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Performance Evaluation in terms of Congestion and Flow Control of Interconnected Token Ring Local Area Networks

by

Muhammad Salim Raza

Thesis submitted to the Faculty of the Graduate School
of New Jersey Institute of Technology in partial
fulfillment of the requirements for the degree of
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ABSTRACT

In an interconnected network, if user demands are allowed to exceed the system capacity, unpleasent congestion effects occur which rapidly neutralize the delay and efficiency advantages. Congestion can be eliminated by using an appropriate set of traffic monitoring and control procedures called ***flow control*** procedures. This thesis first investigates the major technical concepts underlying the token-ring technology, performance and flow-control issues and then gives an approximate analytical solution in terms of mean end-to-end delay in a system of token-ring local area network interconnected through bridges. The analytical solution is based on an approximation of the mean end-to-end delay in a stand alone LAN and then extended by approximating the arrival rates at the bridges as a function of the throughput of each subnetwork . Besides throughput and delay, a more compact form of performance measure called ***power*** has also been in the study.

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2 Analysis and queueing models for Token Ring LANs

3 Interconnection of Token Ring LANs

4 Numerical Results

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CHAPTER 1

INTRODUCTION

1.1 Introduction and background

Interconnection of networks allows a user to access computing resources in different networks. This thesis is concerned with flow and congestion control in interconnected network.

One of the main issues of network interconnection is to resolve the problem of how to control message flow from one network to another in order to achieve the required performance. In an interconnected system a LAN become more congested because of the traffic transmitted from other LANs, even if the traffic generated from itself is net heavy. Moreover the congestion propagates to other LANs (this is referred as "*back pressure effect*") as well. The back pressure effect results in system dead-lock in due course. See fig 1. Therefore a mechanism for controlling the amount of internet-traffic in a LAN, should be required in order to prevent and to relieve the congestion.

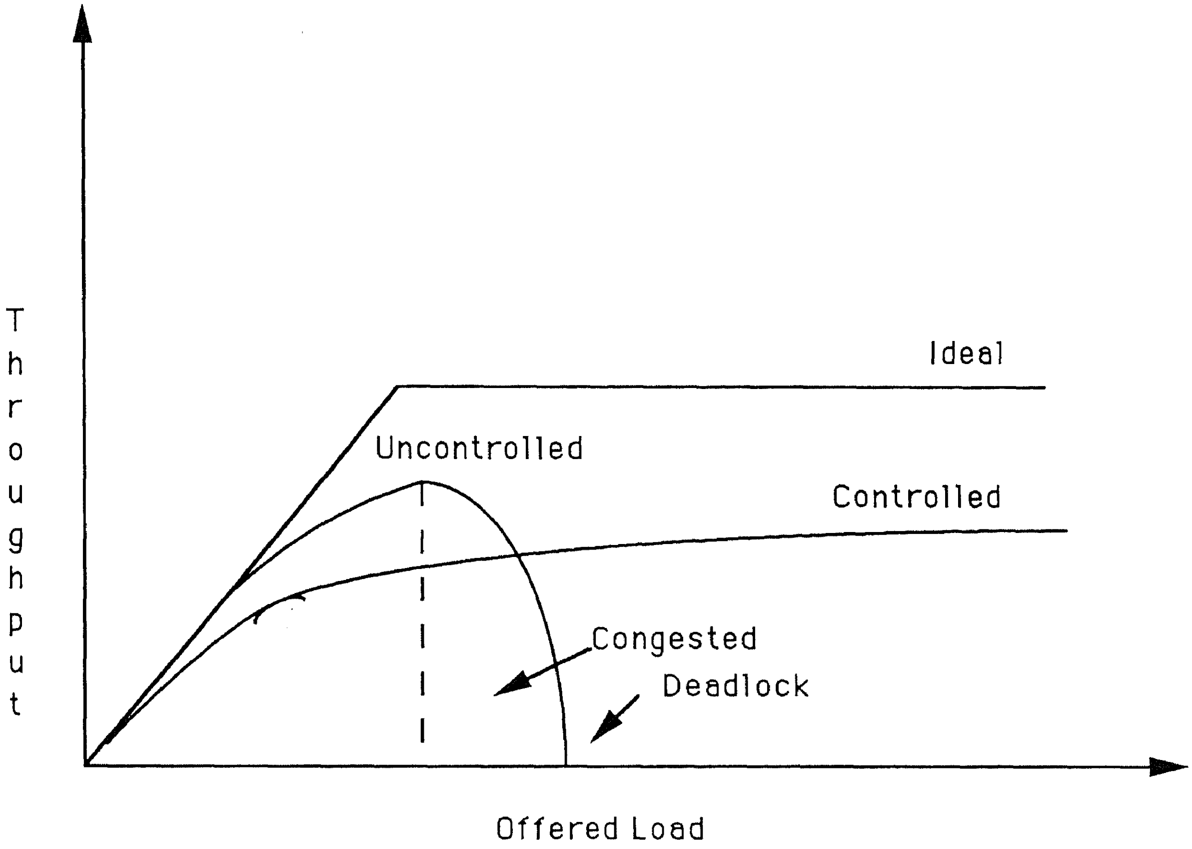


Fig. 1 Tradeoff between througuput and offered load

A local area network (LAN) is a data communication network, whose geographical range is limited to few (usually 1-10) kilometers.

Like any other communication network, a LAN is composed of three basic hardware elements.

- a transmission medium (usually twisted pair, coaxial cable, or fiber optics).
- a mechanism for control of transmission over the medium.
- an interface to the network for every device connected to it.

Besides these hardware components, there is a software component that is a set of protocols, implemented in the devices connected to the network, that controls the transmission of information at various levels.Before giving a more detailed description of various LAN components and technologies, we like to point out that local area networks evolved from long haul networks, due to the decrease in the cost of hardware technology together with the improvement of data communication techniques.On one side the low price of computer hardware caused the concentration of many devices, such as mainframe, minicomputer, and different microprocessor systems, in a localized environment, e.g a building. This situation demand high rate and low cost communication among these devices, in order to make effective use of them. On the other side networking technology evolved in order to make an effective and efficient use of the expensive transmission media (like satellite circuits or wide-band common carrier circuits).

Local area networks therefore are a very successful attempt of combining low-cost hardware with networking techniques and they provide efficient communication between a variety of devices in different environments.

1.2 LANs and ISO-OSI model

Topology of the network is the structure by which nodes are interconnected. The most general topology is an unconstrained graph structure and it is the one typically used in long haul networks because of its flexibility and generality. But it is too complex and not useful in LANs. The most common topologies used in LANs are *bus, star, ring and tree.*

The choice of a particular topology is the first step in the design of LANs and it depends on the environment in which the network has to operate. The choice of a topology is related also to the choice of the transmission medium . For instance broadband coaxial cable is well suited for bus topology, but not for the ring topology, because this would increase the complexity of the ring repeaters. For the ring topology, a twisted pair, a baseband coaxial cable or an optical fiber link is a better choice.

The topology also determines which control mechanism can be centralized. Among the examples of distributed control mechanism, the random access technique are associated with the bus topology and the regulated access technique with the ring topology. Fig 2a, 2b, and 2c shows different networks and their interconnections.

In an effort to standardize network architectures and protocols, the ISO has developed a reference model for use in comparing different architectures and in constructing new networks. This model is called the reference model of open systems interconnection (OSI) and it has seven layers.

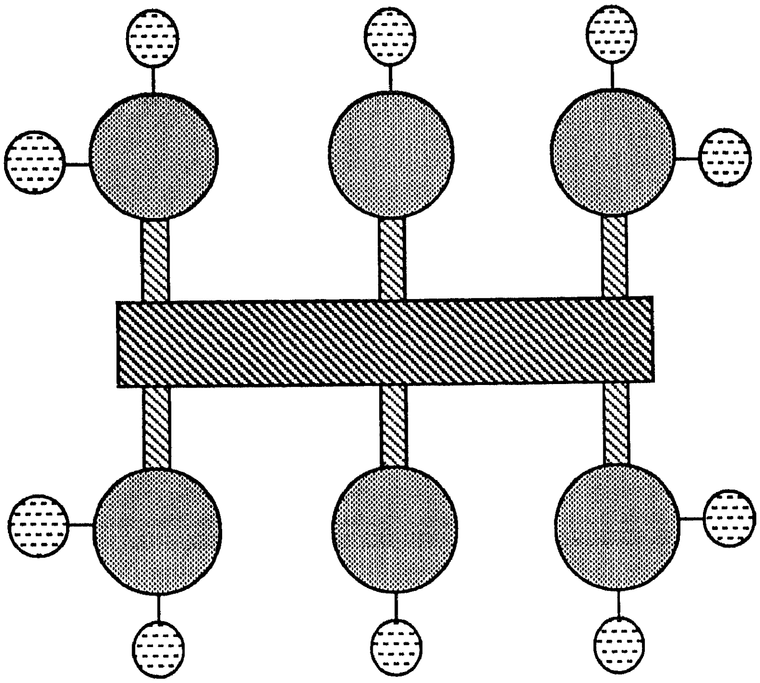
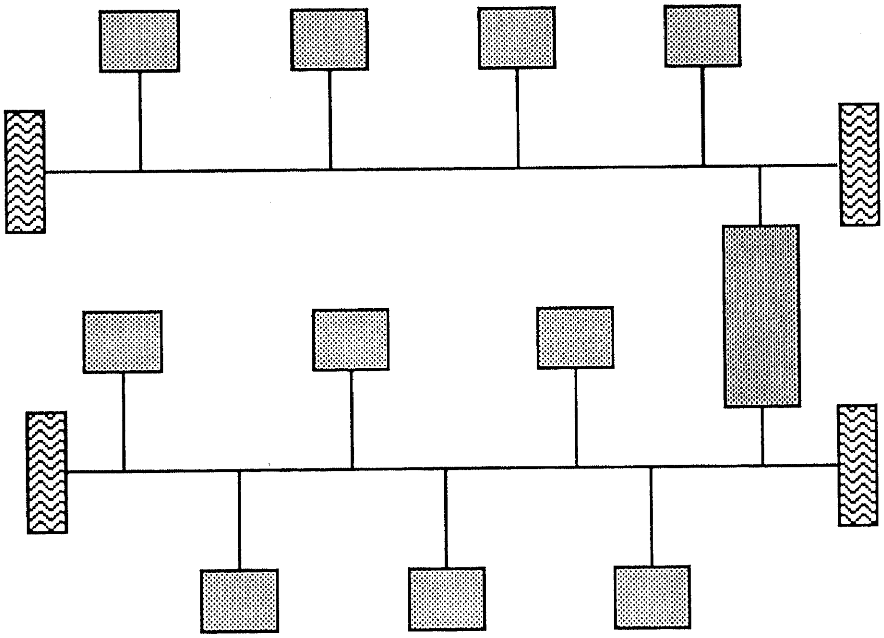


FIGURE 2 (a)



(b)

Fig. 2 b Interconnection of LANs

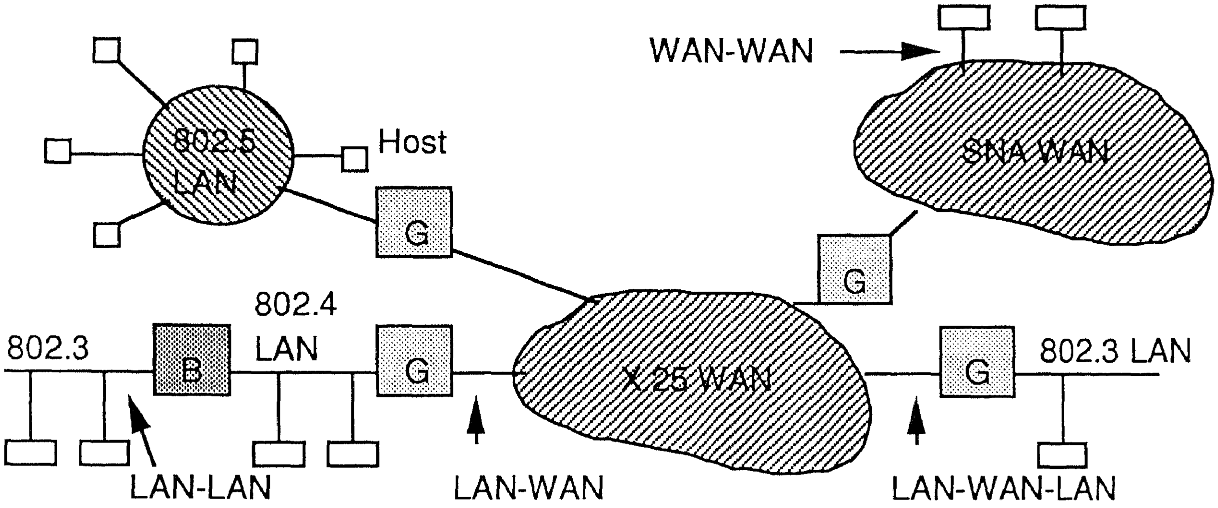


Fig. 2c Network interconnection
The boxes marked B are bridges. The boxes marked G are gateways

The major principles that ISO applied to arrive at the seven layers are the following:

1. A layer should be created where a different level of abstraction is needed.
2. Each layer should perform a well defined function.
3. The function of each layer should be chosen with an eye toward defining internationally standardized protocols.
4. The layer boundaries should be chosen to minimize the information flow across the interfaces.
5. The number of layers should be large enough so that distinct functions need not be thrown together in the same layer out of necessary , and small enough so that the architecture does not become unwieldy.

The OSI model is shown in the figure 3. The figure shows the seven layers in two systems and their relationship:

- protocol in two equivalent layers in two different systems.
- interface between two different layers.

When two systems wish to communicate , they use layer 7 protocol. This protocol requires services from the layer 6 protocol and so on down to the physical layer. That is the only one at which there is a direct communication between the two systems. In the application of the OSI model to LANs it should be noticed that certain high level functions are not required for proper operation of a LAN. Essentially, in a LAN, there is a direct link between any two points and this eliminates the need for routing. Therefore it seems reasonable to restrict the model for LANs to only the two lower layers namely, Physical and data link layer. However, experience gained in observing operating LANs shows that some forms of flow control could greatly improve performance particularly in the case of interconnected LANs. As we will mention in the next chapter , routing and flow control have been proven to be necessary, and this would imply that the network

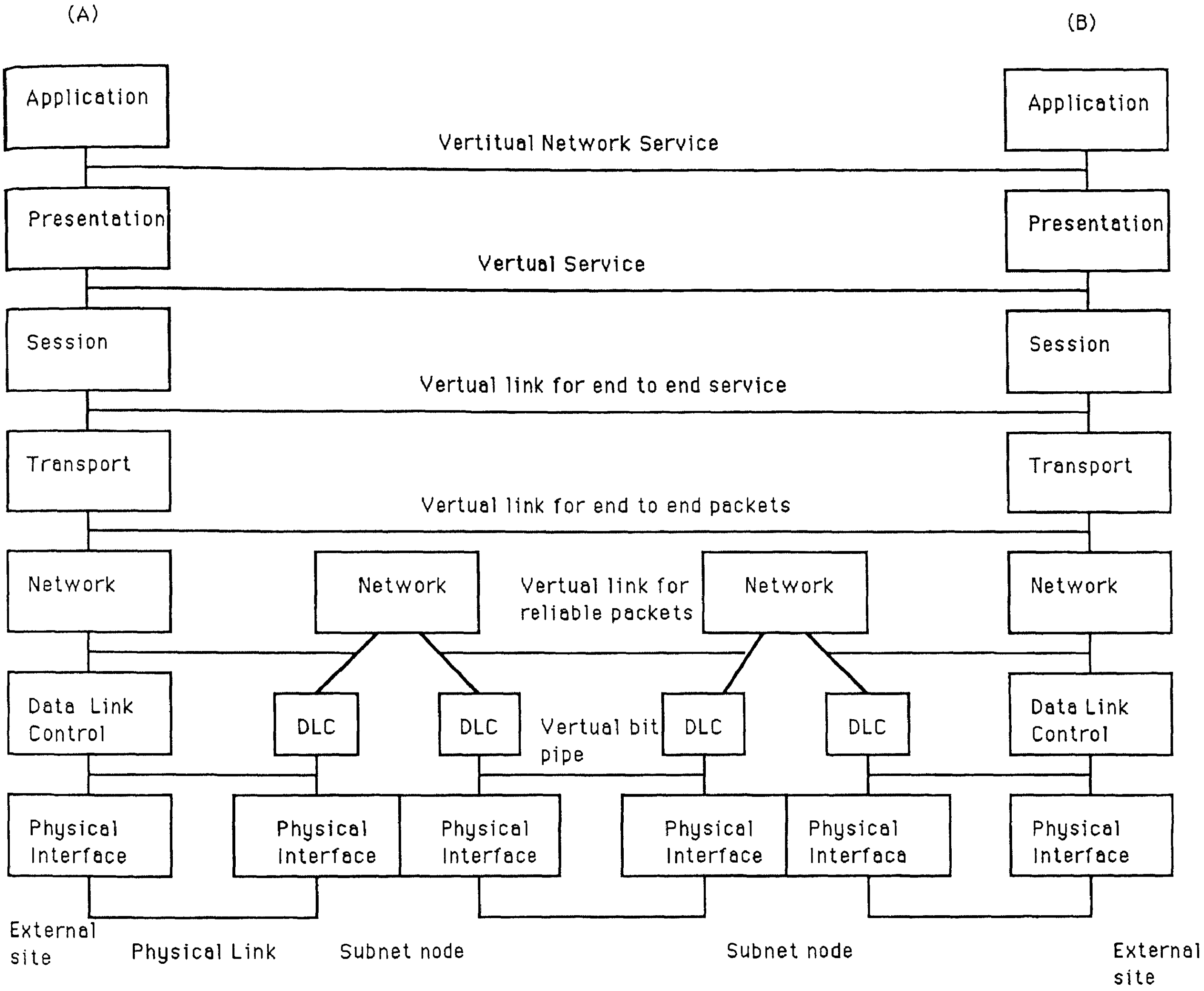


Fig. 3 Seven-layer OSI network architecture. Each layer presents a virtual communication link with given properties to the next higher layer.

control layer (layer 3 of OSI model) also should be implemented in LAN systems.

The IEEE 802 group developed a three-layer architecture for LANs, in which the data link layer of the OSI model is split into two layers, namely LLC (Logical Link Control) and MAC (Medium Access Control). See fig. 4. In the IEEE 802.2 standard a common LLC protocol provides basically the same functions associated with the data link layers i.e assembling data into a frame with address and other control fields, and upon reception of data frames, the disassembling , address recognition and validation of the received data. The MAC layer is necessary to manage access tom a multiple source, multiple destination link and this is not found in a traditional layer 2 link control. This layer has been defined separately also because with the same LLC, many different MAC options can be provided. Figure 4 shows the ISO model and the IEEE LAN reference models compared.

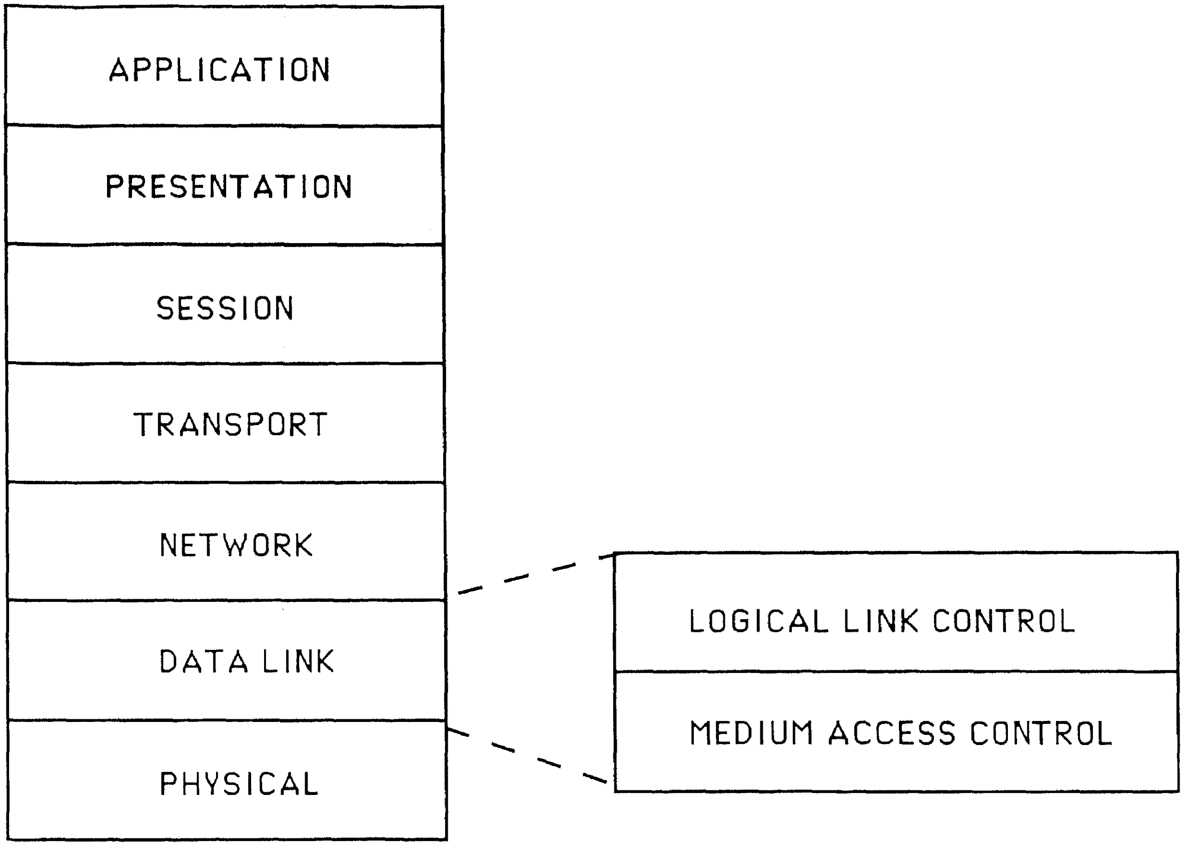
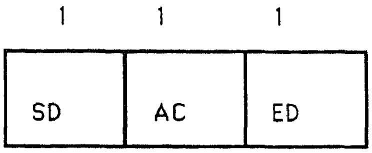


FIG. 4 LAN ARCHITECTURE REFERENCE MODEL

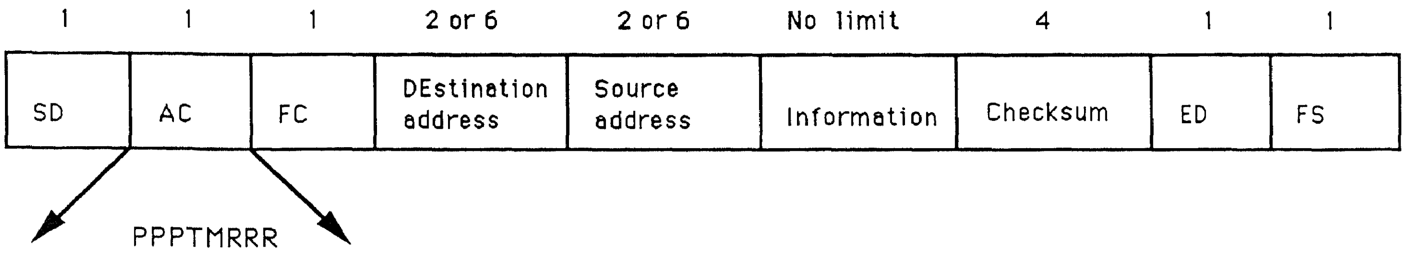
1.3 Token ring protocol

The token ring is the only medium access protocol for rings specified by IEEE 802. The version of token ring that has been adopted by IEEE 802 is an outgrowth of research and development at IBM. The message formats and the protocol has been defined in IEEE 802.5 standard. Token ring protocol is one of the popular access method for communication among devices connected via a local area network. A token ring consists of a set of stations serially connected by a transmission medium. Information is transferred sequentially from one station to the next. Each station on the ring has one a physical component that repeats each bit of information and serves the purpose of attaching the station to the ring. The right of a station to transmit (i.e. its access to the transmission medium.) is determined by the appropriate bit pattern called *token*. A token has the format as shown in the fig. 5. If a station upon detection of a free token has something to transmit, it captures the token by modifying it to indicate that it is busy and appends to it appropriate control and information fields. The destination station (s) copies the information as it passes by and finally the transmitting station removes the information from the ring and generates a new token, which provides other stations the opportunity to gain access to the ring. A token holding timer started at the beginning of data transfer controls the maximum length of time a station can occupy the medium before transmitting a token. In order to make the network more efficient it is possible to define multiple levels of priority to the stations, but this is outside the scope of this thesis.

Figure 5 gives a description of the frame format which is important in our study, because it is the basis for the definition of the service time distribution function in the queuing systems used to model the LAN.



(a)



PPP = PRIORTY BITS
T = TOKEN BIT
M = MONITOR BIT
RRR = RESERVATION BITS

SD = STARTING DELIMETER
AC = ACCESS CONTROL
FC = FRAME CONTROL
DA = DESTINATION ADDRESS
SA = SOURCE ADDRESS
INFO = INFORMATION
FCS = FRAME CHECK SEQUENCE
ED = ENDING DELIMITER
FS = FRAME STATUS

(b)

(a) Token Format

(b) Data frame format

Fig. 5 Token and Data Frame Format

The general format for transmitting on a ring operating according to the IEEE 802.5 standard is called a *frame*. See fig.5 and the following description about the frame format.

1. Starting Delimiter (SD): A unique eight bit pattern used to start each frame.

2.Access Control (AC): Has the format "PPPTMRRR", where PPP and RRR are 3-bit priority and reservation variables, M is monitor bit and T indicates whether this is a token or data frame. In the case of token frame, the only additional field is ED.

3. Frame control (FC): Indicates whether this is an LLC data frame. If not the bits in this field control the operation of the Token ring in the MAC protocol . This also has eight bits.

4. Destination Address (DA): Specifies the station (for which the frame is intended). It may be unique physical address or global address. The choice of 16- or 48- bit address is an implementation decision and must be same for all stations on a particular local network.

5.Source Address (SA): Specifies the station that sent the frame. The SA size must be equal to the DA size.

6.LLC Data (Info): Field prepared at the LLC level.The size is variable up to 133 bytes. there is no predefined limit except that the time necessary to transmit a frame cannot be longer than the token holding time (up to 133 bytes) .v

7. Frame check sequence (FCS): A 32 bit cyclic redundancy check value. based on all fields starting with destination address.

8. Binding Delimiter (ED): Contains the error detection (E) bit and the intermediate frame (I) bit. The I bit is used to indicate that this is the frame other than the last frame of a multiple frame transmission.

9. Frame Status (FS): Contains the address recognized (A) and frame copied (C) bits.

1.4 Flow and Congestion control in lans

Flow control is a mechanism to regulate traffic flowing from source to destination so that the source does not send data at a rate greater than the receiver can process it. The main functions of the flow control are :

- Prevention of throughput degradation and loss of efficiency due to over load.
- Dead lock avoidance
- Fair allocation of resources among competing users.
- Speed matching between the network and its attached users.

Stop-and-wait and the sliding windows are two well known flow control mechanisms. Among the two the, window flow control is usually used in LANs. On the other hand, congestion control is a more global mechanism accomplished by internal network nodes so as to prevent network congestion. Flow control is an end-to-end phenomenon, whereas, congestion control deals with problems occuring at intermediate nodes as well. Some network have attempted to use flow control mechanisms to eliminate congestion. It is difficult to control the amount of traffic in the network using end-to-end flow control rules. Flow control cannot really solve congestion problem for a good reason, computer traffic is busty. Any flow control scheme which is adjusted so as to restrict each user to the mean rate will provide bad service when the user sends a burst of traffic. On the other hand if the flow control limit is set high enough to permit the peak traffic to get through , it has little value as congestion control when several users demand the peak at once. All systems are designed for average traffic, not the worst case.

Various schemes have been proposed for congestion control in an inter-net environment. Gerla and Klienrock reported flow and congestion control mechanisms that can be used in an interconnected LAN environment. Bux and Grillo investigated a dynamic flow control algorithm in a network of interconnected token rings. They employed simulation rather than mathematical modeling. There are several methods to relieve congestion in local area networks. Some of the proposed schemes require the interaction of both flow and congestion control mechanisms. Dropping packets when buffers are full is currently the most popular way to relieve congestion. In this scheme a limit is set on the maximum number of input packets that can be buffered in the packet switch When the limit is exceeded input packets are dropped . The problems with packets is that it does not provide a direct feedback to the traffic sources. When a network become congested , a method that can slow down the sources is to *chock packets..* In this method whenever a network become congested it returns to the source a choke packet containing the header of the packet traveling in the congested direction. The source upon receiving the chock packet, declare the destination is congested, and allows traffic to that destination. Congestion control in interconnected LANs poses more challenging problems than in conventional networks. A combination of both flow and congestion control mechanisms are required to prevent congestion in interconnected LANs and requires some modifications and enhancements of the bridge design. The main challenge will be to retain transparency and throughput of the bridges while introducing these additional features.

1.5 Present research and thesis outline

Various schemes have been proposed for flow and congestion control in an inter-net environment. Gerla and Kleinrock reported flow and congestion mechanisms that can be used in an interconnected LAN environment. Bux and Grillo investigated a dynamic flow control in a network of interconnected token rings. They employed simulation rather than analytical models. Research in an interconnected token ring LAN has not been carried out extensively. So we are focusing to get some analytical results on end-to-end flow control applied to interconnected token ring LANs.

The most popular flow control scheme is window-based flow control which does not admit new packets into a control region reaches its maximum called *window size*. We are using a window flow control mechanism, in which a sender is allowed to transmit not more than a fixed number of frames without having to wait for an acknowledgement. We shall restrict the discussion to two basic performance measures namely throughput and end-to-end delay. We are assuming that no messages are lost and acknowledgements are piggybacked. The acknowledgements have the same end-to-end delay as a message. The packet lengths are assumed to be exponentially distributed. The performance of the system is evaluated by varying the window size.

Thesis outline will be like this. In the next chapter we will summarize the existing results , briefly introduced the basic queuing model that applies to token ring LANs namely the M/G/1 queue with sever vacation times and we will then introduced our results obtained by Bux, i.e an approximate solution to the most general system of queues served in cyclic order, whose exact solution is fairly complicated. The solution is given in terms of the average queuing delay of the packets at each station along the ring and of the average end-to-end delay of a packet from the moment it is generated at a source station to the moment it reaches its destination.

In the third chapter we extend the modelling and analysis introduced in chapter 2 to a set of token ring LANs interconnected via backbone ring. Then we will will investigate some flow control issues in LANs consisting of multiple token rings interconnected through bridges. We are aiming to analyze how window flow control can be implemented in the above system. The bridges are modeled as two queues, one on the sub-network and other on the backbone and the arrival rates for these two queues are found in terms of the throughput of each sub-network and the percentage that is supposed to be directed to the other sub-networks. The delay in each sub-network and in the backbone are found using the approximation presented in chapter 2 and the mean end-to-end delay in the system of interconnected LANs is evaluated as the sum of the average delays in the source ring, in the destination ring and in the backbone ring. The acknowledgements have the same end-to-end delay as a message. Hence the round-trip delay is twice as much as the delay for a message. Chapter 4 will cover the numerical results for different traffic patterns, offered loads and the number of sub-networks in the system. Chapter 5 is the conclusion. The results of chapter 4 will be analyzed and suggestions for future research are proposed.

CHAPTER 2

ANALYSIS AND QUEUING MODELS OF TOKEN RING LANs

2.1. Introduction

In a token ring, access to the transmission channel is controlled by passing a permission token among the ring, when the system is initialized, a designated station generates a free token which travels around the ring until a station ready to transmit changes it to busy and puts its packet onto the ring . The packet can, in principle , be of arbitrary length. The sending station is responsible for removing its own packet from the ring. At the end of its transmission, it passes the access permission to the next station by generating a new free token. Fig. 6 illustrates this operation for a ring with four stations. It shows how station 1 and 3 and again station 1 subsequently access the ring to transmit their packets.

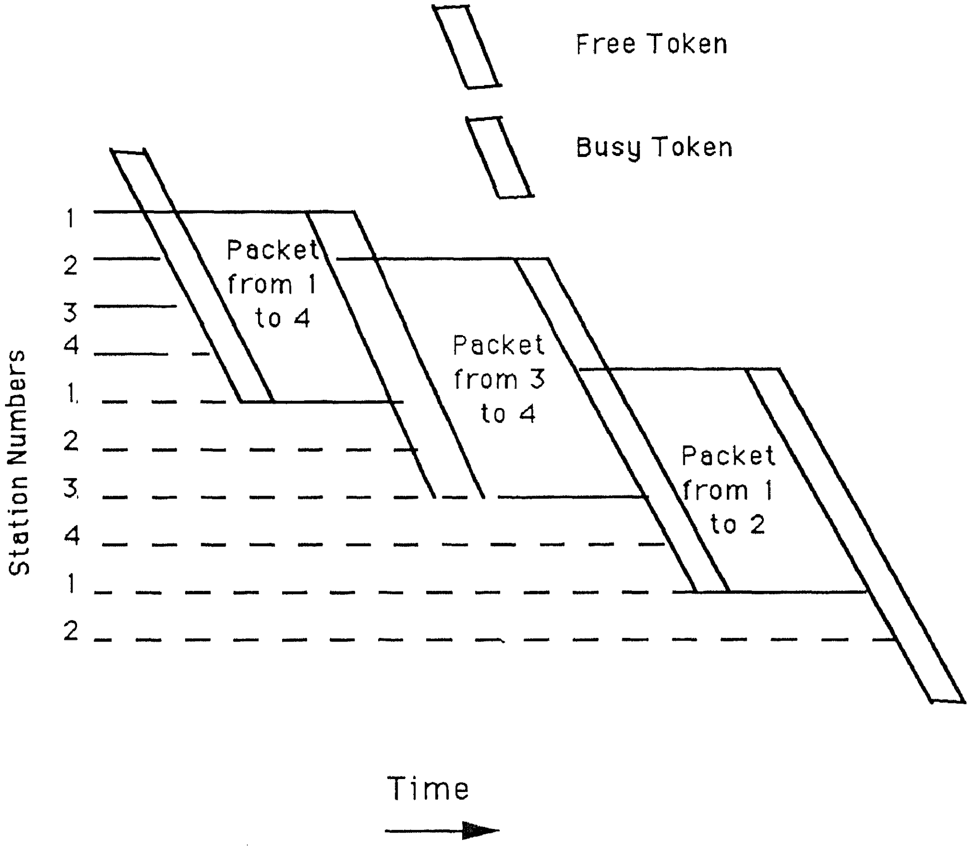


Fig. 6 Token Ring: example of operation (4 stations)

A single service facility serves n buffers of infinite lengths in a cyclic manner. Service is given to any queue until it is emptied, i.e., a service is *exhaustive* . To switch from queue i to queue i + 1 , a constant switch-over times r_i is required. Subsequently, we denote by R the sum of all switch-over times $r_1 + r_2 + \text{---} r_n$, messages arrive at all queues according to poisson processes with rates λ_1, λ_2, λ_3----- λn. The service times (transmission times) H_i of the messages from queue i are generally distributed with mean h_i and second moment $hi^{(2)}$. We denote the utilization $\lambda_i h_i$ due to queue i by ρ_i, and assume that the sum ρ of all ρ_i is less than one.

2.2 Analysis of polling systems

Polling schemes have been analyzed under different parameters, such as zero or nonzero reply interval (i.e., the time needed for the server to switch from one queue to the next), different distributions of message length (service time in the common queuing theory terminology), statistically identical or non identical stations, continues or discrete (i.e., slotted) time, and one-message buffer or infinite buffer systems. Considering the infinite buffer system, i.e., a system in which no message is lost , three types of service discipline have been studied:

a) **EXHAUSTIVE SERVICE:** The server serves each station until the buffer is empty, this means that also the messages arriving during the service time are served.

b) **GATED SERVICE**: The server serves the station for only those messages that are in the buffer when the station is polled.

c) **LIMITED SERVICE:** The server serves the station until either the buffer is empty or a specified number of messages are served , whichever occurs first.

We will restrict our review of the analysis of polling systems to those with continuous time and the first case i.e., *exhaustive* service, and we will always assume that messages at every station are generated according to a poisson process. The early research to find a solution for polling systems was done by many authors and was restricted to the case of only two queues served in cyclic order. Especially Takacs's paper is noteworthy because he derived using the concept of an imbedded Markov Chain. Cooper and Murray in 1968 extended the analysis to the case of an arbitrary number of queues N>= 2. Imbedding the Markov chain at the switch points (i.e., when the server is done serving a queue and switches to the next one, they found the Laplace transforms of the cycle time distribution function and sub sequential expressions for the mean number of messages waiting in the queues and the mean cycle time. In 1969 R.B Cooper extended his previous analysis to obtain the Laplace transform of the order-of-arrival waiting time distribution function and the mean waiting time for a message arriving at the ith queue.In order to calculate the waiting time the following generalization of the M/G/1 queue is introduced.

Whenever there are no messages left in a queue to be served , the server is imagined to go away for a length of time called *vacation.* At the end of the vacation the server comes back and begins to serve the messages that have eventually arrived during the vacation. Of course from the point of view of the whole system of queues, rather than of each individual queue, the time the server spends on vacation is the time necessary to serve all other queues in the system.

2.3 Mean waiting-time analysis

2.3.1. Approximation

In this section we will be describing an approximation of the mean delay in a polling system. The model is very general allowing an arbitrary number of nonidentical queues and nonzero switching overhead but the result has a straightforward numerical evaluation. The other assumptions are the usual ones we mentioned already, i.e., exhaustive service, poisson arrivals, and general service time distributions, that do not need to be identical for every queue.

In our model as shown in the figure 7, the active stations are represented by their transmission queues, r_i denotes the time to pass the token from station i to station i + 1 (physically it corresponds to the propagation delay of the signal between stations i and i + 1 plus the latency within station i due to the repeater and other eventual actions, like change of the token bit, that is usually of the order of one bit)

To determine the waiting time of messages in queue i, it is convenient to consider the model of fig. 7 as an M/G/1 queue with server vacation times. The server vacation time A_i correspond to what is called '*intervisit time*' i.e., the time interval from the server 's departure from queue i until its return to the same queue. This consideration leads direct to the following relation for the mean waiting time of messages in queue i

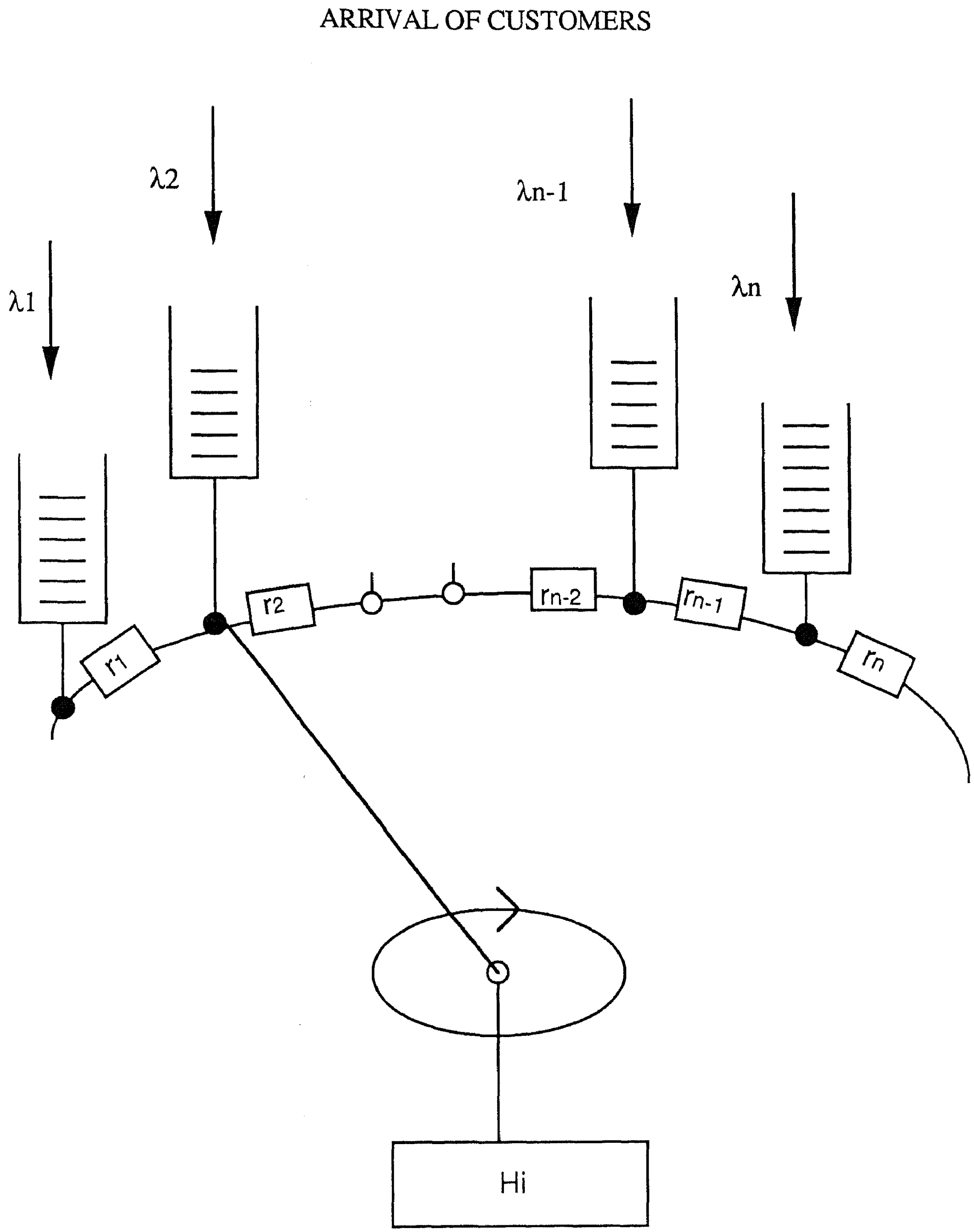


FIG. 7 TOKEN RING QUEUEING MODEL

$wi = E[Ai]/2 + Var[Ai]/2E[Ai] + \lambda i\, hi^{(2)}/2(1 - \rho i)$ (2.1)

Hence the mean delay can be determined if the first two moments of the intervisit times are known . Whereas it is straightforward to determine the first moment of Ai, calculation of the second moment is fairly involved . As subsequently described , our approach is to employ an approximation for the second moment based on a heuristic extrapolation from the exact result for n = 2 queues to the case of an arbitrary value of n.

First moment of Ai. The mean of the intervisit time can be defined as follows:

Let Ci be the total cycle time defined as the time between subsequent visits of the server to queue i. Then, if we define by Ti the time which the server spends to service queue i, the following relationship holds:

$E[Ai] = E[Ci] - E[Ti]$ (2.2)

It can be shown that the average cycle time of all queues is given by

$E[Ci] = R/(1 - \rho)$ (2.3)

The average number of messages serviced during one visit of the server at queue i is given by E[Ti] / hi. For flow conservation reasons this must be equal to the average number of messages arriving at queue i during one cycle time λi E[Ci].

Hence,

$E[Ti] = \lambda i\, E[Ci]\, hi.$ (2.4)

We obtain the mean of the intervisit time of queue i as

$E[Ai] = \{(1 - \rho i)/(1 - \rho)\} * R$ (2.5)

Second moment of Ai. Let Mi be the number of messages waiting in queue i before it is served. Then Ti, the time for which the server is serving queue i, can be conceived as the sum of Mi independent busy periods Bi, the first two moments

$$E[Bi] = hi / (1 - \rho i) \tag{2.6}$$

$$E[Bi^2] = hi^{(2)} / (1 - \rho i)^3 \tag{2.7}$$

In terms of Mi and Bi, the variance of Ti is given by

$$Var[Ti] = E[Mi]\ Var[Bi] + Var[Mi]\ E[Bi]^2 \tag{2.8}$$

Since Mi is equal to the number of arrival at queue i during Ai, we obtain

$$E[Mi] = \lambda i\ E[Ai] \tag{2.9}$$

$$Var[Mi] = \lambda i^2\ Var[Ai] + \lambda i\ E[Ai] \tag{2.10}$$

For an arbitrary number of stations n the above considerations are apparently insufficient to determine the variance of the intervisit times Ai needed to calculate the mean waiting times. In the special case of n = 2 stations , however, the intervisit times of one queue and the server's sojourn times at the other queue differ only by the constant time R; hence, we have

$$Var[Ti] = Var[Ak], \quad i=1,2 \quad k= 3 - i \tag{2.11}$$

From (5) and (11) we obtain, after some algebraic manipulation, the variance of Ai as

$$Var\ [Ai] = R / (1 - \rho)^2\ \frac{\lambda k\, hk^{(2)} (1-\rho i)^2 + \lambda i hi^{(2)}\ \rho k^2}{(1-\rho i-\rho k+2\rho i\rho k)} \quad i = 1,2,\ k = 3 - i \tag{2.12}$$

Returning to our general model with n > 2 queues, we now establish our major assumption. We assume (i) that the impact of the messages from any queue $k \neq i$ on the variance of the intervisit time of queue i can be approximately described by an expression corresponding to (12), and (ii) that the total variance of Ai is obtained by superposition of the individual components of all queues, i.e.,

$$\mathrm{Var}\,[Ai] = R \,/\, (1 - \rho\,)^2 \sum_{\substack{k=1 \\ k \neq i}}^{n} \frac{\lambda k\ hk^{(2)}\,(1 - \rho i)^2 + \lambda i hi^{(2)}\,\rho k^2}{(1 - \rho i - \rho k + 2\ \rho i \rho k,)} \qquad (2.13)$$

By inserting (5) and (13) into (1), we finally obtain the following approximate formula for the mean waiting time at queue i:

$$E[Wi] = \frac{(1-\rho i)R}{2(1-\rho)} + \frac{\lambda i hi^{(2)}}{2(1-\rho i)} + \frac{1}{2(1-\rho)(1-\rho i)} \sum_{k=1}^{n} \frac{(\lambda k\ hk)^2 (1-\rho i)^2 + \lambda i hi^{(2)} \rho k^2}{((1-\rho i-\rho k+2\rho i\rho k))} \qquad (2.14)$$

Bux and Truong discuss the properties of their approximation, that make it so appealing for practical purposes. First of all in the case of any two stations with arbitrary service time distribution and arrival rates it yields the exact result. Moreover, for an arbitrary number of stations, but symmetrical traffic conditions i.e., identical service time distributions with mean h and the second moment $h^{(2)}$ and equal arrival rates λi = ------------$= \lambda n = \lambda / n$) , the following result is obtained and it is the exact result

$$E[Wsym] = \frac{(1 - \rho/n)R}{2(1 - \rho)} + \frac{(\lambda h^{(2)})}{(2\ (1 - \rho))} \qquad (2.15)$$

Their third consideration refers to the average delay in the network, i.e., the summation over station delays weighted with the corresponding utilization, that is given by the following formula,

$$((E[W] = \sum_{i=1}^{n} (\rho i/\rho)\, E\,[Wi] = \frac{(1 - 1/\rho) \sum_{i=1}^{n} \rho_i{}^{2} \quad R}{2(1-\rho)} + \sum_{i=1}^{n} \frac{(\lambda_i h_i{}^{(2)})}{2(1 - \rho)} \qquad (2.16)$$

If R = 0 i.e., zero token circulation time, E[W] is the correct mean delay for the M/G/1 queue . Moreover , the formula shows an intuitively appealing result, the more unbalanced the traffic , the smaller effect the token passing over head has on the end-to-end delay . This seems to be right because , if the station utilizations are very different, there is a high probability that a newly generated message will join a non-empty queue or the one that is currently being serviced and in both cases , such a message will be serviced without any token passing overhead. Bux and Troung compare also their analytical results against simulation results and this proves the accuracy of their approximation .

In particular in the case of unbalanced traffic they noticed that the approximation has the tendency to slightly underestimate the delay of heavy traffic conditions, but on the other hand waiting time averaged over all stations turned out to be quite accurate. When the latency caused by each station increases compare to the message length, the accuracy of the approximation seems to be higher and this indicates that the formula reflects accurately the impact of the token passing overhead on the delay.

All these considerations, including the fact that the average delay in the network is even more accurate than the delays at each station, led us to use this approximation to analyze the model of interconnected token rings that we will described in the next chapter.

CHAPTER 3

Interconnection of Token Ring LANs.

3.1 Introduction

Multiple rings may be required in a LAN when the data transfer requirements exceed the capacity of a single ring or when the attached stations are widely dispersed , as in a multi-floor buildingor a campus environment. Further benefits from LAN interconnection are enhanced total system availibility by running multiple indepenent subnetworks, and the flexibility to map an existing building layout or organization structure onto a corresponding local network structure. Another situation in which the need for having interconnected LANs may arise is the one in which a stand-alone LAN has a poor performance, because it is loaded

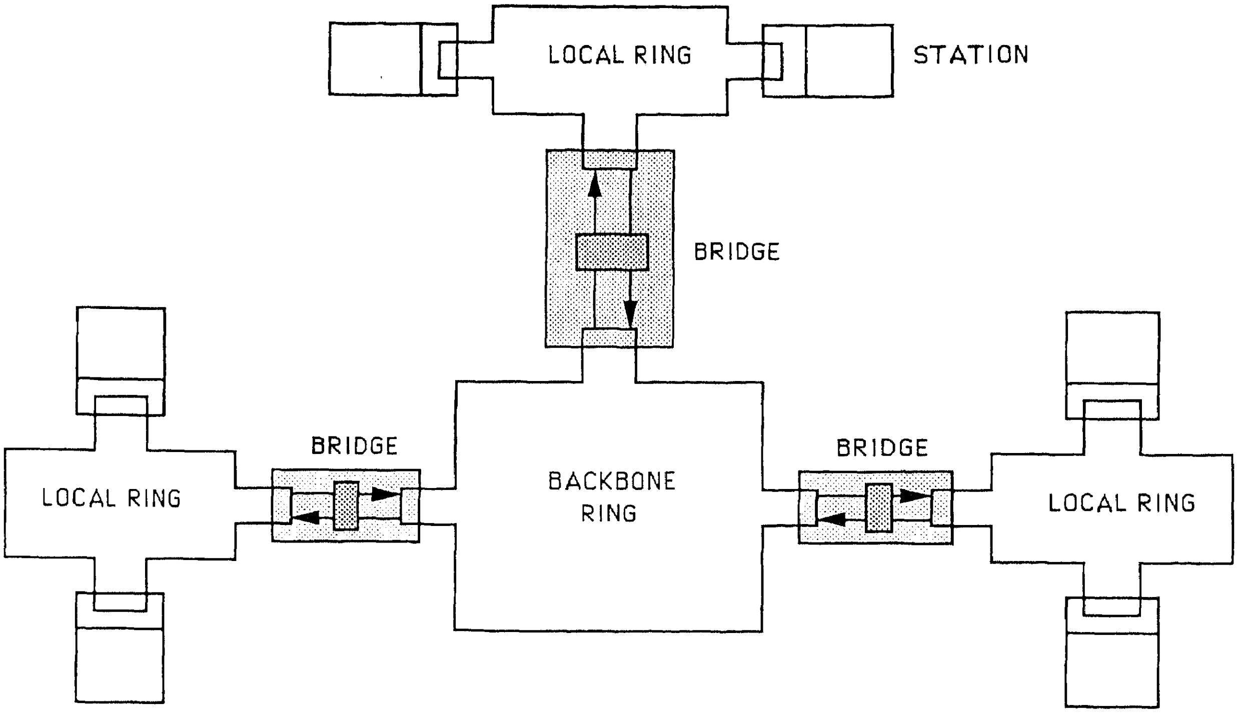


FIG. 8 MULTIRING NETWORK

with too busy traffic. The solution in this case could be the splitting of the LAN in to interconnected sub networks, each with the lower traffic intensity. The improvement may be even more noticeable if the original network is partitioned in such a way that the traffic among the sub networks is kept as low as possible by putting together in the same subnetwork the stations that have a high rate of message transfer among themselves. The basic requirement for the interconnection is that it should be transparent to both the LANs and any other network that might be connected to them. This means when the station wants to transmit the message to another station, it does not need to know whether the destination station is in the same sub-network or not. Analogously when an external network wants to send a message to a station in the LAN, it does not have to know to which sub-network the station belongs. This calls for a uniform addressing and administrative structure within the set of interconnected sub networks. This transparency is what makes interconnection of LANs substantially different from inter networking (i.e., connecting two or more networks together). See fig. 8 for multiring network in which 3 local rings are connected via a backbone ring.

3.2 Bridge:

Rings can be linked together by a high speed device known as a *bridge* . Bridges provide a basic routing and store-and-forward function. Bridges are smart (full of software); they can be programmed to copy frames selectively and make necessary changes while doing so. Frames are buffered in a bridge until they can be transmitted on the local or backbone ring. A bridge consists of two network interfaces, one for each of the two sub networks. It also has a control element that recognizes, based on the destination address in the message header, that a particular message is directed to another sub-network and takes care of storing it in the buffer and re-routing it. Fig. 9 shows the bridge architecture model.

We assumed that the bridge memory space is partitioned into two separate buffer pools, one for each data flow-direction. The buffer pools are structured in segments of a fixed size. Frames which do not find a sufficient number of free segments upon their arrival at a bridge are lost. A bridge is controlled by two processors, each of which handles one direction of data flow. These processors are modeled by two independent single servers which process all frames on a first-come first-served bases.

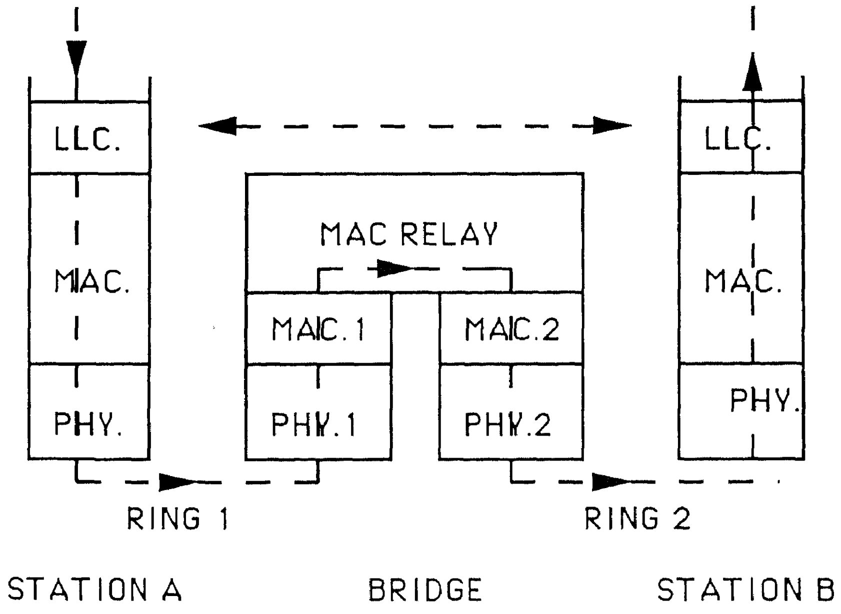


Fig. 9 Bridge Architecture Model

3.3 Gateways:

The gateway is used to connect to a network that does not use the OSI model at all. Gateways performs the necessary operations such as address translation, speed and protocol conversions to interface the LAN to different transmission media.

This study should be restricted to the interconnection of homogeneous LANs, i.e., LANs with the same topology and the same protocol . In this case the devices that connect one sub-network to another are called *bridges*. It should be mentioned here that it is possible to interconnect heterogeneous LANs, but this situation resembles more to the connection between a LAN and a long-haul network and the bridges have to perform gateway functions, such as protocol conversions. There is one more situation in which two sub-networks of the LAN are physically distant, in this case the stations used for the interconnection are called long distance bridges. A long distance bridge is made up of two half-bridges at both ends of a suitable full-duplex point-to-point communication link (e.g. high band width common carrier link or optical link.

3.4 Analytical model of interconnected Token Ring LANs.

Local area network must be capable of interconnecting a large number of stations over maximum distance of several kilometers. Whenever the limitations of a single-ring network are reached with respect to the maximum number of attachments or maximum distance, means for interconnection necessary.

Let's assume that the backbone ring has the same topology and the same protocol as the sub networks , and can therefore be modeled in the same way as the stand-alone token ring has been modeled . This allows a simple and homogeneous analysis and it is therefore very appealing. Fig 10 shows the interconnection model of local rings connected to the backbone ring.

We assume that there is a certain number of token ring sub-networks denoted by m, and also that all the sub-networks have the same number of stations, n. This choice was made just to simplify the parameters on which the numerical evaluation of the delay is based and it does not effect the model.

For the purpose of our study each bridge is modeled as two separate stations, one on the sub-network and other on the backbone. In the analysis of a stand alone token ring LAN in chapter 2, we assumed that every station on the ring generates traffic according to a poisson process and all the stations were supposed to be statistically independent. Now in order to be able to apply the same analysis to this system we need the same assumptions, even though in this case it is evident that the rate at which the bridge generates traffic on the backbone is dependent upon the throughput of each sub-network, and conversely , the rate at which , the rate at

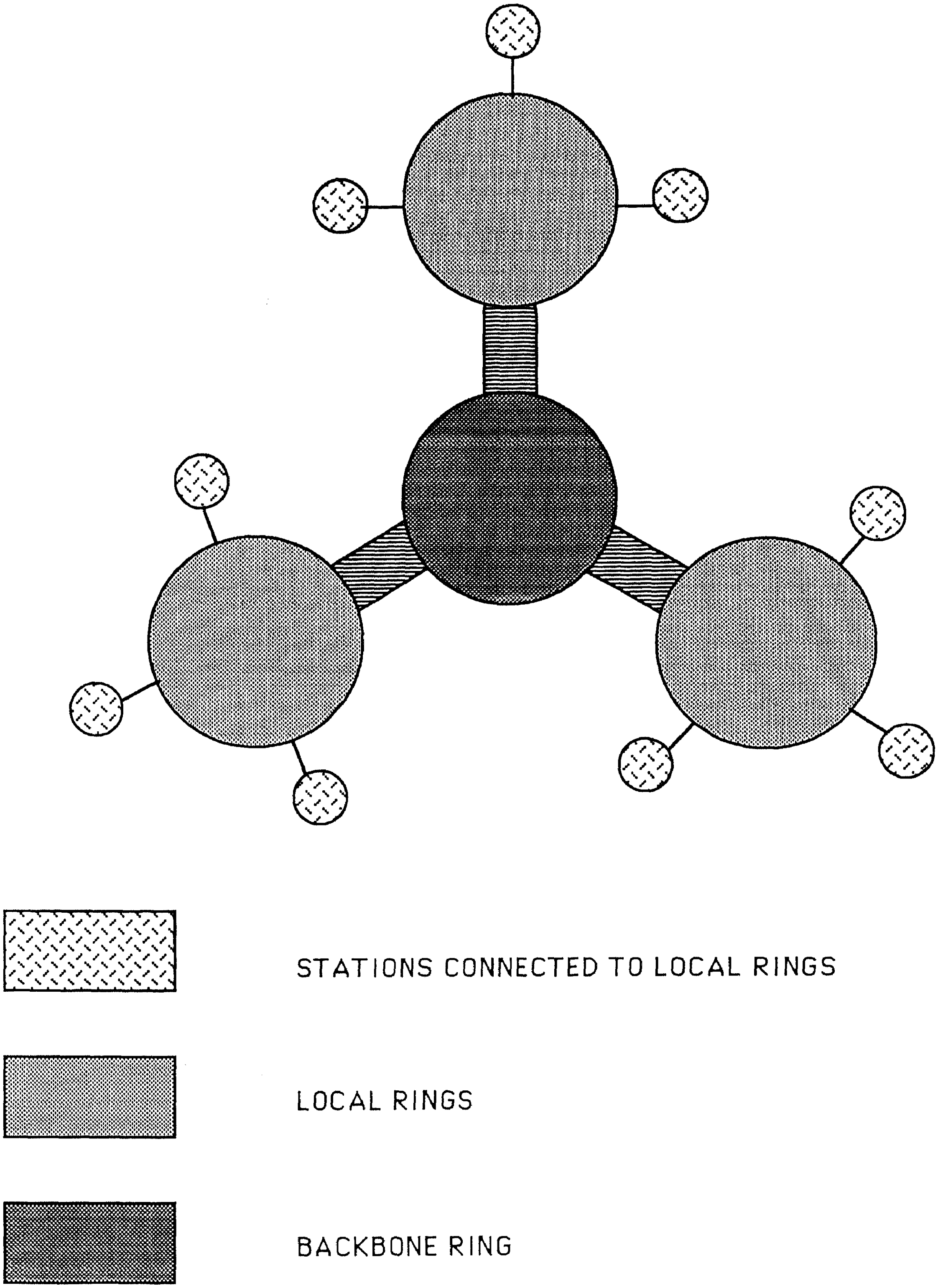


FIG. 10 INTERCONNECTION OF LANs

which the bridges generate traffic within any particular sub-network is dependent upon the throughput of the backbone and ultimately of the other sub networks. It seems resonable that after the network has reached stability conditions, the independence assumptions still hold. We therefore assume that the bridges generate traffic according to a poisson process, and we find the arrival rates of such poisson processes through an iterative algorithm in which the values are adjusted according to the throughput in the other sub networks until stability is reached. In order to describe the method in more detail let us focus on the calculation of the rate at which the bridges generate traffic on the backbone ring.

Let us define two rings i.e., i and j. Suppose Pij as the probability that a message from ring i, is directed to ring j, and Ti be the throughput of ring i i.e., the sum of the arrival rates (λi) of all the stations connected to ring i. The arrival rate at bridge i , connecting ring i to the backbone, can be expressed as follows:

$$\lambda_{Bi} = T_i \sum_{\substack{j=1 \\ j\neq i}}^{m} P_{ij} \qquad (i = 1, 2, 3, 4, \ldots\ldots\ldots\ldots m) \qquad (3.1)$$

With a similar resoning we can express the arrival rates at the bridges, viewed as stations on the local rings. We will denote such arrival rate with λ_{ib} and it is given by the following expression:

$$\lambda_{iB} = \sum_{\substack{j=1 \\ j\neq i}}^{m} T_j P_{ji} \qquad (i= 1, 2, 3, 4, \ldots\ldots\ldots\ldots m) \qquad (3.2)$$

Where Tj is the throughput (arrival rate of messages) of ring j and Pji is the probability of ring j and i.

The iterative procedure assigns an initial value of

$\lambda_{iB} = w/ 2Dn$ ....(3.3)

where n is the number of stations, W is the window size and D is the end-to-end round trip delay of a message.

It is important to note that we are assuming exponentially distributed packet lengths with first moment h and the second moment $h^{(2)}$.
For exponentially distributed packets the second moment is twice the square of the first moment i.e., ($h^{(2)} = 2h^2$). The packet- length distribution is the same at all the stations on every sub-network as well as the bridges. If a packet is sent from a station in a sub-network to a station in the same sub-network its delay can be found as E[w] + h plus the propagation delay, as described in the chapter 2. hence the delay in the sub-network can be expressed as follows

$Di = E[w] + h$ (3.4)

where Di represents the delay in the ring i.

Now if a packet is sent from one sub-network to another sub-network, its delay will simply be the sum of the delay in the source ring, delay in the destination ring and the delay in the backbone ring. If we represent the destination ring as j , then the delay Dj in the destination ring can be expressed as follows:

$Dj = E[W] + h$ (3.5)

and the delay in the backbone ring b, will be

$Db = E[w] + h$ (3.6)

Now we have all the equations to find the mean end-to-end dely. If we represents D as the mean end-to-end delay then D can be find as follows:

$$D = \sum_{\substack{i=1 \\ i\neq j \\ j=1}}^{m} (Di + Db + Dj)(Pij / m) \tag{3.7}$$

In our analytical modeling we are assuming that the acknowledgements are piggybacked and it has the same end-to-end delay as a message and the round trip delay can be defined as follows

$$Dr = 2D. \tag{3.8}$$

Throughput of a network can be defined as the arrival rates at the bridges i.e., λib in messages /sec.The power of a network can be defined as follows:

$$P = \lambda ib / Dr \tag{3.9}$$

in which P is the power of the network and Dr is the round-trip delay.

For all our assumptions in the next chapter, we will assume 4 Mbps transmission rate for local rings and 4 Mbps/ 16 Mbps for the backbone ring. The other parameters we be will find in the next chapter.

CHAPTER 4

NUMERICAL RESULTS

4.1 Assumptions

In this chapter we will present numerical results. We are assuming an average message length of 1000 bits . The arrival rates at each station are the same except for the bridges in which they depend on the inter-network traffic. The other assumptions will be as follows:

1.	Backbone ring capacity:	4Mbps and 16Mbps
2.	Local ring capacity	4Mbps
3.	Total number of interconnected rings	5 10
4.	Total number of stations connected to each ring	10 50

4.2 Results

Results from figure 11:

Figure 11 shows the graph between the throughput and round-trip delay. We have a 4 Mbps local and backbone ring capacity . It is obvious from the figure that as the probability increases from 0.03 to 0.05 , throughput also increases and this in turn increases the delay.

Results from figure 12:

Figure 11 shows the graph between the throughput and the round-trip delay. Again we have 4 Mbps local and backbone capacities . As the delay increases from 0.05 to 0.08, throughput also increases and this in turns increases the delay.

Results from figure 13 :

Figure shows the relationship between throughput and round-trip delay. Here we have 4 Mbps local ring and 16 Mbps Backbone.

Results from figure 14:

Figure 14 shows the backbone utilization verses throughput for probabilities 0.03, 0.05 and 0.08. The backbone capacity is 4 Mbps. It is obvious from the figure that the higher the probability, the sooner the backbone become full.

ROUND-TRIP DELAY Vs THROUGHPUT
4 Mbps Local Ring, 4Mbps Backbone Ring
5 Rings, 10 Stations.

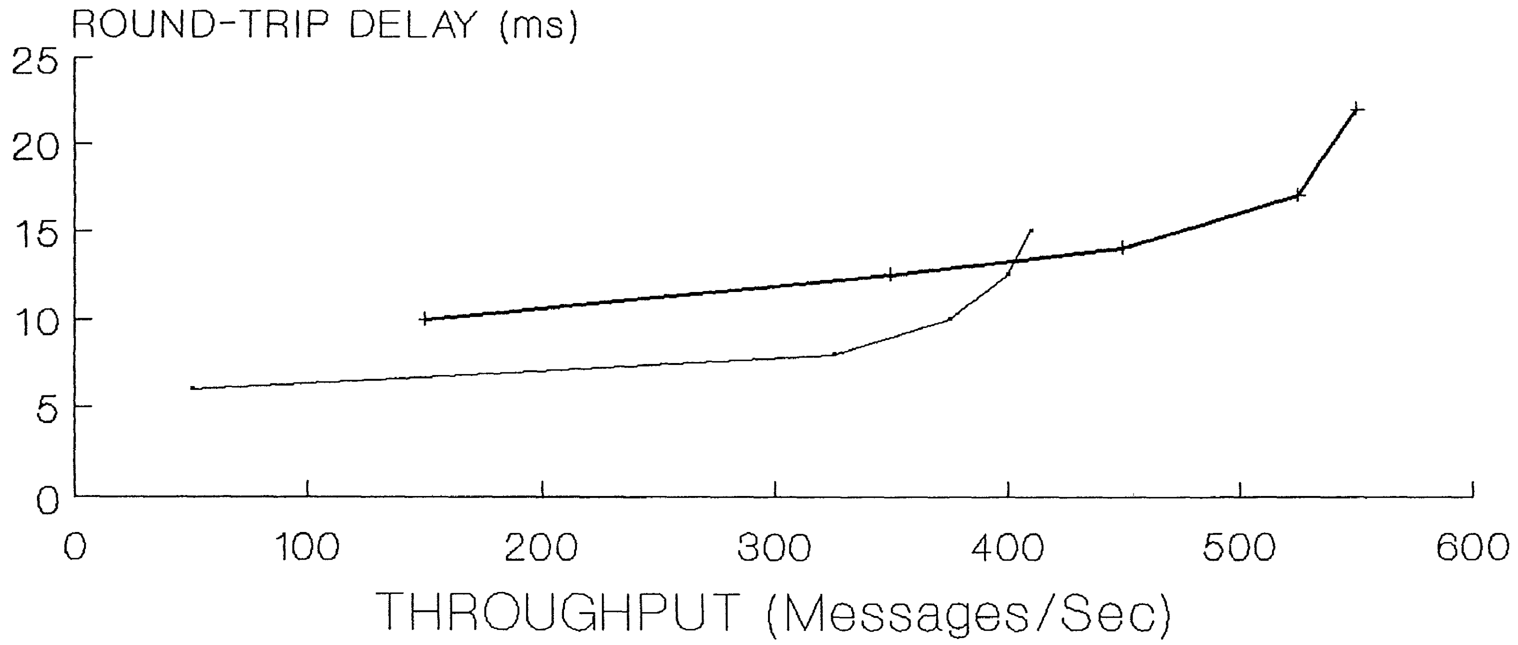


Figure 11 (pg.43)

ROUND-TRIP DELAY Vs THROUGHPUT
4 Mbps Local Ring, 4Mbps Backbone Ring
5 Rings, 10 Stations.

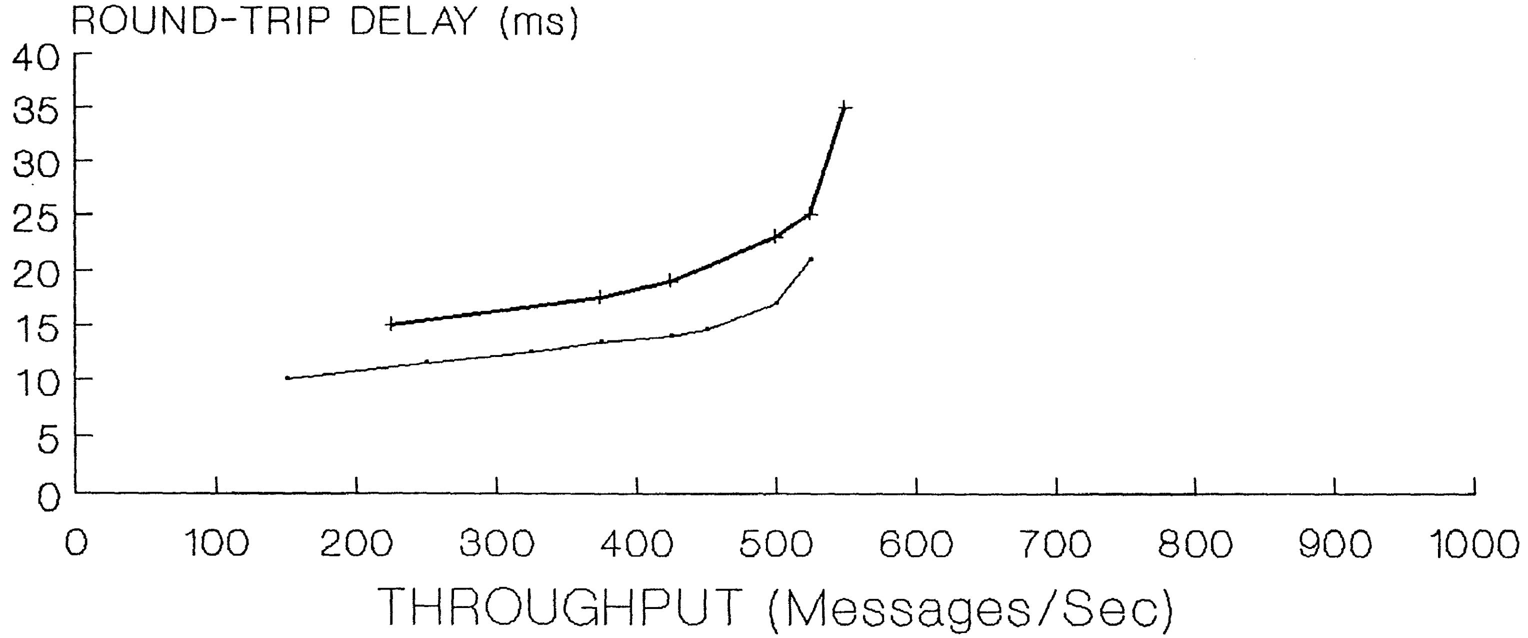


Figure 12 (Pg. 44)

PROB. 0.05 PROB.O.08

ROUND-TRIP DELAY Vs THROUGHPUT
4 Mbps Local Ring, 16Mbps Backbone Ring
5 Rings, 10 Stations.

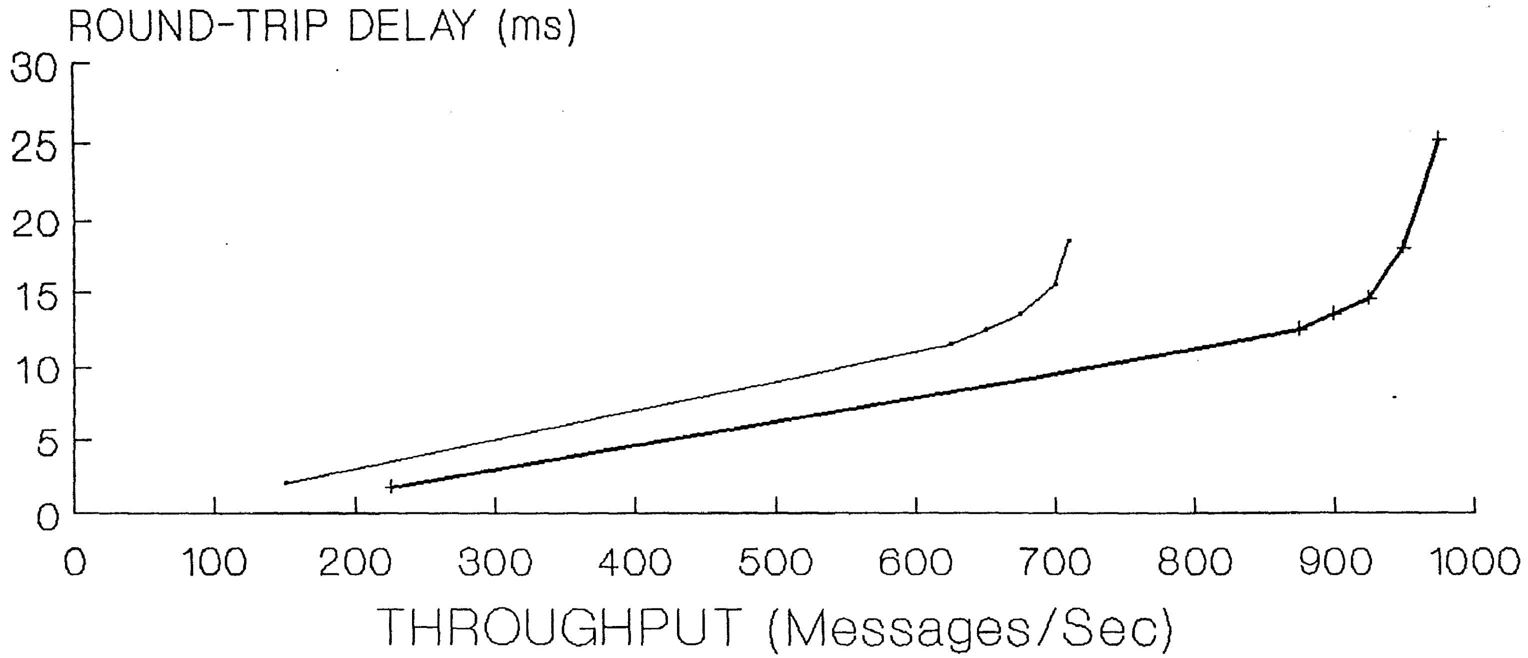


Figure 13 (Pg. 45)

PROBABILITY 0.05 PROBABILITY 0.08

Backbone Utili. Vs Throughput

4 Mbps Local Ring, 4 Mbps BB Ring

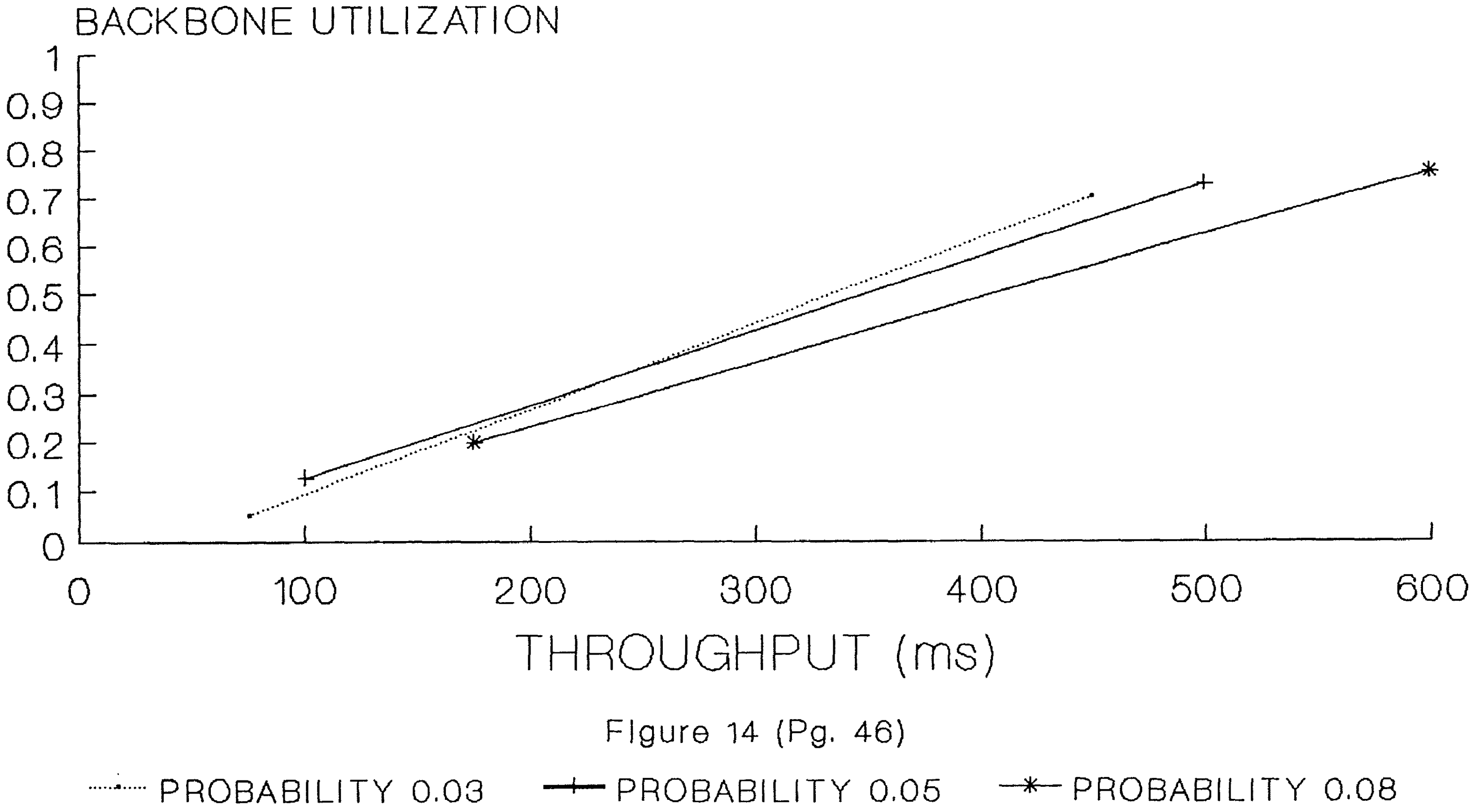


Figure 14 (Pg. 46)

PROB. .03, PROB. .05, PROB. .08

Results from figure 15:

Figure 15 shows the backbone utilization verses throughput for probabilities 0.03, 0.05 and 0.08. By comparing our results with the previous analysis we will see that using a 16 Mbps backbone is under utilized.

Results from figure 16 and 17:

Figure 16 shows the delay verses probability for the window size 1 for 4 Mbps and 16 Mbps backbone. It can be noticed from the figure that as the window size increases from 1 to 16 the utilization increases and that in turn increases the delay.

Results from figure 18:

Figure 18 shows the throughput verses window size for the probability 0.05, and for 4 Mbps and 16 Mbps backbone ring capacities . It is obvious from the figure that as the window size increases the throughput increases. By comparing the throughput with the 4 Mbps ring, it is clear that we have high throughput than in 4 Mbps backbone ring.

Results from figure 19:

Figure 19 gives the relationship between power and throughput.The local ring and backbone has the same 4 Mbps capacity. It is obvious from the figure that we have more power in 0.03 probability and less in 0.08 . This is because of the ratio between throughput and delay.

Results from figure 20:

Figure 20 gives the graph between round-trip delay and throughput. It can be noticed here that when the probability increases from 0.03 to 0.06, the 4 Mbps backbone becomes full.

Results from figure 21:

Figure 21 shows the graph between round-trip delay and the throughput. The backbone has a 16 Mbps capacity. When the probability increases from 0.03 to 0.06, there is a significant difference in the throughput.

Results from figure 22:

Figure 22 shows the graph between throughput and window size. It is obvious that as the backbone capacity increases from 4 Mbps to 16 Mbps but with the same probability , there is a significant improvement in the throughput.

Backbone Utili. Vs Throughput

4 Mbps Local Ring, 16 Mbps BB Ring

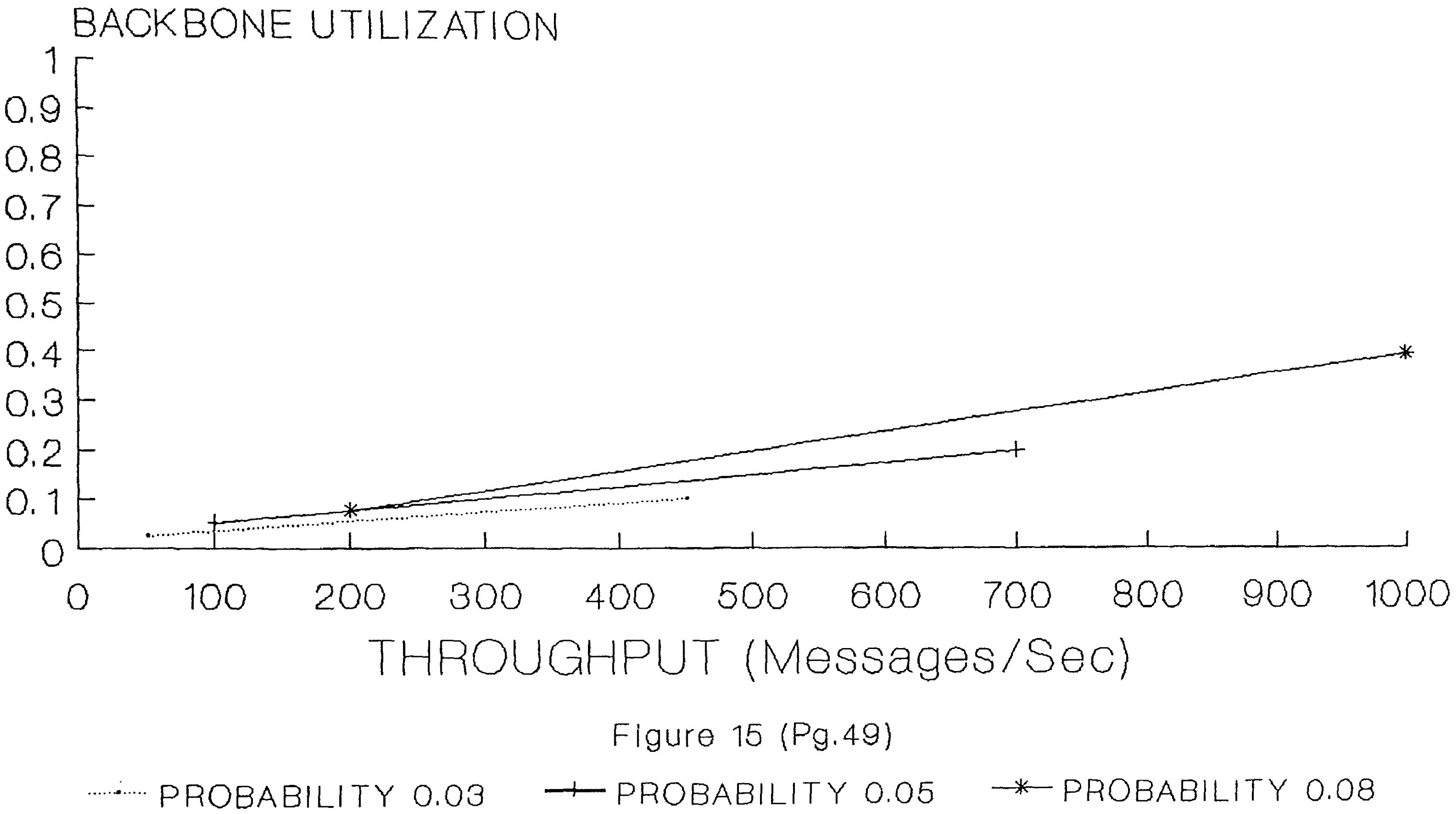


Figure 15 (Pg.49)

PROB. .03, PROB. .05, PROB. .08

ROUND-TRIP DELAY Vs PROBABILITY

5 RINGS, 10 STATIONS, 4 Mbps LOCAL RING

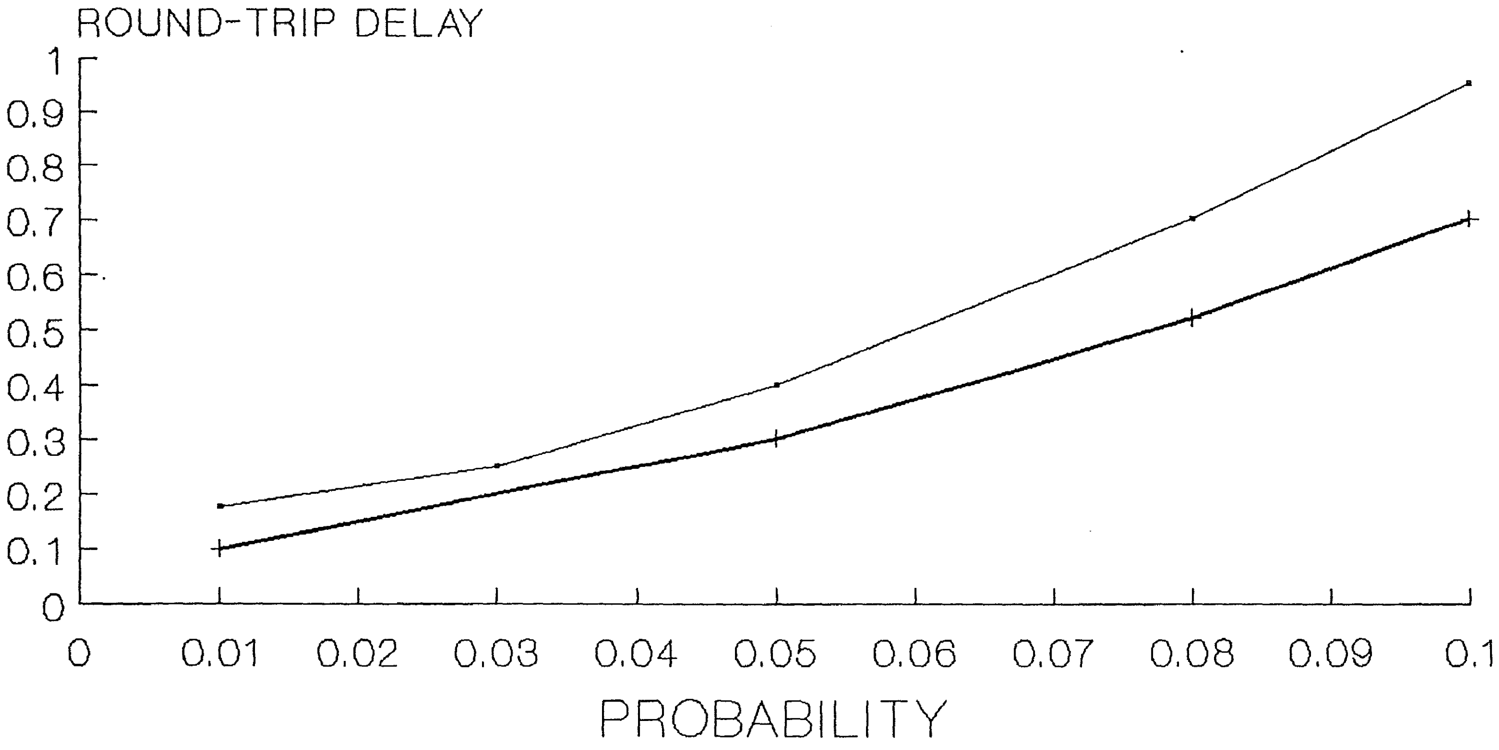


Figure 16 (Pg. 50)

WINDOW SIZE=1

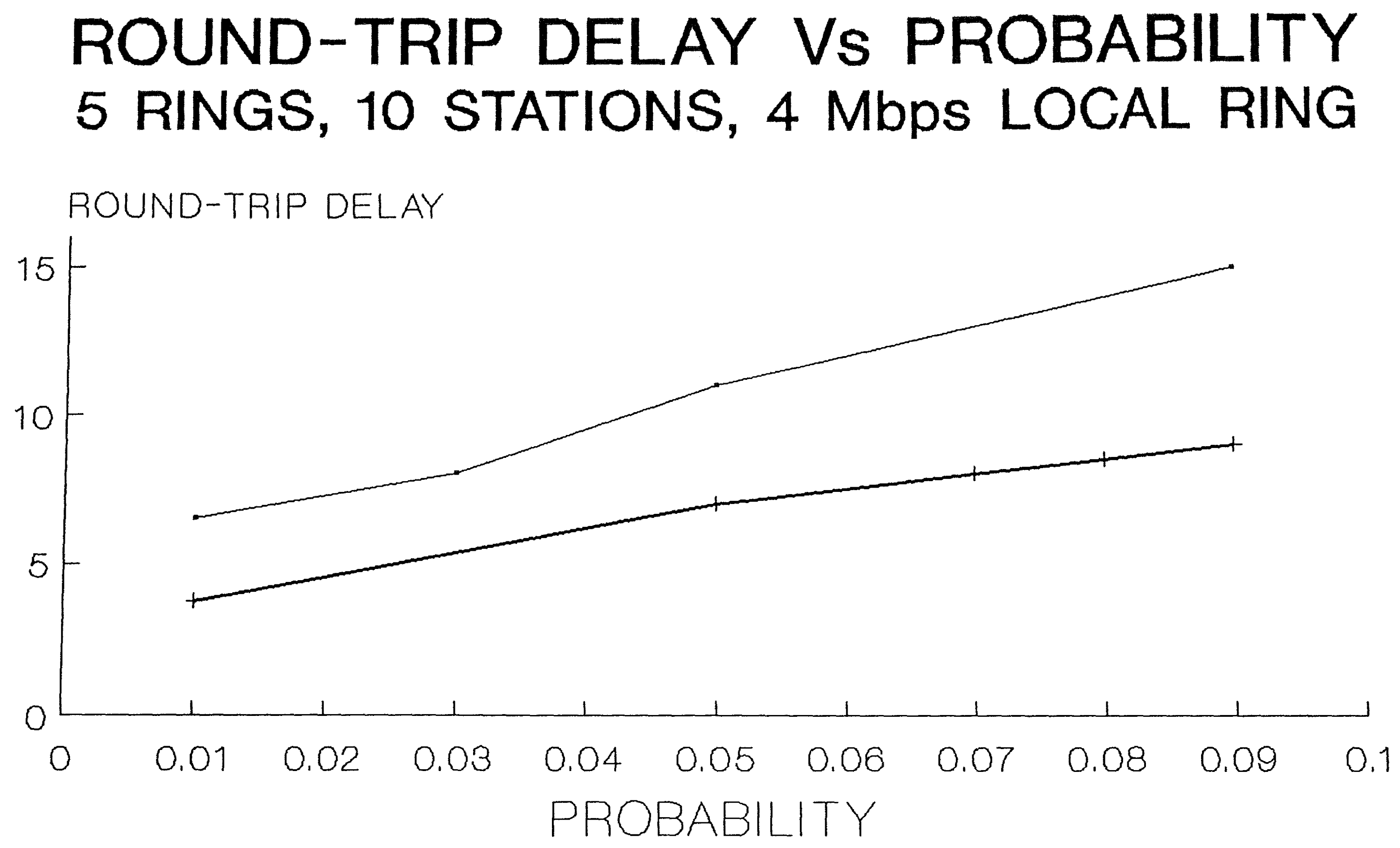


Figure 17 (Pg.51)

—•— 4 Mbps BB Ring —+— 16 Mbps BB Ring

WINDOW SIZE 16

THROUGHPUT Vs WINDOW SIZE

5 Rings, 10 Stat., 4 Mbps Ring, Prob .05

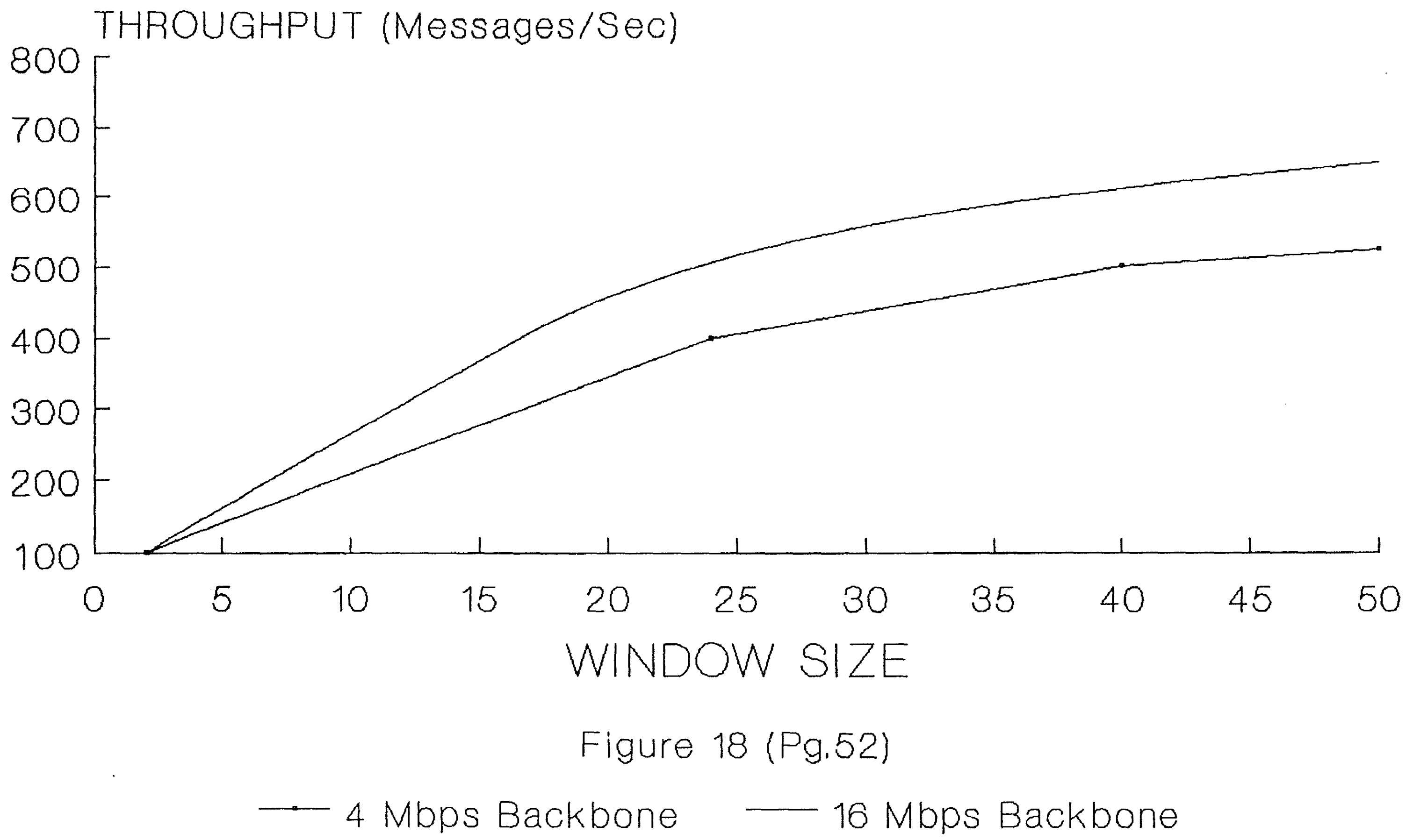


Figure 18 (Pg.52)

PROBABILITY 0.05

POWER Vs THROUGHPUT

5 RINGS, 10 STN., 4Mbps Ring 4Mbps BB

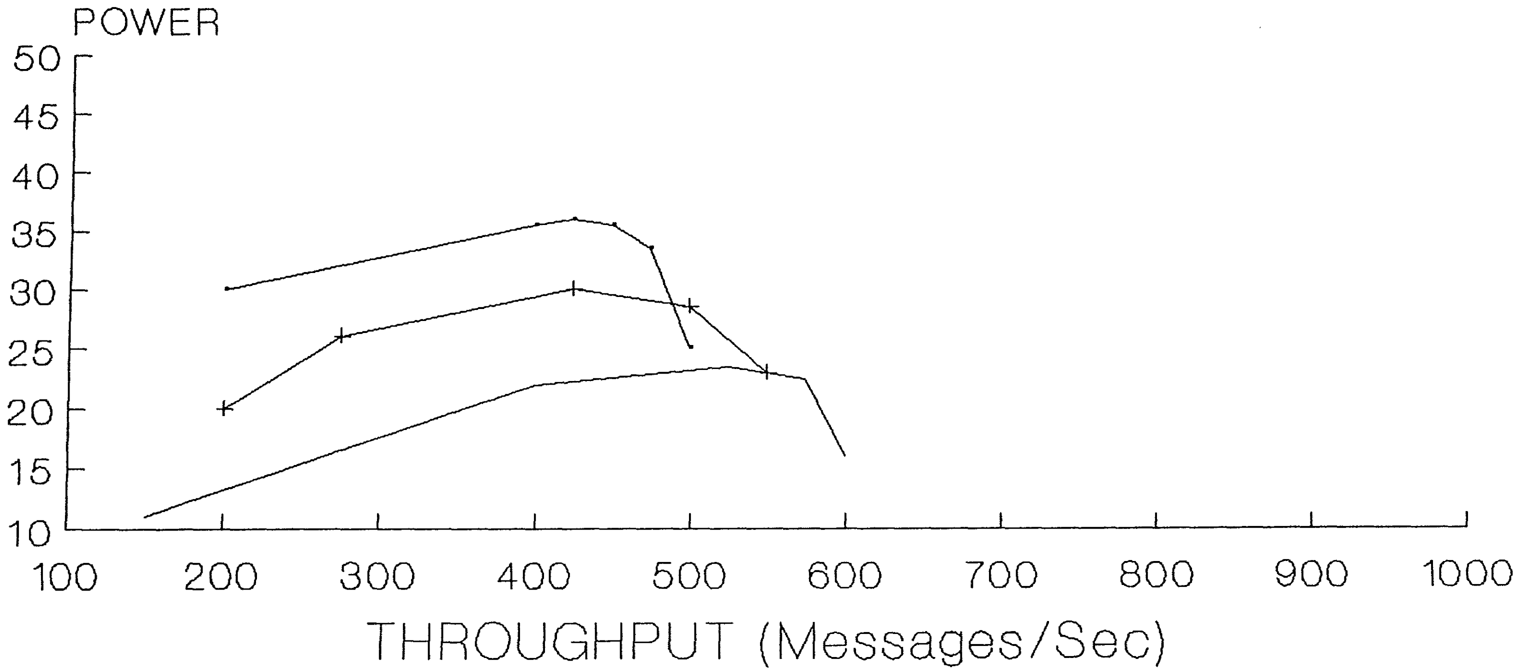


Figure 19 (Pg.53)

—·— PROBABILITY 0.03 —+— PROBABILITY 0.05 —— PROBABILITY 0.08

PROB. 0.03, PROB. 0.05, PROB. 0.08

ROUND-TRIP DELAY Vs THROUGHPUT

10 RINGS, 50 STN., 4 Mbps RING, 4MbpsBB

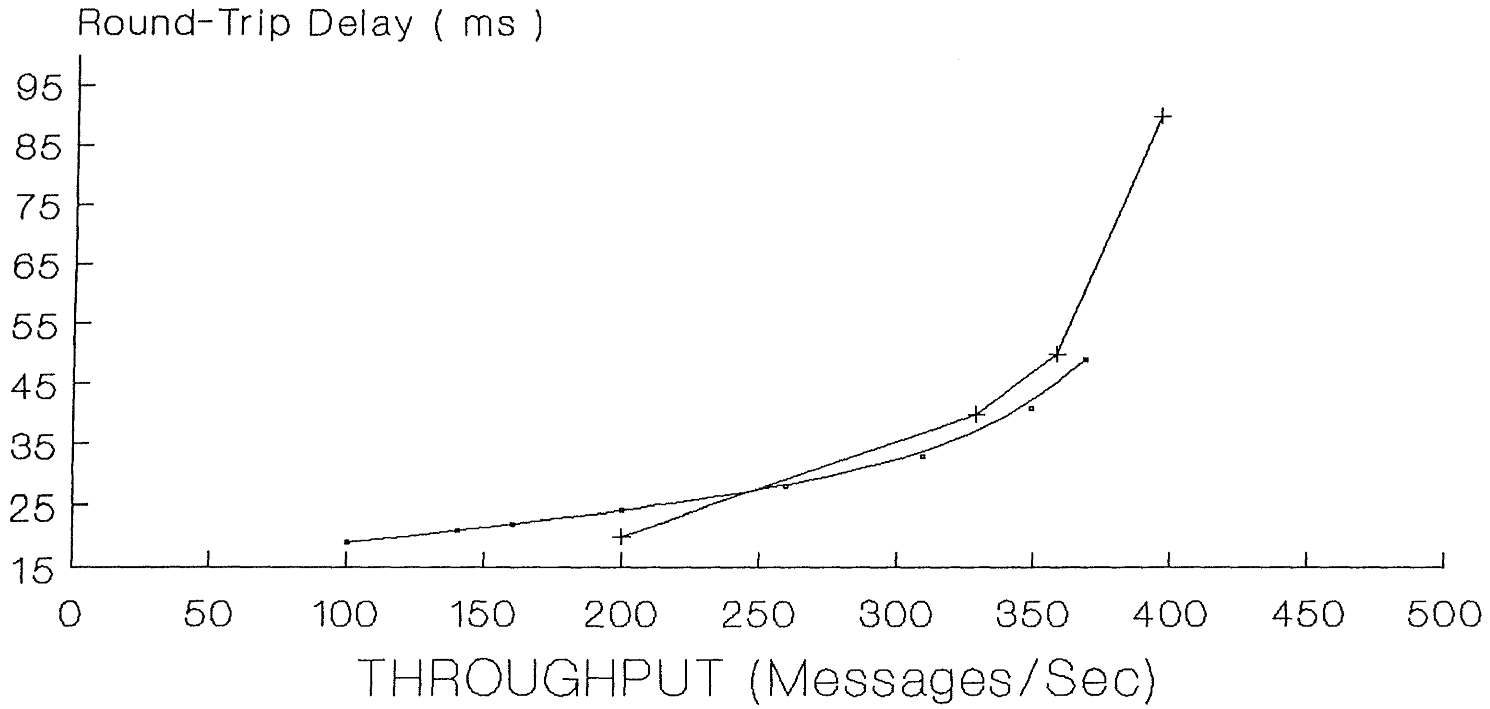


Figure 20 (Pg. 54)

PROB. 0.03 PROB. 0.06

ROUND-TRIP DELAY Vs THROUGHPUT
4 Mbps LR, 16 Mbps BB
10 Rings, 50 Stations

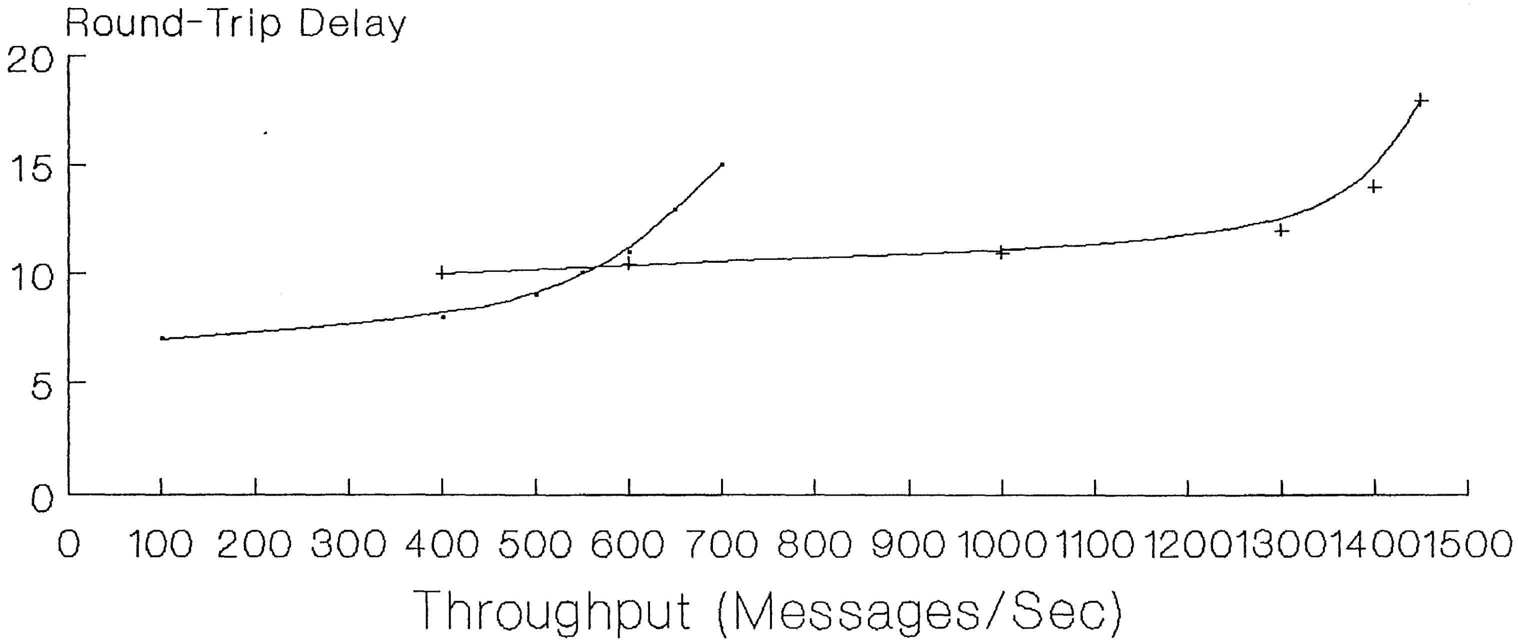


Figure 21 (Pg.55)

Prob. 0.03, Prob. 0.06

THROUGHPUT Vs WINDOW SIZE

4Mbps LR, 4Mbps BB,
10 Rings, 50 Stations

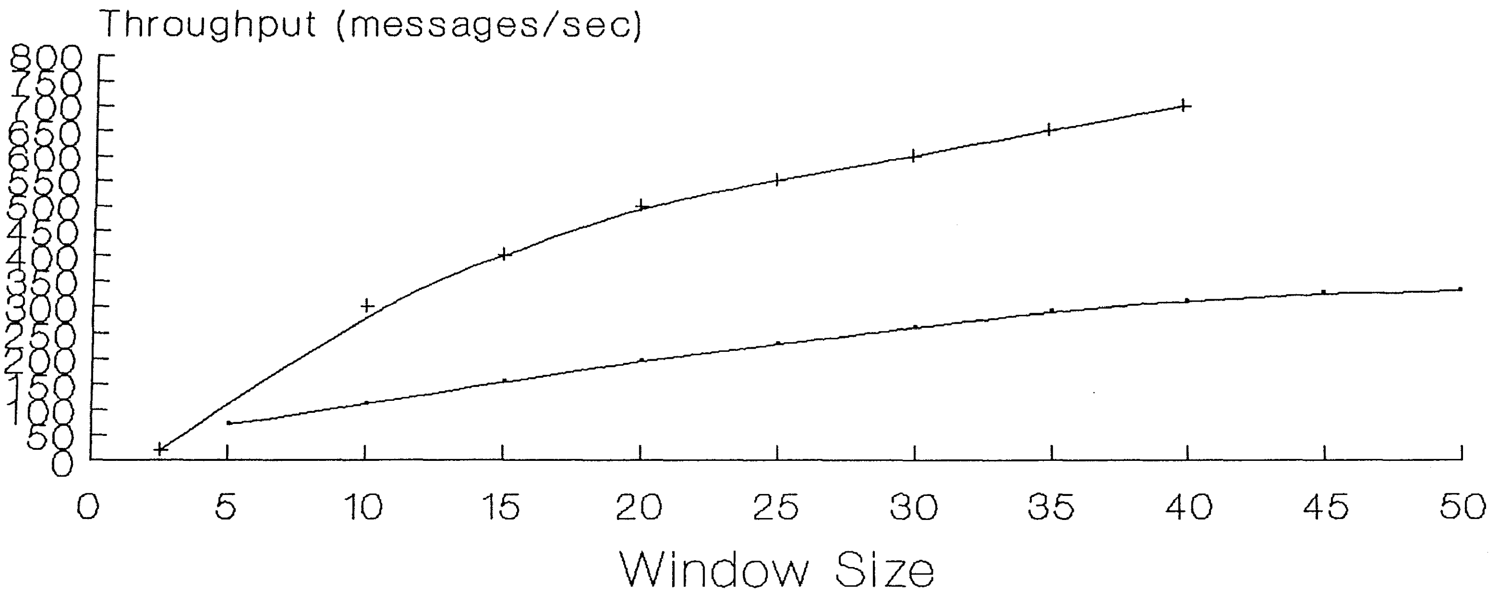


Figure 22 (Pg.56)

Probability 0.03

CHAPTER 5

CONCLUSION

During the past few years, substantial research activity has been devoted to studying the various facets of performance of token-ring based LANs. In the performance evaluation of data-communication systems, particularly local area networks, single-server model with cyclic service among an arbitrary number of queues, represent an important class of queuing system. This work was of significant help in establishing the token-ring technique as a major LAN technology. In this paper we have investigated the performance of local area networks consisting of interconnected token rings. Addition to the performance evaluation, we investigated flow control issues consisting of multiple token rings interconnected through bridges. The results of this thesis is based on the extension of Bux's and Truong's approximate analysis of token ring network, to interconnected token ring network. The key observations of this part of the study as follows:

- Analytic models of the basic-token ring operation are typically limited to poisson arrivals of single frames and very often require symmetry and independance assumptions. Analytically tractable models that lend themselves to numerical evaluation with a more realistic characterization of the traffic would be highly desirable. Equally important would be models yielding more detailed performance measures than just mean delays i. g., delay distributions.

• Congestion can be eliminated by using an appropriate set of traffic monitoring and control procedures called *flow control procedures.* One of the good control mechanism is *window flow control.* In a congested network , large window sizes can lead to severe performance degradation. On the other hand, small windows are unnecessarily restrictive and can lead to a poor performance if the network is not congested.

For the fixed window protocol, we observe that congestion generally becomes worse when the number of stations increases. In contrast , network performance is no longer sensitive to the number of stations, when the dynamic window-size is employed. These observations shed additional light onto the necessity of a dynamic flow control scheme in the multi ring networks. We conclude that the architecture should be enhanced by a suitable flow control mechanism. The proposed solution is to add a dynamic window-size algorithm.

It can be noticed from our work that a 4 Mbps backbone ring can be a bottleneck for the system if the throughput is too high and using a 16 Mbps backbone , the sub-network become the bottleneck at higher throughput. The numerical results we obtained are right but there is no guarantee about there correctness because there are no simulation results against which we could compare our results. This study suggests a simulation study of an exact system like ours.

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