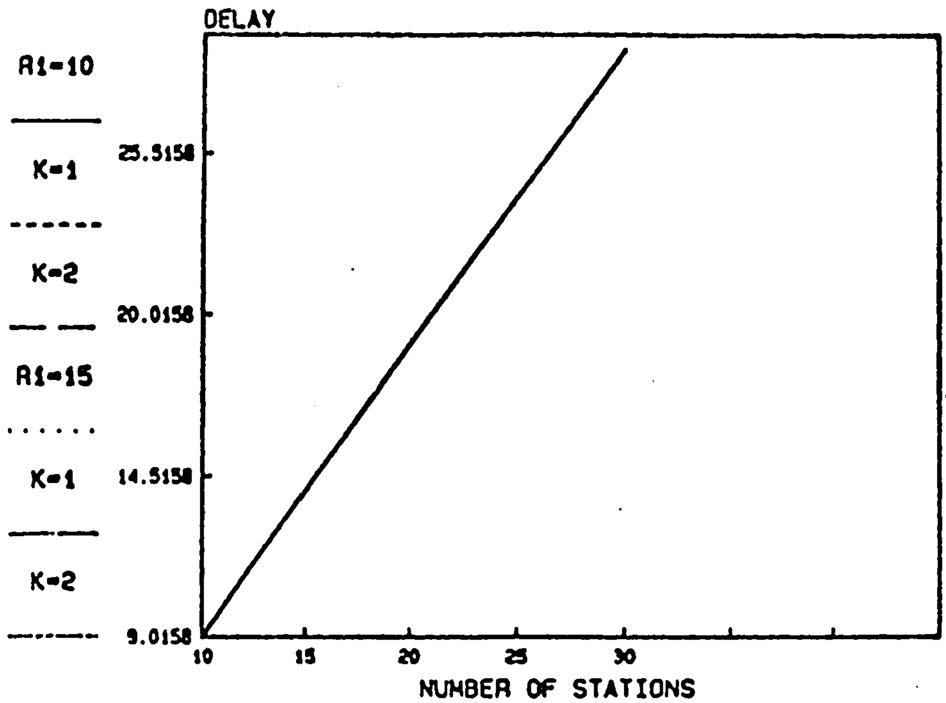

FIGURE 4.20 DELAY AS A FUNCTION OF N FOR THE TOKEN-PASSING PROTOCOL SPOKE-RING
Lp = 1000 bits DR = 50 Mbps

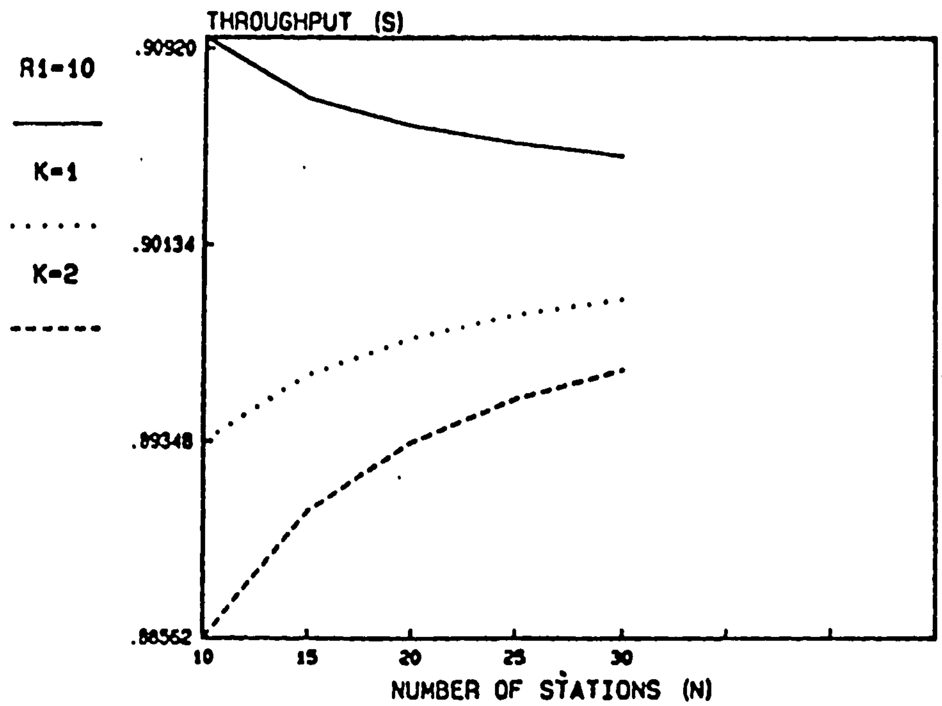

FIGURE 4.21 THROUGHPUT AS A FUNCTION OF N FOR THE CSMA/CD PROTOCOL SPOKE-RING
Lp = 100 bits DR = 10 Mbps

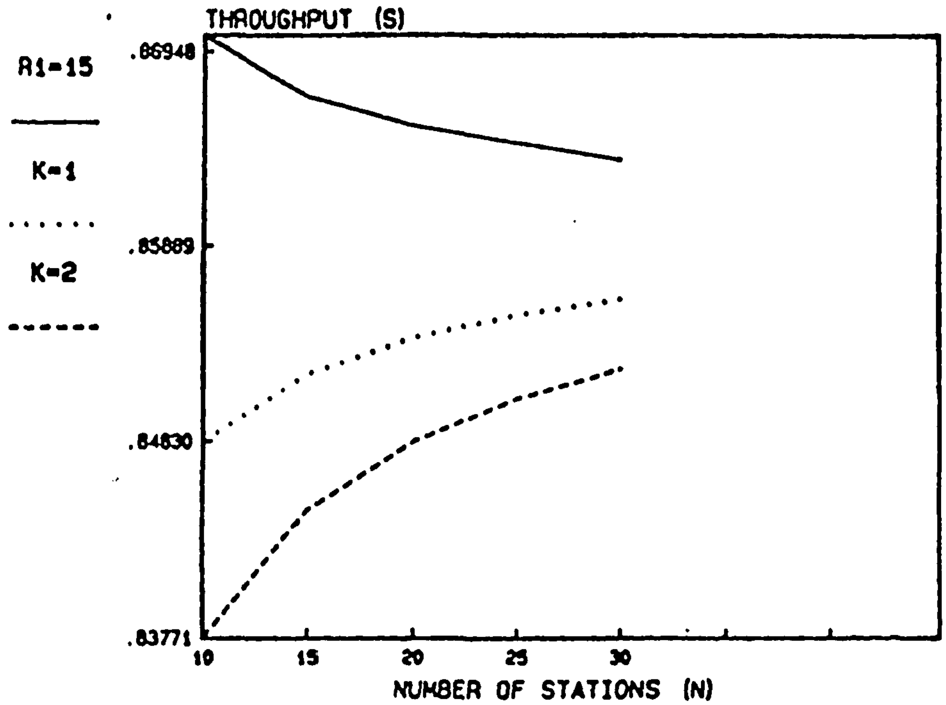

FIGURE 4.22 THROUGHPUT AS A FUNCTION OF N FOR THE CSMA/CD PROTOCOL SPOKE-RING
Lp = 100 bits DR = 10 Mbps

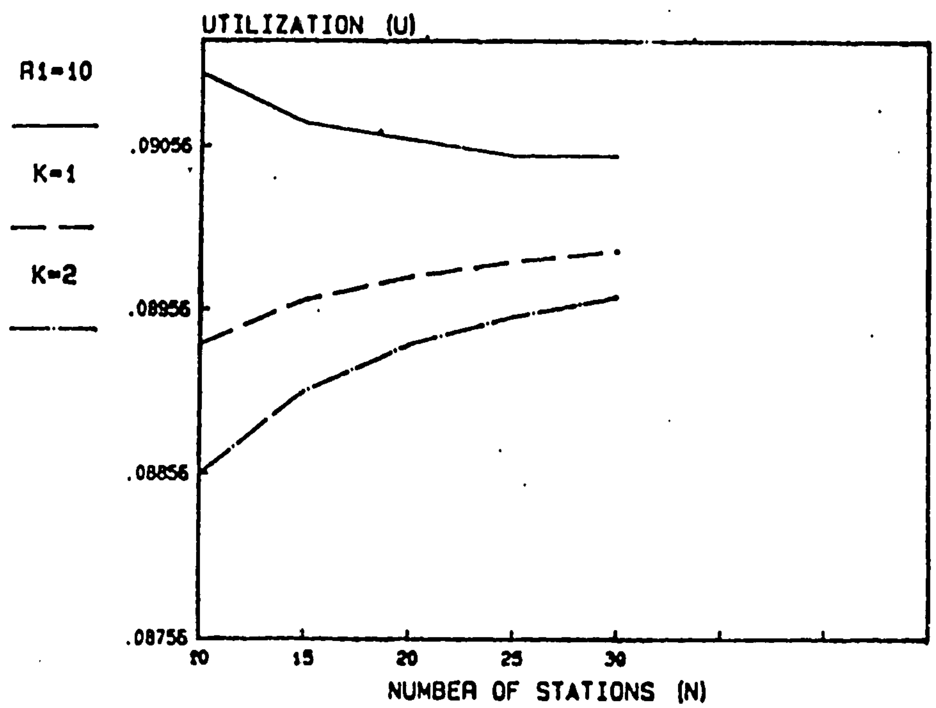

FIGURE 4.23 UTILIZATION AS A FUNCTION OF N FOR THE CSMA/CD PROTOCOL SPOKE-RING
Lp = 100 bits DR = 10 Mbps

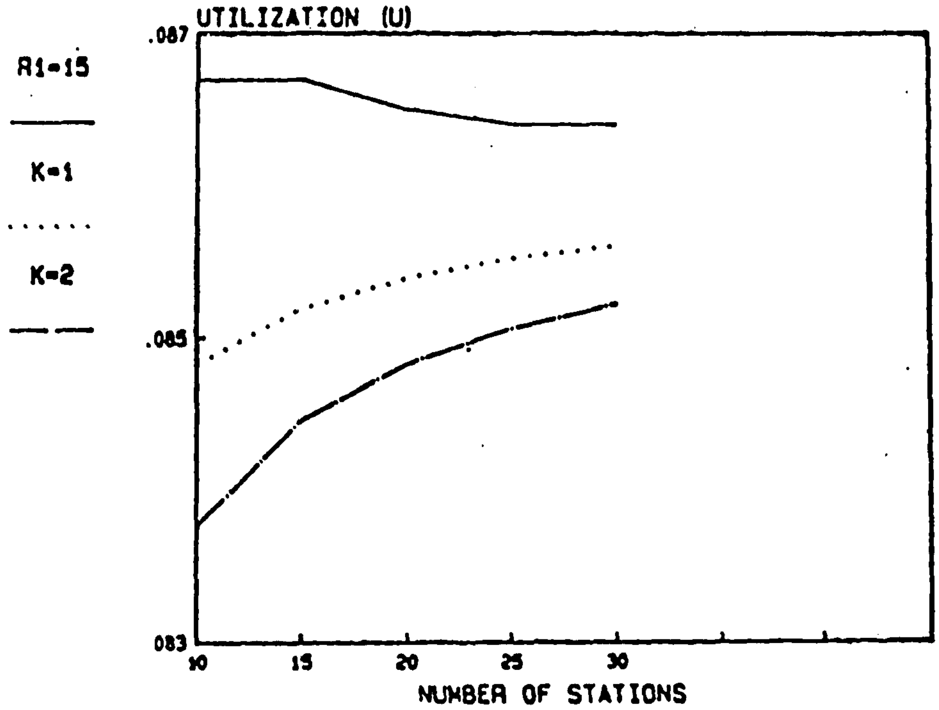

FIGURE 4.24 UTILIZATION AS A FUNCTION OF N FOR THE CSMA/CD PROTOCOL SPOKED-RING
Lp = 100 bits DR = 10 Mbps

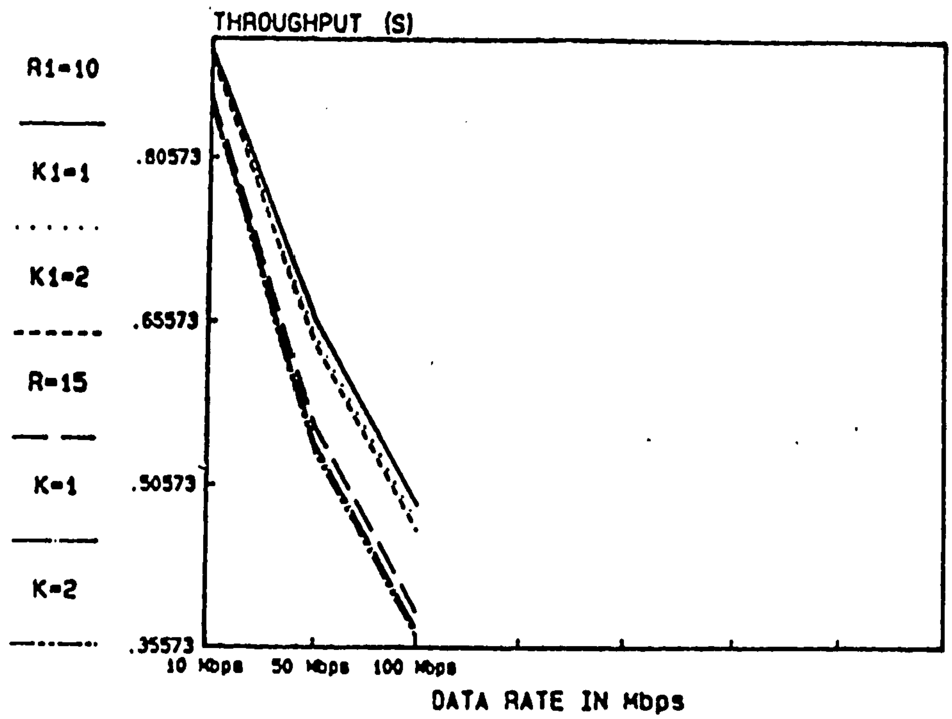

FIGURE 4.25 THROUGHPUT AS A FUNCTION OF N FOR THE FOR THE CSMA/CD PROTOCOL SPOKED-RING N = 30 STATIONS. Lp = 100 bits

$$D = \begin{cases} N + a_p - 1 & a_p < 1 \\ a_p N & a_p > 1 \end{cases} \quad 4.17$$

4.6.2 CSMA/CD PROTOCOL

a. THROUGHPUT

The throughput for the spoked-ring as compared to equation 3.25 will be

$$S_p = \frac{1}{[1 + 2a_p(\frac{1-A}{A})]} \quad 4.18$$

b. CHANNEL UTILIZATION

The channel utilization compared with equation 3.27 will be

$$U_p = \frac{1}{D_r[1 + 2a_p(\frac{1-A}{A})]} \quad 4.19$$

4.6.3 SLOTTED-RING PROTOCOL

From the results (equation 3.32), the spoked-ring delay will then be

$$D_p = (\frac{2}{1-\rho}) [\frac{L_H + L_D}{L_D}] E\{x\} + \tau_p/2 \quad 4.20$$

where

$$\tau_p = \tau_1 [1 + \frac{d}{\pi R_1} - \frac{K}{N}]$$

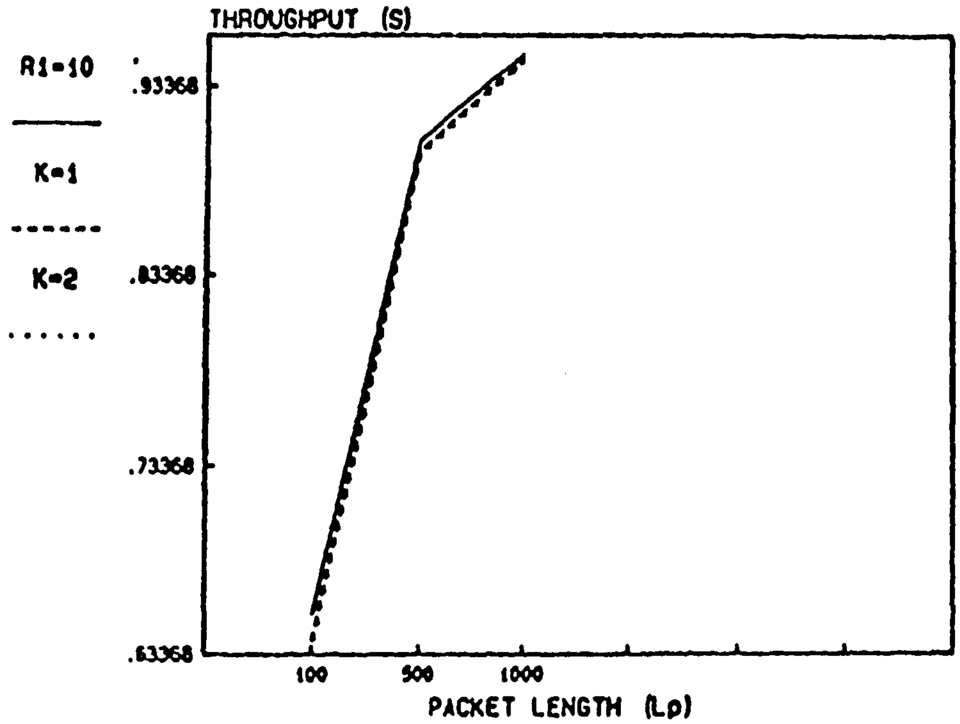

FIGURE 4.26 THROUGHPUT AS A FUNCTION OF PACKET LENGTH FOR THE CSMA/CD PROTOCOL SPOKED-RING N = 30 STATIONS DR = 50 Mbp

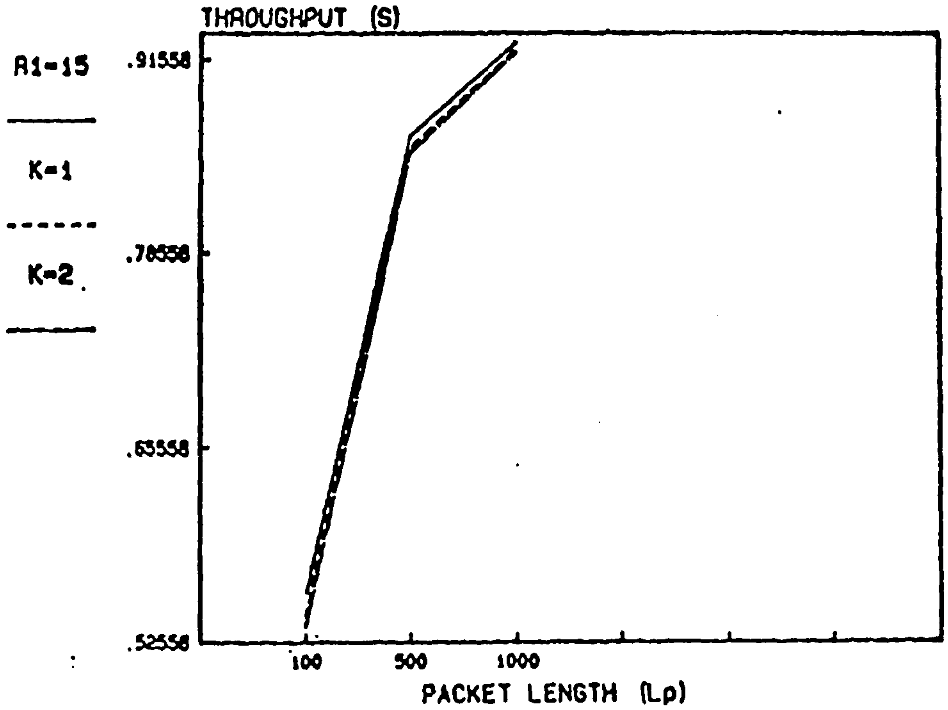

FIGURE 4.27 THROUGHPUT AS A FUNCTION OF PACKET LENGTH FOR THE CSMA/CD PROTOCOL SPOKED-RING N = 30 STATIONS DR = 50 Mbps

For $d = \frac{KL}{N} + \frac{L}{2N}$, τ_p reduces to

$$\tau_p = \tau_1 \left[1 + \left(\frac{K + 1}{N}\right)\right] \qquad 4.21$$

4.7 ANALYSIS OF RESULTS

4.7.1 SCANTIME ANALYSIS

From equations 4.3 and 4.6, the scantime for the spoked-ring is found to be

$$S_p = \frac{2\pi R}{B(1-\rho)}\left[1 + \frac{K+1}{N}\right]$$

when compared to the single-ring network, we find that the change in the scantime is given by

$$\begin{aligned} \Delta S &= S_p - S \\ &= \frac{2\pi R}{B(1-\rho)}\left[1 + \frac{K+1}{N} - 1\right] \\ &= S\left(\frac{K+1}{N}\right) \qquad 4.22 \end{aligned}$$

As we can see, ΔS approaches zero for large N.

Figures 4.7 through 4.10 show these results with respect to the number of stations N on the channel and the ring radius R. The performance of the spoked-ring as expected approaches that of the single-ring with increased number of stations. With the channel delay equal to one half the scantime the conclusion is same as for the scantime analysis.

4.7.2 NETWORK EFFICIENCY UNDER HEAVY LOAD

The network efficiency under heavy load for the spoked-ring is given as

$$E_p = \frac{qNB}{2\pi\mu R}\left[\frac{1}{1+\frac{K+1}{N}}\right]$$

The change in efficiency between the single-ring and the spoked-ring is found to be

$$\Delta E = E - E_p$$

$$= \frac{qNB}{2\pi\mu R}\left[\frac{1}{1+\frac{K+1}{N}}\right]$$

$$= E\left[\frac{K+1}{N+K+1}\right] \qquad 4.23$$

we can easily see that for large N,ΔE approaches zero.

4.7.3 PROTOCOLS PERFORMANCE EVALUATION

a. TOKEN-PASSING PROTOCOL

The results for the same type of analysis performed in Chapter Three are presented in Figures 4.11 through 4.18. The results show that the spoked-ring performance closely approximates that of the single-ring. This is more so with the increase in the number of stations N on the ring. These results are of cause for a less than 1. For "a" greater than 1, it can be easily shown (as in Chapter Three) that the performance of the spoked-ring drops significantly. The

change in throughput for a greater than 1 is given as

$$\Delta S = S - S_p$$

$$= \frac{1}{a_1(1 + 1/N)} - \frac{1}{a_p(1 + 1/N)}$$

where

$$a_p = a_1(1 + \frac{K+1}{N})$$

ΔS becomes

$$\Delta S = \frac{S}{[1 + (K+1)/N]} \qquad 4.24$$

Equation 4.24 approaches zero for large N. This analysis is true for the channel utilization also. The change in delay however is obtained from

$$\Delta D_p = D_p - D$$

$$= a_1(1 + \frac{K+1}{N})N - a_1 N$$

$$\Delta D_p = (K + 1)a_1 \qquad 4.25$$

b. CSMA/CD PROTOCOL

As in Chapter Three, the performance of the spoked-ring is slightly less than that of the single-ring. The results are shown on Figures 4.19 through 4.26. The change in throughput for "a" greater than 1 is given by

$$\Delta S = S - S_p$$

$$\Delta S = \frac{1}{[1 + 2a_1(\frac{1-A}{A})]} - \frac{1}{[1 + 2a_1\,(1 + \frac{K+1}{N})(\frac{1-A}{A})]}$$

$$= \frac{(K + 1)/N}{[\frac{A}{2a_1(1-A)} + 1]\,[\frac{A}{2a_1(1-A)} + (1 + \frac{K+1}{N})]} \qquad 4.26$$

This value reduces to zero for large N.

c. SLOTTED-RING PROTOCOL

As shown in section 3.5.3c the change in the delay here is related to the propagation delay and is given by

$$\tau_p = \tau_1(1 + \frac{K+1}{N})$$

The change in the delay

ΔD is $\qquad K_1 + \tau_p - k_1 - \tau_1$

which yields

$$\Delta D = \tau_1(1 + \frac{K+1}{N} - 1)$$

$$= (\frac{K+1}{N})\tau_1$$

This approaches zero as N becomes large.

CHAPTER FIVE

DUAL ACCESS BI-RING NETWORK

(DABNET)

5.1 INTRODUCTION

While the double-ring and the spoked-ring networks address the issue of fail-safe networking, their utility depends on the frequency of breakdown and/or expansion of the network. For a low failure rate (which includes low expansion rate), the added reliability may be costly in terms of wasted bandwidth. This is true especially for the double-ring network where the inner loop is used only during the network reconfiguration in the event of a station breakdown or expansion. For the spoked-ring the spokes are activated only when there is a station failure. The network operation is limited by the combination of failures. As long as the number of consecutive station failures is limited to K (the number of stations between any spoke-connected stations), the network works fine. However, if for example, station number N is transmitting information to all stations, a breakdown in station N+K results in the skipping of station N+K+1. With a breakdown of station N+1 and N+K+1, the network ceases to function.

If the network grows moderately, we observe that both the double-ring and the spoked-ring have limited flexibility. For the spoked-ring, growth areas could be anticipated and incorporated in the design of the network before installation. This implies that some costly equipment will remain unused until stations are installed at

these locations. The spokes at the anticipated locations would not be activated in the absence of terminal equipment to implement spoke-activation. This approach will limit the connectivity of the installed network when a node failure occurs. Another approach would be to rearrange the spokes everytime a new node is added to the network. This method is impractical since every new arrangement essentially represents a dismantling and rebuilding of the spoked-ring network.

The double-ring network, however, has an advantage over the spoked-ring in this regard since it can adapt to a working network in the event of a station failure or expansion. The design of the double-ring allows it to sustain any number of consecutive failures and still maintain an operational network where the remaining stations can communicate with each other. Any other combination of failures results in the fragmentation of the double-ring network where partial operation of the network is feasible. This mechanism would depend on the protocol in use for a particular network. The double-ring is limited during expansion to the circumference of the outer ring since the inner ring is introduced solely for the purpose of network reconfiguration during node failures and or expansion.

These inherent problems (flexibility of network expansion and waste of bandwidth), can be addressed by a dual access bi-ring network (DABNET) shown in Figure 5.1.

5.2 OPERATION

The DABNET in figure 5.1 is made up of two independent ring networks connected by means of gateways. Under normal operations both

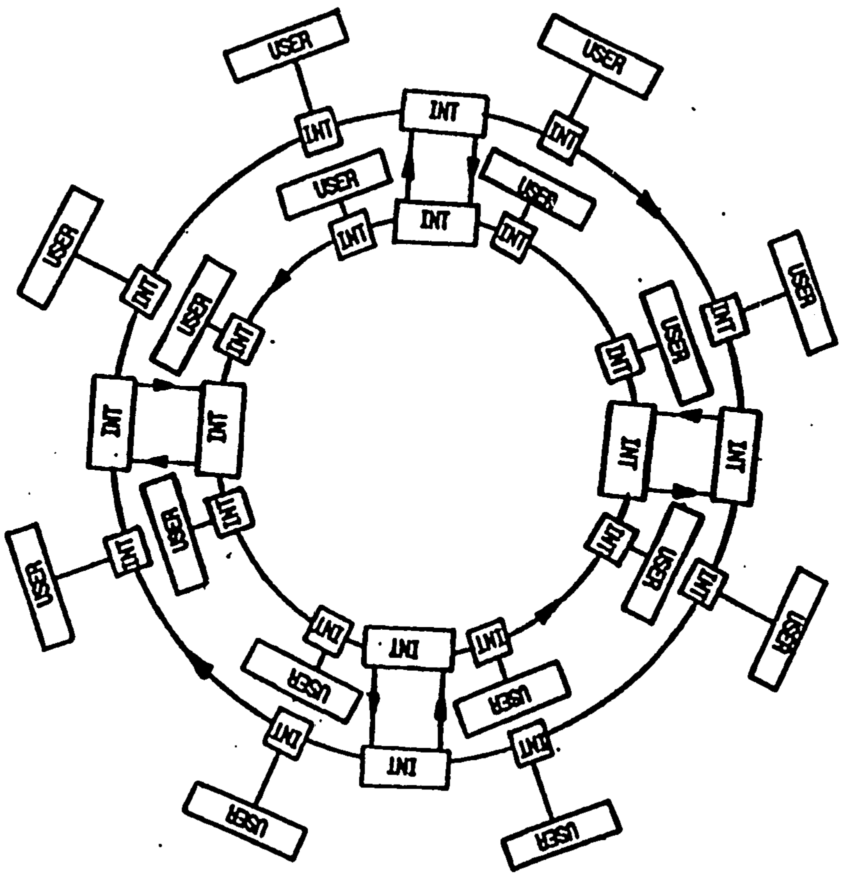

FIGURE 5.1 THE DABNET CONFIGURATION

rings are considered to be independent. The radii of the inner and outer rings are completely arbitrary. Transmission and communication is primarily between stations on the same ring. The station numbering on both rings is also arbitrary. This implies that stations with the same number assignment on both rings can communicate if they so desire. When a station breakdown on any ring or communication between stations on both rings is desired, the gateway nearest to the transmitting station is activated which results in a new ring that incorporates stations on both rings for an inter-ring communication. We therefore achieve total utilization of the inner ring, which is in contrast to the standard double-ring network by reconfiguring the network via gateways. Furthermore, the problem of expansion associated with the spoked-ring does not exist.

5.3 CONNECTIVITY

The DABNET concept is not to connect all stations on the outer ring to the stations in the inner ring via gateways. Such a connection would require that the number of stations on both be equal and the gateway connections to be 2N where N is the number of stations on each ring. As the number of stations increase, we observe that the number of gateways become prohibitively large and expensive. Too few a number of gateways, however, would mean that a large segment of stations will be isolated during a breakdown of more than one station. The proposal for the DABNET is to divide the number of stations on the outer ring by some factor such that the maximum number of stations that can be isolated during any combination of breakdowns would be at an acceptable level. The acceptable level, however, depends on the user

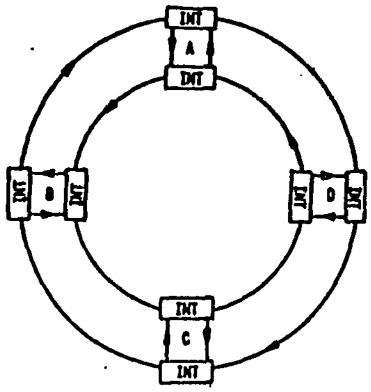

FIGURE 5.2 THE DABNET INTERFACE

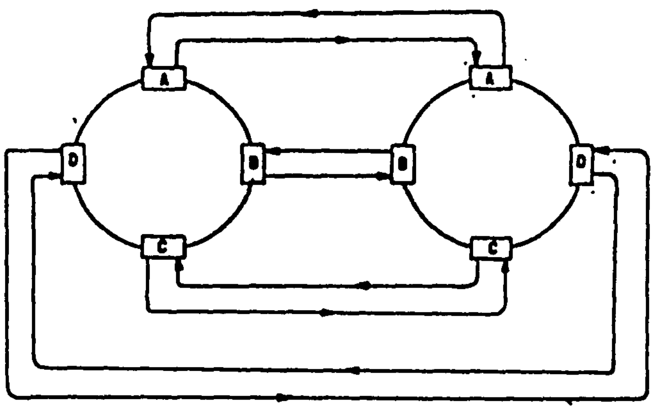

FIGURE 5.3 THE DABNET LOGICAL CONFIGURATION

requirements. For example, if the outer ring has thirty-five stations, it would be better to have between five and seven gateways rather than two gateways where a greater number of stations will be isolated.

5.4 PROTOCOLS

The issues to be addressed by the DABNET protocol are the following:

1. How does a station know where its message is coming from?
2. What scantime to use in determing the network operation?
3. Is prioritization of messages allowed? If so how can it be accomplished?
4. Is network reconfiguration under a breakdown the same as network reconfiguration without a station breakdown?
5. How does a station determine which gateway to use?

To answer these questions, we present the following protocol for the operation of the network.

Since transmission can be between stations on the same ring or across rings, it is necessary to have tokens (for the token-passing protocol) that both rings identify and which are not duplicated as messages by either ring. Within the token, a bit position will be dedicated to supply the information needed to know which ring the communication is intended for. Since we are dealing with only two rings, the representation would be 0 for same ring transmission and 1 for transmission on the other ring. Thus at the node interface, each

station determines if communication is between stations on the same ring or between rings. A communication between stations on the same ring would require the transmitting station to transmit its message without any change to the token. In this way, we establish a zero at the designated bit location on the token to maintain the default transmission between stations to be on the same ring. If the communication is between stations across rings, then the transmitting station changes the designated bit position on the token to 1 and it will be so recognized at the node interface during transmission. Under normal operation where transmission is between stations on the same ring, the scantime used will be the scantime associated with the ring. This information is automatically obtained during transmission. If a transmitting station does not receive an acknowledgement within this period of time, it retransmits its message by changing the designated bit position value from 0 to 1. An ambiguity arises, however, in this case since the numbering of stations on both rings is arbitrary and the message may not be received by the intended station. To resolve such conflicts, the next bit to the right of the designated bit position is used to convey this information. Using the same format for transmission between stations on the same ring or different rings, 0 will indicate transmission to a station on the same ring via the second ring, while 1 would mean transmission to a station on the second ring. During such a transmission, the second bit position is used to determine if the network reconfiguration is due to ring station breakdown or communication between stations on both rings. Prioritization and transmission between stations on both rings will be classified as the same. During such operations, priority is assigned

to the transmitting station. If a station on the second ring was transmitting at the time, it will terminate its transmission as soon as it receives the new message and attach its station number to the message so that control can be returned to it after transmission from the prioritized station. However, if transmission through the reconfigured network is due to the breakdown of a station or the ring, then priority is assigned to the station on the unbroken ring that was transmitting at the time. This information will be obtained from the designated two bits positions in the token. Since the first bit position indicates transmission between rings or same ring and the second bit position indicates transmission to stations on same ring or different ring, 10 will mean that a ring breakdown had occurred. In this case, a station on the broken ring is now retransmitting the message to a station on the same ring via the second ring. On the other hand, a deliberate transmitting of information between stations on both rings will require that these bit positions contain the value 11. In any case, the gateway used is assumed to be the first gateway from the transmitting station that is in the direction of signal flow. We will now summarize the DABNET protocol as follows:

1. Both rings will have different tokens recognizable by each ring and not duplicated in the form of data by either ring.
2. Two bit positions on the tokens will be designated to determine the mode of transmission on the rings. These positions will be the same on both rings.
3. The first designated bit position will indicate whether transmission is across rings or on the same ring. A "0"

represents same ring transmission and a "1" represents bi-ring transmission.

4. The second designated bit position will indicate whether transmission is between stations on the same ring or stations on both rings. Thus, a "00" bit representation in these locations will represent transmission between stations on the same ring. A "10" bit representation means that transmission is between stations on the same ring but through the second ring or the reconfigured ring. The "11" bit representation indicates the communication between stations on both rings. This is a priority/tariff situation and it does not matter whether the ring is reconfigured due to network failure or simple communication between stations under normal operations. The bit combination "01" cannot occur and will be recognized as an error. The reset value will be "00". A "01" bit representation means that a station on the first ring wants to communicate with another station on the second ring using the first ring.
5. When the designated bit positions indicate a 10, a breakdown has occurred on the transmitting station's ring. This transmission is completed through the reconfigured ring when there is no transmission by any station on the unbroken ring.
6. If there was an ongoing transmission on the unbroken ring at the time, the transmitting station stops transmission and a time out equal to the sum of the scantime of both rings is initiated. During this period, the station on the damaged ring recognizes the problem and stops transmission.

7. Priority under these conditions, is given to the station that was transmitting on the unbroken ring. So a new token which in this case is the same as that of the unbroken ring is generated by this station and passed on in a round robin fashion. All stations on this new ring will now use the same token.
8. After repairs on the broken ring, a token is generated which will then initiate the restructuring of the rings to individual rings.
9. When the designated bits indicate 11, communication is intended to be between stations on both rings. If the second ring is in the idle state (no transmission) the transmission goes through.
10. If the second ring is in the busy state, the station transmitting at that time ceases to transmit and priority is assigned to the station wishing to transmit to a station on the other ring.
11. At the end of the communication, the rings are disengaged by changing the designated bit positions to 00. This is the reset value.

5.5 AN EXAMPLE OF DABNET PROTOCOL IMPLEMENTATION

Figure 5.4 is an example of the DABNET with eight stations on the outer ring and eight stations on the inner ring. Let us assume now that the tokens for the outer ring and the inner ring are 11100111 and

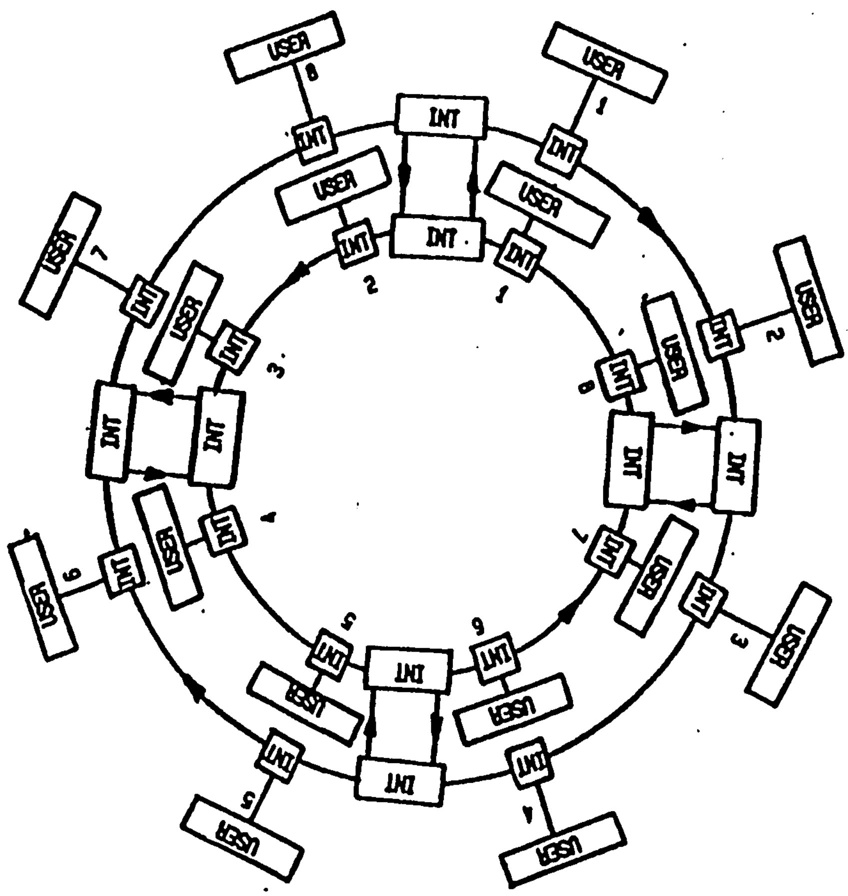

FIGURE 5.4 THE DABNET WITH 8 STATIONS ON EACH RING

10100101 respectively. The bit positions with two consecutive zeros indicate the designated bit positions where information about the mode of transmission is derived. For these tokens, the bit positions have been arbitrarily chosen to be four and five. It is important that the designated bit position be the same for all rings connected to one another in this fashion. We are also assuming here that tokens of both rings cannot be duplicated. Therefore any similar message or interrupt signal will have to be stuffed. There are techniques used to solve this problem and will not be further considered in our exposition. Let us now create different transmission modes and work through the protocol to determine the network response.

CONDITION 1: Stations 1 and 5 on the outer ring are communicating while stations 1 and 5 on the inner ring are doing the same.

Since this case examplifies the transmission between stations on the same ring, the tokens will remain the same. It is interesting to note that the permissible bit combinations at the designated bit positions cannot be duplicated by any form of message on both rings. This additional condition ensures that errors introduced during transmission will not redundantly trigger false alarms and reconfigurations of the network when no failure has actually occurred. Therefore for the condition we have just considered, normal operation implies that the fault detection time out is equal to the scantime of the individual rings.

CONDITION 2: Same as Condition 1 except that ring number 1 is broken between stations 4 and 5 in a manner that has not effected the nearest gateways.

Since the information being transmitted to station 5 on the outer ring cannot be received, station 1 waits for a time out period equal to the ring scantime. It realizes there is a problem and proceeds to retransmit the message using the inner ring. So its token now changes to 11110111, which means transmission to a station on the same ring via the second ring. The network reconfiguration is initiated by the first gateway in the direction of transmission which in this case is right after station number 4. The new token is then recognized by station 1 on the inner ring which stops transmission. Meanwhile due to the reconfiguration, the token in use by the inner ring now gets to station 1 of the inner ring only after it has gone through the station 1 of the outer ring and back through the gateway after station 4 of the outer ring. While going through station 1 of the outer ring, the station recognizes that the inner ring was busy at the time of the reconfiguration and stops transmission immediately. Both stations wait for a time out period equal to the sum of the scantime of the two rings. Now since this network reconfiguration is due to a break or station failure on the outer ring, priority of transmission is assigned to the station that was transmitting in the inner ring. For this example, it is station number 1. So it generates its token which is 10100101 and reinitiates transmission. This token is now the new ring's token. The ring is the reconfigured ring and all stations attached to it recognize this token. After transmission, the token is advanced to station number 2 in the inner ring and this process is continued until the token gets to station number 1 of the outer ring. Before transmission, station 1 now changes the first position bit among the designated bits to 1. Thus the token now becomes 10110101, meaning

transmission through the inner ring to a station on the same ring with the transmitting station. Thus, the station that will receive this message is not station 5 of the inner ring but station 5 of the outer ring. Once station 1 is through transmitting, it changes the token back to 10100101 before forwarding it to the next station.

CONDITION 3: The inner ring was in the idle state (no transmission on the ring) when a break occurred on the outer ring. Communication is still between stations 1 and 5 and the cut on the ring is between stations 4 and 5.

In this case, after waiting for a time out period equal to the scantime of the outer ring, station 1 retransmits by changing its token to 11110111 which means transmission through the second (inner) ring to a station on the outer ring. Since there was no transmission in the inner ring, there will be no disruption of service in this case and the message will go through. However stations on the second (inner) ring now recognize the new token and by default have assigned priority to the stations on the outer ring. The new token for the reconfigured ring is the token for the outer ring (11100111). A transmission between stations on the inner ring will mean using the token without any modification of the designated bit positions. This approach ensures that the modification of the token under these circumstances (conditions 2 and 3) is done by the station on the damaged ring regardless of the token being used. We therefore maintain consistency in the protocol format. The network breakdown is not limited to the outer ring. The reverse of the above description of the protocol is true when the failure is in the inner ring rather than the outer ring. The network under conditions 3 and 4 can be restored to the original

form (two independent rings) after repairs have been made. Once the disabled ring or station is repaired, the gateway is deactivated by the first station to the left of the gateway. In the above example, station 4 on the outer ring deactivates the gateway by generating the outer ring's token as soon as it is directly connected to station 5 of the outer ring. This token will be 11100111 and it will automatically disengage both rings. The new token now circulates through the outer ring and is taken only by the station on the outer ring that was transmitting at the time. There will be no ambiguity here since before the disengagement of the rings, all stations on the reconfigured ring knew if the ring was busy or idle. From this information, the stations also know who was transmitting at the time. Therefore the order or sequence of equal access to the token has not been destroyed. The inner ring, seeing that its original link to the outer ring has been cut, generates its token immediately. This is achieved in this example by station 6 on the inner ring generating the token which now circulates only in the inner loop. Both rings wait for a time out equal to the scan time of the individual rings and proceed with normal transmission. The scantime reset is established by the circulating tokens after generation since each station resets its scantime as soon as it encounters the generated tokens.

CONDITION 4: Communication between stations on both rings without a break or failure on either ring while there is an ongoing transmission on the ring with the intended recipient of the inter-ring communication. We shall assume that station 1 on the outer ring wants to communicate with station 5 while stations 1 and 5 in the inner ring are communicating.

In this situation, station 1 on the outer ring changes its token to the pattern 11111111 and then transmits the information. The gateway between stations 4 and 5 on the outer ring is activated and station 1 in the inner ring recognizes that this is a priority transmission and stops transmission. A time out equal to the scantime of both rings is initiated during this process by all stations to clear the reconfigured ring of all outstanding information on the ring. Station 1 on the outer ring now communicates with station 5 on the inner ring. At the end of the transmission, station 1 on the outer ring resets the token and forwards it to the next station. The resetting of the token disables the gateways and the rings become independent again. However the regenerated token by station 6 in the inner ring is passed on station 1 of the inner ring which immediately resumes transmission between it and station 5 as it existed before the interruption. This approach is maintained for any number of station combination that the transmitting station wishes to communicate with on both rings.

CONDITION 5: Same as condition 4 except now the inner ring is in the ideal state. Here there is no obstruction and only a reset of the time out to the sum of the scantimes of both rings is required. At the end of the communication the same procedure as in condition 4 is used to restore the rings independence from each other.

CONDITION 6: Same as conditions 2 and 3 but with an additional failure on either ring.

In this instance, which ever token that was in circulation is used to activate the necessary gateways and then passed on to the transmitting station. Communication protocol between stations just as

in conditions 2 and 3 is followed and the network is restored to its original state as described above.

5.6 FURTHER ISSUES

One of the many issues facing the DABNET designer is that of the number of gateways the network must have. While this issue is user dependent, it is necessary to understand the nature of the problem. As we have seen in the example in section 5.5, it is possible not to have a gateway in the event of a network breakdown. Assuming the same transmission between stations 1 and 5 on the outer ring, a breakdown on any of the stations 2, 3 and 4 totally disables the network reconfiguration process. This shows that the best connection will be to have two rings with equal number of stations and have gateways between two opposite stations on both rings. This will result in total access to the reconfiguration process where each transmitting station uses its gateway to activate network reconfiguration. This approach works well as long as the number of stations is small. As the networks grow (the individual ring networks), it becomes very expensive to practice the one-to-one connection between stations. In most cases, the growth of the network will not be the same, since their use could be prioritized. As an example, in a university setting, the outer ring could be the local area network connecting all the different academic entities such as Engineering, Business, Science, Music and law for the purpose of coordinating common interests shared by these colleges and schools while the inner ring is the local area network connecting the different administrative areas such as finance, academic, chancellor, President and student government. These networks as we can observe,

will not grow at the same rate since the outer ring would be primarily a function of the number of professors wishing to have or already having a common area of interest with their peers in other areas. With the introduction of more colleges, we can observe a rapid growth in the number of stations on the outer ring. This will not be the case for the inner ring which more or less remains the same except when new administrative positions are created. In any case, network reconfiguration will be activated more frequently by the inner ring than the outer ring since the administration will from time to time send information down to the lower levels in event of policy changes and to further inform the faculty on new research opportunities. It will however be impossible to connect both networks on station to station basis.

One way to approach this problem will be to look at the number of stations on each ring and come up with a relative size relationship. Let the number of stations on the outer ring and inner ring be N and M respectively. Then the ratio N/M establishes the number of stations that can be connected to each other. Thus for N=40 and M=20, the optimum station connection will be N/M=2. This implies that for every two stations on the outer ring, there should be a gateway to a station in the inner ring. The above example has been chosen just to simplify the calculation. N and M could be a combination of odd and even numbers. In this case we assume the nearest integer to the value of the ratio of N/M to be the optimum connection needed. For an example, N=40 and M=13; N/M = 13 1/13 which means that twelve of the thirteen inner ring stations will be connected to every third station on the outer ring and the thirteenth station to four stations. Any suboptimal

solution as to the number of connections or gateways can be obtained simply by dividing optimum number by the fraction of "success" attainable by the user. Using the above examples, for N=40 and M=20, a 50% connectivity would mean dividing the optimum number (2) by 0.5 which results in a connection of 4 stations to 1 in the inner ring. A 75% connectivity will be 3 to 1 connection. Similar results can be obtained for any combination of N and M.

The efficiency of the network with respect to the network connectivity can be improved in a fiber optical realization by using fail-safe nodes at the station locations. This approach always provides for a bypass whenever a station fails. This means that a minimum connection between rings can also achieve the same results as would be obtained from N=M case which is the one to one connection.

Another issue is with respect to the cluster of rings that can be obtained by activating selected gateways. These new rings will be a combination of stations from both rings. It turns out that if two stations from both rings are within the same segment, (bounded by same gateways) it will be shorter to circulate the information through that segment of the ring. The issue here will be that of the complexity of the protocol to be employed. Such isolations are generally possible when the network reconfiguration is due to a breakdown in two or more locations in an arbitrary fashion. In this instance the normal protocol as established in section 5.4 is followed. However if it is due to station's request which does not include network reconfiguration due to station breakdown, the complication may be too great. It will be left up to the software engineer to address this issue. A third issue is that of expandability. How many rings can be connected in

this fashion. As shown in figures 5.2 and 5.3, logical ring connections can be obtained. The question is how many? To have a feel for the issue, one has to look at the connection between the number of rings and the number of obtainable tokens. There are 2^8 combinations of tokens available. Out of these, certain combinations will be designated for communication between stations on different rings and its not feasible to obtain that many rings. Even when we say that we could have 2^8 rings, they have to by constraint operate independently without any choice of inter-ring connection since a change in any bit results in a token for any other ring. Removing that number of bit combinations for the data and control field would require a very complicated system that would be almost infeasible to implement. Nevertheless, the concept of a multiple access multiple ring network (MAMNET) may have applications in certain situations. If the rings are few in number, connectivity and protocols might be obtained without too great a difficulty.

CHAPTER SIX

CONCLUSION

We first compare single access double-ring and spoked-ring networks.

6.1 SCANTIME ANALYSIS COMPARISON

From the analysis of the results in Chapters three and four it is shown that the change in scantime for the double-ring network is given by

$$\Delta S = S[(1 - 1/N)]$$

whereas for the spoked-ring change in scantime is found to be

$$\Delta S = S\left[\frac{K+1}{N}\right]$$

As the number of stations N on the rings increase, the change in scan time changes to

$$\Delta S = S$$

for the double-ring, whereas it approaches zero for the spoked-ring. This shows that the spoked-ring is superior to the double-ring network in scantime performance. This is also true for the channel-acquisition delay which is one-half the scantime.

6.2 NETWORK EFFICIENCY COMPARISON

The change in efficiency for the double-ring is found to be

$$\Delta E = E\left[1 - \frac{1}{[(1 - 1/N) + 1]}\right]$$

while for the spoked-ring,

$$\Delta E = E \left[\frac{K + 1}{N + K + 1} \right]$$

The common parameter to both efficiency changes is N, the number of stations on the ring. As N increases, ΔE approaches zero in the case of the spoked-ring while ΔE approaches

$$\Delta E = E \left[1 - \frac{1}{[1 + R_1/R_1]} \right]$$

$$= \frac{E}{2}$$

showing a superior performance for the spoked-ring.

6.3 PROTOCOL PERFORMANCE COMPARISON

In all three protocol analysis performed, the spoked-ring performance is superior to the double-ring network. This however may not be a good indication of both networks performance since a key element in this analysis is that the spoke activation time has been considered to be very negligible. It is assumed in this analysis that the spoke-activation time and the double-ring reconfiguration are very negligible. For the spoked-ring, the proposed spoke activation time is given in equation 4.1c as

$$T_{SA} = \frac{L}{2NB} [2K + 3]$$

$$= \frac{\pi R}{NB} [2K + 3] \text{ sec.}$$

This value depends on the data rate, radius of the ring and the number

of stations connected to the ring for any given configuration. The double-ring, if optically coupled, will have a reconfiguration time equal to the time taken to detect failure and the time needed to eliminate the failed node or station from the ring. If the delay associated with the double-ring reconfiguration time is not comparable to the delay associated to the spoked-ring activation time delay, the protocol analysis may show a break even point where each design out performs the other. However, this aspect of the analysis depends on the switching technology which is changing rapidly.

6.4 COMPARISON OF THE DABNET TO THE DOUBLE-RING AND SPOKED-RING NETWORKS

The DABNET design minimizes the waste in bandwidth associated with the doubling network. Since computer networks do not breakdown very frequently, the key issue with the double-ring is that of expandability. If the rate of expansion is high, then limitation will be in the outer-ring. Once a maximum achievable size is obtained (number of stations on the outerring), the second ring becomes an expensive piece of equipment that is hardly used. The DABNET addresses this issue by connecting gateways to both rings and achieving the same rate of expansion on both rings as would be obtained from the double-ring. The scantime during network reconfiguration are similar. However, the protocol used for the DABNET is more complete. The double-ring design's primary goal is to maintain simplicity while achieving some measure of added reliability. In this regard, the DABNET's design is superior. A comparison between the DABNET and the

spoked-ring shows that the DABNET is a more general purpose network than the spoked-ring. The difficulty associated with the expandability of the spoked-ring limits it to a special purpose network. During breakdowns, the network reconfiguration and transmission is achieved after a loss of a scantime plus the spoke activation time. Thus, the time is takes to reconnect the network is less than that of the DABNET which is approximately the sum of the scantimes of the individual networks.

6.5 SUGGESTIONS FOR FURTHER RESEARCH

One of the suggestions for further research is to identify the effect of network reconfiguration time on the analysis of the protocols. A cost analysis based on network channel capacity is also suggested as a possible research area. For the DABNET, it is important to find out if there is an acceptable or optimum number of rings that can be connected to intercommunicate. It is possible to use the gateways as divisions or segmentations of the rings and during prioritization, the network can activate the necessary gates and transmit information over shorter distances. This approach may improve the efficiency of the network. The question of how hardware redundancy could be put to use for other computer networks also remains a significant open problem.

APPENDIX

```
:    ****************************************************************
:    SIMULATION OF RING NETWORKS USING DOUBLE RING AND SPOKE TOPOLOGIES
:    THE SIMULATION LOOKS AT VARIOUS PARAMETER RESPONSES. THE DOUBLE
:    RING NETWORK PARAMETERS ARE FOLLOWED BY THE SPOKED-RING PARAMETERS
:    AND THE NETWORKS PERFORMANCES ARE COMPARED.
:    ****************************************************************
      DIMENSION SP(100),SDP(100),SDAP(100),SB(100),SDB(100),SDAB(100)
      DIMENSION SR2(100),SDR2(100),SDAR2(100),SR1(100),SDAR1(100)
      DIMENSION SN(100),SDN(100),SDAN(100),SDR1(100),ESB(100)
      DIMENSION ESR1(100),EU(100),EDU(100),DK(100),EQ(100),EDQ(100)
      DIMENSION ESQ(100),TSAK(100),ESNK(100),SPP(100),SPNK(100)
      DIMENSION SPR1(100),SPB(100),SPN(100),ESN(100),TSAB(100)
      PI = 3.142857143
      DO 10 N =5,25,5
      DO 60 NR = 5,15,5
      DO 70 NK = 1,2
      DO 80 IQ = 5,15,5
      DO 90 IU = 5,20,5
      DO 20 IR1 = 10,15,5
      XL = 2* PI*IR1
      DO 30 IR2 = 5,15,5
      DO 40 IB = 100,200,50
      DO 50 K = 1,9,2
      P = K/10.0
      SP(K) = (2*PI*IR1)/((1-P)*IB)
      DP = SP(K)/2
1     FORMAT ( 1X,'SCAN TIME = ', 2F10.5)
      SDP(K) = (2*PI*((1-1.0/N)*IR1 + IR2))/((1-P)*IB)
      DDP = SDP(K)/2
2     FORMAT ( 1X,'DOUBLE-RING SCANTIME = ',2F10.5)
      SDAP(K) = (2*PI*(IR1 + IR2))/((1-P)*IB)
      DDAP = SDAP(K)/2
3     FORMAT ( 1X,'DOUBLE-RING APPROX SCANTIME = ',2F10.5)
50    CONTINUE
      SB(IB) = (2*PI*IR1)/((1-P)*IB)
      TSAB(IB) = (L*(2*NK+3.0)*IB)/(2*IB*N)
      WRITE(6,4) SB(IB),TSAB(IB)
4     FORMAT (1X,'PROPAGATION SPEED SCANTIME =',2F10.5)
      SDB(IB) = (2*PI*((1-1.0/N)*IR1+IR2))/((1-P)*IB)
5     FORMAT ('DOUBLE-RING PROP. SPEED SCANTIME =',F10.5)
      SDAB(IB) = (2*PI*(IR1 + IR2))/((1-P)*IB)
      ESB(IB) = (IQ*N**2.0*IB)/(IU*2*PI*IR1*(N+NK+1.0))
      WRITE(6,6) SDAB(IB),ESB(IB)
6     FORMAT ('DOUB.RING APPX SCANTIME = ',2F10.5)
40    CONTINUE
      SR2(IR2) = (2*PI*IR1)/((1-P)*IB)
      WRITE(6,7) SR2(IR2)
7     FORMAT (1X,'SCANTIME DUE TO R2 = ',F10.5)
      SDR2(IR2) = (2*PI*((1-1.0/N)*IR1 + IR2))/((1-P)*IB)
8     FORMAT (1X,'DOUB.RING SCANTIME DUE TO R2 = ',F10.5)
      SDAR2(IR2) = (2*PI*(IR1 + IR2))/((1-P)*IB)
9     FORMAT (1X,'DOUB.RING APPX SCANTIME DUE TO R2 = ',F10.5)
30    CONTINUE
      SR1(IR1) = (2*PI*IR1)/((1-P)*IB)
      WRITE(6,11) SR1(IR1)
```

```
 20   CONTINUE
      EU(IU) = (IQ * N * IB)/(2*PI*IR1*IU)
      WRITE(6,17) EU(IU)
 17   FORMAT(2X, ' SINGLE RING EFFICIENCY DUE TO Q = ',F10.5)
      EDU(IU) = (IQ*N*IB)/(2*PI*IU*((1-1.0/N)*IR1+IR2))
      WRITE(6,18)EDU(IU)
 18   FORMAT(2X,'DOUB.RING EFF. DUE TO U =',F10.5)
 90   CONTINUE
      EQ(IQ) = (IQ*N*IB)/(2*PI*IR1*IU)
      WRITE(6,19) EQ(IQ)
 19   FORMAT(2X,'SING.RING EFF. DUE TO Q =',F10.5)
      EDQ(IQ) = (IQ*N*IB)/(2*PI*IU*((1-1.0/N)*IR1+IR2))
      ESQ(IQ) = (IQ*N**2.0*IB)/(IU*2*PI*IR1*(N+NK+1.0))
      WRITE(6,21) EDQ(IQ),ESQ(IQ)
 21   FORMAT(2X,'DOUB.RING EFF. DUE TO Q =',2F10.5)
 80   CONTINUE
C    ***********************************************************
C                     SPOKE - RING SIMULATION.
C    ***********************************************************
      TSAK(NK) = (XL*(2*NK+3)*NR)/(2*IB*N)
      DK(NK) = ((NK+1.0)*L)/N
      ESNK(NK) = (XL*(N**2*IB))/(IU*2*PI*IR1*(N+NK+1.0))
      WRITE(6,22) TSAK(NK),DK(NK),ESNK(NK)
 22   FORMAT(2X,'SPOKE ACTIVATION TIME DUE TO K =',3F10.5)
 70   CONTINUE
 60   CONTINUE
      SPR1(IR1) = (XL*(N*2.0*NK+1.0))/(IB*N*(1-P))
      DS = SPR1(IR1)/2.0
      SPNK(NK) = (XL*(N*2.0*NK+1.0))/(IB*N*(1-P))
      SPP(K) = (XL*(N*2.0*NK+1.0))/(IB*N*(1-P))
      DS = SPP(K)/2
      SPB(IB) = (XL*(N*2.0*NK+1.0))/(IB*N*(1-P))
      DS = SPB(IB)/2
      SDAN(N) = (2*PI*(IR1 + IR2))/((1-P)*IB)
      ESN(N) = (IQ*N**2.0*IB)/(IU*2*PI*IR1*(N+NK+1.0))
      WRITE(6,16) SDAN(N)
 16   FORMAT (1X,'DOUB.RING APPX SCANTIME DUE TO N = ',F10.5)
 10   CONTINUE
      STOP
      END
```

CSMACD2 PROGRAM

```
      DIMENSION S1(100),S2(100),U1(100),U2(100),T2(100)
      DO 200 ISAM = 500,1000,500
      N = 30
      DR = 50
      B = 200
      LP = ISAM
      PI = 3.141592654
      WRITE(6,100)
 100  FORMAT(/////,5X,' OUTPUT FOR "CSMACD2" PROGRAM: ')
      WRITE(6,110)
 110  FORMAT(/////,1X,'S1(IR2) S2(IR2) U1(IR2) U2(IR2)    A1    A2   /
     &'  IR1  IR2   T1   T2(IR2) ',//)
      AA = (1-1.0/N)**(N-1)
      DO 10 IR1 = 10,15,5
      DO 20 IR2 = 5,15,5
      A1 = (2*PI*IR1*DR)/(B*LP)
      X2 = IR2
      X1 = IR1
      A2 = A1*(1.0 + X2/X1)
      S1(IR2) = 1.0/(1 + 2* A1*(1-AA)/AA)
      S2(IR2) = 1.0/(1 + 2* A2*(1-AA)/AA)
      U1(IR2) = S1(IR2)/DR
      U2(IR2) = S2(IR2)/DR
      WRITE(6,30) S1(IR2),S2(IR2),U1(IR2),U2(IR2),A1,A2,AA,IR1,IR2
 0    FORMAT(1X,7(F7.4),2(1X,I2))
 :    SLOTED RING DELAY
      T1 = (PI*IR1)/B
      T2(IR2) = ((PI*IR2)/B)*(1.0 + X2/X1)
      WRITE(6,40) T1,T2(IR2)
 10   FORMAT(57X,2(F8.4))
 20   CONTINUE
 10   CONTINUE
 200  CONTINUE
      STOP
      END
```

SPOKRNG PROGRAM

```
      DIMENSION SP(100),UP(100),DP(100)
      DO 200 ISAM = 500,1000,500
      DO 300 JSAM = 50,100,50
      DO 400 KSAM = 10,30,5
      N = KSAM
      DR = JSAM
      LP = ISAM
      B = 200
      PI = 3.141592654
      WRITE(6,55) ISAM, JSAM, KSAM
55    FORMAT(/////,5X' OUTPUT FOR "SPOKRNG" PROGRAM:',
     & '    LP =', I5, '    DR =', I5, '    N =', I5)
      WRITE(6,50)
50    FORMAT(/////,5X,'    SP(K)      ',2X,'      UP(K)     ',2X,
     &'      DP(K)   ',5X,'       AS       ',2X,' K',2X,'IR1',///)
      DO 10 IR1 = 10,15,5
      DO 20 K = 1,2
      A1 = (2*PI*DR*IR1)/(B*LP)
      AS = A1*((1.0 + K)/N + 1)
      IF(AS.GT.1) GO TO 30
      SP(K) = 1.0/(1.0 + AS/N)
      UP(K) = SP(K)/DR
      DP(K) = N + AS -1
      GO TO 31
30    SP(K) =1.0/(AS*(1.0 + 1.0/N))
      UP(K) = SP(K)/DR
      DP(K) = AS*N
31    WRITE(6,40) SP(K),UP(K),DP(K),AS,K,IR1
40    FORMAT(3(2X,(F15.5)),2X,F15.4,2(2X,I2))
20    CONTINUE
10    CONTINUE
400   CONTINUE
300   CONTINUE
200   CONTINUE
      STOP
      END
```

BIBLIOGRAPHY

1. L. G. Roberts, "The Evolution of Packet Switching." IEEE Proceedings, vol. 66, pp. 1307-1313, November 1978.
2. L. Kleinrock, "Principles and Lessons in Packet Communication." IEEE Proceedings, vol. 66, pp. 1320-1329, November 1978.
3. F. A. Tobagi, "Multi Access Protocols in Packet Communication Systems." IEEE Transactions on Communication, vol. com-28, pp. 468-687, April 1980.
4. D. E. Sproule and F. Mellor "Routing Flow, and Congestion Control in the Data-Pac Network." IEEE Transactions on Communication, vol. com-29, pp. 386-391, April 1981.
5. U. O. Pouch, W. H. Green, and G. G. Moss, "Telecommunications and Networking." Little, Brown Computer System Series, 1983.
6. V. K. Bharagava, D. Haccoun, R. Matyas, and P. Nuspl, "Digital Communications by Satellite." Modulation, Multiple Access and Coding. John Wiley and Sons, Inc. 1981.
7. J. C. Bellamy, "Digital Telephony." John Wiley and Sons, Inc. 1982.
8. J. W. Atkins, N. J. P. Cooper, and K. E. Gleen, "The Evolving Art of Packet Switching." British Telecommunication Technol Journal, vol. 2, no. 2, April 1984.
9. F. M. Luis and I. Rubin, "Message Delays for a TDMA Scheme Under a Nonpreemptive Priority Discipline." IEEE Transactions on Communication, vol. com. 32, no. 5, May 1984.
10. V. G. Cerf and R. E. Kahn. "A Protocol for Packet Network Interconnection." IEEE Transactions on Communication, vol. com-22, no. 5, May 1974.

11. T. A. Hamdy. "Operations Research: An Introduction." 2nd Edition. MacMillan Publishing Company, 1976.

12. A. S. Tanenbaum, "Computer Networks." Prentice Hall Inc., Englewood Cliffs, NJ, 1981.

13. C. R. Dhas, "Network and Flow Control." ACM Computer Communications Review, pp. 6-15, March 1984.

14. F. A. Tobagi, M. Gerla, R. W. Peebles and E. G. Manning, "Modeling and Measurement Techniques in Packet Communication Networks." Proceedings of the IEEE, vol. 66, no. 11, pp. 1423-1443, November 1978.

15. J. C. R. Licklider and A. Vezza "Applications of Information Networks." Proceedings of the IEEE, vol. 66, no. 11, pp. 1330-1346, November 1978.

16. L. Klienrock, W. E. Naylor and H. Opderbeck "A Study of Line Overhead in the Arpanet." Communications of the ACM, pp 3-12, January 1976.

17. A. Albanese, "Fail-Safe Nodes for Lightwave Digital Networks." Bell System Tech. J. vol. 61, no. 2, pp. 247-256, February 1982.

18. L. G. Roberts, "Data by the Packet." IEEE Spectrum, pp. 46-51, February 1974.

19 R. W. Klessing, "Overview of Metropolitan Area Network." IEEE Communications Magazine, vol. 24, no. 1, pp. 9-15, January 1986.

20. P. I. P. Boulton and E. S. Lee, "Bus, Ring, Star and Tree Local Area Networks." ACM Computer Communication Review, pp. 19-23, March 1984.

21. T. H. Pyke, Jr., and R. P. Blanc, "Computer Networking Technology - A State of the Art Review." Computer Networks: A Tutorial by M. Abrams, R. P. Blanc and I. Cotton, 3rd. edition, 1980, pp. 2-6 to 2-11.

22. W. F. Glozza and G. Noguez, "FIPNET: A 10 MBPS Fiber Optics Local Network." ACM Computer Communications Review, pp. 6-20, June 1984.

23. K. T. Pogram, "Selection of a Local Area Network for the Cronus Distributed Operating System." ACM Computer Communications Review, pp. 25-35, June 1984.

24. T. Total, "Totally Awesome Protocol." ACM Computer Communications Review, pp. 43-47, June 1984.

25. H. K. Pung, P. A. Davies, "Fibre-Optic Local-Area Network with Arbitrary Topology." IEE Proceedings, vol. 131, pt. H, no. 2, April 1984.

26. S. C. Kak, "A Spoked-Ring Network." Tech Report, Louisiana State University, June 1984.

27. A. Ralston, E. D. Reilly, Jr., "Encyclopedia of Computer Science and Engineering." 2nd edition, pp. 1270-1372.

28. J. O. Limb, L. E. Flamm, "A Distributed Local Area Network Packet Protocol for Combined Voice and Data Transmission." IEEE Journal on Selected Areas in Communications, vol. Sac-1, no. 5, pp. 926-934, November 1983.

29. I. W. Cotton, "Computer Networks: Capabilities and Limitations." Structural Mechanics Computer Programs, Pilkey et al, eds., Charlottesville, University Press of Virginia, 1974, pp. 1043-1056.

30. W. Stallings, "Local Networks: An Introduction." Macmillan Publishing Company, 1984.

31. P. K. Verma, "A 1984 Perspective on Developments in Computer Communication Systems." Computer Networks and ISDN Systems 9, pp. 43-48, 1985.

32. P. Hughes, "Communications: The Future Computer Networks and ISDN". Computer Networks and ISDN Systems 9, pp. 49-53, 1985.

33. T. Brooks, "New Technologies and Their Implications for Local area Networks." Computer Communications, vol. 8, no.2, pp. 82-89, April 1985.

34. B. Rich, "Communications and the Personal Computer." Computer Communications, vol. 8, no. 2, pp. 64-73, April 1985.

35. M. DePrycker, "LANS in an ISDN: Consistency with the OSI Reference Model." Computer Communications, vol. 8, no. 2, pp. 74-78, April 1985.

36. K. R. Crooke, "Local Network Strategy - Today's Plans for Tomorrow's Network." British Telecommunication Engineering, vol. 3, pp. 297-299, January 1985.

37. S. Heap and J. D. Arthur, "Network Management in the Digital Network." British Telecommunications Engineering, vol. 3, pp. 308-310, January 1985.

38. I. Watson, "The Integrated Services Local Network." British Telecommunications Engineering, vol. 4, pp. 21-26, April 1985.

39. M. Raghupati, K, Ramkumar and M. Satyam, "Priority - Oriented Reservation Multiple Access Scheme for Data Communication Through Satellites." Computer Communications, vol. 8, no. 1, pp 22-26, February 1985.

40. T. K. Apostolopoulos and E. N. Protonotarios "Queueing Analysis of Buffered Slotted Multiple Access Protocols." Computer Communications, vol. 8, no. 1, pp. q-21, February 1985.

41. S. Serpell, "Networking Secure Electronic Funds Transfer at Point of Sale." Computer Communications, vol. 8, no. 1, pp. 3-8, February 1985.

42. T. K. Apostolopoulos, E. D. Sykas and E. N. Protonotarios, "Performance Analysis of a Retransmission Control Algorithm for Local Area Networks." Computer Communications, vol. 8, no. 3, pp. 128-140, June 1985.

43. T. Reed, "Operating and Managing a Commercial Worldwide Network." Computer Communications, vol. 8, no. 3, pp. 141-147, June 1985.

44. A. L. Chapin, "Computer Communications Standards", Computer Communication Review, vol. 15, no. 2, pp. 60-68, April/May, 1985.

45. G. J. Fosclini, "Uninstrusive Communication of Status in a Packet Network in Heavy Traffic." AT&T Technical Journal, vol. 64, no. 2, pp. 463-479, February 1985.

46. D. L. Jagerman, "Waiting Time Convexity in the M|G|1 Queue." AT&T Technical Journal, vol. 64, no. 1, pp. 33-41, January 1985.

47. C. Courcoubetis and P. Varaiya, "Optimal Resource Allocation for Two Processes." AT&T Technical Journal, vol. 64, no. 1, pp. 1-15, January 1985.

48. B. Fadini, A. Marcelli and A. Mazzeo, "Approach to the Performance Evaluation and Tuning of Communication Protocol Implementation." Computer Communications, vol. 8, no. 2, pp. 59-63, April 1985.

49. J. S. Grant, "Multilink - An Open Network for Personal Computers." Computer Communications, vol. 8, no. 1, pp. 27-34, February 1985.

50. B. N. Garbutt, "Digital Restructuring of the British Telecommunications Network." British Telecommunications Engineering, vol. 3, pp. 300-305, January 1985.

51. M. T. Liu, W. Hilal and B. H. Groomes, "Perfromance Evaluation of Channel Access Protocols for Local Computer Networks." Proceeding of the CompCon Fall 82 Conference, pp. 417-426.

52. W. Bux, "Local-Area Subnetworks: A Performance Comparison." IEEE Transactions on Communications, vol. 29, no. 10, pp. 1465-1473, October 1981.

53. M. Imai, K. Takagi and Nakatani, "Development of a Medium-/Small-Scale Local Area Optical Fiber loop Network." IEEE Journal of Lightwave Technology. vol. LT3, pp. 532-526, June 1985.

54. K. Ohno, M. Masuda, Y. Watanabe and K. Kubo, "Multipurpose Optical Loop Network - Pana - Lan". IEEE Global Tellecommunications Conference, pp. 1042-1047, December 1985.

55. J. Minowa, N., Tokura, K. Nosu, "Development of Fiberoptic Local Area Networks in Japan." IEEE Journal of Lightwave Technology, vol. LT3, pp. 448-458, June 1985.

56. J. Mollenauer, "Standards for Metropolitan Area Networks." Proceedings of the National Communication Forum, vol. 1, pp. 561-565, September 1986.

57. J. H. Saltzer and D. D. Clark, "Why a Ring." Proceedings of the Seventh Data Communication Symp. 1981, pp. 211-217.

58. J. M. Metcalfe and D. R. Boggs, "Ethernet: Distributed Packet Switching for Local Computer Networks". Communications of the ACM, July 1976.

59. IEEE Project 802. Local network standard, Draft C, May 17, 1982.

60. Digital Equipment Corp., Intel Corp; and Xerox Corp. The Ethernet: A Local Area Network Data Link Layer and Physical Layer Specifications, September 30, 1980.

61. J. F. Shoch, Y. K. Dalu, and D. D. Redell, "Evolution of the Ethermer Local Computer Network." Computer, August 1982.

62. J. E. McNamara, "Technical Aspects of Data Communications." 2nd Edition, Digital Press, 1982.

63. W. D. Farmer and E. E. Newhall, "An Experimental Distributed Switching System to Handle Busty Computer Traffic." Proceedings, ACM Symposium of Problems in the Optimization of Data Communications 1969.

64. J. R. Pierce, "Network for Blocked Switching of Data." Bell Systems Technical Journal, July/August 1972.

65. E. R. Hafner, Z. Nendal, and M. Tschantz, "A Digital Loop Communications System." IEEE Transactions on Communications, June 1974.

66. A. Hopper, S. Temple and R. Williamson, "Local Area Network Design." Addison-Wesley Publishing Company, 1986.

67. P. Zafiropulo, "Perfirmance Evaluation of Reliability Improvements. Techniques for Single-Loop Communication Systems." IEEE Transaction on Communications, vol. 22, June 1974.

68. R. M. Dritz, "Enhanced Local Area Network Growth." National Communications Forum Proceedings, vol. 1, pp. 550-551, 1986.

VITA

Born in Okrika, Nigeria on April 22, 1955, Ibibia Karisemie Dabipi attended Government Comprehensive Secondary School Port-Harcourt in 1969 and obtained his high school diploma (West African School Certificate) in 1973. His Bachelor of Science in Electrical Engineering and Bachelor of Science in Physics and Mathematics were all obtained in 1979 from Texas A & I University, Kingsville, Texas. He came to Louisiana State University, Baton Rouge, Louisiana, in January 1980 and successfully completed his masters degree program in Electrical Engineering in August 1981. Since graduation, he has been an Assistant Professor in the Electrical Engineering Department at Southern University, Baton Rouge, Louisiana while pursuing a Ph.D in Electrical Engineering at Louisiana State University. He has also worked at Bell Communications Research in New Jersey during the summers of 1984 and 1985, where he worked on fiber optics projects with the local communications district. He is a member of Eta Kappa Nu (Electrical Engineering Honorary Society). He is presently a candidate for the degree of Doctor of Philosophy in Electrical Engineering.

EXAMINATION AND THESIS REPORT

Candidate: Ibibia Karisemie Dabipi

Major Field: Electrical Engineering

Title of Thesis: Fail-Safe Local Area Networking Using Channel Redundancy

Approved:

Subhash Kak

Major Professor and Chairman

[illegible]

Dean of the Graduate School

EXAMINING COMMITTEE:

[illegible]

M. Zohdi

[illegible]

R. A. Kinney

P. Klinkhachorn

Date of Examination:

March 11, 1987