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PERFORMANCE OF DEMAND ASSIGNMENT MULTIPLE ACCESS SCHEMES IN BROADCAST BUS NETWORKS

Michael Fine

Technical Report: CSL-TR-87-316

March 1987

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PLICIT tokens, i.e., control messages, are used to provide the required scheduling. Other schemes use implicit tokens whereby stations in the network rely on information deduced from the activity on the bus to schedule their transmissions. In this work, three basic access mechanisms, according to which these implicit-token DAMA schemes can be classified, are identified. We describe these three mechanisms and the various service disciplines achieved by them, together with their network topologies. We present some representative schemes for each class, discussing their performance and other important attributes. We show that these schemes overcome some of the performance limitations of existing random access schemes, making them particularly suited to the high bandwidth requirements of an integrated-services digital local network. For a more extensive and detailed evaluation of the performance of these DAMA schemes, we examine two of them, Expressnet and Fasnet, operating under various service disciplines. Throughput and delay characteristics over a range of operating conditions are presented and discussed. Lastly we propose a combined voice/data protocol suitable for DAMA broadcast networks. Such networks are well suited to the integration of voice and data since they guarantee bounded delay and provide high utilization of the channel even at high bandwidths. Using Expressnet as a representative scheme, we examine the characteristics of the service that the voice traffic experiences under this protocol. We show that the access protocol is able to utilize the channel efficiently to support a large population of voice sources while maintaining low packet delay and guaranteeing some prespecified minimum bandwidth for data traffic. In addition, we show the advantages of silence suppression, i.e., discarding speech that constitutes silent periods, and we look at the effects of overloading the network.

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Abstract

Local area communications networks based on packet switching techniques provide simple architectures and efficient and flexible operation. Various ring systems and CSMA contention bus systems have been in operation for several years. More recently, a number of demand assignment multiple access (DAMA) schemes suitable for broadcast bus networks have emerged which provide conflict-free broadcast communications by means of various distributed round-robin scheduling algorithms. In some of these schemes, explicit tokens, i.e., control messages, are used to provide the required scheduling. Other schemes use implicit tokens whereby stations in the network rely on information deduced from the activity on the bus to schedule their transmissions. In this work, three basic access mechanisms, according to which these implicit-token DAMA schemes can be classified, are identified. We describe these three mechanisms and the various service disciplines achieved by them together with their network topologies. We present some representative schemes for each class, discussing their performance and other important attributes. We show that these schemes overcome some of the performance limitations of existing random access schemes, making them particularly suited to the high bandwidth requirements of an integrated-services digital local network. For a more extensive and detailed evaluation of the performance of these DAMA schemes, we examine two of them, Expressnet and Fasnet, operating under various service disciplines. Throughput and delay characteristics over a range of operating conditions are presented and discussed. Lastly we propose a combined voice/data protocol suitable for DAMA broadcast networks. Such networks are well suited to the integration of voice and data since they guarantee bounded delay and provide high utilization of the channel even at high bandwidths. Using Expressnet as a representative scheme, we examine the characteristics of the service that the voice traffic experiences under this protocol. We show that the access protocol is able to utilize the channel efficiently to support a large population of voice sources while maintaining low packet delay and guaranteeing some prespecified minimum bandwidth for data traffic. In addition, we show the advantages of silence suppression, i.e., discarding speech that constitutes silent periods, and we look at the effects of overloading the network.

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Chapter 1

Introduction

1.1 Historical Background

At the end of the 1960's, store-and-forward packet switching technology emerged as a cost effective alternative to circuit switching techniques for providing communications to "bursty traffic" such as that generated by computer communications [1], [2].

Packet switching is based on the idea of providing part or all of the communication resource to one user at any given time but only for the amount of time that that user requires it. A store-and-forward network consists of a number of switches or *nodes*, partially connected by point-to-point links. Each user or *station* divides its messages into *packets* which are individually addressed and handled. The station passes the packets to the node to which it is connected. This node then forwards the packets to their destination stations via some intermediate nodes. Statistical multiplexing of messages from multiple source/destination pairs is achieved by queueing

the packets at the individual nodes and transmitting them when the outgoing channel becomes free. At the present time, these systems are reasonably mature and are now in widespread use.

Local area networks is a relatively new field that emerged around the mid 1970's [3]. It was intended to bring the advantages of packet switching communication technology to bear on the communications needs of the expanding in-house computer facilities. They can generally be differentiated from long-haul store-and-forward networks by the following characteristics: local area networks span a geographical area of a few kilometers such as a single building or cluster of buildings; the data rate on a local area network is high, typically 1-100Mb/s; they are usually privately owned rather than publicly available; the cost of transmission and controlling the network is low compared with long-haul networks.

1.2 Existing Local Area Network Schemes

Local area networks can be broadly categorized into two basic types. These are broadcast busses and ring systems [2], [3], [4], [5]. In ring systems the data flow is unidirectional propagating around the ring from station to station. The interface between a station and the network is an active device which receives the signal from the incoming line and retransmits it on the outgoing line. Various techniques for accessing the channel exist which give rise to various types of ring networks such as token rings, slotted rings, and register insertion rings.

In a token ring, a data structure consisting of a delimiter and a busy/free indicator, called a *token*, propagates around the ring from station to station. When a station wishes to send a packet it waits until it receives the token. If the token is free, the station sets it to busy and transmits its packet preceded by the busy

token. When the packet returns to the sender, it is removed from the ring. The sending station also has the responsibility to retransmit a new free token.

In the slotted ring, the channel is divided into a fixed number of slots which travel around the ring. A station that wishes to send a packet waits for a free slot, sets the busy/free indicator of that slot to "busy" and inserts its packet into this slot. When the packet returns to the sender it is removed from the ring and the slot is set to "free."

In a buffer insertion ring, a station transmits its packet onto the ring by inserting into the ring sufficient buffer space to accommodate the additional data. Incoming data can be placed in this buffer as the packet is transmitted out onto the ring. This buffer space is reclaimed by the station when it becomes empty due to normal inter-packet gaps. In contrast to token rings and slotted rings, in the buffer insertion ring it is the responsibility of the receiving station to remove the packet from the ring.

In [4], the maximum channel throughput or network capacity of a token ring is shown to be

$$C = \begin{cases} \frac{1}{1 + \frac{\tau W}{B}/N} & (\tau W/B) \leq 1 \\ \frac{1}{\frac{\tau W}{B}(1 + 1/N)} & (\tau W/B) > 1 \end{cases} \quad (1.1)$$

where N is the number of stations transmitting, τ is the propagation delay of the signal around the ring in seconds, W is the bandwidth of the channel in bits per second, and B is the number of bits in a packet. The ratio $\tau W/B$ is referred to as the "a factor." In Fig. 1.1, the channel capacity for a token ring is plotted versus a . We see that the channel utilization falls off rapidly for a greater than about 1. We note that ring systems always guarantee an upper bound on the packet delay.

Multiaccess broadcast bus systems have been popular since, by combining the

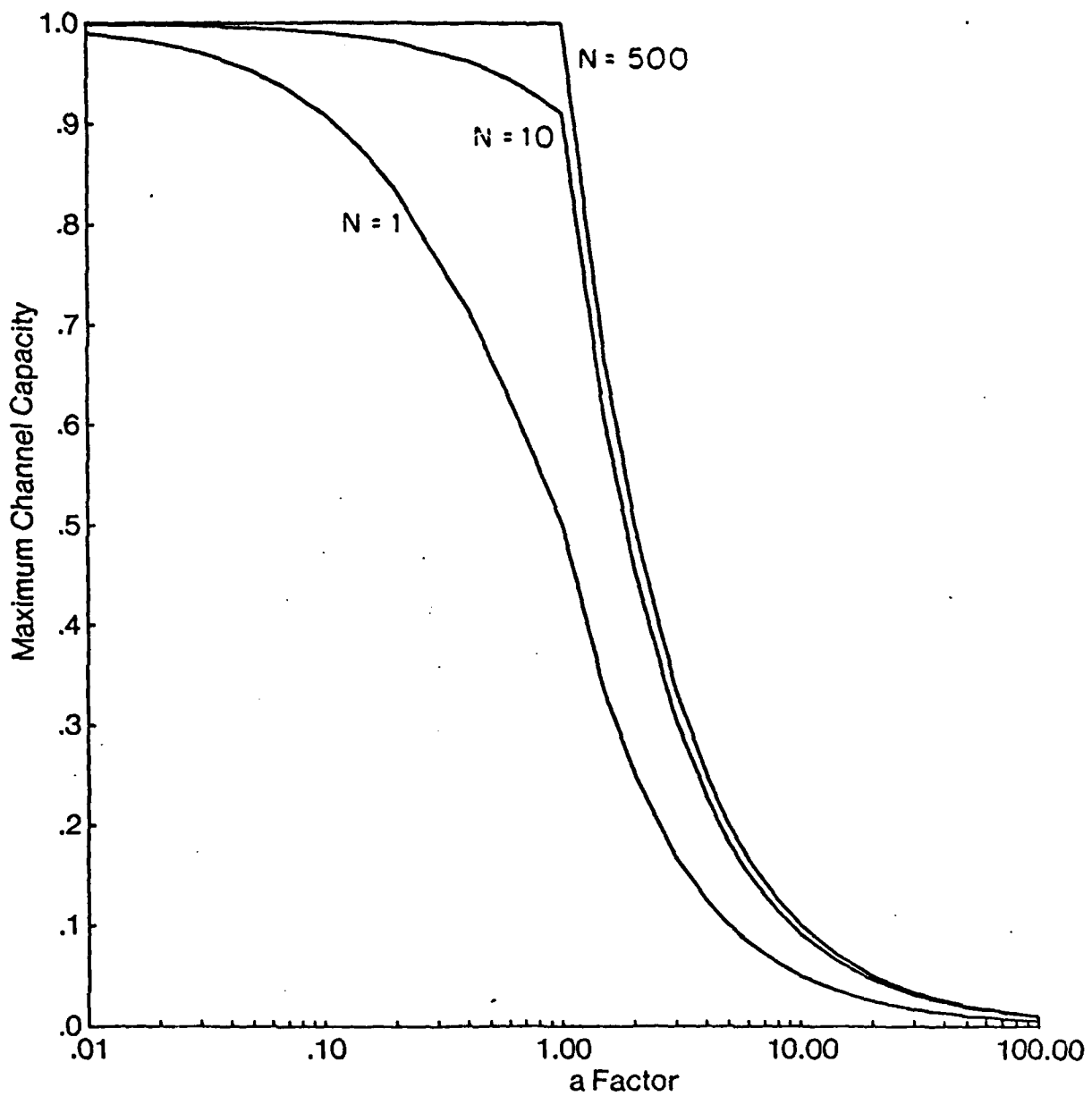


Fig. 1.1 Maximum channel capacity versus a for a token ring with various values of N .

advantages of packet switching with broadcast communications, they offer efficient solutions to the communication needs of the local area environment both in simplicity of topology and flexibility in satisfying growth. Random access methods such as Carrier Sense Multiple Access (CSMA) [6], [7] have been effectively employed. The Ethernet [8] is a common example. In these systems, a station that wishes to transmit senses the channel to see if it is currently being used. If the channel is busy, the station in question reschedules its transmission for some later time. If the channel is idle, the packet is transmitted. Note that it takes a finite amount of time for the signal to propagate across the network. Therefore, it is possible for a station to detect the channel to be idle and to begin transmission when, in fact, another station has already begun to transmit. In this circumstance, two transmissions occur simultaneously and a *collision* occurs. The efficiency of the channel can be improved if stations listen for such collisions and abort their transmissions when a collision occurs [9]. This variant of the access protocol is called CSMA with collision detection (CSMA/CD).

While reliable operation of a ring network relies on the integrity of explicit information such as a unique token, or slot boundaries and slot status, as well as on the proper operation of the active taps in relaying the packets and removing them at either the receiver or the sender, random access schemes are simple to implement, robust, and are considered more reliable than ring networks since the taps and medium used are generally passive. However, due to the *random conflicts*, a fraction of the bandwidth is wasted and packet delay is unbounded. Moreover, they do exhibit performance limitations particularly when the channel bandwidth is high or the geographical area to be spanned is large. For example, in [9], [11] it has been shown that the maximum channel throughput for the infinite population

slotted CSMA/CD scheme is given by

$$S = \begin{cases} \frac{1}{1 + Ka} & a \leq 0.5 \\ \frac{1}{(2 + K)a} & a > 0.5 \end{cases} \quad (1.2)$$

where K is a constant (in the neighborhood of 3 to 6) which depends on the particular version of the protocol, and as for rings, $a = \tau W/B$ where τ is the end to end propagation delay of a signal across the network in seconds, W is the bandwidth of the channel in bits per second and B is the number of bits in a packet. In Fig. 1.2, the channel utilization for the slotted non-persistent CSMA/CD protocol with infinite population is plotted versus a . The behavior is similar to that experienced with token rings. We see that, for CSMA/CD, the channel utilization falls off rapidly for a greater than about 0.1. Any effort to push the technology to higher data rates with the hope to achieve a network throughput proportional to the increase in channel bandwidth is unfortunately rewarded only by a marginal improvement.

Local area networks have registered significant advances in recent years. Currently many ring and random access broadcast bus networks are commercially available and these networks have proven to work well in the environments for which they have been designed, i.e., operating in the 1-10Mb/s range and spanning a couple of kilometers. However, as the communication network is required to service a larger number of stations and as new services such as voice, facsimile and perhaps video are integrated onto a single network, more stringent requirements in terms of high bandwidth and real-time response will be placed on the network. It is clear that the existing broadcast bus schemes will not perform adequately in the new environments. Thus, there is the need to look beyond existing schemes for new methods and techniques suitable for future local area network requirements.

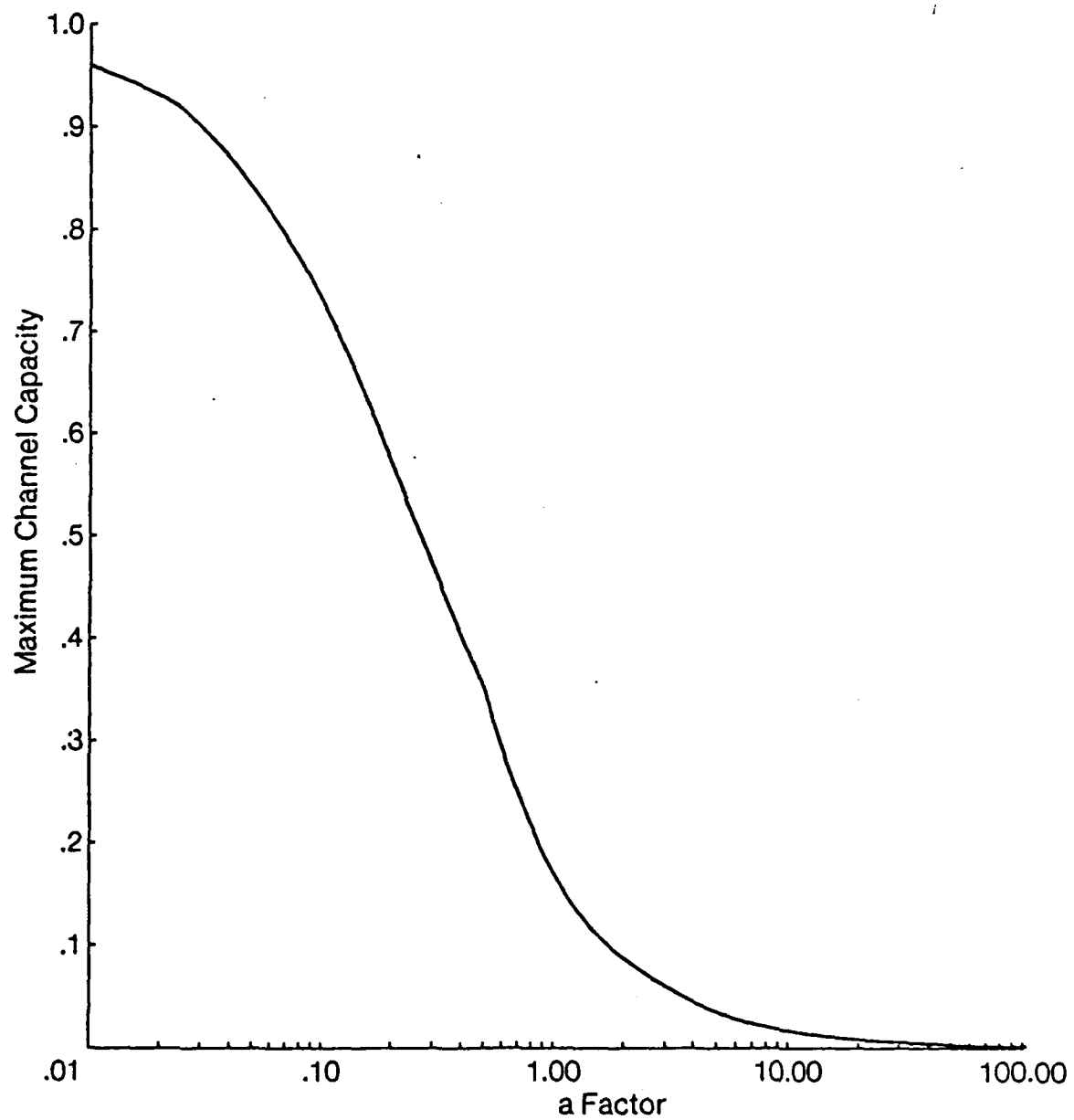


Fig. 1.2 Maximum channel capacity versus a for CSMA/CD with infinite population.

1.3 New Proposals and Trends for Local Area Networks

Recently, a number of new schemes have been proposed for broadcast bus networks. These schemes provide conflict-free transmission using distributed access protocols with *round robin scheduling functions* which thus lead to bounded delay. The stations that are "alive" are ordered so as to form what is called a *logical ring* according to which they are given their chance to transmit. We refer to these network proposals as demand assignment multiple access (DAMA) schemes. In some of these schemes, such as the Token-Passing Bus Access Method [15] and the Sound Off Control Scheme [16], an *explicit message* gets sent around the logical ring to provide the required scheduling; the station holding the token at any instant is the one that has access to the channel at that instant. It relinquishes its right to access the channel by transmitting the token to the next one in turn. Unfortunately, as in rings, the robustness of these networks depends on the integrity of the token and on the proper operation of the involved stations. In addition, the performance degrades significantly with α [4].

In contrast to those schemes where a station transmits an explicit token to the next in turn, in others the stations rely on various events due to activity on the channel to determine when to transmit. Since the token passing operation is *implicit*, the overall robustness of the network is improved over token bus networks. Here too, packet delay is bounded; but in addition both throughput and delay can be made much less sensitive to α , thus rendering these schemes particularly suitable to networks with high bandwidth, small size packets (such as those arising from real time applications), and long distances.

While packet switching has become well established as a means of providing efficient communications for bursty traffic characteristic of computer systems, there has also been considerable interest in the application of such networks to the

communication requirements of stream traffic, such as digitized speech [17], [18], [19]. In particular, the integration of voice and data onto a single local area network has received considerable attention [20], [21], [22], [23], [11], [24]. There are a number of reasons for this: the features of local area networks, such as their simple architectures and flexible operation, can be exploited for voice communication systems; a single distributed communication system can be used to satisfy the communication needs of both data and voice traffic (and, in the future, other types of communications that services may require); one can more easily implement services that depend on both data and voice communication systems, such as a voice mail facility; the rapidly declining costs of digital electronics and processing capability makes distributed digital communications not only feasible but cost effective.

The communication requirements of voice traffic are significantly different from those of data traffic. Data sources usually, require an error free transmission channel as well as some minimum bandwidth. Voice sources, on the other hand, require some bound on the delay of the speech signal, i.e., the time taken to deliver a speech sample to the listener after the instant at which it is generated, since real time speech that is excessively delayed is worthless, and they require continual access to the channel once a connection has been made. However, they are able to tolerate some corruption or loss of the speech signal with unnoticeable or little degradation of the intelligibility of the received speech signal.

Demand assignment multiple access networks are particularly well suited to the integration of voice and data traffic since they provide round robin service thus leading to bounded packet delay. In addition, as exemplified by the performance of Expressnet and Fasnet to be discussed in chapter 3, many of these DAMA schemes provide high utilization at high bandwidths which is required if one wishes to support a large number of voice sources while at the same time providing sufficient

bandwidth for data traffic and other services.

1.4 Contributions of This Work

The purpose and usefulness of this work is to investigate these implicit token DAMA schemes and evaluate their performance. We will show that, if they are designed correctly, they can achieve close to full utilization of the channel at arbitrarily high bandwidths and, due to the round robin nature of the service, are well suited to satisfy the demands of real time traffic.

This investigation consists of three major thrusts. In chapter 2 we present a number of implicit token DAMA schemes. Most of these implicit-token DAMA schemes have been proposed independently and, from reading their descriptions, may appear to be unrelated. However, with careful examination, basic commonalities can be identified. These commonalities as well as the unique features of each scheme are made explicit by presenting the schemes in a unified manner. It is possible to identify three basic access mechanisms according to which these schemes can be classified. These are the *scheduling-delay access mechanism*, the *reservation access mechanism* and the *attempt-and-defer access mechanism*. In this chapter, we describe in general terms these three access mechanisms, their underlying network topologies, and several different conflict free round robin service disciplines that can be identified. We present an in depth analysis of the schemes belonging to each of the three classes, highlighting their similarities and differences.

In chapter 3 we examine two of these DAMA schemes, namely Expressnet [10], [11] and Fasnet [12], [13], and present a detailed analysis of their performance in terms of their throughput/delay characteristics. In these schemes, the access overhead per packet in a round is independent of both the end-to-end propagation

delay and the number of stations connected to the network. Due to this feature, they overcome some of the performance limitations of random access schemes, token rings and busses, as well as the less efficient of the implicit token DAMA schemes discussed in chapter 2. In this chapter, we describe a mathematical model for these systems followed by the analysis for each of the service disciplines that can be achieved by Expressnet and Fasnet. Thereafter, we discuss numerical results showing the performance of these systems.

In chapter 4 we present and evaluate a combined voice/data access protocol for Expressnet. While we use Expressnet as the representative scheme, the protocol is suitable for any network where stations are serviced in a round robin fashion and where variable size packets can be accommodated. We describe the characteristics of voice traffic and the method by which voice and data are integrated on Expressnet. We present an analysis of the system which allows us to examine the performance achieved by voice traffic and its effect on the bandwidth available to data stations. First we analyze a system where there is no silence suppression, i.e., the discarding of speech that constitutes silent periods in the speech signal, and with a constant load from data stations; such a system is a deterministic one. Then we analyze a system with silence suppression; such a system is stochastic. Thereafter, we present some quantitative numerical results for the performance.

Chapter 2

Classification of Implicit Token DAMA Schemes

Each of the DAMA schemes considered in this work provides network access via a distributed conflict-free round-robin scheduling function without the use of explicit tokens. As stated in chapter 1, these schemes can be grouped into three subsets according to the basic mechanisms used in accomplishing the objective; these are the *scheduling delay access mechanism*, the *reservation access mechanism*, and the *attempt-and-defer access mechanism*. In this chapter, we begin with some preliminaries that describe the various network topologies and various service disciplines that can be achieved by these DAMA schemes. Thereafter, we describe in general terms the three basic access mechanisms followed by a discussion of the various schemes grouped according to their basic access mechanisms. For clarity we avoid a chronological presentation, but rather the schemes are described in an order which allows a logical development thereof and a clear understanding of their features.

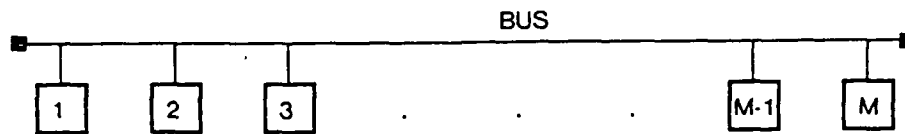


Fig. 2.1 Topology of the bidirectional bus system.

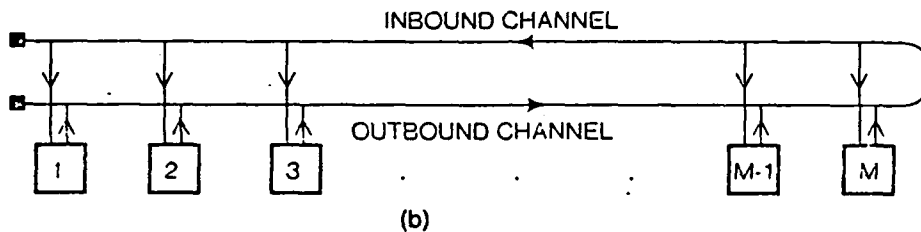
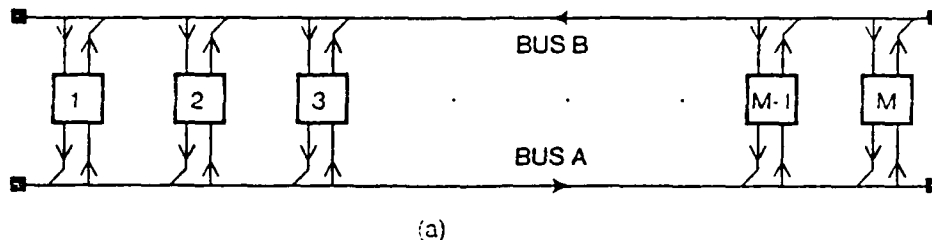


Fig. 2.2 Two configurations of the unidirectional bus system.

2.1 The Network Configurations

In presenting these basic mechanisms, three distinct broadcast bus network configurations can be identified. The first is the *bidirectional bus system* (BBS) in which, as with Ethernet, the signal transmitted by a station propagates in both directions to reach all other stations on the bus. (See Fig. 2.1.) The second is the *unidirectional bus system* (UBS) in which the transmitted signal propagates in only one direction. In this case, broadcast communications can be achieved in various ways. One way is by means of two unidirectional busses with signals propagating in opposite directions as shown in Fig. 2.2(a), so as to provide each station with

a direct path to every other station. Another way is to fold the cable onto itself (or to use a separate frequency channel in the case of broadband signaling) so as to create two channels, an outbound channel onto which the users transmit packets and an inbound channel from which users receive packets, and such that all signals transmitted on the outbound channel are repeated on the inbound channel. (See Fig. 2.2(b).) The third configuration is the *bidirectional bus with control* (BBC) which consists of a bidirectional bus along with an auxiliary control wire used to control the allocation of the bus.

2.2 The Various Service Disciplines

All of the DAMA schemes provide conflict free transmission with round robin scheduling functions. A round is defined by a time interval during which each station on the network is given the chance to transmit a single packet. The order in which stations are serviced within a round is not the same for all networks, and depends on the particular access protocol as well as the instants at which stations become backlogged. We identify three variants of round robin service which are described below. These are *Non-Gated Sequential Service* (NGSS)[†], *Gated Sequential Service* (GSS) and *Head Of Line Service* (HOLS) [25]. These three service disciplines are extensively analysed for their throughput/delay characteristics in the next chapter. In this chapter, we use these service disciplines as yet another identifying characteristic that will be useful in highlighting differences in the various access protocols.

NGSS is a "conventional" round robin discipline where stations are serviced in a predetermined order. If a station has no message to transmit when its turn comes up, it declines to transmit and then must wait for the implicit token to be

passed among all the other stations before getting another turn. GSS also achieves sequential service from round to round in the same predescribed order. However, with this service discipline, only those stations that are backlogged at the beginning of a given round are serviced in that round. For an access scheme to achieve GSS, a station, upon becoming backlogged, must wait for the beginning of the next round before attempting to access the channel. In HOLS, stations are ordered by some means such as by their index numbers, or their relative locations on the control wire or on the unidirectional bus. With this service discipline, the next station to transmit, after a given transmission, is the one that is at the head of the line, according to the fixed order, and is backlogged but has not yet transmitted in the current round. In some schemes, the sequence in which stations are offered service within a round is always sequential according to some order, however, this order is reversed from round to round. This we call *reversing sequential service*. When applied to NGSS it leads to *Non-Gated Reversing Sequential Service* (NGRSS) and when applied to GSS leads to *Gated Reversing Sequential Service* (GRSS).

2.3 The Basic Access Mechanisms

We now describe the three different basic access mechanisms. We consider that there are M stations in the network. We assume the stations to be numbered 1 through M . As it will be apparent in the sequel, for some schemes this numbering is a requirement and is explicitly made use of in the algorithm, while for other schemes it merely serves the purpose of clarity in presentation. We shall denote by S_i the station with index i . Furthermore, a station which has a message to transmit is said to be *backlogged*. Otherwise, it is said to be *idle*.

2.3.1 The Scheduling Delay Access Mechanism

This class is suitable for the BBS configuration where the only means for coordinating the access of the various users following the end of a transmission is by staggering the potential starting times of these users. More specifically, each station is assigned a unique index number. These indices form a logical ring which determines the order in which stations are allowed to transmit. Included with each transmission is a field for the index number of the sending station. Let S_i be the station currently transmitting. Let $EOC(i)$ denote its *end-of-carrier*. Following the detection of $EOC(i)$, station S_j assigns itself a *scheduling delay* $H_j(i)$, function of both i and j , according to which it schedules its potential transmission. $H_j(i)$ is sufficiently long such that, if at least one of the stations with indices between S_i and S_j is backlogged, then that backlogged station which is the next in sequence following S_i would have begun to transmit its packet and would have been detected by S_j before the scheduled transmission time of S_j , thus resulting in a round-robin scheduling. The network schemes considered in this work that use this access mechanism are BRAM [26], MSAP [27], [28], SOSAM [29], [30], BID [31], Silentnet [32], and L-Expressnet [33].

2.3.2 The Reservation Access Mechanism

This class is mainly suitable for the BBC configuration in which the stations use the control wire to place reservations and to reach a consensus on the next station to transmit, prior to the transmission on the bus. The next station to transmit is determined according to some measure, such as the relative positions of the stations on the network, or their addresses. Examples of such schemes are DSMA [34], [35], and the control wire system [36], [37]. The reservation access mechanism can also be implemented on a UBS configuration. For robustness purposes, reservations

consist of unmodulated bursts of carrier. These are transmitted on the same bus interleaved with packet transmissions. Consensus here can be reached due to the ordering of the stations, implied by the unidirectionality in transmission and the stations' positions on the bus. An example of this is UBS-RR [39], [40].

2.3.3 The Attempt-and-Defer Access Mechanism

This mechanism can only be implemented on UBS configurations where there is an implicit ordering of the stations. Using this access mechanism, a station wishing to transmit waits until the channel is idle. It then begins to transmit thus establishing its desire to acquire the channel. However, if another transmission from upstream is detected, then this station aborts its transmission and defers to the one from upstream. The upstream transmission is therefore allowed to continue conflict-free. In slotted systems, a station wishing to transmit waits for the next slot to arrive, and subsequently asserts its desire to transmit in that slot by marking the slot full. However, if the station finds that the slot has already been marked full, it defers and waits for the next slot. Examples of network schemes that use the attempt-and-defer access mechanism are Expressnet [10], [11], D-Net [41], Fasnet [12], [13], U-Net [42], Token-Less Protocols [43], MAP [44], CSMA-DCR [45], and Buzznet [46].

2.4 Definitions and Model

For all schemes, we consider that the bandwidth W is the same, and that all packets contain a fixed number of bits, B , giving a constant packet transmission time equal to $T = B/W$. In asynchronous schemes the transmission of each packet

is preceded by the transmission of a preamble needed for receiver synchronization. The transmission time of such a preamble is denoted by Ω . We consider that it takes a nonzero amount of time Δ for a station to detect the presence or absence of carrier on the bus. We also consider that, for the scheduling delay access schemes, it takes a nonzero amount of time Φ for a station to decode the index of the last station to have transmitted and to compute the scheduling delay. Due to the different amount of computation involved, it is possible that Φ takes on different values for different schemes. While τ denotes the maximum bus end-to-end propagation time, we let $\tau_{i,j}$ denote the signal propagation time between S_i and S_j . Normalizing time to T , we let $\alpha \triangleq \tau/T$, $\delta \triangleq \Delta/T$, $\omega \triangleq \Omega/T$, and $\phi \triangleq \Phi/T$. In all schemes, there is an overhead incurred in the transfer of access right from one station to the next backlogged station in sequence. The amount of overhead associated with each scheme has a primary effect on the performance of that scheme. To keep the performance evaluation simple and yet be in a position to adequately compare the various schemes, we consider the situation in which a subset of stations of size N , $N \leq M$, is continuously backlogged, whereas the remaining $M - N$ users are idle. The case $N = M$ is referred to as the *heavy traffic* condition. The performance of a scheme is given in terms of the *channel utilization* $C(M, N)$ representing the fraction of time spent in packet transmission (as opposed to synchronization and protocol overhead), and in terms of the *packet delay* $D(M, N)$ for the head of the queue at each station, (that is, the time separating two consecutive packet transmissions from the same station). From these results one could also derive the network capacity as well as a bound on delay, by considering the heavy traffic condition.

In the following sections we present the various schemes grouped according to their basic access mechanisms. We discuss their similarities and differences, and examine their performance.

2.5 Schemes Using the Scheduling-Delay Access Mechanism

In this section we describe those schemes that use the scheduling delay technique as their basic access mechanism. They differ according to (i) the way the delay function $H_j(i)$ is computed, (ii) the extent to which the scheme is distributed, that is, the extent to which it makes use of particular stations to perform specific functions, (iii) the need for (or lack thereof) a correspondence between the indexing of the stations and their relative positioning on the bus, and (iv) the performance achieved.

2.5.1 BRAM (Chlamtac, Franta, Levin, 1979) [26]

Stations are indexed arbitrarily, independent of their positions on the cable. Furthermore stations are assumed to have no knowledge of their respective positions, nor of the distances that separate them. As indicated in section 2.3, the delay $H_j(i)$ should be sufficiently long such that, if any station with index k , $i < k < j$, happened to transmit, then station j would be in a position to detect that transmission prior to its scheduled time. For BRAM, $H_j(i)$ is given by

$$H_j(i) = (2\tau + \Delta)[(j - i + M - 1) \bmod M]. \quad (2.1)$$

We now show the correctness of (2.1). To illustrate how events occur on the network, we consider *time-space diagrams* in which the vertical axis represents distance along the network, and the horizontal axis represents time increasing from left to right. (See Fig. 2.3.) The dots represent the *origin* of an event, such as the beginning of carrier (BOC) or end of carrier (EOC) for a transmission. The diagonal lines emanating from each dot represent the *time-space locus* of that event.

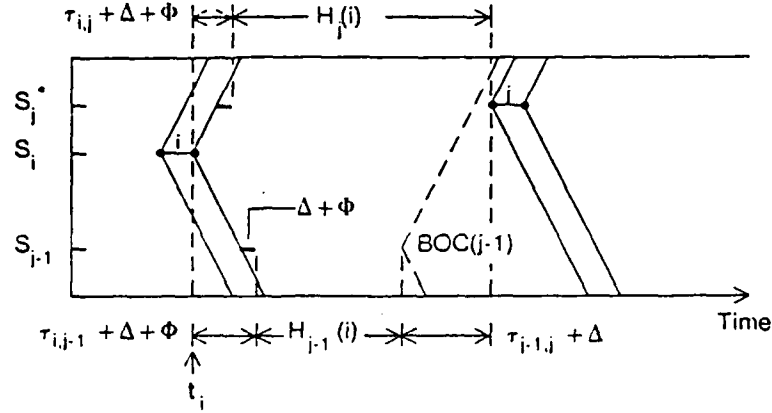


Fig. 2.3 Time-space diagram showing the recursive nature of the computation of $H_j(i)$ in BRAM.

Consider a transmission from S_i . Clearly, S_{i+1} , being the next in turn,* may transmit immediately when it detects $\text{EOC}(i)$; hence $H_{i+1}(i) = 0$. In the general case $H_j(i)$ can be computed recursively in terms of $H_{j-1}(i)$ and the end-to-end propagation delay τ . Suppose that $\text{EOC}(i)$ is generated by S_i at time t_i . As can be seen from the time-space diagram in Fig. 2.3, S_{j-1} will have detected $\text{EOC}(i)$ and evaluated $H_{j-1}(i)$ at time $t_i + \tau_{i,j-1} + \Delta + \Phi$. Were S_{j-1} to transmit, it would do so after a scheduling delay $H_{j-1}(i)$, and $\text{BOC}(j-1)$ would be detected by S_j at time $(t_i + \tau_{i,j} + \Delta + \Phi) + H_{j-1}(i) + (\tau_{j-1,j} + \Delta)$. To guarantee collision free transmissions, S_j must schedule its transmission for a time later than this time. Since S_j detects the synchronizing event $\text{EOC}(i)$ at time $t_i + \tau_{i,j} + \Delta$ and takes Φ s to compute $H_j(i)$, we require that

$$H_j(i) \geq H_{j-1}(i) + \tau_{i,j-1} + \tau_{j-1,j} - \tau_{i,j} + \Delta. \quad (2.2)$$

Without the knowledge of exact propagation times between consecutively indexed

*More precisely, the next station in turn is $S_{(i \bmod M)+1}$. However, this notation is cumbersome. So for the purpose of the discussion we will ignore the modulo nature of the index numbers.

stations, $H_j(i)$ is computed by using the maximum value possible, that is, the bus end-to-end propagation delay τ . Under this condition, the inequality above becomes $H_j(i) = H_{j-1}(i) + (2\tau + \Delta)$ and the scheme will accommodate all possible layouts. This recursive expression is easily expanded to give the scheduling delay function for BRAM as in (2.1). Note that the detection time Δ must be accounted for in the computation of $H_j(i)$. The processing time Φ , however, which is incurred by all stations following the detection of EOC, does not affect $H_j(i)$ and thus need not be accounted for in its computation.

As stations are given their turn according to the sequence determined by their indices, the service discipline is NGSS. Given that a subset of N out of M stations is continuously backlogged, a round can be defined as the time since the start of transmission of some station in the backlogged subset until the next start of transmission by that same station. The round length is equal to the cumulative packet transmission times of all stations in the round, $N(T + \Omega)$, plus the cumulative channel overhead incurred in the round. Denoting the latter by $Y(M, N)$, the channel utilization is then given by

$$C(M, N) = \frac{NT}{N(T + \Omega) + Y(M, N)}. \quad (2.3)$$

The packet delay, as defined in section 4, is simply the total length of the round,

$$D(M, N) = N(T + \Omega) + Y(M, N). \quad (2.4)$$

While $H_j(i)$ is, by design, independent of the relative physical locations of the stations, the exact timing of the transmissions on the channel and the associated overheads are not. This is the case because the time until the next transmission following an EOC is based on the time at which that EOC is detected by the next backlogged station in sequence. To compute the overhead associated with

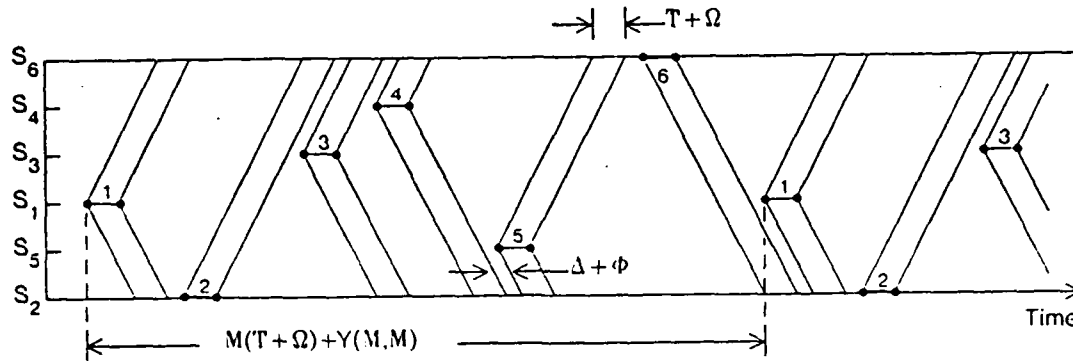


Fig. 2.4 Time-space diagram showing the activity on the channel for a typical six station BRAM network under heavy traffic conditions.

this scheme, which in turn allows us to evaluate the performance, we consider in Fig. 2.4 a time-space diagram for a network with six stations operating under the *heavy traffic* assumption. Thus, we see a round beginning with the transmission of S_1 and ending with the next transmission of S_1 . The overhead between two consecutive transmissions is the time taken for the EOC from one transmitter to propagate on the channel to the tap of the following transmitter plus $\Delta + \Phi$.^{*} The total overhead in a round is the sum of the propagation delays between consecutively indexed stations plus $M(\Delta + \Phi)$. Thus

$$Y(M, M) = \sum_{i=1}^{M-1} \tau_{i,i+1} + \tau_{M,1} + M(\Delta + \Phi) \quad (2.5)$$

The round overhead is maximized, and hence the network capacity is minimized, when $\tau_{i,i+1} = \tau$, $\forall i$. This situation arises in the case where all even numbered stations are collocated on one side of the network and all odd numbered stations on the other. Under these conditions $Y = M(\tau + \Delta + \Phi)$ and

$$C(M, M) = \frac{1}{1 + \omega + \delta + \phi + a} \quad (2.6)$$

^{*}The value of Φ may be zero if it is possible for a station to decode the index of an on-going transmission and compute the resulting scheduling delay during the time of that transmission.

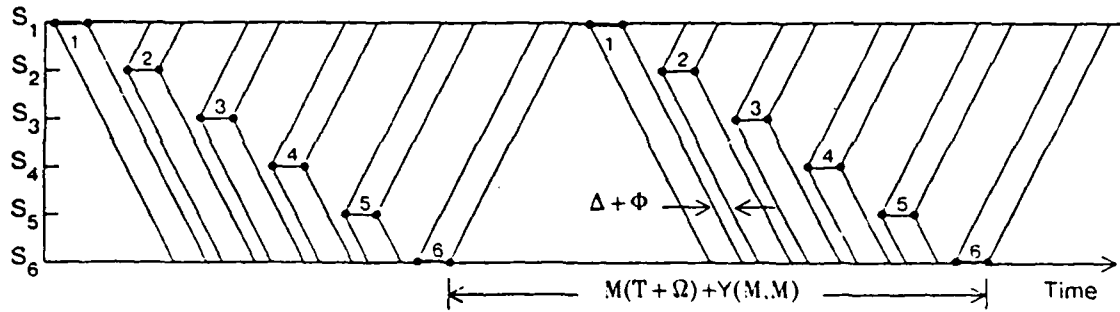


Fig. 2.5 Time-space diagram showing activity on the channel for BRAM when the stations' logical order is the same as their physical order on the channel.

Clearly, the minimum overhead is incurred for a layout in which all stations are collocated, since then, in the limit, $\tau_{i,i+1} = 0, \forall i$. In this case $Y(M, M) = M(\Delta + \Phi)$ and $C(M, M) = 1/(1 + \omega + \delta + \phi)$. If, on the other hand, we insist that the layout be such that the farthest two stations are τs apart, then $Y(M, M)$ is minimized when the stations are ordered in such a way so that their logical order is the same as their physical order on the bus. (See Fig. 2.5.) In this case $Y(M, M) = 2\tau + M(\Delta + \Phi)$ resulting in a throughput given by

$$C(M, M) = \frac{1}{1 + \omega + \delta + \phi + 2a/M} \quad (2.7)$$

which is almost independent of a if M is sufficiently large.

In Fig. 2.6, we plot the capacity $C(M, M)$ versus a for BRAM. In this and subsequent performance figures we assume that $\omega = \delta = \phi = 0$. The dashed curve shows $C(M, M)$ as specified by (2.6) versus a . This corresponds to the worst case layout for which the capacity is independent of M . The solid lines show $C(M, M)$, as specified by (2.7), versus a . This corresponds to the best case layout given that the farthest two stations are τs apart. In this case, the capacity is not insensitive to M and increases with increasing M . Thus, for large a , a high utilization can be

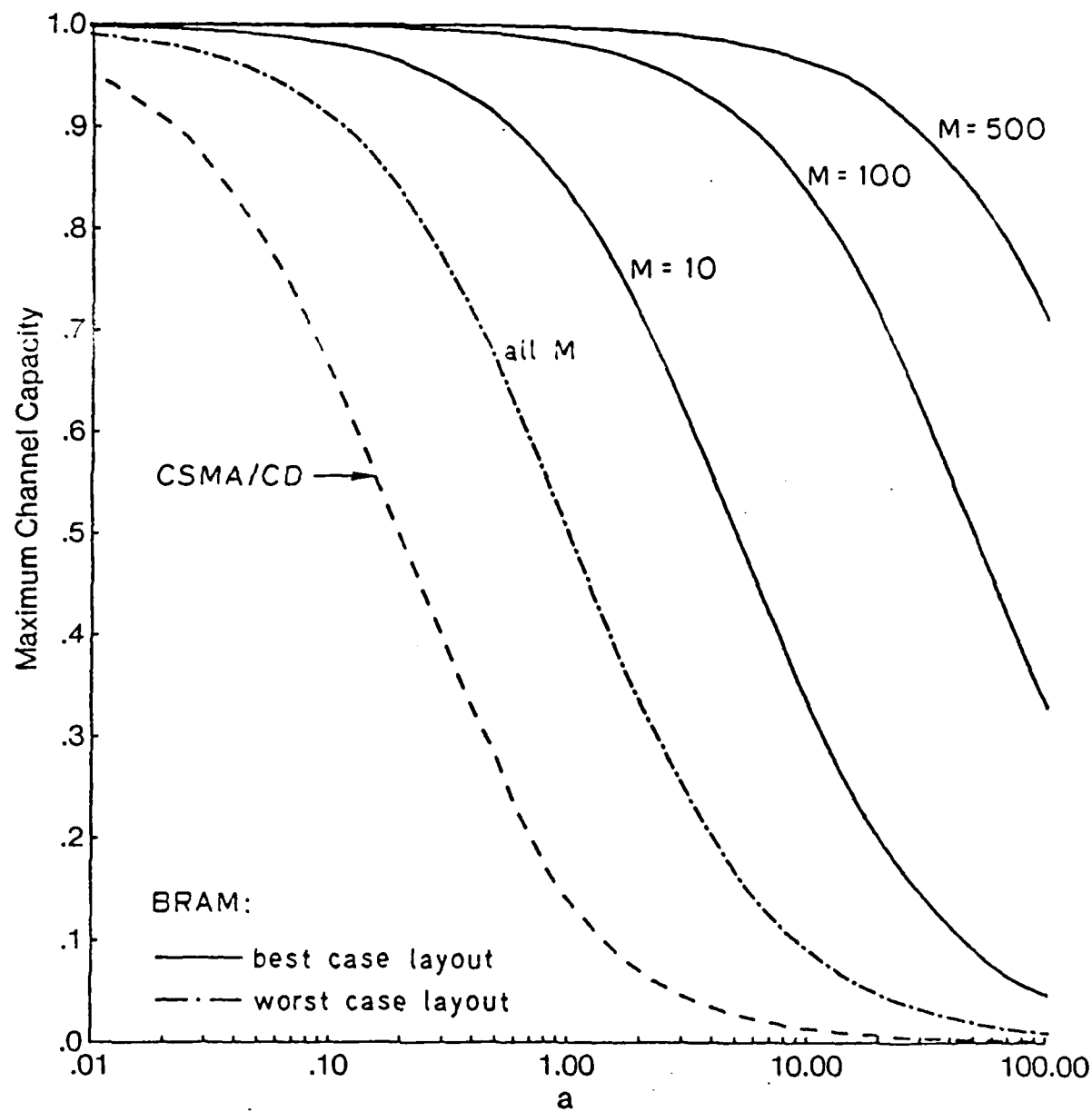


Fig. 2.6 Network capacity versus a for BRAM under the heavy traffic condition, and for CSMA/CD with infinite population.

achieved if M is sufficiently large. In the worst case when $M = 2$,^{*} the capacity for this layout coincides with the capacity for the worst case layout. The dotted curve corresponds to the capacity of CSMA/CD[†]. Clearly, the capacity of BRAM is superior to CSMA/CD for all values of a , even with the worst case layout.

The question now is how the overhead is affected when some stations do not transmit. Consider three stations numbered consecutively i , $i + 1$ and $i + 2$. If, in a given round, all three of these stations transmit when their turns come up, the overhead between these transmissions is $\tau_{i,i+1} + \tau_{i+1,i+2} + 2(\Delta + \Phi)$. Suppose now S_{i+1} does not transmit. S_{i+2} will transmit $2\tau + 2\Delta + \Phi$ s after EOC(i) reaches it. In this case the overhead is $2\tau + \tau_{i,i+2} + 2\Delta + \Phi$. These two cases are shown in Fig. 2.7. The effect of the missing transmission is to cause a *virtual time-space locus* for EOC(i) which is delayed in time by $2\tau + \Delta$ from the actual one. The interesting point is that, by missing a turn, S_{i+1} has not only reduced the total number of transmissions in a round but in addition has caused an increase in the overhead in this round. More generally, consider the case where N out of M stations are continuously backlogged with packets to be transmitted. We note that $Y(M, N)$ depends on the stations' layout and the particular choice of the subset of backlogged stations. The maximum possible value is

$$Y(M, N) = N\tau + (M - N)(2\tau + \Delta) + N(\Delta + \Phi) \quad (2.8)$$

giving the minimum value for capacity of

$$C(M, N) = \frac{1}{1 + \omega + \delta + \phi + a + \frac{M-N}{N}(2a + \delta)} \quad (2.9)$$

^{*}In BRAM, the case $M = 1$ is not the worst case since in this case the single station can transmit packets back to back, and assuming that $\omega = \delta = \phi = 0$, can achieve a capacity of 1.

[†]The CSMA/CD scheme considered here is the slotted p -persistent CSMA/CD with infinite population as analysed in [9] and [11].

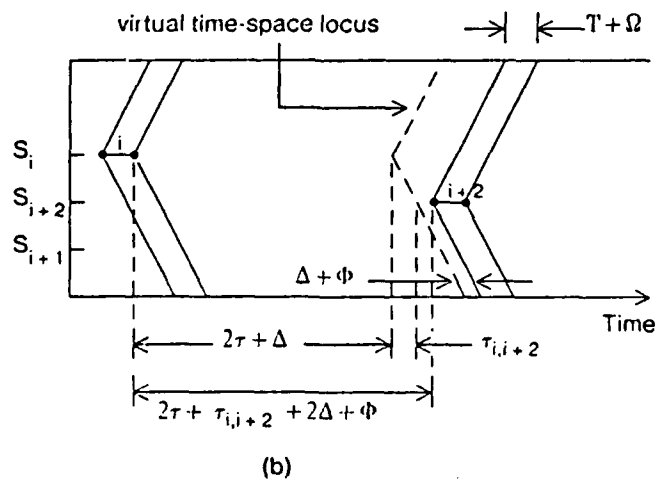
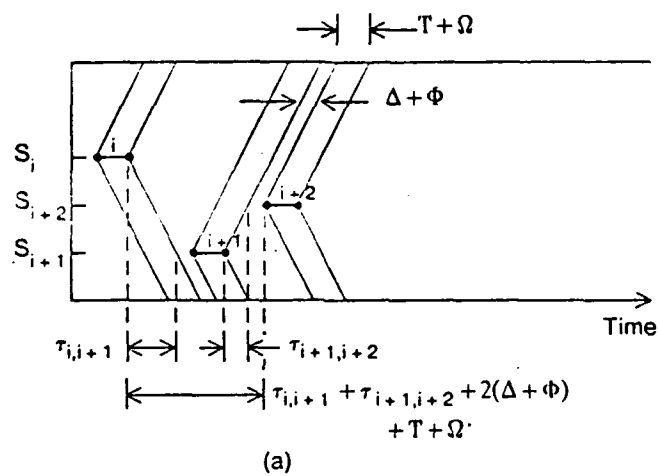


Fig. 2.7 The effect on the overhead of a station missing its turn to transmit in BRAM. In (a) all three stations S_i , S_{i+1} and S_{i+2} transmit. In (b) S_i and S_{i+2} transmit while S_{i+1} misses its turn.

The minimum value of the overhead is

$$Y(M, N) = (M - N)2\tau + \Delta + N(\Delta + \Phi) \quad (2.10)$$

giving a maximum value for capacity of

$$C(M, N) = \frac{1}{1 + \omega + \delta + \phi + \frac{M-N}{N}(2a + \delta)} \quad (2.11)$$

Assuming that $\omega = \delta = \phi = 0$, we show in Fig. 2.8 the capacity versus a for the best case and worst case layouts and various values of the ratio N/M . As predicted by Fig. 2.6, a high capacity is achieved for N/M close to 1 and the best case of layout. However, for small values of the ratio N/M , the performance degrades significantly and the capacity achieved by BRAM, even for the best case layout is worse than that of CSMA/CD.

Comments: (i) BRAM accommodates all layouts without requiring knowledge by each station of the layout, paying a price in performance. The algorithm is entirely distributed. However, the original description of it in [26] fails to address important issues pertaining to the loss of the synchronizing event EOC which would occur were all stations to become temporarily idle, nor does it describe how the algorithm gets started. The robustness of the scheme is furthermore dependent on the ability to properly and accurately decode the index of each transmitting station by all stations in the network. Other schemes discussed in this section address these issues (and their solutions certainly can be applied to BRAM), and provide improved performance. Nevertheless, BRAM and its cousins, MSAP and MSRR* (Kleinrock,

*In [27] and [28], Kleinrock and Scholl present several access schemes for radio channels which are in essence variations of the scheduling delay access mechanism proposed for BRAM. One of these, MSRR, is identical to BRAM. Others apply different service disciplines. For example, MSAP allows a station, having gained access to the channel, to transmit until its buffer is empty. Yet another variation is proposed for slotted systems. In this variation, a slot is subdivided into M minislots, each of duration τ , followed by a fixed length data field for the packet transmission. If, for a given slot, a station's scheduling delay expires and the channel is idle, this station transmits unmodulated carrier in the remaining minislots, and then transmits its packet in the data field of the slot.

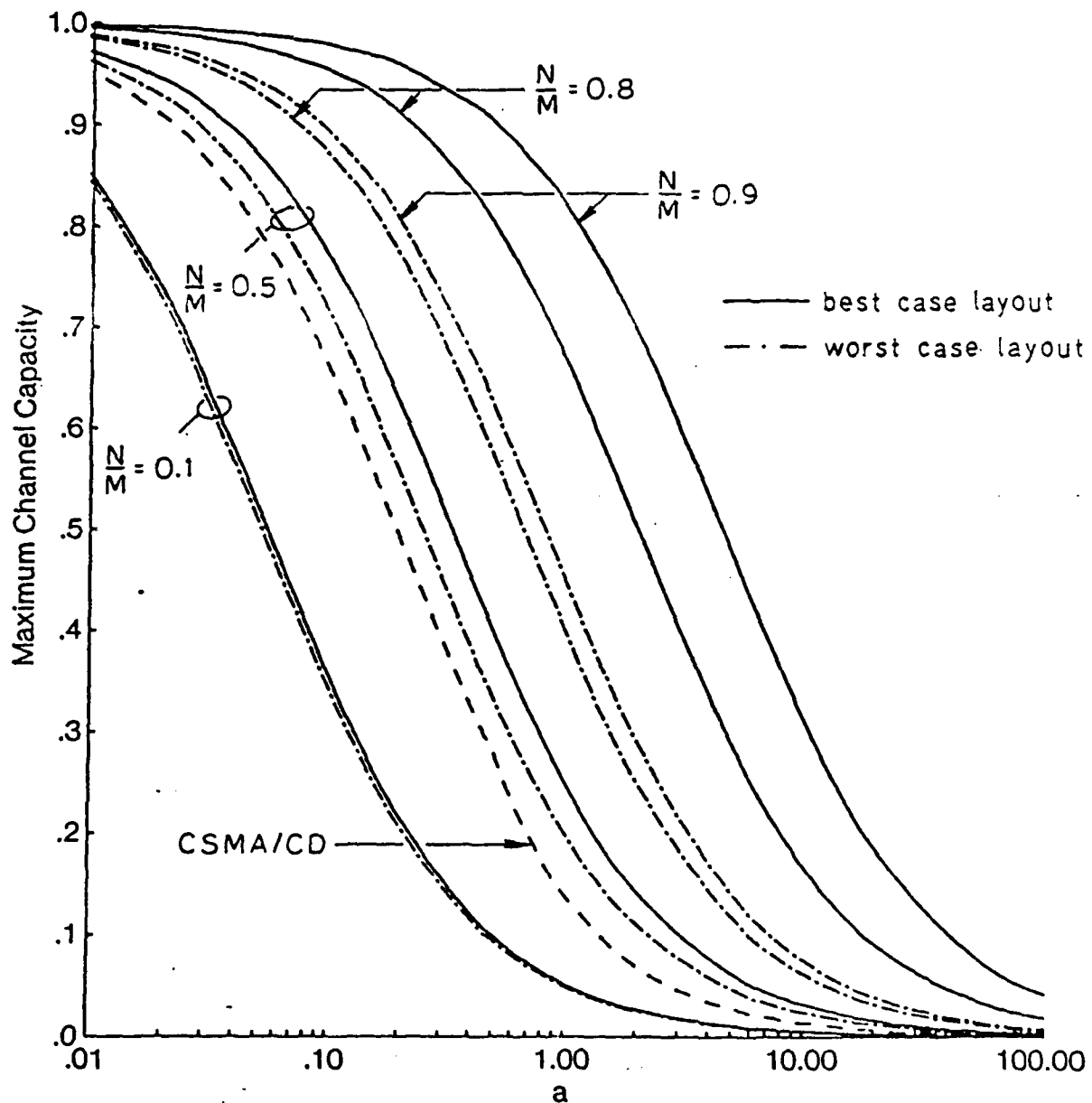


Fig. 2.8 Network capacity versus a for BRAM showing the effect of some stations missing their turns.

Scholl, 1977) [27], [28] which bear great resemblance to BRAM, are among the first conflict-free algorithms for distributed broadcast networks.

(ii) In the description of BRAM in [26] the detection time Δ and processing time Φ are ignored. This in effect is equivalent to assuming that they are zero. However, in deriving the expression for the scheduling delay function, we proved that, to accommodate all possible layouts, the detection time Δ must be included in the computation. If Δ is neglected, a value of τ which is larger than the actual value by at least $\Delta/2$ must be used. In addition, in the original descriptions of BRAM and MSAP [26], [27], [28], the stations' scheduling delays are staggered by τ instead of $2\tau + \Delta$. Such a scheduling delay would work only in a network where $\tau_{i,j} = \tau \forall i, j, i \neq j$. Obviously, such a restriction is not desirable for a local area network. In our opinion, the scheduling delay function given in (2.1) is correct and complete.

(iii) In the discussion above we considered only the case where each station transmits a single packet. However, one could allow a station to transmit multiple packets when its turn comes up. Ultimately, one could envision allowing a station to empty its buffer when its turn comes up. Such a service discipline is referred to as *prioritized BRAM* in [26] and *alternating priorities* in [27]. In such a case, a change must be made to the scheduling delay function $H_j(i)$ to reflect this new service order. Suppose that S_i is currently transmitting. In order that S_i has highest priority for accessing the channel at the end of its own transmission, it is required that $H_i(i)$ be the shortest scheduling delay. Let $H_i(i) = 0$. Assuming some gap g between consecutive transmissions by S_i , (which accounts for processing time to evaluate $H_i(i)$, time to turn off and on its transmitter, load buffers etc.), the next station in sequence, S_{i+1} , will have detected the next transmission by S_i a time $g + \Delta$ after $\text{EOC}(i)$ reaches it.[†] Since it takes S_{i+1} a time $\Delta + \Phi$ to detect $\text{EOC}(i)$ and

[†]We assume that $g > \Delta$. If this is not the case, then the gap between consecutive packets from S_i

evaluate $H_{i+1}(i)$, its scheduling delay must be at least $g - \Phi$ to guarantee conflict free transmissions. For any station S_j , where $j \neq i$ or $i + 1$, $H_j(i)$ can be computed using the same recursive argument given above. In this case, however, the initial condition on the recursion is $H_{i+1}(i) = g - \Phi$ instead of $H_{i+1}(i) = 0$.

2.5.2 SOSAM (Gold, Franta, 1982) [29], [30]

This scheme, called the source synchronized access method (SOSAM), is similar to BRAM in that it provides the NGSS discipline, requires no correspondence between the stations' indices and their locations, and achieves the same performance when *all* stations are backlogged. It provides, however, improved performance when stations miss their turns. To accomplish that, all stations must have explicit knowledge of the propagation delay between every pair of stations. Given this knowledge, S_j can determine the minimum time required, after detecting $EOC(i)$, to detect a potential transmission from S_{j-1} and set $H_j(i)$ to this amount of time. This minimum time is given by (2.2). In particular, $H_{i+1}(i) = 0$. In the general case for $j \neq i + 1$, $H_j(i)$ is defined recursively by

$$H_j(i) = \begin{cases} H_{j-1}(i) + \tau_{i,j-1} + \tau_{j-1,j} - \tau_{i,j} + \Delta & j \neq 1 \\ H_M(i) + \tau_{i,M} + \tau_{M,j} - \tau_{i,j} + \Delta & j = 1. \end{cases} \quad (2.12)$$

As with BRAM, the overhead in a round for SOSAM, $Y(M, N)$, varies depending on the relative locations of the stations on the bus and their logical ordering, and this can take on a range of values depending on the topology. However, for a given configuration, this overhead is constant regardless of how many or which stations transmit in the round, and is computed as in (2.5). Given $Y(M, N)$, the network capacity $C(M, N)$ and delay $D(M, N)$ can then be easily derived. (See (2.3) and

would not be detectable, and these packets would appear to constitute a single transmission.

(2.4.) If we assume that $\omega = \delta = \phi = 0$, $C(M, N)$ for SOSAM is identical to $C(N, N)$ for BRAM. Thus the curves shown for BRAM in Fig. 2.6 are representative of the capacity of SOSAM.

Comments: (i) To gain this improvement in performance over BRAM, in SOSAM each station S_k must store in its network interface either all the inter-station propagation delays $\tau_{i,j}$ or precomputed values of its scheduling delay $H_k(i)$, $\forall i$. Either option requires substantial memory if the network is large (say 1000 stations). Also this memory would have to be updated at every station every time one is moved, added to, or removed from the network. This makes the task of maintaining such a network a difficult one.

(ii) It was indicated in BRAM that the synchronizing event EOC is lost and the network stalls if all stations are idle at the time that their respective scheduling delays expire. SOSAM proposes a mechanism to prevent this. If a station is idle when its scheduling delay expires, that station resets the latter to some predetermined constant which is larger than any scheduling delay, thereby maintaining the staggering of the potential transmission times. Clearly, the smallest constant that can be used is $\max_j \{H_j(i)\} + \Delta = H_i(i) + \Delta$. Furthermore, it can be shown that $H_i(i) = Y(M, M)$ and hence is independent of i . While this ensures that the network will operate under zero load, the robustness of the scheme nevertheless depends on the ability to correctly decode the index of each transmitting station.

2.5.3 BID (Ulug, White, Adams, 1981) [31]

We indicated that in SOSAM, just as in BRAM, the overhead is minimized and the network capacity maximized by numbering the stations such that their logical order is the same as their physical order on the bus. In this case $\tau_{i,j-1} + \tau_{j-1,j} = \tau_{ij}$, and the scheduling delay function given for SOSAM becomes

$$H_j(i) = H_{j-1}(i) + \Delta = (j - i - 1)\Delta \quad j > i \quad (2.13)$$

which is independent of the propagation delays between stations. In fact, this is in essence what the scheduling delay in BID, a predecessor of SOSAM, is. The stations are numbered 1 through M as shown in Fig. 2.1, and the scheduling function is essentially that given in (2.13). We now describe those features specific to BID. The end stations (S_1 and S_M) perform a special function called the *start-of-round*. A round or cycle is the time between two consecutive start-of-rounds. Within a round, stations transmit in sequential order; however, this order varies from round to round. In a *left-to-right round*, backlogged stations transmit in turn starting with S_1 and ending with S_M . In a *right-to-left round*, backlogged stations transmit in turn starting with S_M and ending with S_1 . Each station can determine the "direction" of the current service order by an indicator which is transmitted along with the index number in each packet. Suppose that round r is a left-to-right round. Round r ends with the end of S_M 's transmission if S_M is backlogged in this round, or, if S_M is idle, at the time that S_M would have transmitted were it backlogged. At this time S_M initiates round $r + 1$ by transmitting a packet which has the direction indicator set to "right-to-left." If S_M is backlogged, this packet would be a data packet; if S_M is idle, this packet would be an explicit token or start-of-round packet. By symmetry, round $r + 2$ is initiated by S_1 at the end of round $r + 1$. From the direction indicator and the index number of the transmitting station, each station can compute the scheduling delay as

$$H_j(i) = \begin{cases} (j - i - 1)\Delta & \text{left-to-right round and } j > i \\ (i - j - 1)\Delta & \text{right-to-left round and } j < i \\ \infty & \text{otherwise.} \end{cases} \quad (2.14)$$

Obviously, the service discipline is NGRSS.

In Fig. 2.9 we show a time-space diagram of the activity on the channel in BID. Due to the nature of the order of transmissions within a round, and by reversing

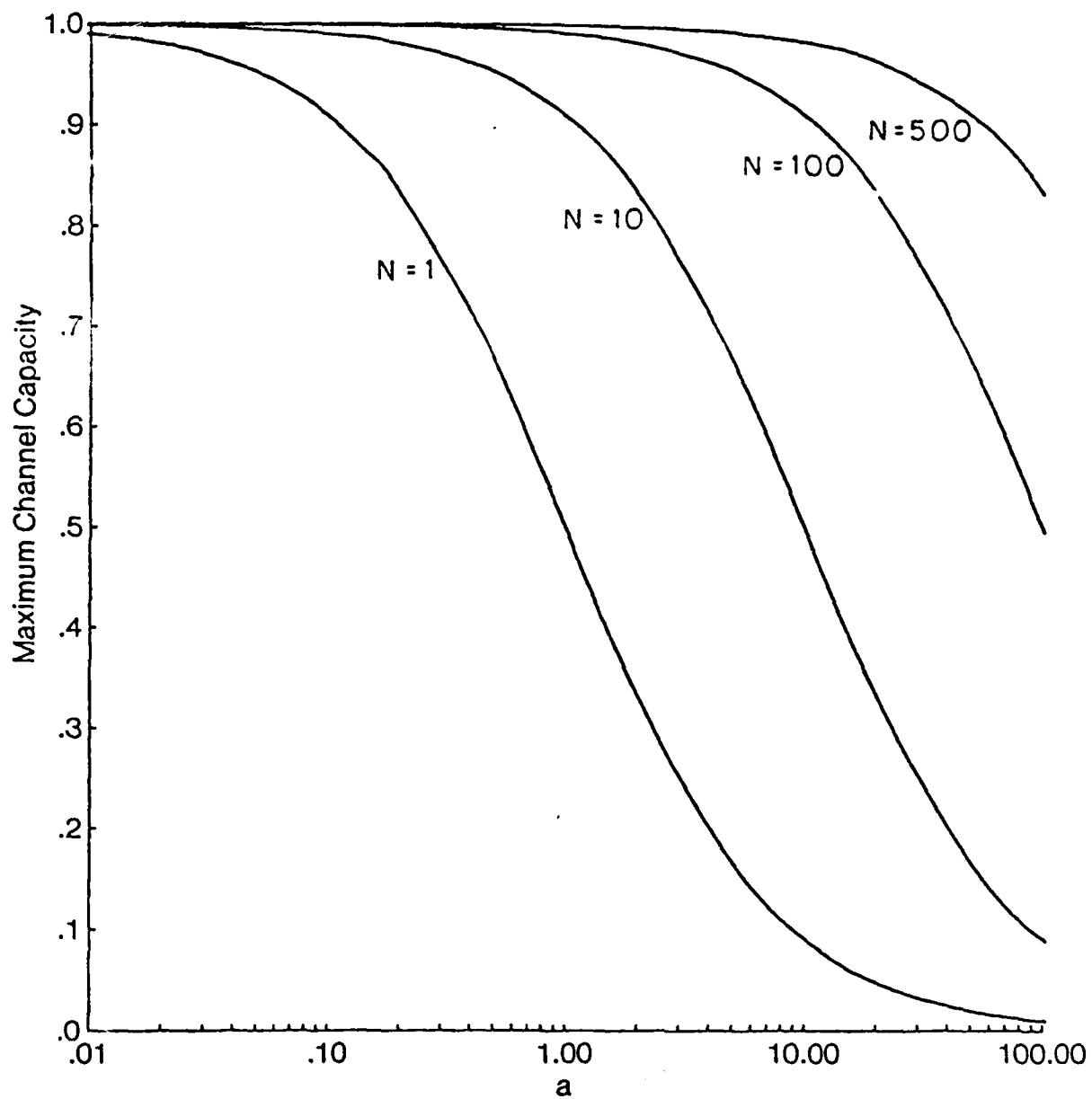


Fig. 2.10 Network capacity versus a for BID.

while at the head of the queue at its station plus the transmission time of that packet. Since in BID the order of service is reversed from round to round, $D(M, N)$ varies with each station and with the direction of the sweep. Bounds on $D(M, N)$ are given by the delay incurred at an end station where, normalized to T , $D(M, N)$ alternates between a low value of

$$D_{\min}(M, N) = 1 + \omega + \delta + \phi \quad (2.16)$$

and a high value of

$$D_{\max}(M, N) = 2N(1 + \omega + \phi) + 2M\delta + 2a. \quad (2.17)$$

For any other station, $D(M, N)$ lies between these two values.

Comments: (i) Since the logical ordering of stations is the same as their physical order, BID is able to achieve a performance which is almost independent of τ if N is sufficiently large. However, this restriction on the ordering of stations makes it difficult to add stations to the network or move existing ones.

(ii) BID is partially centralized in that end stations are required to initiate new rounds. As a result, the network is robust in the sense that synchronizing events are periodically generated even when all stations are idle, and in the sense that one end station can initiate a new round if the index number or direction indicator of a transmission cannot be decoded. In the event of an end station failure the adjacent station can assume the functions of the end station. More generally, if, in a left-to-right round, stations $M, M-1, \dots, k+1$ all fail simultaneously, then S_k will perform the functions of the end station on the right. S_k will determine that stations $M, M-1, \dots, k+1$ have failed if it does not detect any activity on the network for a sufficiently long time after it has had its turn in this round. This time-out period is determined as follows. (See Fig. 2.11.) Consider the time reference at S_k to be

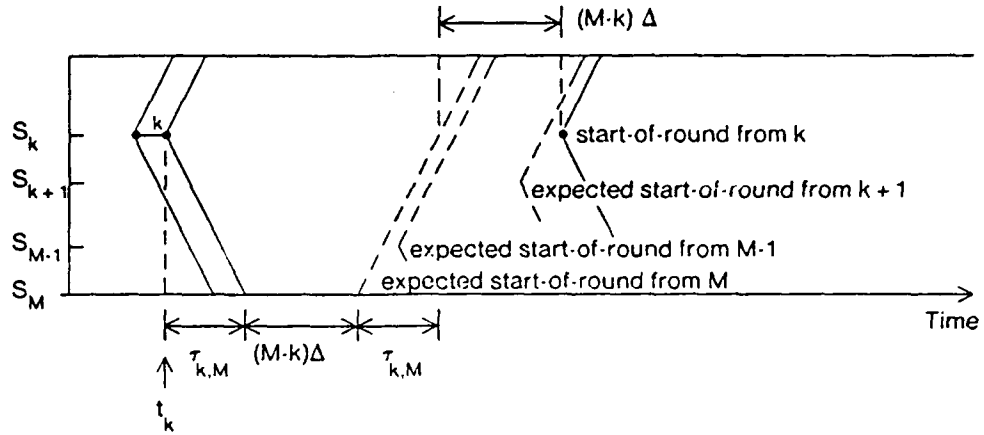


Fig. 2.11 Time-space diagram for BID showing the components of the time-out period used by S_k to determine that stations $S_j, j > k$, have all failed.

the end of its transmission (or, if S_k is idle, the time that it would have transmitted were it backlogged). S_k will expect to have detected a start-of-round packet from S_M after an interval of $2\tau_{k,M} + (M-k)\Delta + \Delta$. Allowing another $(M-k-1)\Delta$ for each of the stations $M-1, M-2, \dots, k+1$ to possibly start the new round, S_k must detect no activity for $2\tau_{k,M} + 2(M-k)\Delta$ to determine that stations $M, \dots, k+1$ have failed. By symmetry one can determine the appropriate time-out to determine that the stations on the left have all failed. To simplify the installation of the network, the quantity $2(M-k)\Theta$ can be substituted for $2\tau_{k,M}$ where Θ is a constant greater than the maximum propagation delay between adjacent stations. In the description of BID in [31] the latter approach has been adopted. However, in [31], Δ has been ignored and so, in order for the preceding algorithm to be correct, Θ must have implicitly included Δ . Except for this recovery algorithm, no knowledge of τ or $\tau_{i,j}$ is required in BID.

(iii) By having the end stations alternately initiate the rounds, the order in which stations are served within a round is reversed from round to round. While this

improves the network capacity, it means that the upper bound on $D(M, N)$ is given by the length of two rounds as opposed to the length of one as in BRAM and SOSAM where the service order is the same from round to round. However, the average value is the length of one round and is less than that of BRAM and SOSAM due to the reduction in the overhead.

2.5.4 Silentnet (Jensen, Tokoro, Sha, 1980) [32]

Silentnet is similar to BID in that the stations' logical ordering is the same as their physical order on the bus, and thus can apply the same efficient scheduling delay function. As in BID, the service discipline is NGRSS. The distinction in Silentnet is the distributed, as opposed to centralized, mechanism used to initiate a round. While BID makes use of the end stations for this purpose, in Silentnet this functionality is part of the scheduling delay function. Although in this system there are no explicit start-of-round events, we nevertheless define a round to be the sequence of transmissions which are in a given order, either left-to-right or right-to-left. Consider a round in which the service order is from left to right, and S_i is the last station to have transmitted. For $j > i$, $H_j(i)$ is computed as is done in BID. For $j \leq i$, S_j has already had its turn in the current round; it schedules its next transmission for a time at which it will be the first in the next round, on the assumption that, following the transmission of S_i , stations $S_{j+1}, \dots, S_i$ do not start the new round. We now show how this time is evaluated. The reader is referred to Fig. 2.12. Were some station S_k , $i < k \leq M$, ready to transmit after S_i , the latter would detect $\text{BOC}(k)$ an amount of time $\Delta + \Phi + (k - i)\Delta + 2\tau_{i,k}s$ after completing its transmission. Thus, if S_i detects no activity for $\Delta + \Phi + (M - i)\Delta + 2\tau_{i,M}s$ after it completes its transmission, none of the stations S_k , $i < k \leq M$ could have been backlogged; hence S_i can transmit at this time (beginning the new round in which the service order will be from S_i to S_1). If we assume that it takes Φs for S_i

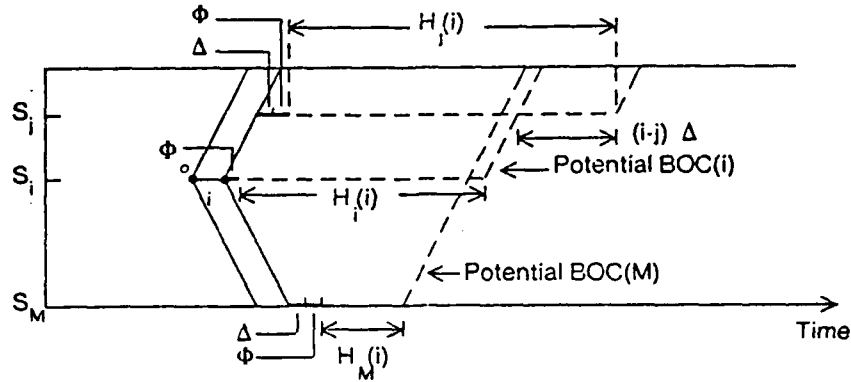


Fig. 2.12 Time-space diagram for Silentnet showing the relationship between the scheduling delays $H_i(i)$, $H_M(i)$ and $H_j(i)$, $j < i$.

to re-evaluate its scheduling delay at the end of its own transmission, $H_i(i)$ for a left-to-right round is $H_i(i) = 2\tau_{i,M} + (M - i + 1)\Delta$. Given this potential BOC(i), stations S_j , $j < i$, must stagger their potential transmission times appropriately. Thus, synchronizing to the actual event EOC(i), the scheduling delay for S_j , $j < i$, is $H_j(i) = (H_i(i) - \Delta) + (i - j)\Delta$. The scheduling delay for a right-to-left round can be deduced by a symmetrical argument. Thus $H_j(i)$ can be computed as

$$H_j(i) = \begin{cases} (j - i - 1)\Delta & j > i \\ 2\tau_{i,M} + (M - i + 1)\Delta & j = i \\ 2\tau_{i,M} + (M - j)\Delta & j < i \end{cases} \quad \text{left-to-right round} \quad ; \quad H_j(i) = \begin{cases} (i - j - 1)\Delta & j < i \\ 2\tau_{1,i} + i\Delta & j = i \\ 2\tau_{1,i} + (j - 1)\Delta & j > i \end{cases} \quad \text{right-to-left round.} \quad (2.18)$$

Using this scheduling delay, the performance of Silentnet is identical to BID. If, at the cost of some efficiency, one desires that the scheduling delay be independent of the stations' locations on the network, one could replace $\tau_{i,M}$ and $\tau_{1,i}$ in (2.18) by τ . In this case, with S_1 and S_M backlogged, an overhead of 3τ is incurred between

consecutive rounds leading to a network capacity of

$$C(M, N) = \frac{1}{1 + \omega + \phi + \frac{M}{N}\delta + 3a/N} \quad (2.19)$$

Comparing (2.19) with (2.15) we see that the capacity of Silentnet is identical to BID except that the overhead incurred in a round is greater by an amount $2a$. In Fig. 2.13 we plot the capacity versus a for Silentnet and BID and various values of N . The effect of this additional overhead is clearly shown.

Comments: (i) Three variants of Silentnet are presented in the original description [32]. The first, called the "basic algorithm," is the one described above but with $\tau_{i,M}$ replaced by τ . The second, called the "distance algorithm," is the one described above in (2.18), and its performance is superior to that of the basic algorithm. It is given the name distance algorithm since each station must have knowledge of the distance of all the stations from one end of the network. In the third, called the "see-saw algorithm," the start-of-round function is assigned to the end stations. This variant of Silentnet is identical to BID.

(ii) As in SOSAM, a mechanism is provided in Silentnet to maintain the existence of the synchronizing event ($EOC(i)$) when the network load is low. This is achieved by having the last station to have transmitted, if idle at the end of its transmission, set its scheduling delay to a constant sufficiently large such that it will have detected a transmission by any backlogged station before this scheduling delay expires. If the scheduling delay does expire, a dummy packet is transmitted, thereby regenerating the event $EOC(i)$. This continues until some other station transmits, in which case that station becomes the last to have transmitted. The minimum value of the constant is given by $\lceil \max_{i,j} \{H_j(i)\} + 2\tau \rceil$ which is $4\tau + M\Delta$. In the descriptions of the "basic algorithm" and "distance algorithm" given in [32], $H_i(i)$ is given by this amount regardless of whether S_i is backlogged or idle. As a result the last station to

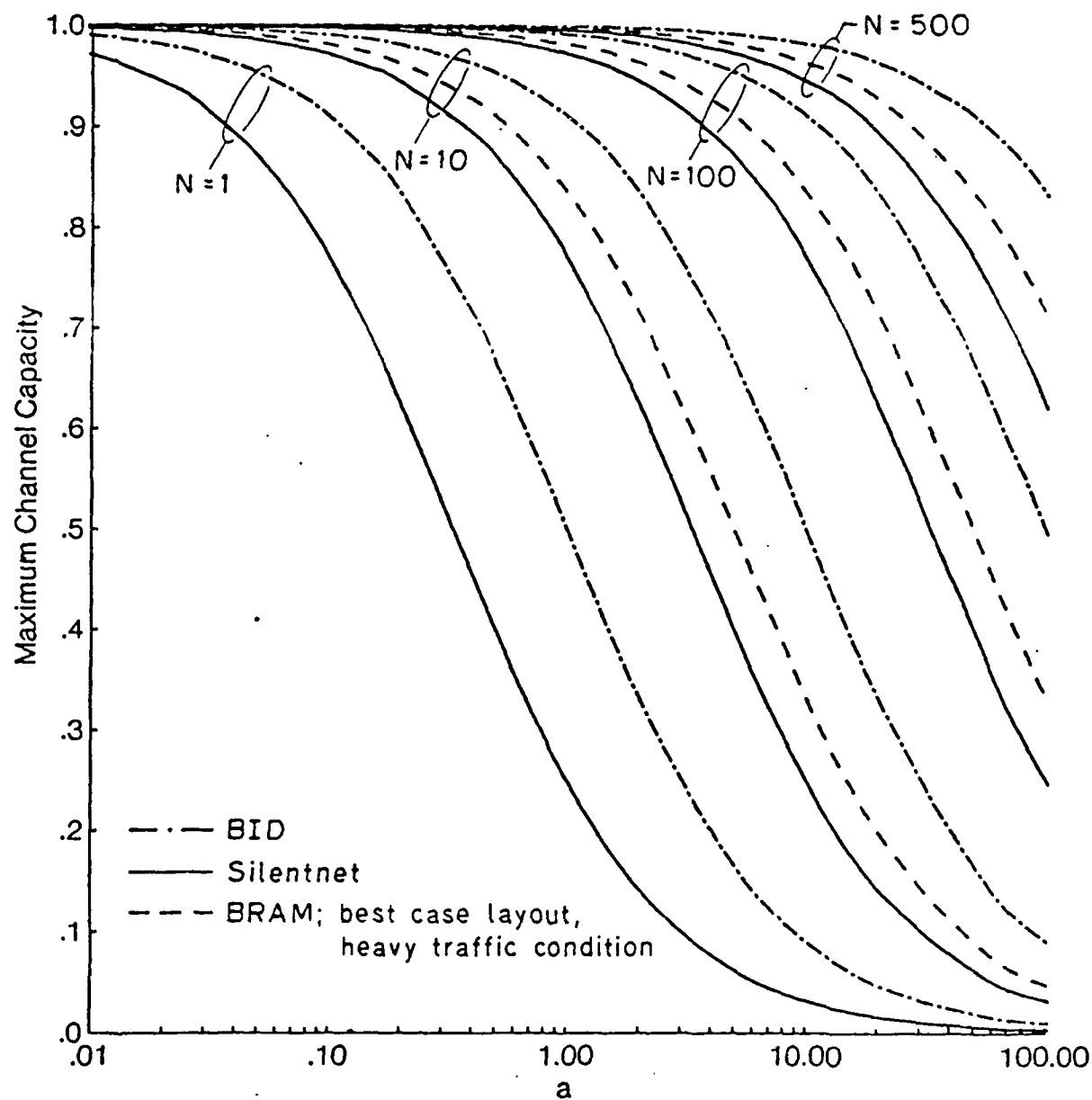


Fig. 2.13 Network capacity of Silentnet and BID. The effect of a larger inter-round overhead in Silentnet is apparent.

transmit in a given round misses its turn in the next, leading to an unfair scheduling function. The scheduling delay function in (2.18) overcomes this limitation in the original scheme and provides fair service to all stations.

2.5.5 L-Expressnet (Borgonovo, Fratta, Tarini, Zini, 1983) [33]

As in BID and Silentnet, the stations in L-Expressnet are numbered sequentially according to their physical locations. But contrary to the former two, in L-Expressnet the order of transmissions is always in one direction (say left-to-right), and the scheduling delay is computed only once, at the beginning of the round with respect to an explicit start-of-round token (which need not contain any information but may consist merely of a burst of carrier). The scheduling delay in question then represents the cumulative period of idle time, counted from the start-of-round token, that a station has to observe before it is allowed to transmit. To show how this scheduling delay is evaluated, let us revisit BID and consider a left-to-right round starting with the start-of-round token (or packet) generated by S_1 . After detection of EOC(1), S_j computes $H_j(1) = (j - 2)\Delta$. By definition $H_j(1)$ is the amount of idle time that S_j must observe, following the detection of EOC(1) and the computation of $H_j(1)$, before initiating a transmission. Assume now that some station S_k , $1 < k < j$, (and only that station), is backlogged and hence transmits before S_j . At S_k , the event BOC(k) occurs a period of time $H_k(1) + \Phi s$ following the detection of EOC(1). Since the stations are numbered according to their physical order, BOC(k) is detected at S_j a period of time $H_k(1) + \Delta s^*$ following the detection of EOC(1). Following the detection of EOC(k), S_j computes $H_j(k) = (j - k - 1)\Delta$ and thus starts its transmission $H_j(k) + \Phi s$ after the detection of EOC(k). The observation here is that the cumulative idle time that S_j

*Just as it takes Δs to detect EOC, we assume that it takes Δs to detect the BOC event. That is, the channel is assumed to be in the idle state for Δs following the actual start of signal reception; hence the additional Δ in the expression.

must have observed since the detection of EOC(1) and the computation of $H_j(1)$ is precisely $H_k(1) + \Delta + H_j(k) + \Phi = H_j(1) + \Phi$. This argument can be easily generalized: if n stations transmit between S_1 and S_j , the cumulative idle period counted from EOC(1) and observed by S_j would be $H_j(1) + n\Phi$. This suggests that, instead of having to compute a new scheduling delay following each transmission, one could compute $H_j(1)$ once at the beginning of each round, and then count idle time on the channel. In fact this eliminates the additional idle time Φ needed for each computation. This is the basic idea behind L-Expressnet.

The other important difference between L-Expressnet and BID is that the former does not rely on the end stations for the generation of the synchronizing token. The latter is transmitted by that station which is the leftmost of the participating stations,[†] thus rendering the scheme totally distributed. For this station to determine the time at which it should transmit, the following mechanism is proposed in L-Expressnet. It makes use of the previous explicit token as a time reference. Let r denote the current round and let S_{k_r} be the station that transmitted the start-of-round token in round r . Upon observing EOC(token), all stations count a cumulative idle time up to $2r + M\Delta$. This amount is sufficient, even in the worst case where $k_r = 1$, for all stations $j > k_r$ to have had their chance to transmit. As can be seen in Fig. 2.14, this creates a virtual time reference which has the property that it occurs after the last transmission in the round. (A tight time reference, i.e., one which corresponds exactly to the potential event EOC(M), could be achieved but this would require knowledge of the $\tau_{i,j}$'s and the index number of the station that transmitted the start-of-round token.) Using this virtual time reference, each participating station S_k schedules a time at which to transmit the next start-of-round token using a similar approach to the procedure to recover from an end

[†]By participating station we mean a station that is "alive" and in sync with the activity on the channel. Note that a participating station need not be backlogged.

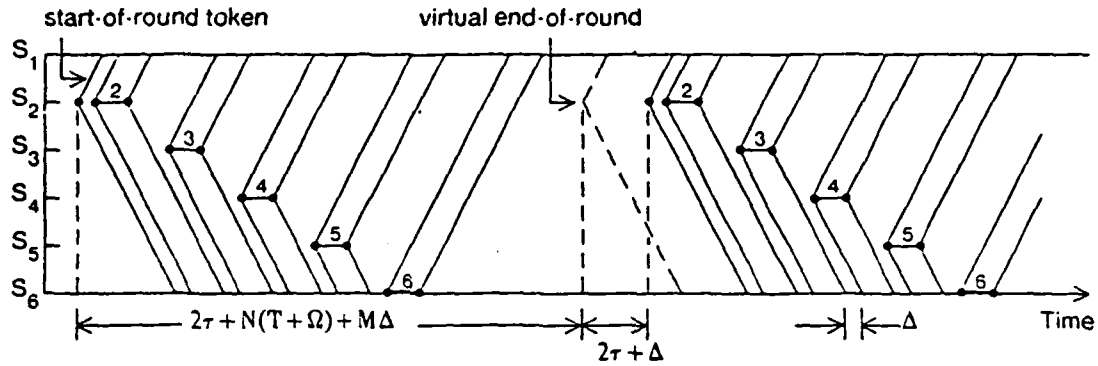


Fig. 2.14 Time-space diagram for L-Expressnet showing typical activity on the channel. S_2 is the leftmost participating station and therefore it transmits a start-of-round packet before its data packet, hence the two time-space loci for this station in each round.

station failure in BID. That is, if S_k fails to detect activity within $2\tau_{1,k} + (k-1)\Delta$ s as measured from this virtual time reference, it transmits the start-of-round token. In the description of L-Expressnet in [33], this scheduling delay is computed as $2(k-1)\Theta$ where, as in BID, Θ is a constant greater than the maximum propagation delay between adjacent stations and (although not mentioned in [33]) must include an amount Δ for the detection time. Obviously, the overhead for this scheme, neglecting the start-of-round token, is

$$Y(M, N) = 2\tau + M\Delta + 2(k-1)\Theta \quad (2.20)$$

where S_k is the leftmost of the participating stations. Thus $C(M, N)$ and $D(M, N)$ can be easily computed from (2.3) and (2.4). Curves showing the capacity versus α for L-Expressnet would be similar to those for BID and Silentnet.

Comments: (i) In L-Expressnet, each station uses a scheduling delay which is a function only of that station's index number. Hence no need exists to read the index of each transmission. In addition, since the scheduling delay is not re-evaluated after

each EOC, the processing overhead is incurred only once in a round.

(ii) In L-Expressnet all stations participate in the start-of-round procedure. As a result, the scheme is robust in the sense that the synchronizing event EOC(token) is continuously regenerated.

(iii) While, in general, stations take Δ s to detect EOC(token) plus Φ s to recognize it as such, the station that transmits the start-of-round token does not incur this overhead. Thus there is a discrepancy in the time references between this station and the rest which has been overlooked in the above description of L-Expressnet. This can be compensated for by the former, having transmitted the start-of-round token, delaying any further activity for $\Delta + \Phi$ s.

2.6 Schemes Using the Reservation Access Mechanism

In this section we describe those schemes that use the reservation method as their basic access mechanism. Some of the characteristic features, by which these schemes may differ from each other, are (i) the network architecture, (ii) the method by which a reservation is made, (iii) the need for stations to be uniquely numbered or lack thereof (contrast this to the scheduling delay access mechanism class where all schemes require the stations to be numbered), (iv) the need to synchronize the reservation signals, or lack thereof, which can be accomplished in an entirely distributed or partially centralized fashion, and (v) the performance achieved.

2.6.1 DSMA (Mark, 1980) [34], [35]

The BBC network configuration used in this scheme is shown in Fig. 2.15. It consists of a data bus and a set of control wires inter-connected by OR circuits.

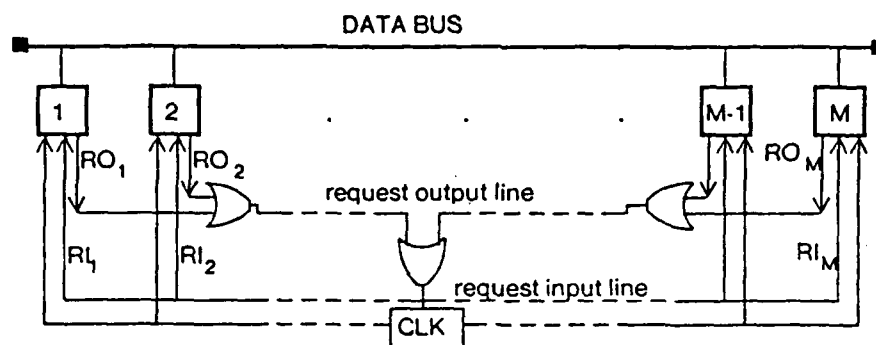


Fig. 2.15 BBC configuration used in DSMA.

With this control-wire assembly, signals transmitted by individual stations can be “ORed,” and the result can be broadcast to all stations on a return path. Each station is assigned a unique binary address. In brief, synchronized with the start of a transmission on the data bus, stations wishing to transmit attempt to reserve the channel by simultaneously transmitting their addresses onto the *request-output line* (*RO*) of the control-wire assembly. (Note that reservations can be done simultaneously with packet transmissions.) Due to the particular OR function implemented by the control-wire assembly, it is possible to identify the station with the highest address, which is then the next to access the data bus.

We now present the details of a station’s operation. Assume for the moment that the reservation operation, and consensus as to the station that successfully reserves the channel, can be completed prior to the end of the current packet transmission on the data bus. Consider that some station, say S_i , has reserved the channel. As soon as S_i detects that the data bus is idle, it begins to transmit its packet on the data bus and simultaneously sets RO_i to the logical ‘1’ state. This action initiates a new *reservation period* (RP) to determine the station to transmit after S_i . During the RP, each participating station transmits its address serially and synchronously

with the other stations onto the reservation wire, beginning with the most significant bit. Synchronous transmission of the stations' address fields is accomplished by a clocking signal which consists of a sequence of clock pulses issued by a *centralized clock* at the time that it detects the $0 \rightarrow 1$ transition on the *RO* line. Assume that the logic in each station is clocked on the rising edge of each clock pulse. For each rising edge of the clock signal, one bit of each station's address is placed on the *RO* line, and the logical OR of the signals from all the participating stations is returned to all stations by means of the *request-input (RI)* line. If a station detects that the *RI* line is '1' while its transmitted bit is '0', then it knows that it has unsuccessfully contended for the channel and ceases to transmit any more of its address bits in this RP. Only one station will correctly receive all its bits and this station is the next to access the data bus. It does so as soon as the data bus is sensed idle.

For each RP, the station that successfully reserves the channel is the one, among those making reservations, that has the highest address. Hence, the service discipline is HOLS. In order to ensure fair allocation of the bandwidth to all stations, each station can be in one of two states: *active* or *dormant*. A station is initially active and in this state attempts to reserve the channel as described above. Having transmitted, it becomes dormant and in this state does not contend for the channel. This allows other stations to take their turns. Eventually, all stations will be either dormant or idle and so no station will reserve the channel. This condition can be detected by reading the '0' address during the RP. When they detect this event, all dormant stations reset their respective states to active and immediately initiate a new RP by asserting their respective *RO* lines. Thus the first reservation of each round requires a double reservation period.

To ensure the proper operation of the scheme, the clock rate of the centralized clock must be sufficiently slow so as to allow the address signal from each

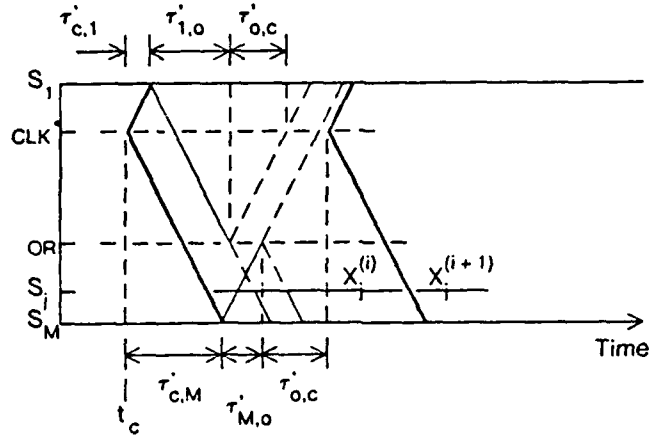


Fig. 2.16 Time-space diagram showing the minimum time required between two consecutive clock pulses for arbitrary positions of the clock and final OR gate. $X_j^{(i)}$ represents the value of RO_j after clock pulse i . The dashed diagonal lines represent the time-space locus of events on the RI line.

participating station to propagate through the control-wire assembly in the time between successive clock pulses. In order to compute the clock rate, and hence the duration of the RP, we consider the general case where the clock and the final OR gate are located at arbitrary positions on the network. Let $\tau'_{i,c}$ and $\tau'_{i,o}$ denote the propagation times on the *control wire* between S_i and the clock, and S_i and the final OR gate, respectively. Let $\tau'_{o,c}$ denote the propagation time on the *control wire* between the clock and the final OR gate. Consider a clock pulse occurring at time t_c as shown in Fig. 2.16. As this clock signal reaches a given station, this station places its next address bit on its respective RO line. These signals from all stations propagate towards the final OR and then back along the RI line to all stations. The next clock pulse must occur at each station after the time that this complete OR result reaches that station. As can be seen in Fig. 2.16, this requires a separation between clock pulses of

$$\max\{\tau'_{c,1} + \tau'_{1,o} + \tau'_{o,c}, \tau'_{c,M} + \tau'_{M,o} + \tau'_{o,c}\} \quad (2.21)$$

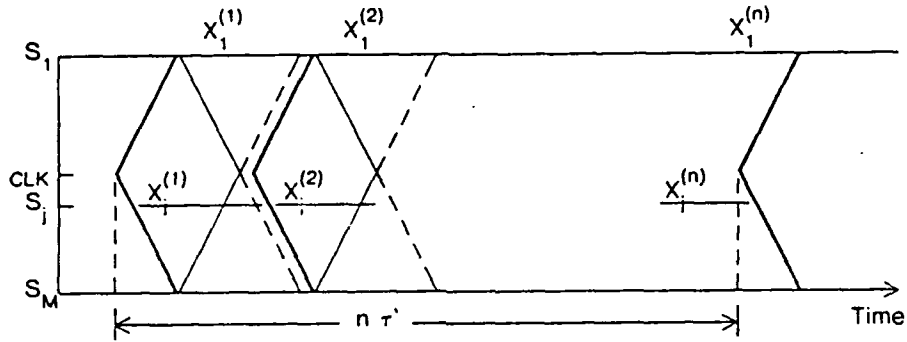


Fig. 2.17 Time-space diagram showing the clocking on the DSMA reservation channel during a reservation period. The clock is located in the center of the network. $X_j^{(i)}$ represents the value of RO_j after clock pulse i .

Obviously, the optimum position of both the clock and final OR gate (so as to minimize the time between clock pulses and hence the duration of the RP) is in the center of the network as shown in Fig. 2.15. Under these conditions, the expression in (2.21) reduces to τ' . From here on, we assume that this optimal condition is true; i.e., the clock and final OR gate are collocated in the center of the network, and the separation of pulses from the central clock is τ' . Now that we have the period of the clock, we can compute the duration of the RP.

Let n denote the number of bits in the address. With M users, $n = \lceil \log_2(M + 1) \rceil$. (The '0' address is used to indicate an end of round.) As can be seen in Fig. 2.17, $n + 1$ clock pulses are required to transmit and receive the entire n bit address, and so the duration of the RP, denoted by h , is given by

$$h = n\tau'. \quad (2.22)$$

At the end of the RP, one station will have successfully reserved the channel. This station will be the one that has the highest address among those making reservations.

Now that we have established the events over one RP let us examine the activity on the channel over consecutive transmissions. Let S_i be the station currently transmitting. Let $\tau_{i,c}$ denote the signal propagation delay on the data bus between S_i and the clock. Assume that S_j is the station to transmit after S_i . Depending on the relative values of h and T , and on the locations of S_i and S_j , the end of the RP (EORP) may occur at S_j either before or after the event $\text{EOC}(i)$. In Fig. 2.18(a), we show the case where the event EORP occurs *before* the event $\text{EOC}(i)$ at every point on the network. In Fig. 2.18(b), we show the case where EORP occurs *after* $\text{EOC}(i)$ at every point on the network. In Fig. 2.18(c), we show the case where the time space loci of EORP and $\text{EOC}(i)$ intersect. In this case, depending on its location, the synchronizing event for S_j could be either EORP or $\text{EOC}(i)$.

In general, given that S_i is currently transmitting, EORP reaches S_j a time period of $\tau_{i,c} + h + \tau_{j,c}$ after the time at which S_i begins to transmit; and $\text{EOC}(i)$ reaches S_j a time period of $T + \Omega + \tau_{i,j}$ after S_i begins to transmit. Consequently, the overhead between two consecutive transmissions in the same round can be expressed in general terms as

$$\max\{\tau_{i,j}, h + \tau_{i,c} + \tau_{j,c} - T - \Omega\} + \Delta \quad (2.23)$$

where the Δ accounts for the time required to detect either $\text{EOC}(i)$ or to sense that the bus is idle after detecting EORP. Between the last transmission in round r and the first in round $r + 1$, an addition RP is incurred. The overhead between these two transmissions is*

$$\max\{\tau_{i,j}, 2h + \tau_{i,c} + \tau_{j,c} - T - \Omega\} + \Delta \quad (2.24)$$

We now derive some performance measures. Under the assumption that the clock (together with the final OR gate) is located in the center of the network, we

*We assume here that the clock can also detect an all '0' address and then immediately initiate a new RP. Hence, there is no overhead between these two RPs due to propagation delays.

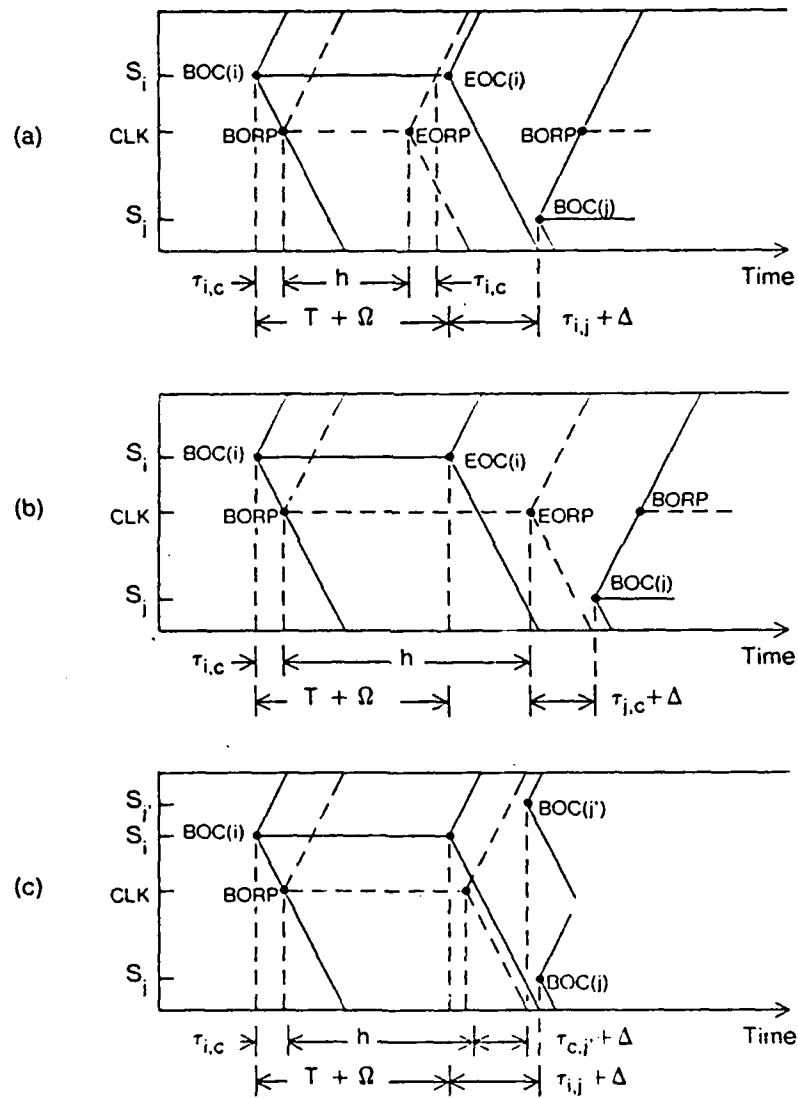


Fig. 2.18 Time-space diagram showing the execution of a reservation occurring simultaneously with a transmission. In (a) the end of the RP (EORP) is always seen before EOC(i), in (b) EORP is always seen after EOC(i), and in (c), depending on the relative positions of the stations, EORP may be seen either before or after EOC(i).

consider values of h such that EORP occurs either (i) before or (ii) after EOC(i) at every point in space. To satisfy case (i) requires

$$\begin{aligned}\tau_{i,j} &\geq 2h + \tau_{i,c} + \tau_{j,c} - T - \Omega \\ h &\leq \frac{1}{2}(T + \Omega + \tau_{i,j} - \tau_{i,c} - \tau_{j,c}) \\ \Rightarrow h &\leq \frac{1}{2}(T + \Omega - \tau)\end{aligned}\tag{2.25}$$

where the last expression is the necessary bound to satisfy all possible positions of i and j . Assuming $\tau' = \tau$ this condition can be expressed as

$$a \leq \frac{1 + \omega}{2\lceil \log_2(M+1) \rceil + 1}.\tag{2.26}$$

If the above condition holds, the overhead between consecutive transmissions by S_i and S_j is $\tau_{i,j} + \Delta$, and the performance of DSMA depends on the layout in the same way as the performance of SOSAM does. Briefly we mention two extremes, which are the bounds on the overhead for the case where EORP occurs before EOC(i). The first is where stations are located on the extremes of the network and where, given a transmission by S_i , the next station to transmit is on the opposite side of the network from the side of S_i . In this case the overhead over one round is

$$Y(M, N) = N\tau + N\Delta\tag{2.27}$$

The second extreme is the case where the stations' logical order is the same as their physical order on the bus.[†] In this case, like SOSAM, the overhead over one round is

$$Y(M, N) = 2\tau + N\Delta\tag{2.28}$$

[†]There is also the degenerate case where all stations are located at the same point. In this case the overhead over one round is $Y(M, N) = N\Delta$.

To satisfy case (ii), i.e., where EORP occurs after EOC at every point on the network, we require that

$$\begin{aligned}\tau_{i,j} &\leq h + \tau_{i,c} + \tau_{j,c} - T - \Omega \\ h &\geq T + \Omega + \tau_{i,j} - \tau_{i,c} - \tau_{j,c} \\ \Rightarrow h &\geq T + \Omega\end{aligned}\tag{2.29}$$

which, assuming that $\tau' = \tau$, can be expressed in terms of a as

$$a \geq \frac{1 + \omega}{\lceil \log_2(M + 1) \rceil}.\tag{2.30}$$

The overhead between consecutive transmissions by S_i and S_j is given by (2.23) and in this case, evaluates to be $n\tau' + \tau_{i,c} + \tau_{j,c} - (T + \Omega) + \Delta$. Clearly, the overhead over one round where N arbitrary stations transmit depends on the distance of each of these stations from the clock. A lower bound on this overhead occurs when all N stations and the clock are collocated in the center of the network and, including the extra RP incurred between rounds, is given by

$$Y(M, N) = N[n\tau' - (T + \Omega) + \Delta] + n\tau'.\tag{2.31}$$

An upper bound on the overhead occurs when each of the stations is located at one of the extreme ends of the network, and is given by

$$Y(M, N) = N[n\tau' + \tau - (T + \Omega) + \Delta] + n\tau'.\tag{2.32}$$

For a more realistic value, we assume that the stations are uniformly distributed over the length of the network such that the average value of $\tau_{i,c}$ is $\tau/4$. For such a layout, the expected overhead in a round, where N arbitrary stations transmit, is

$$Y(M, N) = N[n\tau' + \tau/2 - (T + \Omega) + \Delta] + n\tau'.\tag{2.33}$$

Using one of the above expressions for the overhead, it is straightforward to compute $C(M, N)$ and $D(M, N)$ for the case when EORP occurs after EOC at every point in space.

Neglecting ω and δ , we plot in Fig. 2.19 the capacity versus a using for the overhead (2.28) for values of $a \leq 1/[2\lceil \log_2(M+1) \rceil + 1]$ and (2.33) for values of $a \geq 1/\lceil \log_2(M+1) \rceil$. In the region given by $1/[2\lceil \log_2(M+1) \rceil + 1] < a < 1/\lceil \log_2(M+1) \rceil$, $C(M, N)$ depends on both the ratio h/T and the relative locations of the stations, making it difficult to obtain quantitative values. Instead, we fill in this small region by eye so as to maintain continuity between the other two regions. While $C(M, N)$ does show slight sensitivity to different combinations of M and N , the capacity does not increase with increasing M . In fact, it decreases with increasing M for a greater than about 0.1. This effect occurs because, for such values of a , the overhead incurred with each transmission depends on the length of the reservation period, and the latter increases with increasing M .

Comments: (i) The network architecture is a cumbersome arrangement of three control wires in addition to the data bus. The request-output line requires active taps thus compromising reliability by making the network sensitive to a single station failure.

(ii) Since stations are required to transmit their network addresses synchronously onto the *RO* line, a centralized clock is provided rendering the scheme partially centralized.

(iii) If $a \leq (1 + \omega)/(2\lceil \log_2(M+1) \rceil + 1)$, the overhead between consecutive transmissions in the same round is proportional to Δ , and is independent of a . If, on the other hand, $a \geq (1 + \omega)/\lceil \log_2(M+1) \rceil$, the overhead between consecutive transmissions is proportional to a , and it increases linearly with increasing a . As a result, the performance degrades rapidly with increasing bandwidth and (to a lesser degree)

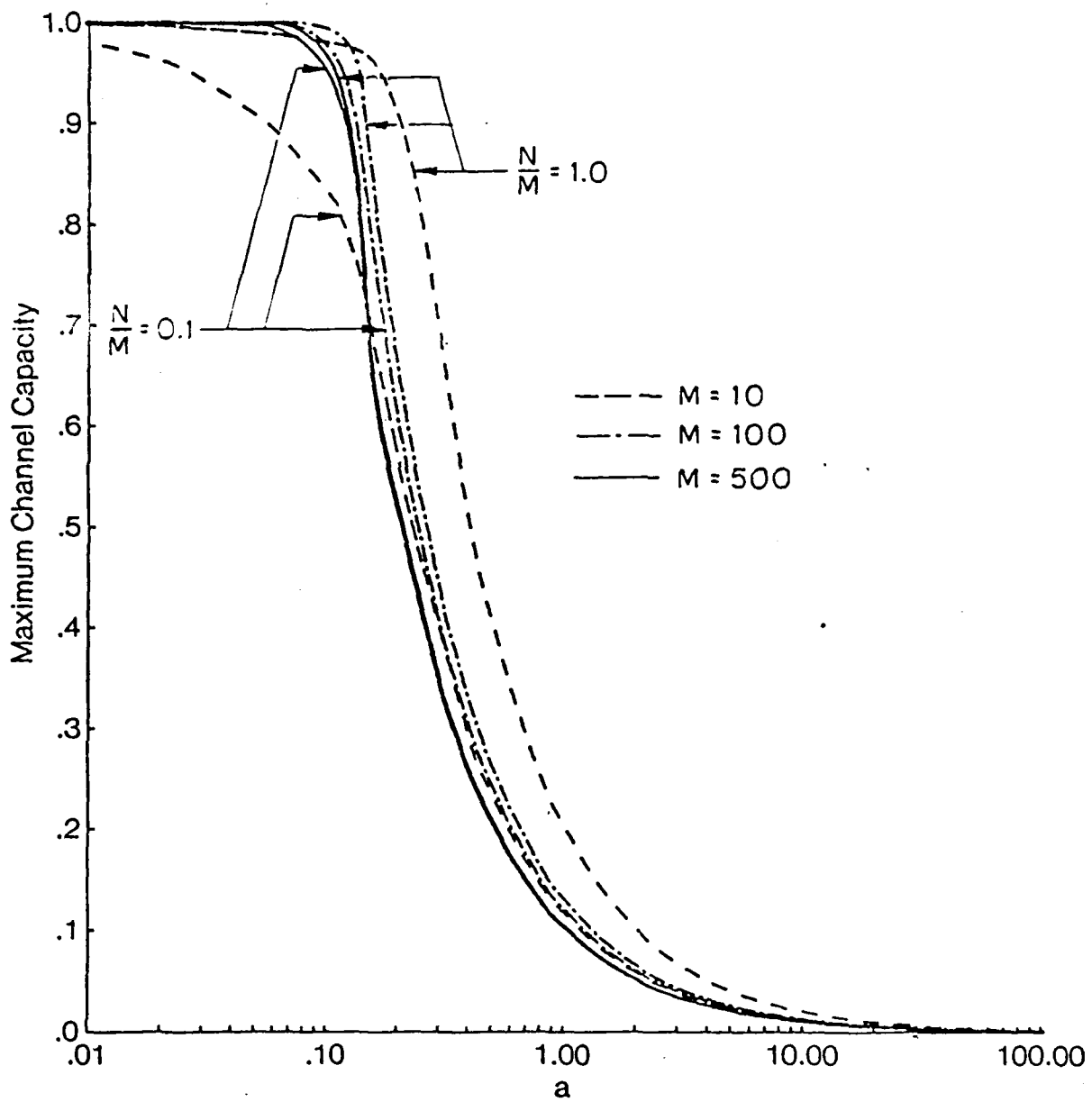


Fig. 2.19 Network capacity versus α for DSMA.

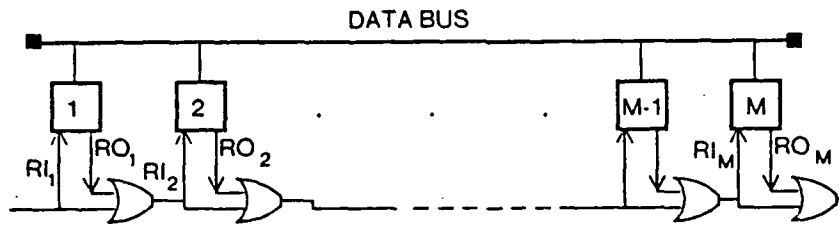


Fig. 2.20 BBC configuration used in the Control-Wire scheme.

with increasing M . Even a reasonably small value of $M = 31$ together with a large value of $\omega = 0.5$ requires $\alpha < 0.3$ to be below this bound.

(iv) Instead of using active/dormant states to achieve fair scheduling, one could instead require a station, upon becoming backlogged or having already transmitted in the current round, to wait for the beginning of the next round before attempting to access the channel. If the latter method is used, the service discipline becomes GSS instead of HOLS.

2.6.2 The Control-Wire (Eswaran, Hamacher, Shedler, 1979) [36], [37]

The network configuration used for this scheme is shown in Fig. 2.20. As in DSMA, it consists of a data bus, which is used for packet transmission, and a set of control wires inter-connected by OR gates, which is used to arbitrate access to the bus. While in DSMA, signals on all of the RO lines of the control-wire assembly are ORed and the result returned on the RI line to every station, in the Control-Wire scheme, each station receives on its RI line the logical OR of the signals from the RO lines of the stations to its left only. Thus, at any time, S_i can determine whether any station to its left is asserting its RO line.

A station wishing to transmit, say S_i , operates as follows. S_i places a reservation by setting RO_i , and waits a period of time (which is determined below) sufficient

for this reservation signal to propagate across the network. After this amount of time has elapsed, S_i is ready to transmit but may be inhibited from doing so by one of two signals. Either the data bus is sensed busy, meaning that some other station is currently using the channel, or $RI_i = 1$, meaning that some station to the left of S_i has placed a reservation for the channel. If, however, both the data bus is idle and $RI_i = 0$, S_i may transmit its packet conflict free. Having completed its transmission, S_i sets $RO_i = 0$, thereby allowing stations to its right to take their turns.

We consider now the method whereby conflict free transmission is guaranteed. Consider two stations S_i and S_j where $j > i$. Suppose that S_i sets $RO_i = 1$ at time t_i . If $RI_j = 0$, this signal makes a transition from 0 to 1 at time $t_i + \tau'_{i,j}$. If S_j is not yet transmitting, it will be inhibited from doing so until S_i has taken its turn. Suppose that S_j began to transmit just before time $t_i + \tau'_{i,j}$. In this case, $BOC(j)$ reaches S_i at time $t_i + \tau'_{i,j} + \tau_{i,j}$. Therefore, if S_i waits $t_i + \tau'_{i,j} + \tau_{i,j}$ after setting $RO_i = 1$ and only then senses the data bus for carrier, it will be appropriately inhibited from transmitting by the activity from S_j . Thus, to guarantee conflict free transmission in general, each station must wait $\tau' + \tau_s$ after setting $RO = 1$ before attempting to access the channel, i.e., before it reads the state of RI and senses the data bus for activity. Having successfully gained access to the channel and completed its packet transmission, S_i must relinquish its reservation so as to allow those stations on its right to take their turns. This is achieved by simply setting $RO_i = 0$.

Note that the Control-Wire access protocol described above is unfair in that it gives priority to stations on the left. This is the first of two algorithms proposed in [36] and is consistent with the earliest version of the scheme [37]. The second algorithm proposed in [36] allocates the channel fairly among all stations by means

of a round robin service discipline. From the above description of the protocol, it is clear that the maximum time separating two consecutive transmissions is $2\tau + \Delta$. Hence, the end of a round can be detected when the channel remains idle for longer than $2\tau + \Delta$ s. To ensure fair scheduling, each station, upon becoming backlogged or having already transmitted, waits for the end of the current round before it contends for the channel. This leads to the GSS discipline. Alternatively, the method of active/dormant states, which was proposed for DSMA and leads to the HOLS discipline, could be used.

In Fig. 2.21 we show the activity on the data bus and the control wire for a network with six stations operating under heavy traffic. Clearly, the maximum channel overhead over one round is

$$Y(M, N) = 5\tau + \tau' + (N + 1)\Delta \quad (2.34)$$

and hence, letting $a' \triangleq \tau'/T$,

$$C(M, N) = \frac{1}{1 + \omega + \delta + (5a + a' + \delta)/N} \quad (2.35)$$

$$D(M, N) = N(1 + \omega + \delta) + 5a + a' + \delta.$$

The relationship between the capacity and a for this scheme is similar to that for BID and Silentnet. However, the overhead over one round is on the order of 6τ as opposed to τ and 3τ for BID and Silentnet, respectively. As a result, $C(M, N)$ for the Control-Wire scheme is slightly lower for given values of a and N .

Comments: (i) Like DSMA, the control-wire taps are active components, and therefore, reliable operation of the network depends on reliable operation of all stations. Nevertheless, the topology of this network scheme is simpler than that of DSMA since only a single control wire is required. While in DSMA synchronous transmission of all the bits in the address field is required, in the Control-Wire scheme

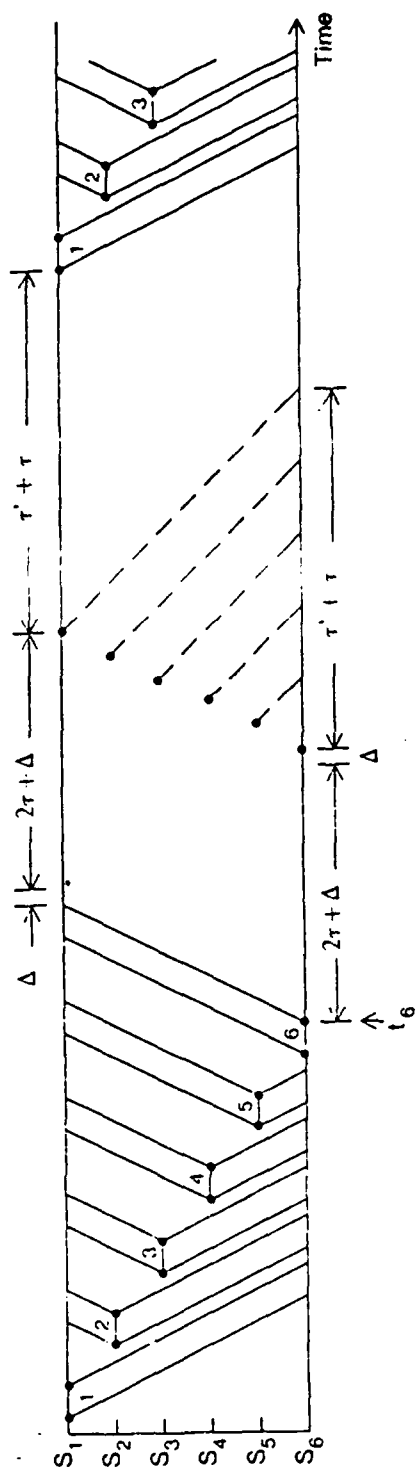


Fig. 2.21 Activity on the data bus and control wire for the Control-Wire scheme with six stations operating under heavy traffic conditions. The time-space locus of a $0 \rightarrow 1$ transition on the RO line is represented by a dashed line.

the stations make reservations asynchronously merely by asserting their respective RO lines. Since no centralized timing is required, the Control-Wire scheme is fully distributed.

(ii) In the access protocol described above, each station, having set $RO = 1$, is required to wait $\tau' + \tau$ before attempting to access the channel. This however is an excessive amount of time except for stations on the extreme left of the network. In fact it is sufficient for S_i to wait $\tau'_{i,M} + \tau_{i,M}$ to guarantee conflict free transmission. This, however, requires S_i to have knowledge of its position on the network. In the original description of the Control-Wire, this time interval is specified as $\tau'_{i,M} + \tau$ which is not the minimum and also requires knowledge of the station's position on the network.

(iii) Recall that stations recognize an end-of-round event when, after some EOC, the channel remains continuously idle for $2\tau + 2\Delta$ s. We note that all stations detect the EOC a time period of Δ after its actual occurrence except for the one that is the last to transmit in a given round. As a result, there is a discrepancy of Δ between the time reference of the station that is the last to transmit in a given round and that of the other stations. We discuss the possible ramifications of this discrepancy. With reference to Fig. 2.21 where S_6 is the last to transmit in the round, let t_6 be the time at which EOC(6) occurs at S_6 . S_6 will be prepared to transmit at time $t_6 + (2\tau + \Delta) + (\tau' + \tau)$ which, as is evident from the figure, is a time period of Δ before the reservation signal from S_1 reaches it. For the scenario depicted in Fig. 2.21, this is of no consequence since S_6 is already inhibited from transmitting by reservations from S_2 to S_5 . If however, those stations did not place reservations (or, alternatively, if S_1 to S_5 were collocated on the left side of the network) then S_6 would have been the first to transmit in this round, followed by S_1 . While conflict free transmission is guaranteed, the service discipline is not GSS. Nevertheless, GSS can be guaranteed by increasing the time that stations are required to wait after

setting $RO = 1$ and before attempting to transmit by Δ to $\tau' + \tau + \Delta$.

(iv) In the description of the scheme given above, we stated that a station relinquishes its reservation by setting $RO = 0$ at the instant that it *ends* its packet transmission. In the description of the scheme in [36], a station relinquishes its reservation at the instant that it *begins* its transmission. Using the method proposed in [36], it is possible for a packet collision to occur under some circumstances which we now describe. Consider the following scenario: two stations, S_j and S_k , $j < k$, have made reservations by asserting their respective RO lines; a third station, S_i , $i < j$, is currently transmitting. Suppose that $EOC(i)$ occurs and is detected by S_j at time t_j . At this time, S_j sets $RO_j = 0$ and begins to transmit. Clearly, S_k detects $EOC(i)$ at time $t_j + \tau_{j,k}$, $RI_k = 0$ at time $t_j + \tau'_{j,k}$, and $BOC(j)$ at time $t_j + \tau_{j,k} + \Delta$. In order to guarantee conflict free transmission, it is essential that $BOC(j)$ be detected by S_k *before* $RI_k = 0$ is detected. To satisfy this constraint requires that $t_j + \tau_{j,k} + \Delta < t_j + \tau'_{j,k}$, or

$$\Delta < \tau'_{j,k} - \tau_{j,k}. \quad (2.36)$$

It is not clear that this constraint can always be satisfied (for example, what if $\tau' < \tau$). Thus, the method of relinquishing the reservation described in [36] may lead to erroneous operation of the network, whereas the method described above always leads to correct operation of the network.

2.6.3 UBS-RR (Tobagi, Rom 1980) [39], [40]

In each of the previous two schemes, a control-wire assembly is used to place reservations for the data bus. By using the UBS network configuration shown in Fig. 2.2(b), the UBS-RR scheme implements the reservation access mechanism by transmitting both reservation signals and packets on the data bus. Just as

the unidirectional signalling along the control wire in the Control-Wire scheme establishes a natural ordering among stations, the unidirectional signal propagation on the *outbound channel* of the UBS also establishes a natural ordering among stations and allows a given station to indicate to those on its right its desire to transmit; however, each station must have the capability to sense activity on the outbound channel due to stations on the upstream side of its transmit tap. We now describe the access protocol which shows how stations place reservations and transmit packets in the conflict-free environment.

Assume for the purpose of this discussion that stations are numbered sequentially from left to right. With respect to a given round, any station, as in DSMA, may be either active if it has not yet transmitted in the current round or dormant if it has. A station that is idle or dormant does not contend for the channel. Only an active backlogged station may contend for the channel. According to this scheme, such a station, say S_i , waits for the next EOC on the inbound channel (EOC(in)). Then, to place a reservation, it transmits a short burst of unmodulated carrier of duration Δ and simultaneously begins listening to the outbound channel for a period of time of at least $2\tau_{1,i} + \Delta$. This is sufficient time for EOC(in) to propagate to the end of the inbound channel and then for a possible reservation burst from the beginning of the outbound channel to propagate to and be detected by S_i . If the outbound channel is sensed idle during this entire period then there are no stations to the left of S_i reserving the channel. S_i transmits its packet and goes to the dormant state. If S_i does sense activity on the outbound channel during this period, it defers to the station on its left and waits for the next EOC(in) at which time it repeats the protocol.

It is clear that according to this algorithm the most upstream station of those making reservations transmits conflict free. (While it is possible for reservation

bursts to overlap, packet transmissions never incur a collision.) Like DSMA, this leads to the HOLS discipline. Since dormant stations do not contend for the channel, we are assured that no station will transmit more than one message in a round and thus fair scheduling is attained. Looking at the activity on the channel (Fig. 2.22), one will observe that the time separating two consecutive packets in the same round is one *round trip delay* (i.e., twice the propagation delay between two end stations) plus 2Δ . In this gap are the reservation bursts of the active backlogged stations attempting to gain access to the channel. (These burst may overlap.) When the inbound channel is idle for longer than this time, then all stations are either idle or dormant meaning that the round has ended. At this time, all dormant stations set their states to active and contend for the channel by transmitting reservation bursts. As a result, an overhead of $4\tau + 3\Delta$ is incurred between two consecutive rounds as shown in Fig. 2.22(b).

Clearly, the total overhead in a round where N stations transmit is

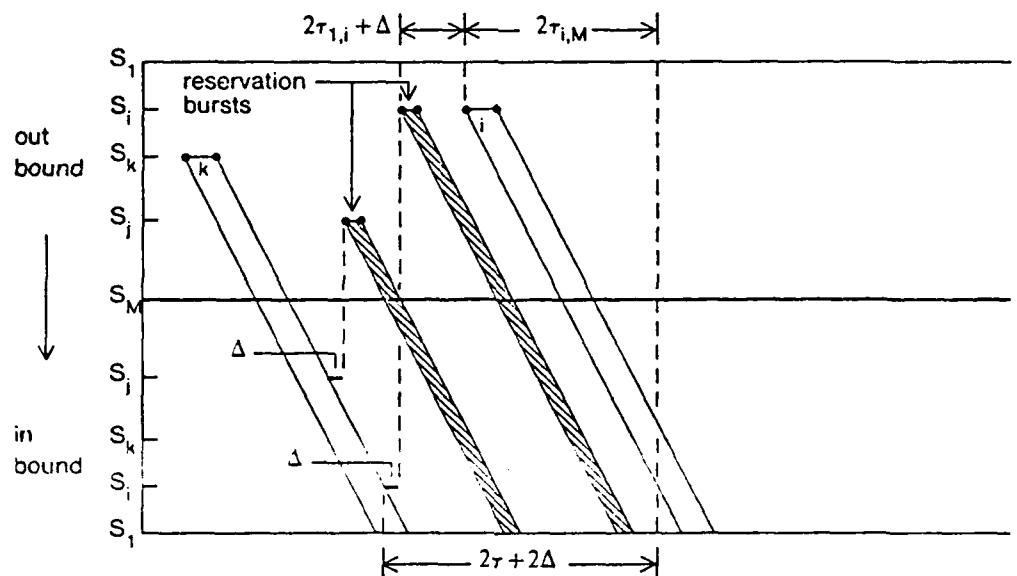
$$Y(M, N) = N(2\tau + 2\Delta) + 2\tau + \Delta \quad (2.37)$$

and the network capacity and packet delay are

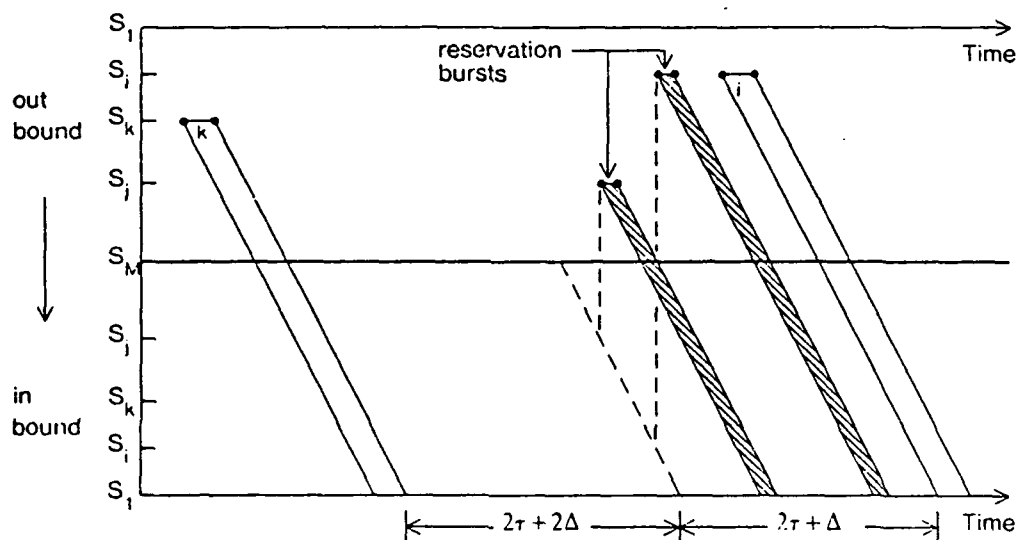
$$C(M, N) = \frac{1}{1 + \omega + 2\delta + 2a + (2a + \delta)/N} \quad (2.38)$$

$$\text{and } D(M, N) = N(1 + \omega + 2\delta + 2a) + 2a + \delta.$$

Like DSMA, the overhead per transmission increases linearly with increasing a . As shown in Fig. 2.23, $C(M, N)$ decreases sharply as a becomes larger than about 0.3 which makes this scheme unsuitable for applications requiring high bandwidth or small packets, as is the case with other schemes suffering from this performance limitation (such as DSMA, BRAM, and CSMA).



(a)



(b)

Fig. 2.22 Activity on the channel between two consecutive transmissions in UBS-RR. In (a) S_i and S_j are both active and backlogged at the time EOC(in) is detected. In (b) they are both dormant and backlogged at this time. S_i , being the more upstream station on the outbound channel, transmits first.

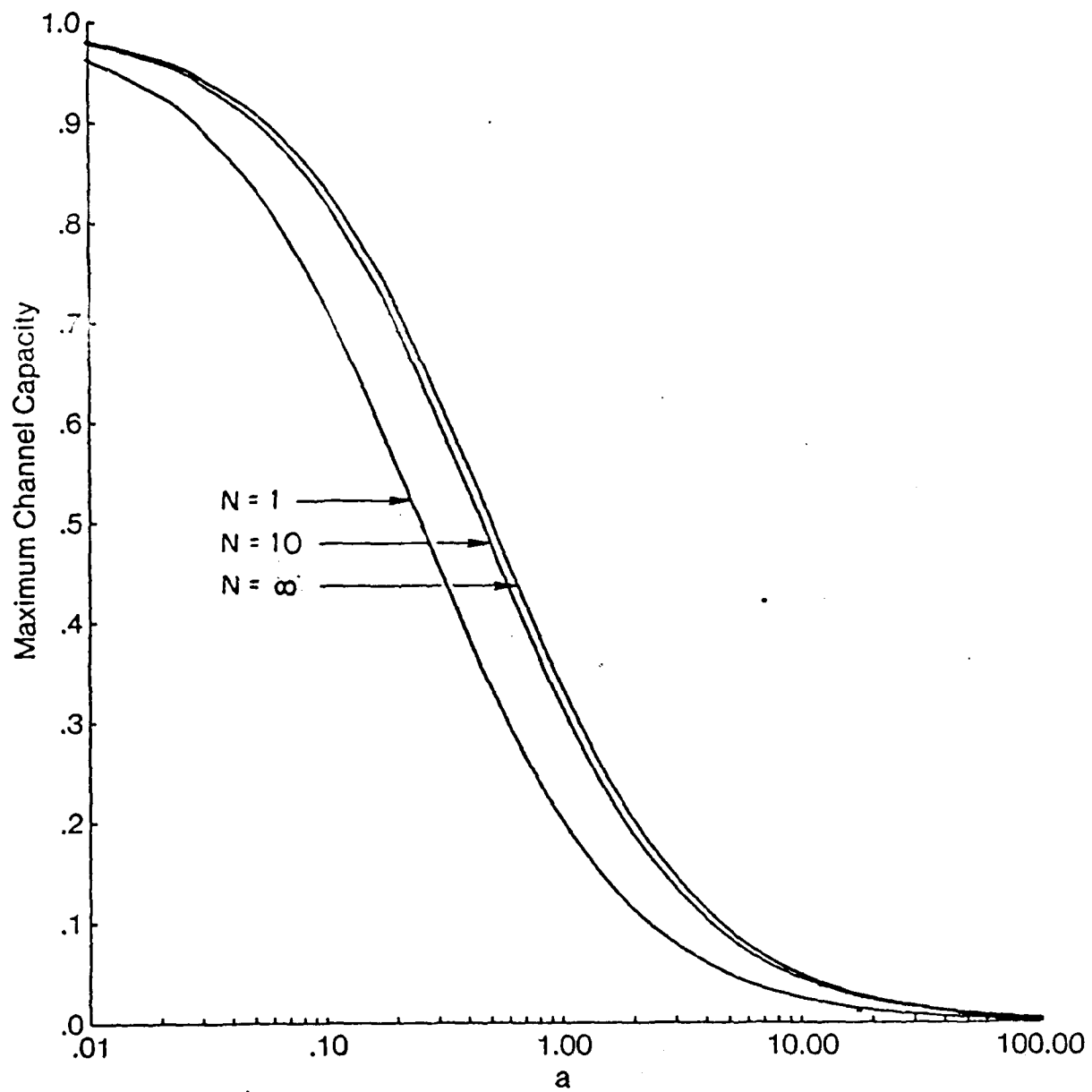


Fig. 2.23 Capacity versus a for UBS-RR.

Comments: (i) As in the Control-Wire scheme, stations do not need unique index numbers but are naturally ordered according their positions on the network. Each station is connected to the channel via passive taps. Hence, the network is less vulnerable to single station failures than the previous two schemes.

(ii) As described above, each station is required to wait at least $2\tau_{1,i} + \Delta$ after detecting EOC(in) so as to detect a possible reservation burst from some station on its left. This is the minimum time required. If it is undesirable that the access protocol requires knowledge of the stations' positions on the network, a constant time-out period of $2\tau + \Delta$ can be used by all stations at some cost in performance. If this approach is used, the channel overhead between consecutive transmissions by S_i and S_j is $2\tau + 2\Delta + 2\tau_{j,M}$ (as opposed to $2\tau + \Delta$ if the minimum time-out period is used). The maximum idle time between two consecutive transmissions is $4\tau + 2\Delta$. Since Δ is incurred in detecting EOC(in), dormant stations must measure an idle period of $4\tau + \Delta$ to determine that a round has ended.

(iii) This scheme is robust in the sense that the algorithm is completely distributed and conflict free, any station can start a new round when the current one ends and, if the network goes idle for an extended period of time, any station can begin to transmit by detecting this idle condition and making a reservation. If there are no reservations from the left within the time-out period, such a station may transmit.

(iv) Some improvement in performance can be gained if each active backlogged station, upon detecting EOC(in), immediately transmits its packet instead of transmitting its reservation burst and waiting for the time-out period. A transmitting station must continue to monitor the outbound channel for a transmission from upstream while the packet transmission is in progress. If such a station detects activity from upstream, it aborts its transmission, allowing the upstream one to continue conflict free. Thus we see how the reservation mechanism used on a UBS configuration is similar to the attempt-and-defer basic access mechanism which has

been briefly described in section 2.3.3, and will be discussed in detail in the next section.

2.7 Schemes Using the Attempt-and-Defer Access Mechanism

In this section we describe those schemes that use the attempt-and-defer method as their basic access mechanism. Only the UBS configuration can support this access mechanism. None of these schemes require stations to be indexed. Rather, the unidirectionality of the signal propagation provides a natural ordering of the stations which is exploited by the round robin scheduling functions. Characteristic features of these schemes include (i) synchronous or asynchronous operation, (ii) completely distributed or partially centralized access protocols, (iii) the service discipline and (iv) the performance achieved. Some of these schemes allow random access techniques to be used when the channel is lightly loaded but revert to DAMA when collisions are incurred. We classify such schemes as *hybrid forms* of the attempt-and-defer access mechanism. In this section we first discuss the pure forms of the attempt-and-defer access mechanism and then we discuss the hybrid forms.

2.7.1 Pure forms of the Attempt-and-Defer Access Mechanism

2.7.1.1 Expressnet (Fratta, Borgonovo, Tobagi, 1981) [10], [11]

The topology of Expressnet is shown in Fig. 2.24. As in the UBS-RR, stations access the outbound channel to transmit and the inbound channel to read

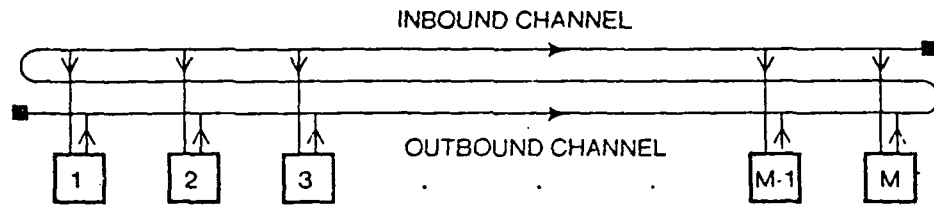


Fig. 2.24 UBS configuration used with Expressnet.

the transmitted data. Each station also has the ability to sense activity on the outbound channel due to stations on the upstream side of its transmit tap. While in the UBS-RR, EOC(in) is used as the synchronizing event, in the access protocol of Expressnet, the end-of-carrier on the outbound channel (EOC(out)) is used to determine when to transmit. In accordance with the attempt-and-defer access mechanism, all backlogged stations attempt to transmit when they detect EOC(out) and defer to upstream transmissions. More specifically, S_i waits for EOC(out), and then transmits carrier while simultaneously listening for a transmission from upstream. If, within Δ_s , EOC(out) is detected, then some station upstream from S_i is transmitting. S_i defers its transmission and waits for the next EOC(out). The overlap of these two transmissions is limited to the first Δ_s of each, thus affecting only the beginning part of the preamble. On the other hand, if S_i does not detect any activity from upstream within Δ_s , it proceeds with its transmission. Note that there is a single station which does not have to abort its transmission, and hence, it transmits successfully. Moreover, a station that has completed the transmission of a packet in a given round will not encounter the event EOC(out) again in that round, thus guaranteeing that no station will transmit more than once in a given round.

We now describe the mechanism for initiating a round. Define a *train* to be a

succession of transmissions in a given round. A train is generated on the outbound channel and entirely seen on the inbound channel by all stations. The end of a train on the inbound channel (EOT(in)) is detected whenever the idle time exceeds Δ . Using a topology for Express-net as shown in Fig. 2.24, EOT(in) will visit each station in the same order as they are permitted to transmit. To start a new round, EOT(in) is used as the synchronizing event, just as EOC(out) was used in the above description. Thus stations synchronize their transmissions to the first of the two events EOC(out) or EOT(in). (Note that only one such event can occur at a given point in time.)

Now we discuss the performance of the system. In Fig. 2.25 we show the activity in Expressnet over one round under the heavy traffic condition. Consecutive transmissions in the same round are separated by a gap of Δ . In addition, as with all schemes using the attempt-and-defer mechanism, the first Δ seconds of each packet is likely to be corrupted by an overlapping transmission. In this figure, and in other time-space diagrams showing the activity of a scheme using this access mechanism, these Δ s are shown by the shaded portion at the beginning of each transmission. The time separating two consecutive trains is the propagation delay between the transmit tap and the receive tap of a station plus 2Δ and is the same for all stations. For the topology shown in Fig. 2.24 this amounts to $2\tau + 2\Delta$. Thus the capacity of Expressnet and the maximum delay are

$$C(M, N) = \frac{1}{1 + \omega + 2\delta + (2a + \delta)/N} \quad (2.39)$$

$$\text{and} \quad D(M, N) = N(1 + \omega + 2\delta) + 2a + \delta.$$

The capacity versus a for Expressnet with $\delta = \omega = 0$ is plotted in Fig. 2.26. As with the more efficient schemes that have been discussed so far (such as BID, Silentnet, and the Control-Wire scheme), the capacity of Expressnet does not depend on M

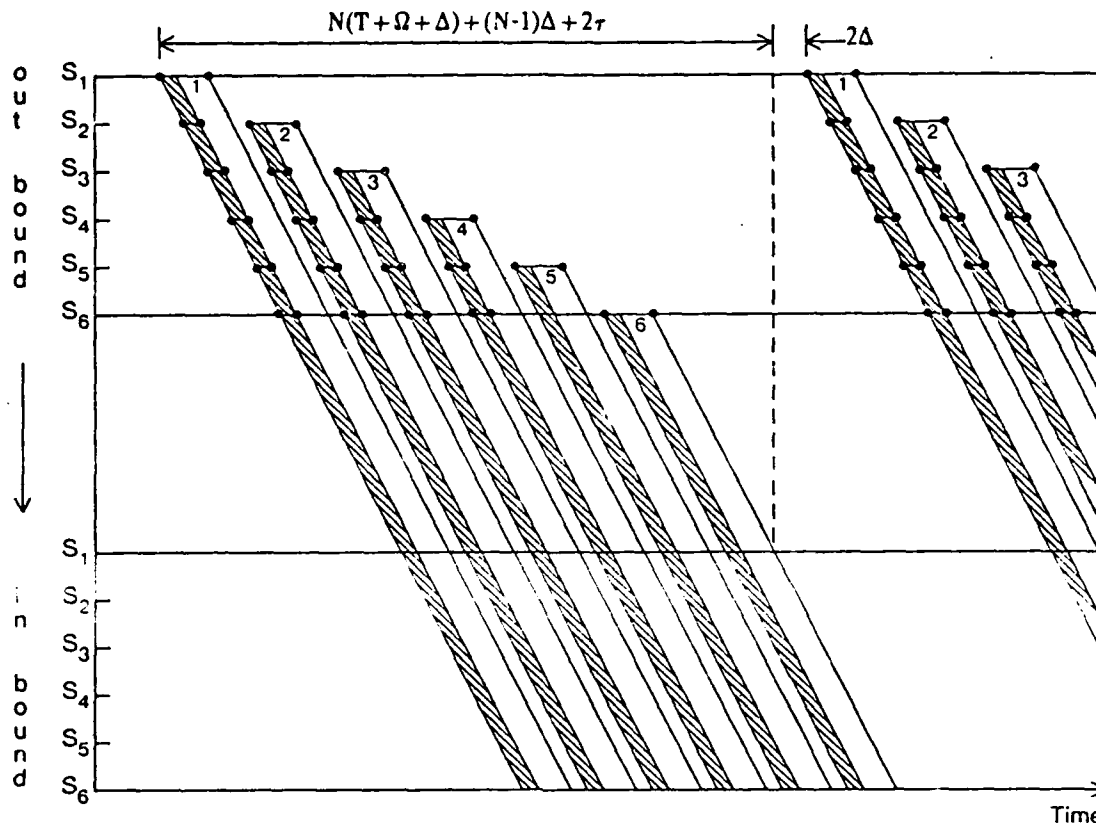


Fig. 2.25 Activity on Expressnet over one round under the heavy traffic condition. The shaded portion at the beginning of each transmission represents the portion of the preamble that is likely to be corrupted by an overlapping, aborted transmission.

but only on N , and it increases with increasing N .

Two variants of Expressnet have been proposed in the literature and we mention them briefly. The first is a partially centralized version of Expressnet called D-Net (Tseng, Chen, 1983) [41]. In this variant, the most upstream node on the outbound channel undertakes the responsibility of transmitting the start-of-round token each round. This approach simplifies the station design; no cold start procedure is required, it is not necessary for a station to listen for EOT on the inbound channel, and a station need never transmit the start-of-round token. In addition, since it is

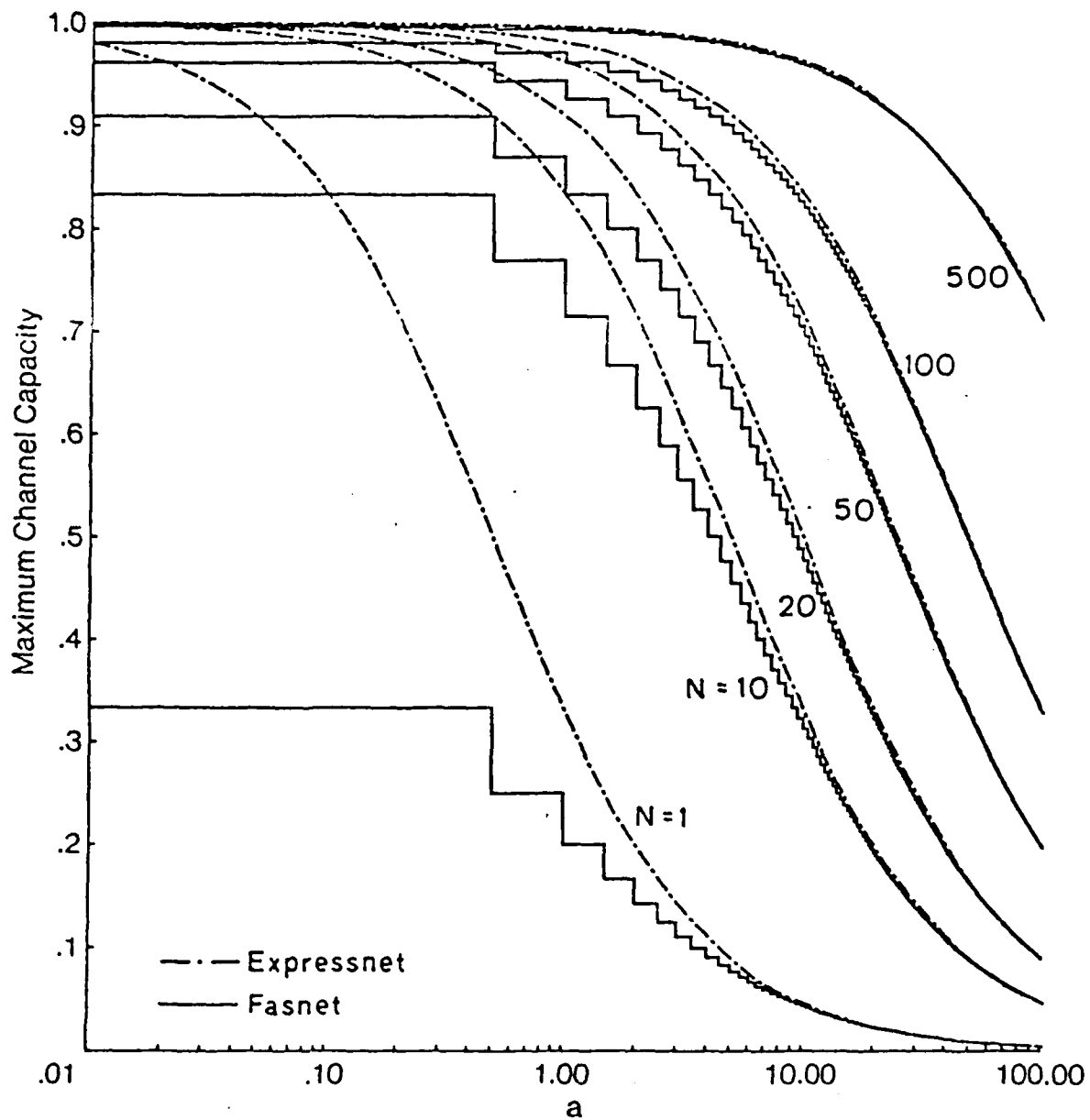


Fig. 2.26 Network capacity versus a for Expressnet and Fasnet.

no longer a requirement for the EOT to visit each station on the inbound channel in the same order as their physical order on the channel, the topology can be reduced from a double folded channel to a single folded channel as shown in Fig. 2.2(b). The functionality and performance of D-Net are identical to that of Expressnet; D-Net merely simplifies the stations' access protocol by centralizing the functions associated with the generation of the start-of round token and obviates the need for the cold-start procedure.

The second variant uses the BBC configuration shown in Fig. 2.20. In this variant, (Fratta, 1983) [38], the synchronizing event is a $1 \rightarrow 0$ transition, which propagates from left to right along the unidirectional control wire. A station wishing to transmit, say S_i , examines RI_i . If $RI_i = 1$, then S_i sets $RO_i = 1$, waits for the $1 \rightarrow 0$ transition on RI_i , and then begins to transmit. Having transmitted, S_i sets $RO_i = 0$, thus allowing the $1 \rightarrow 0$ transition to propagate along the control wire to S_{i+1} . If, when S_i examines the state of the control wire, it determines that $RI_i = 0$, then S_i has had its turn, and will not see another $1 \rightarrow 0$ transition on the control wire in the current round. In this case, S_i must wait for the next round before contending for the channel. An end of a round occurs when the data bus is detected idle for a time period of $\tau' + \tau$. At this time all backlogged stations set $RO_i = 1$ and, by means of an appropriate time-out at each, the new round is begun by the leftmost backlogged station. This station transmits and then sets $RO = 0$, thereby regenerating the synchronizing event. While the Expressnet access protocol does not make any use of information regarding the topology, this variant requires knowledge of τ' and τ . Also, the performance of the latter is the same as that of the Control-Wire scheme which is inferior to that of Expressnet.

Comments: (i) Expressnet is fully distributed and asynchronous. It does not require that stations be indexed or ordered in any way; ordering is automatically achieved

due to the unidirectionality of the signal propagation. Furthermore, stations require no knowledge of the topology or size of the network. Not even the value of τ is required.

(ii) To avoid losing the synchronizing event EOT(in) which happens if no packets are ready when it sweeps the inbound channel, all stations (whether idle or backlogged) transmit a short burst of unmodulated carrier of duration Δ whenever EOT(in) is detected. (If the station is in the backlogged state it does so before attempting to transmit a packet.) We refer to this burst as the start-of-round token; in [10] and [11] it is referred to as a *locomotive*. If the train were to be empty, then the end of the start-of-round token constitutes EOT(in). Hence the system is robust and is periodically synchronized by the event EOT(in). The description of Expressnet in [10] and [11] also presents a distributed *cold-start procedure* whereby this event can be generated if the network is completely silent, such as when it is powered on. A station undertakes the cold-start procedure when it detects that the channel is idle for a time of $2\tau + 2\Delta$. At this time it transmits carrier on the outbound channel for $2\tau s$, deferring if necessary to another station undertaking the cold-start procedure on its left. The end of this transmission provides the unique synchronizing events EOC(out) and EOT(in).

2.7.1.2 Fasnet (Limb, Flores, 1981) [12], [13]

The topology of Fasnet consists of the UBS configuration shown in Fig. 2.2(a). While Expressnet is a completely distributed, asynchronous scheme, Fasnet is partially centralized with the most upstream station (or head station) and the most downstream station (or end station) on each bus performing special functions. All users can read from and write to both channels. A user wishing to send a packet transmits on one of the channels such that the recipient is downstream from the

sender. As the two channels are identical we consider events on channel A. For channel A the head user is user 1 and the end user is user M . The head user transmits a clock signal which keeps the system bit synchronous. From this clocking information and a synchronization pattern transmitted at regular intervals, stations listening to the channel are able to identify fixed length slots travelling downstream. Each slot begins with an access control field (AC) which determines how and when each station may access the channel. The structure of the AC field consists of three bits. The *start* bit (SB), when set, indicates the start of a new round or cycle (SOC). The *busy* bit (BB), when set, indicates that a packet has been written into the slot. After each of these bits is a dead time which allows the station some time to read and process them as the slot is travelling by. The third bit, called the *end* bit (EB), is located in the dead time between the start and busy bits. This bit is used by the end station to instruct the head station *via channel B* to initiate a new cycle on channel A.

The details of this access protocol are as follows. A station wishing to transmit a packet on channel A waits for a new round which begins when $SB=1$ is detected. It then reads and sets the BB of each slot (setting an already set bit is assumed to have no effect). When an empty slot is detected, the station writes its packet into it. If the station has another packet to transmit it waits for $SB=1$ before attempting to transmit again. This ensures that each station transmits only once in every round. In particular, the GSS discipline is achieved. If, instead, each station maintains an active/dormant flag which is set to dormant after transmitting and reset to active when the station reads $SB=1$, then fair round robin scheduling can be achieved without requiring idle stations, upon becoming backlogged, to wait for a new round before transmitting. This approach was used in an earlier version of Fasnet which is described in [12] and it leads to the HOLS discipline. Thus the basic access mechanism of Fasnet can be classified as the attempt-and-defer access mechanism

since a station attempts to transmit by writing the BB but defers if it determines that BB is already set. A new round is initiated by the head station in cooperation with the end station. The end station examines all slots on channel A, decoding the status of SB and BB. Upon detecting SB=1, the end station looks for the first slot in which BB=0 (indicating that all users are idle or waiting to detect SB = 1), at which time it sets EB=1 in the next slot on channel B. The head user, detecting EB=1 on channel B, then sets SB=1 in the next slot on channel A.

Due to the slotting of the time axis, the inter-round overhead is an integral number of slots and on the average is $\lceil 2\tau/T \rceil T + T$ where T is the transmission time of a slot. This implies that the inter-round overhead is at least two slots, even if τ is close to zero. The interpacket overhead is merely the length of the AC field which is on the order of a few bit times [13], [14]. Denoting the length of the AC field normalized to T by ac , the network capacity of Fasnet is given by

$$C(M, N) = \frac{1}{1 + ac + (\lceil 2a \rceil + 1)/N} \quad (2.40)$$

Curves showing the capacity versus a for Fasnet and Expressnet with $\delta = \omega = ac = 0$ are plotted in Fig. 2.26. Since the overhead over one round in Fasnet is never less than two slots, Fasnet achieves a poor utilization for small values of N . In Expressnet, there is no slotting of the time axis, and so as $a \rightarrow 0$ the overhead becomes zero and the capacity approaches 1. (The same is true for all non-slotted schemes.) Note that in Fasnet there are two channels which operate independently. Equation (2.40) and the capacity curves in Fig. 2.26 represent the network capacity as a fraction of either the bandwidth per channel or of the combined bandwidths of both channels.

In Fasnet, transmissions are bit synchronous with the clock signal from the head station and so no preamble is required. All other schemes, being asynchronous,

require a preamble. The effect of a nonzero preamble on the network capacity is shown for Expressnet in Fig. 2.27. A preamble which is on the same order of magnitude as the packet transmission causes a significant degradation in the capacity. Although this figure only shows the effect of the preamble on Expressnet, all asynchronous schemes suffer this degradation in performance as a result of the preamble.

As with all schemes using the GSS discipline, the maximum delay incurred by a packet is given by the maximum length of two rounds; a station may become backlogged just after the SOC has gone by, and hence have to wait one whole round before it sees the next SOC, plus, if it is the farthest downstream, another round before it can transmit this packet. Thus $D(M, N)$ is

$$D(M, N) = (2N - 1)(1 + ac) + \lceil 2a \rceil + 1. \quad (2.41)$$

If the protocol implements the HOLS discipline such as with an active/dormant flag, a station never has to wait more than one round before being able to transmit, and in this case, $D(M, N)$ is

$$D(M, N) = N(1 + ac) + \lceil 2a \rceil + 1 \quad (2.42)$$

Comments: (i) Stations are synchronized to a common clock generated by the head station on each bus. Consequently, there is no need to have a preamble preceding each transmission. This can improve the channel utilization as compared to an asynchronous system by 5 to 50 percent depending on the expected packet length. (For example, assuming that the length of the preamble in an asynchronous system is 64 bits and the length of the packet is 700 bits, the channel utilization is increased by about 10 percent in an asynchronous system due to the lack of the preamble.)
(ii) Due to the slotted channel, this scheme is best suited for fixed length packets.

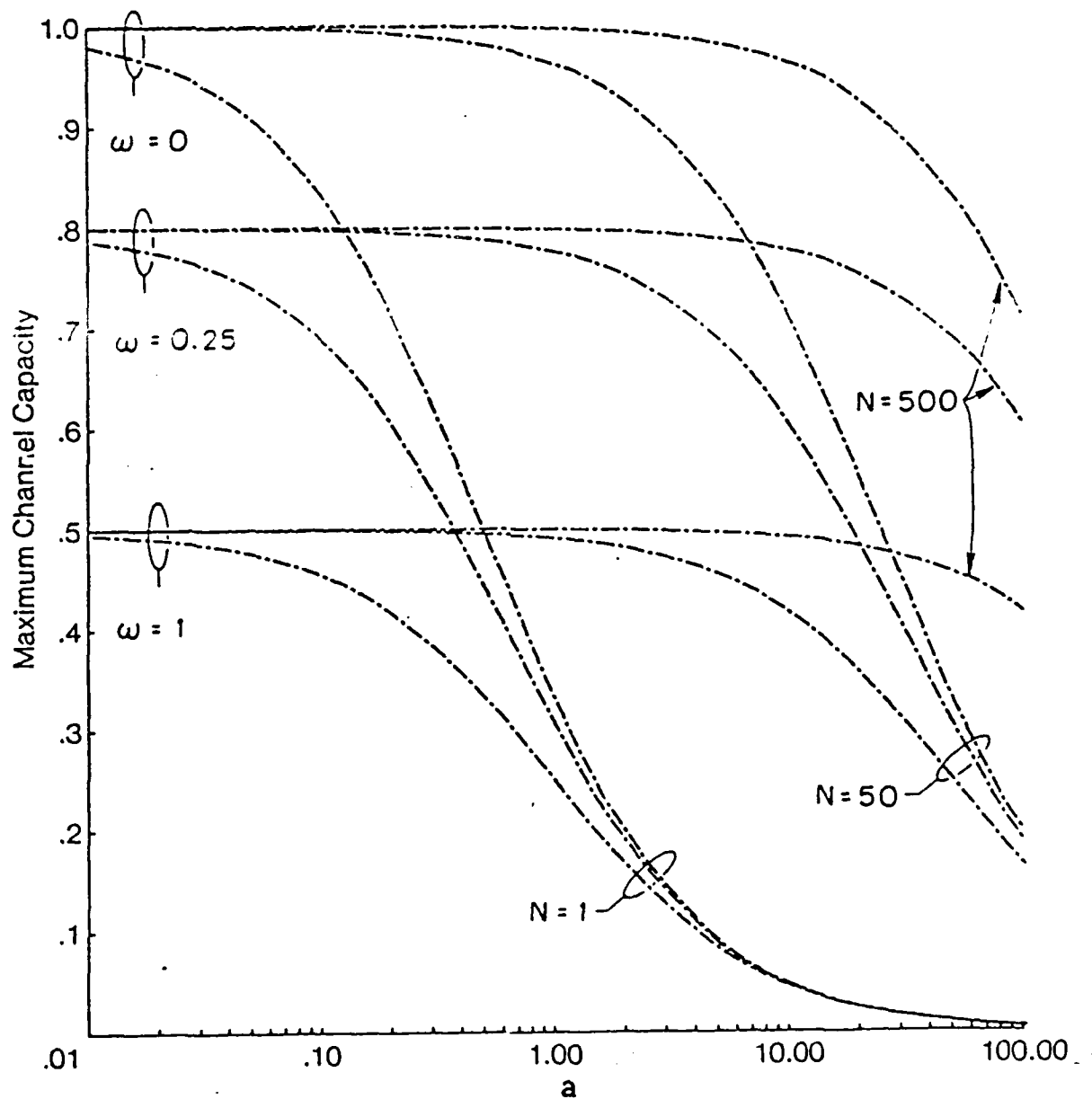


Fig. 2.27 Network capacity versus a for Expressnet with $M = 1, 50$, and 500 , and with three values of the preamble corresponding to $\omega = 0, 0.25$ and 1 .

Unless packets are always exactly the size of a slot, some fraction of the slot will not be used, and this constitutes wasted capacity. Also, in the case where a packet requires multiple slots, the overhead associated with the AC field and headers is incurred for each slot.

(iii) S_i , wishing to send a message to S_j , must select one bus on which to transmit such that S_j is downstream from S_i . Therefore each station must know which stations are to its left and which are to its right. This information could easily be stored in a table and maintained by the station itself. If S_i were unsure of the location of S_j , it could transmit on both channels. As soon as an acknowledgment were received, S_i could determine the relative location of S_j .

(iv) Since this scheme has two channels which operate independently, the total bandwidth available is twice the bandwidth of one channel.* If we were to compare Fasnet to the other schemes, we would, however, consider only one of these two channels. The reason is the following. Fasnet requires two transmitters and transmit taps, two receivers and receive taps, and two finite state machines to execute the access protocol. For all intents and purposes, this amounts to two networks with some arbitrary mechanism whereby a station selects one of the two networks on which to transmit (in this case, the relative location of the recipient is the selection criterion). For any of the other networks one could achieve the same effect of doubling the bandwidth by supplying another bus and doubling the hardware at each station.

(v) As long as the head and end stations are functioning properly, this scheme is completely robust, since stations can always achieve bit and frame sync from the head station. The scheme can also be made robust against a head station failure by providing each station with the ability to perform the functions of the head station.

*This assumes that the bandwidths of each of the two channels are equal and that the expected traffic is the same on both of them.

Then, an arbitrary station can provide the functions of the head station if it does not detect the clock signal from any other station upstream.

2.7.1.3 U-Net (Gerla, Yeh, Rodrigues, 1983) [42]

U-Net is an asynchronous scheme implemented on the UBS configuration shown in Fig. 2.2(a). Like Fasnet, this scheme is partially centralized. The end stations are responsible for initiating new rounds by transmitting start-of-round tokens. A station, S_i , wishing to transmit, listens to both busses for a start-of-round token. Denote the bus on which this token is detected as the *forward bus*, the other as the *reverse bus*. Suppose that S_i detects a token on bus A. (This token would have been transmitted by S_1 .) S_i then attempts to access the forward bus using the attempt-and-defer access mechanism described in Expressnet. Once S_i establishes on the forward bus that it has access right to the channel, it transmits its packet on both busses simultaneously.

As in Expressnet, activity on the forward bus consists of a sequence of transmissions separated by gaps of Δ . S_M is able to determine the end of the round on bus A when it detects an idle period greater than Δs (or immediately after its own transmission if it transmits in this round). At this time, S_M initiates a new round on bus B. Clearly, the service discipline of this scheme is GRSS. This is due to the fact that stations are serviced sequentially, the order of service reverses from round to round, and gating is in effect by requiring stations to wait for a start-of-round token before attempting to transmit. A time-space diagram showing the activity on the channel for this scheme is shown in Fig. 2.28. Neglecting the start-of-round token, the capacity is given by

$$C(M, N) = \frac{1}{1 + \omega + 2\delta + (a + \delta)/N} \quad (2.43)$$

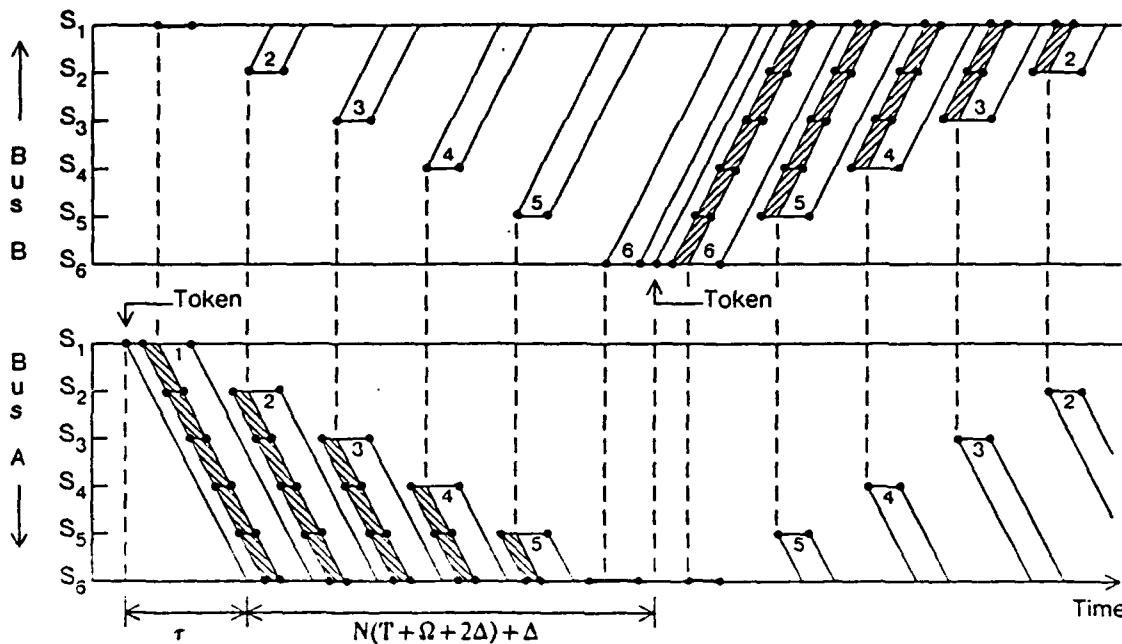


Fig. 2.28 Activity on the channel for U-Net operating under the heavy traffic condition.

Neglecting ω and δ , the network capacity of this scheme is the same as for BID and is plotted as a function of a in Fig. 2.10. Like BID, the delay $D(M, N)$ varies with each station and with the order of service. The bounds are given by the delay incurred at an end station where, neglecting the start-of-round token, $D(M, N)$ alternates between the values

$$D_{\min}(M, N) = 1 + \omega + 3\delta, \quad (2.44)$$

$$D_{\max}(M, N) = 2[N(1 + \omega + 2\delta) + a + \delta].$$

For any other station, the delay lies between these two values.

Comments: (i) U-Net is similar to BID in the way that the end stations are used to start new rounds. In BID, a "direction indicator" is transmitted with each packet to establish the order of service in a given round; the end station that is the last to

transmit in a given round toggles the direction indicator, thereby starting the next round. U-Net achieves the same result by means of a token, transmitted by an end station, which establishes the forward bus for a given round; however, this method requires the token to be a uniquely identifiable signal such as a reserved bit pattern. In BID, the end station is required to transmit a token to start a new round if it does not have a packet to transmit, but this token is simply an empty packet with the appropriate value in the direction-indicator field.

(ii) This access protocol requires two distinct busses, two transmitters and two receivers. This almost doubles the cost of the hardware. While this is true also of Fasnet, in Fasnet each packet is transmitted on only one of the two busses whereas in U-Net each packet is transmitted on both busses. In addition, U-Net is asynchronous and requires that each packet be preceded by a preamble. Thus, if the transmission rate on each of the busses is W , the bandwidth available in Fasnet is $2WC_{\text{Fasnet}}(M, M)$. The bandwidth available in U-Net is $WC_{\text{U-Net}}(M, M)$; less than half that of Fasnet. The bandwidth available in U-Net is approximately the same as that available in Expressnet but at twice the cost.

(iii) The issues concerning robustness discussed for BID are equally applicable to U-Net. As long as both end stations operate correctly, the network will remain operational. A recovery technique similar to the one discussed for BID could be used to accommodate end station failures in this scheme.

2.7.1.4 Token-Less Protocol (Rodrigues, Fratta, Gerla, 1984) [43]

In section 2.5.4 we described how Silentnet implemented an access protocol similar to BID but in a distributed fashion, i.e., without the need for end stations. The Token-Less Protocol scheme (TLP) similarly implements a completely distributed version of U-Net. To achieve this, stations use the event EOC instead of an explicit token to determine which bus is the forward bus, as explained below.

A station S_i , wishing to transmit, senses both busses for activity. Due to the unidirectionality of the signal propagation, and the conflict-free nature of transmissions, only one of the two busses can be sensed busy by S_i at any given time. Suppose S_i senses carrier on bus A. (Hence, it senses bus B to be silent.) Bus A is denoted the *forward bus* and plays the same role as the forward bus in U-Net. S_i waits for EOC(forward) and then attempts to access the channel by transmitting for Δs on the forward bus. As in U-Net, S_i transmits its packet on both busses simultaneously once it has established that it has acquired the right to access the channel. Having completed its transmission, S_i continues to transmit carrier on the reverse bus without interruption, until it detects BOC(reverse). We call this transmission of carrier on the reverse bus the *blocking signal* (BS). Hence, EOC(forward) propagates to the next station in turn after S_i , but no EOC event is generated on the reverse bus. As seen in the time-space diagram in Fig. 2.29, this continues until the forward bus (bus A in Fig. 2.29) becomes completely idle while the reverse bus (bus B) carries the blocking signal, which is being transmitted by the last station to have transmitted in the current round. Suppose that S_i is that station, i.e., it is the one transmitting the blocking signal on bus B. (In Fig. 2.29, the station in question is S_6 .) As soon as it determines that no stations downstream from it on the forward bus are going to transmit, S_i makes bus B be the forward bus (hence, bus A is the reverse bus), and assuming that S_i is backlogged, transmits its packet on both busses. (If S_i is idle at this time, it skips the last step.) Then, S_i transmits the blocking signal on the reverse bus (now bus A), while EOC(forward) propagates to S_{i-1} , allowing the latter to take its turn.

The question is how does S_i determine that it is the last to transmit in the current round. As shown in Fig. 2.30, if S_i transmits the blocking signal for $2\tau_{ij} + 3\Delta$ and has not yet detected BOC(reverse), then S_i can be certain that S_j has seen EOC(forward) but did not transmit. If S_i transmits the blocking signal

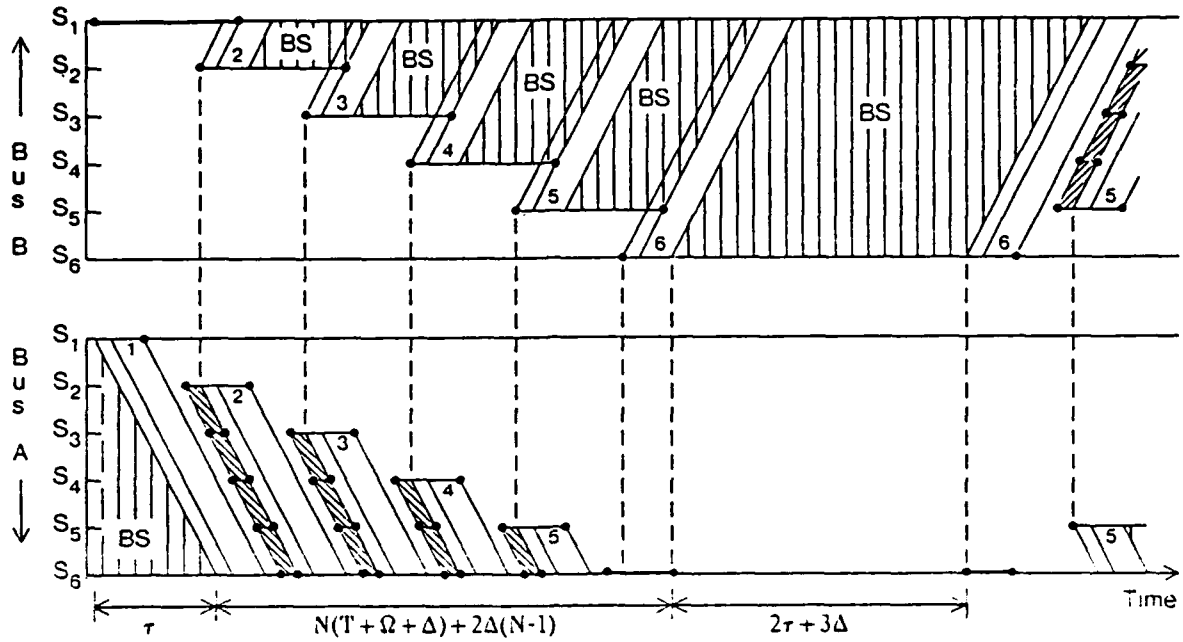


Fig. 2.29 Activity on the channel over one round on both busses in TLP. The heavy traffic condition is shown. Bus A is the forward bus for this round. Note how the first Δ s of each packet transmission on the reverse bus is destroyed due to the blocking signal.

for $2\tau_{i,M} + 3\Delta$ and does not detect BOC(reverse), then it can be certain that all stations downstream from S_i on bus A have detected EOC(forward) but none have transmitted. Hence S_i may initiate the new round on bus B*. Clearly, S_i can maintain the blocking signal for longer than this time. So, if it is undesirable that the time-out period for each station be dependent on that station's position, the worst case value of $2\tau + 3\Delta$ (or even a larger value) could be used for all stations. If the minimum time-out period is used at each station the capacity is given by

$$C(M, N) = \frac{1}{1 + \omega + 3\delta + (a + \delta)/N} \quad (2.45)$$

*By symmetry, if bus B is the forward bus, then S_i must transmit the blocking signal for $2\tau_{1,i} + 3\Delta$ before it may initiate a new round on bus A.

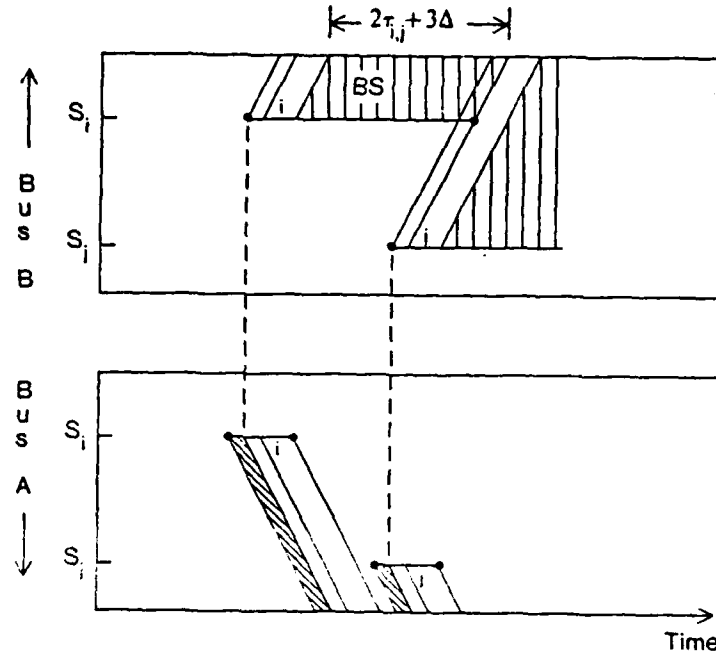


Fig. 2.30 Time-space diagram showing the length of time for which S_i transmits the blocking signal in TLP, before detecting the transmission from S_j .

However, if all stations use a time-out of $2\tau + 3\Delta$ then the capacity, assuming that S_1 and S_M are among the N backlogged stations, is

$$C(M, N) = \frac{1}{1 + \omega + 3\delta + (3a + \delta)/N} \quad (2.46)$$

The capacity in both of the above cases is almost independent of a if N is sufficiently large. Like BID, the service discipline is NGRSS and hence the delay $D(M, N)$ is different at each station and depends on the order of service. The bounds are given by the two values of delay incurred at an end station. Assuming all stations use a time-out of $2\tau + 3\Delta$, the bounds on delay are

$$D_{\min}(M, N) = 1 + \omega + 4\delta + 2a, \quad (2.47)$$

$$D_{\max}(M, N) = 2N(1 + \omega + 3\delta) + 6a + 2\delta.$$

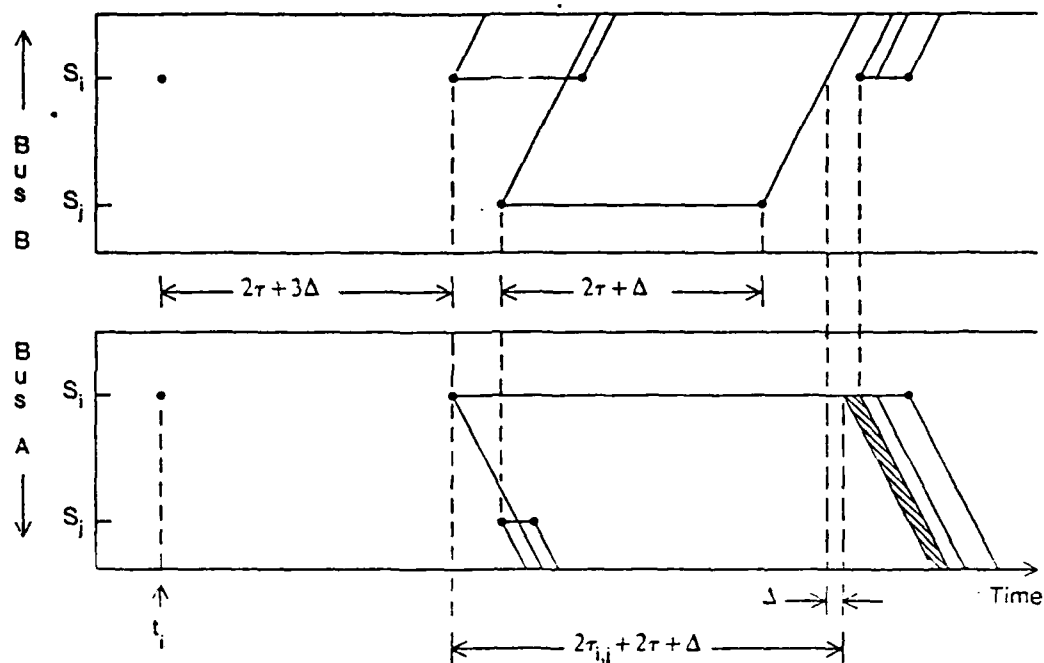


Fig. 2.31 Activity on the channel in TLP when S_i and S_j undertake the cold-start procedure.

As in Expressnet, the description of TLP in [43] includes a distributed *cold-start procedure* to initialize the network. A station will invoke this procedure if it determines that there is no activity on the network. It can easily be shown that, if there is at least one station that is transmitting, then any arbitrary station will detect some activity on the channel within a time interval of $2\tau + 3\Delta$. If any station does not detect any activity on either bus for $2\tau + 3\Delta$ s, then it must imply that there is no activity on the network, and S_i must undertake the cold-start procedure. With reference to Fig. 2.31, we now describe this procedure under the assumption that two stations simultaneously invoke a cold start.

Suppose that S_i is powered on at time t_i , and invokes the cold-start procedure at time $t_i + 2\tau + 3\Delta$. Then S_i transmits carrier on both busses for $2\tau + \Delta$ s, deferring to an upstream transmission from any other station also performing the cold start

procedure. As seen at any point on the network, at least one bus is busy after time $t_i + (2\tau + 3\Delta) + (2\tau + \Delta)$. This prevents any more stations from undertaking the cold start. The remaining steps of the procedure ensure that the leftmost participating station can begin a round and transmit conflict-free. We take the convention that bus A is the forward channel for the first round following a cold-start procedure. After transmitting on both busses for $2\tau + \Delta$, S_i stops transmitting on bus B but continues to transmit on bus A. Eventually, the left-most participating station is the only one transmitting. When it detects that both busses are idle, this station transmits its packet. Following this first transmission, all stations will be correctly synchronized as in a regular round. The overhead incurred by this procedure, measured from the time at which S_i begins to transmit (time $t_i + 2\tau + 3\Delta$ in Fig. 2.31) is at least $2\tau + \Delta$. However, it may be as much as $4\tau + \Delta$ if S_1 and S_M both undertake the cold-start procedure and S_M begins to transmit just before it detects activity from S_1 .

A large fraction of the overhead in this scheme is the 2τ overhead incurred by a station to determine that it is the end station in a given round. In [43], an algorithm is described whereby this part of the overhead can be eliminated. The underlying premise of this algorithm is that the end station changes infrequently; that is, if station S_i is the rightmost participating station in say, round r in which bus A is the forward bus, it assumes that it will be the end station in round $r + 2$. Consider now round $r + 2$. In order for station S_i to start round $r + 3$ (with bus B as the forward bus), it must operate as follows. It waits for its turn in round $r + 2$. If it has a packet to be transmitted, it transmits it and then transmits a BS on bus B for a period of 2Δ (while on bus A an idle gap of duration 2Δ appears). Note that if it did not have a packet to transmit, upon its turn, S_i immediately transmits the BS of duration 2Δ on bus B. In either case, immediately following this step, it initiates round $r + 3$ with bus B as the forward channel. This is accomplished either by the

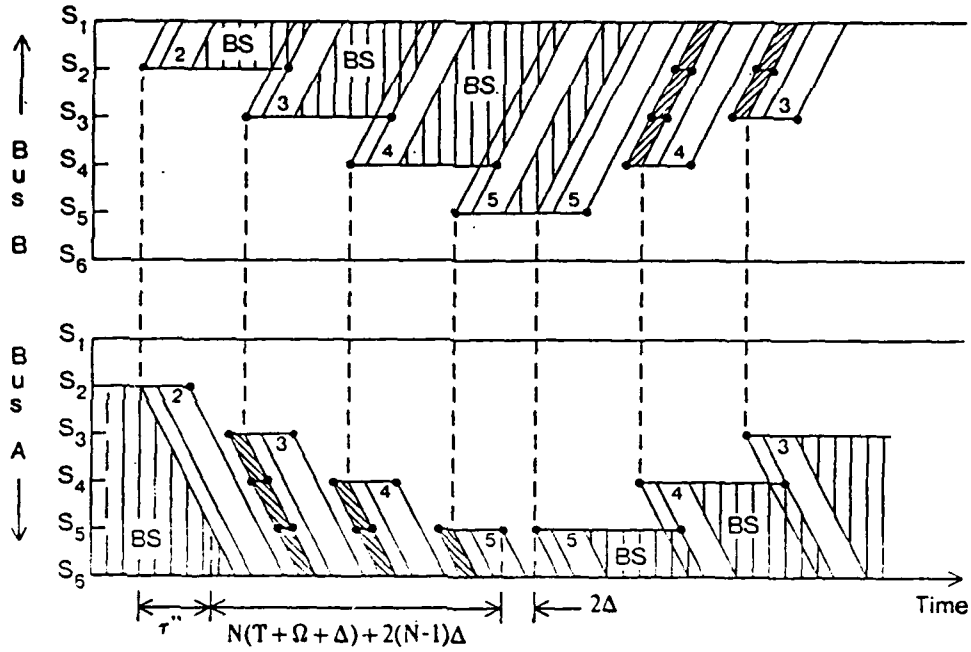


Fig. 2.32 Activity on the channel over one round for the variation of TLP that assumes that the end stations are the same from round to round. In this figure, S_2 and S_5 are currently the end stations.

transmission of a packet on both busses followed by the BS transmitted on bus A, or by the termination of the BS on bus B in case S_i has no packets. Note that the 2Δ gap on bus A at the end of round $\tau + 2$ allows a station to the right of S_i to begin transmission of a packet, should it be backlogged, thus causing a collision and forcing all stations to reinitialize by means of the cold-start procedure.

The activity on the channel when this variant of the Token-Less Protocol is being used is shown in Fig. 2.32. The capacity of this variant of the scheme is

$$C(M, N) = \frac{1}{1 + \omega + 3\delta + (a'' + \delta)/N} \quad (2.48)$$

where $a'' = \tau''/T$, and τ'' is the propagation delay between extreme participating stations. As long as the end station changes rarely, the overhead associated with

the re-initialization process is incurred infrequently. However, by comparing (2.48) and (2.46) we see that, if N is sufficiently large, the performance improvement is minimal. This fact, together with the additional complexity of the access algorithm, may make this version of the protocol unwarranted.

Comments: (i) This scheme is fully distributed and does not require a special token. However, there are no silent intervals between consecutive packets on the reverse bus. Since there is always some activity on the bus between consecutive packet, some means must be devised to delimit the beginning and end of each packet. A code violation or special bit pattern could be used.

(ii) This scheme is robust in the sense that an arbitrary station can initialize the network by means of the cold-start procedure. In addition, if a station detects any abnormal events, such as simultaneous activity on both busses, or a BOC event when it is already transmitting, then that station can undertake a cold-start. The cold-start procedure is guaranteed to bring the network to a coherent state.

(iii) Like U-Net and Fasnet, this scheme requires two busses and two transceivers at each station. As in the case of U-Net, packets are transmitted on both busses simultaneously and so the bandwidth of this scheme is the bandwidth of one bus whereas in Fasnet, where each packet is transmitted on only one bus, the bandwidth available is twice the bandwidth of one bus.

2.7.2 Hybrid Forms of the Attempt-and-Defer Access Mechanism

2.7.2.1 MAP (Marsan, Albertengo, 1982)[44]

The topology used by MAP is shown in Fig. 2.2(b). While in Expressnet a transmission is synchronized to one of the events EOC(out) or EOT(in), in MAP stations can transmit at any time, as long as there is no activity from upstream.

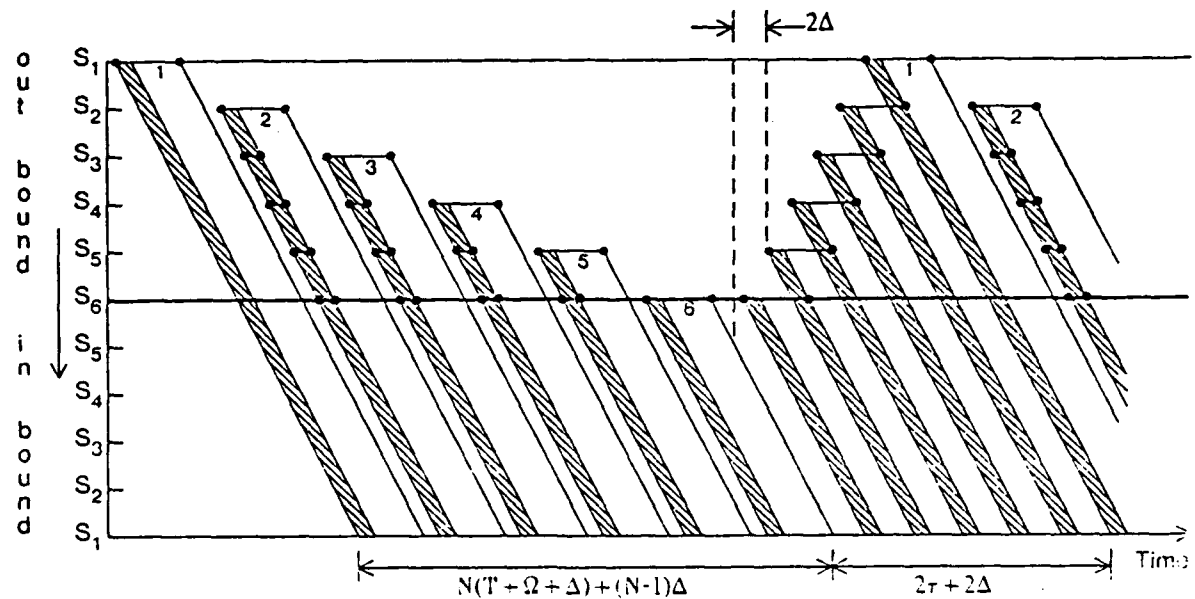


Fig. 2.33 Activity on the channel in MAP over one round. The heavy traffic condition is assumed. Successful transmissions are numbered; unsuccessful ones are not.

If a station detects BOC(out) while it is transmitting, it merely aborts its transmission, allowing the upstream one to continue conflict free. To allow downstream stations their turns, a station becomes dormant after it transmits and becomes active again when it detects the end-of-round condition (described below) on the inbound channel. Clearly, the service discipline of this scheme is HOLS.

As with other attempt-and-defer schemes, the activity on the channel consists of a sequence of transmissions separated by a gap of Δ , forming a "train". Hence EOT(in) can be detected when the inbound channel is idle for longer than Δ s. At this time, dormant stations set their states to active and begin transmitting if they are backlogged. The overhead between consecutive rounds depends on which station begins the new round and, under the heavy traffic condition, will be 2τ plus 2Δ . This can be seen in Fig. 2.33 where we show the activity on a network with six

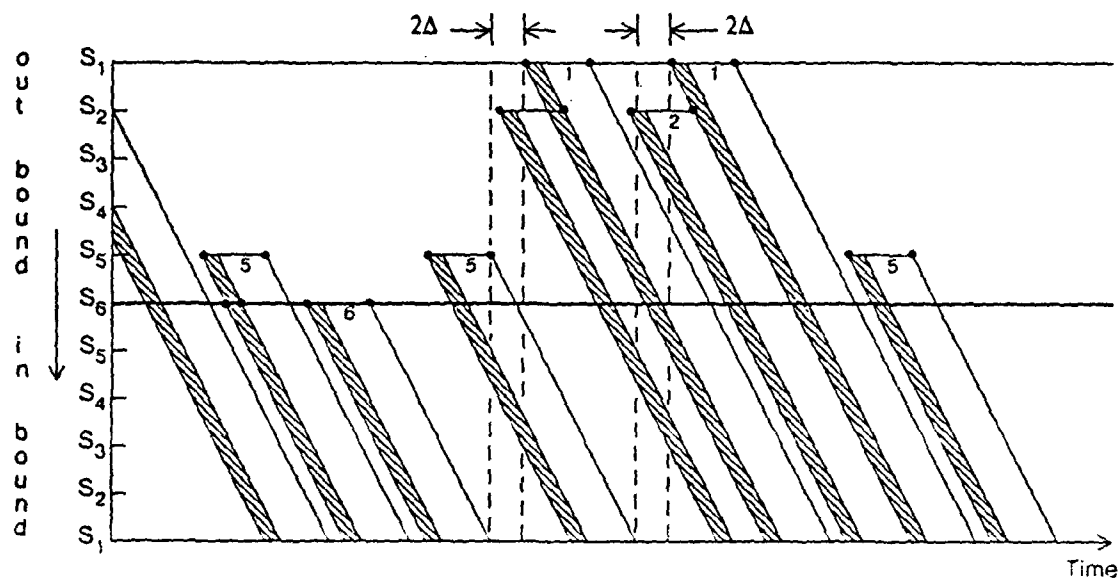


Fig. 2.34 Time-space diagram for MAP showing how S_5 transmits a packet during the inter-round overhead period.

stations operating under the heavy traffic condition. Notice how S_i , $i > 1$, detects EOT(in), begins to transmit, and then aborts its transmission when it detects activity from S_{i-1} . Assuming that S_1 is among the N participating stations, the channel capacity and packet delay for MAP are given by

$$C(M, N) = \frac{1}{1 + \omega + \delta + 2a/N} \quad (2.49)$$

$$D(M, N) = N(1 + \omega + \delta) + 2a.$$

Except for some terms in δ , the capacity and delay in (2.49) are identical to those for Expressnet. (The capacity for Expressnet is plotted versus a in Fig. 2.26.)

If the distance between S_i and S_{i-1} is sufficiently large (or if T is sufficiently small), it is possible that S_i completes its transmission before activity from S_{i-1} reaches it. In this case, S_i is able to successfully transmit a packet during the inter-round overhead period. To illustrate this condition, we show in Fig. 2.34 a

network with six stations as before, however S_3 and S_4 are idle. The figure shows that S_5 , having detected EOT(in) after the transmission of S_6 , is able to complete a packet transmission during the inter-round overhead period. S_2 and S_1 also detect this EOT event. S_1 transmits the first packet of the new round, but then detects EOC(in) due to the transmission of S_5 which is interpreted to be another EOT(in). As a result, S_1 transmits a second packet, possibly pre-empting those stations to its right. To avoid this erratic and unfair behavior, the access protocol can be modified to require each station to detect first its own transmission on the inbound channel and then EOT(in), before changing its state from dormant to active. In [44] this requirement has been imposed.

Depending on the instants at which stations become backlogged, it is possible for the situation to occur which results in an access delay which is larger than that given in (2.49). This extremely unlikely event, an example of which is shown in Fig. 2.35, occurs when S_i , each time it attempts to transmit, has to abort its transmission when this transmission has almost been completed, so as to defer to a transmission from upstream. This can repeat for at most $i - 1$ attempts, after which S_i is guaranteed to be able to transmit successfully. If $i = N$, the delay incurred by this packet is

$$D'(M, N) = (2N - 1)(1 + \omega + \delta) . \quad (2.50)$$

Since this sequence of events is so unlikely, it is reasonable to represent the maximum delay for MAP by (2.49). However, the reader should be aware that the events leading to the maximum delay given in (2.50) could occur with non-zero probability.

Comments: (i) In a network which is lightly loaded, a station, upon becoming backlogged, is likely to find the outbound channel to be idle and hence is able to transmit its packet immediately. Thus, under light loads, MAP achieves the performance advantage of zero waiting delay which is characteristic of random access

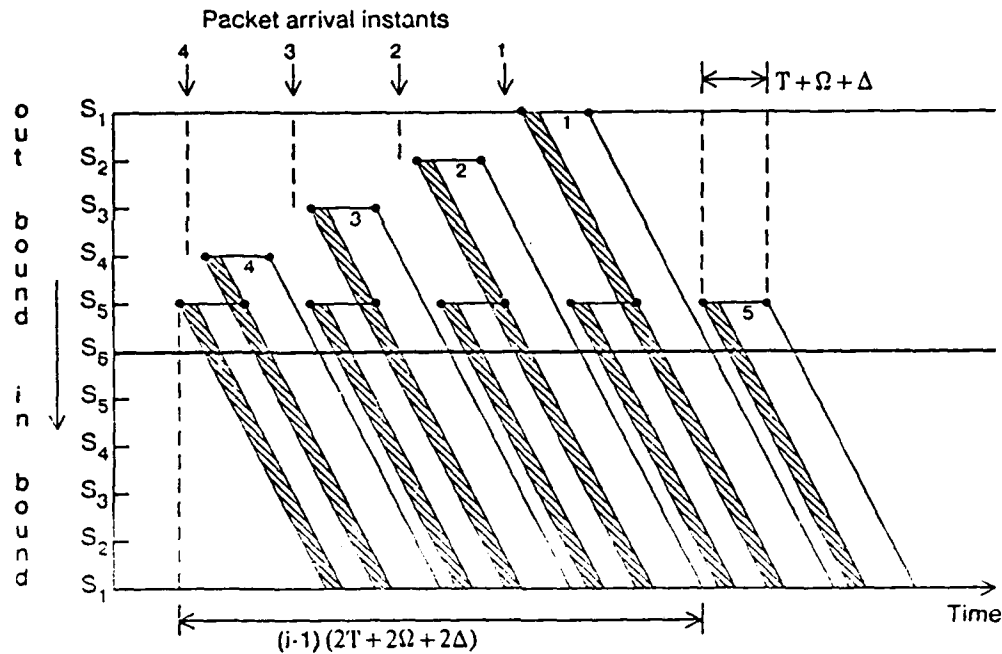


Fig. 2.35 Activity on the channel in MAP, which leads to the worst case delay for S_5 .

schemes, while achieving the throughput, bounded delay, and stability characteristic of DAMA schemes at high loads. Since each station, having transmitted a packet, must detect its own packet and then EOT(in) before it can send another packet, an overhead of at least $2\tau_{i,M} + 2\Delta$ is incurred between consecutive packets from S_i even when this station is the only one accessing the channel. Therefore, a single continuously backlogged station cannot utilize the full channel bandwidth by transmitting its packets back to back. The next two hybrid schemes overcome this limitation of MAP.

(ii) Since a given packet may be immediately preceded by the aborted transmission of another packet, some means of delimiting the beginning of a packet is required so that it can be successfully received on the inbound channel. As suggested for the Token-Less Protocol, a code violation or special pattern could be used.

(iii) Recall that each station is required to first detect its own packet on the inbound channel and then detect EOT(in) before it can move from the dormant to active state. However, suppose that a packet, say from S_i , is destroyed due to noise on the channel. S_i will be unable to receive this packet on the inbound channel and will remain indefinitely in the dormant state. To avoid this deadlock, we propose a slight modification to the access protocol which uses the fact that the beginning of its transmission reaches S_i at most 2τ after the instant at which it began to transmit: Having transmitted, S_i waits either until it detects its transmission on the inbound channel, or until it times out a period of 2τ . Thereafter, S_i waits for EOT(in) before moving to the active state. This modification requires that each station have knowledge of τ which otherwise would be unnecessary.

2.7.2.2 CSMA-DCR (Takagi, Yamada, Sugawara, 1983)[45]

The UBS configuration used by this scheme is shown in Fig. 2.2(a). As in U-Net and the Token-Less Protocol scheme, packets are transmitted on both channels simultaneously. In general, stations contend for the channel using CSMA as the basic access mechanism. That is, a backlogged station, say S_i , waits until both busses are idle and then transmits its packet. If a collision occurs, S_i aborts its transmission. As can be seen from the time-space diagram shown in Fig. 2.36, an aborted transmission is at most $2\tau + \Delta s$. By requiring that packets be longer than this time, a means is provided whereby all stations (idle or backlogged) are able to identify a collision. Indeed, a transmitting station identifies a collision when it detects activity from upstream on one or both of the busses; a non-transmitting station identifies a collision if it detects a transmission of duration less than or equal to $2\tau + \Delta$. When a collision occurs, a DAMA scheduling procedure, which is described below, is invoked to resolve the contention.

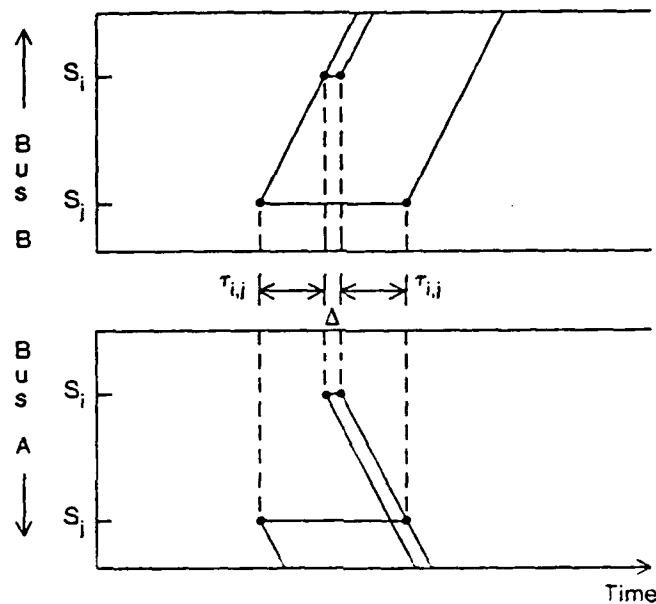


Fig. 2.36 Time-space diagram showing simultaneous transmissions by two stations and resulting packet collisions in CSMA-DCR.

Due to the network configuration being used, the leftmost (by convention) of all stations whose packets collided can easily be identified. If S_i detects no activity from the left within $2\tau + \Delta s$, measured from the time that it *began* to transmit, then it is the leftmost of those stations whose packets collided, and it retransmits its packet. Any other station, say S_j , waits for the EOC on bus A (on which S_i is the leftmost). Using this event, it attempts to transmit with bus A being the forward bus. Following such activity, when a station senses the channel idle for $2\tau + \Delta s$, then it knows that all stations have had their turns. If backlogged, it can immediately attempt to transmit using CSMA. In Fig. 2.37, we show the activity on the channel under the heavy traffic condition. Note how each station begins to transmit $2\tau + 2\Delta s$ after it detects the EOC of the last transmission in the round. Since more than one station is backlogged, packet collisions take place. S_1 begins to transmit conflict free $2\tau + \Delta s$ after the beginning of its unsuccessful transmission.

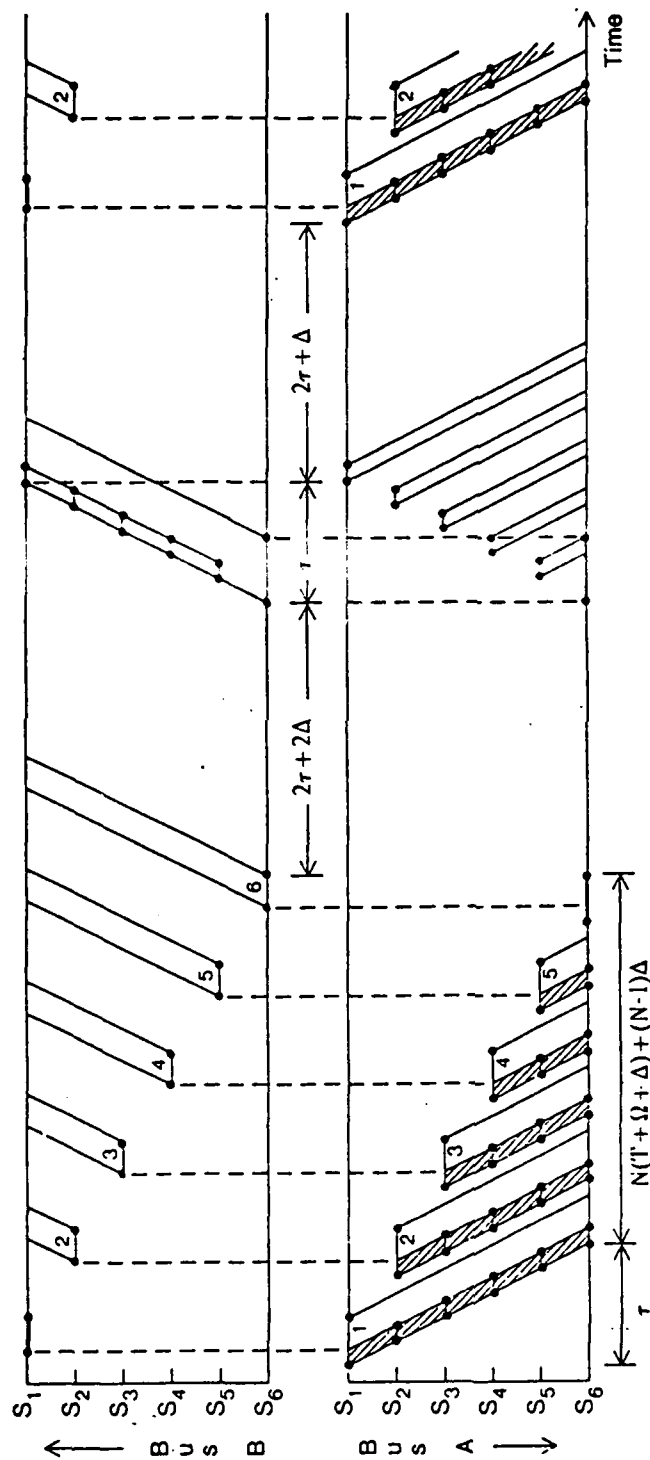


Fig. 2.37 Activity on the channel for CSMA-DCR under the heavy traffic condition.

Clearly, the service discipline of this scheme is NGSS.

As stated above, in order that all stations can recognize a collision, CSMA-DCR requires that $T + \Omega > 2\tau + \Delta$, i.e., $a < 0.5 + \omega/2 - \delta/2$. As a result, this scheme suffers the same performance limitations as CSMA with collision detection (CSMA/CD). In particular, for the case where $T + \Omega \leq 2\tau + \Delta$ (such as when the channel bandwidth is high or the number of bits per packet is small), the packet size must be artificially increased with dummy bits. Therefore, the channel capacity and packet delay are given by

$$C(M, N) = \begin{cases} \frac{1}{1 + \omega + 2\delta + (6a + 2\delta)/N} & a \leq 0.5 + \omega/2 - \delta/2 \\ \frac{1}{2a + 3\delta + (6a + 2\delta)/N} & a \geq 0.5 + \omega/2 - \delta/2 \end{cases} \quad (2.51)$$

and

$$D(M, N) = \begin{cases} N(1 + \omega + 2\delta) + 6a + 2\delta & a \leq 0.5 + \omega/2 - \delta/2 \\ N(2a + 3\delta) + 6a + 2\delta & a \geq 0.5 + \omega/2 - \delta/2 \end{cases} \quad (2.52)$$

The capacity versus a with $\omega = \delta = 0$ is plotted in Fig. 2.38 for various values of N .

Comments: (i) This access protocol provides low delay typical of CSMA at low loads. In addition, a single station may transmit any number of packets consecutively, as long as no collisions occur, thereby achieving a throughput close to the channel bandwidth. At the same time, CSMA-DCR exhibits the advantages of DAMA schemes at high loads; the network remains stable and the packet delay is bounded.

(ii) As a result of the particular UBS configuration used in the scheme, simultaneous transmissions by two stations results in both aborting their respective transmissions. In MAP, which uses the alternative UBS configuration, only the rightmost of the

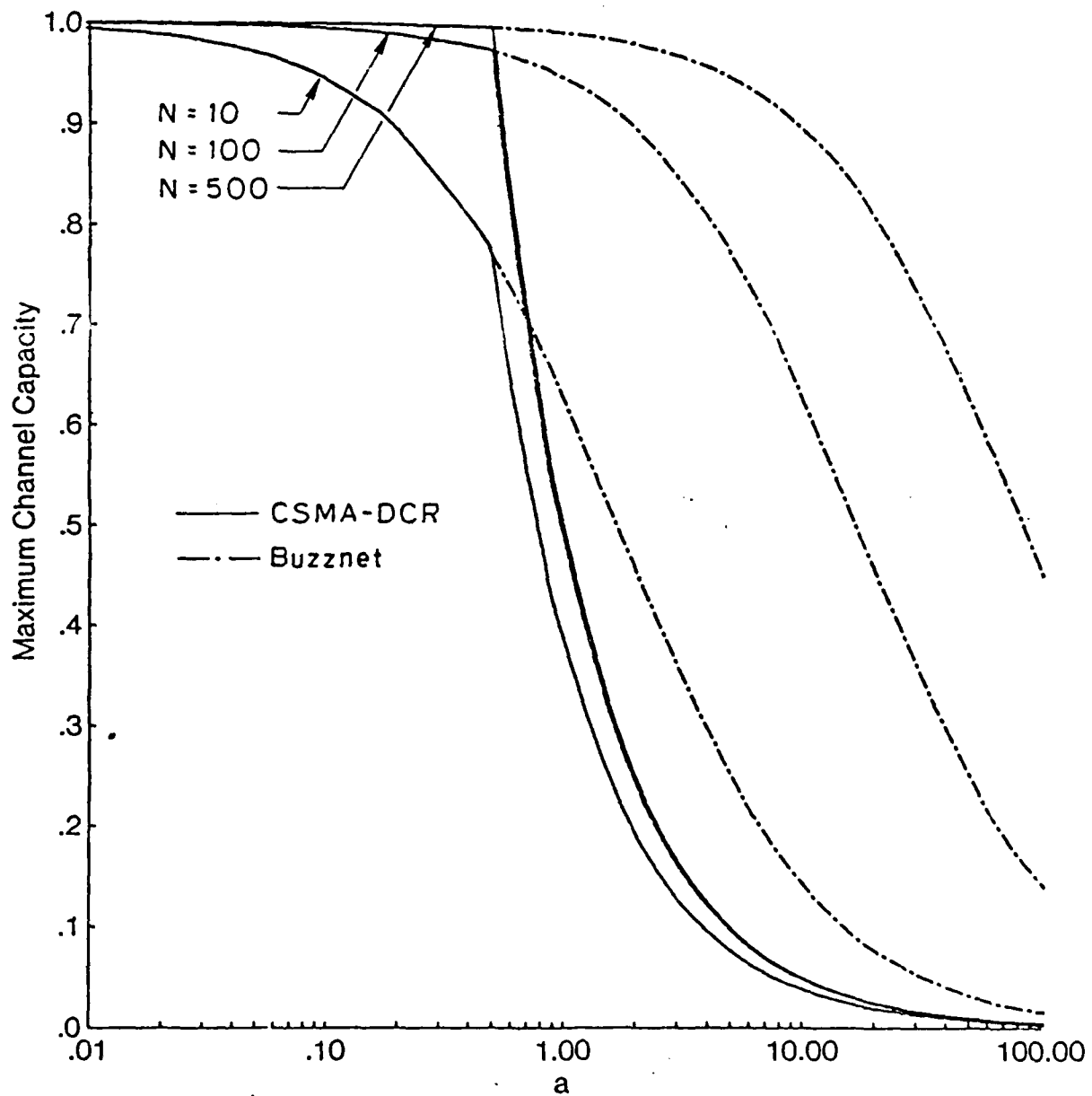


Fig. 2.38 Network capacity versus a for CSMA-DCR and Buzznet.

two stations aborts; the leftmost is able to complete its transmission successfully.

(iii) Once a collision occurs and DCR scheduling is invoked, an idle station, if it becomes backlogged, must refrain from accessing the channel using random access mode as this will disrupt the DAMA scheduling. Thus, all stations, even if idle, must continuously monitor the channel so that they are always aware of its current state. Furthermore, a station that becomes backlogged while DAMA scheduling is in effect must wait for the channel to revert to random access mode before attempting to transmit.

2.7.2.3 Buzznet (Gerla, Rodrigues, Yeh, 1983) [46]

The drawback of CSMA-DCR is the requirement that $T + \Omega > 2\tau + \Delta$. This requirement is necessary so that all stations can identify a collision and invoke the DAMA round robin scheduling protocol. In Buzznet, an out-of-band *buzz signal* is used to indicate the occurrence of a collision and to synchronize stations so as to invoke the round robin scheduling function. Consequently, Buzznet provides identical service to CSMA-DCR, without the restriction on the minimum size of a packet. We now describe this access protocol.

Initially, stations attempt to transmit using CSMA as the basic access mechanism. While this is satisfactory at light loads, collisions will occur for medium to high loads. Suppose that some packet interferes with the transmission by S_i . S_i informs all other stations of this collision by transmitting the buzz signal on both channels. All stations detect this signal and together they initialize the network so as to invoke the conflict-free round robin access mechanism as follows. Suppose that S_i begins transmitting the buzz signal at time t_i . When S_j detects this signal at time $t_i + \tau_{i,j} + \Delta$, it too transmits the buzz signal. This "buzz" from S_j reaches S_i at time $t_i + 2\tau_{i,j} + \Delta$. Therefore, to ensure that both channels are being buzzed at every point on the network, and hence there are no EOC events on the channel, S_i

must buzz both channels for a period of $2\tau + \Delta$. After this period, S_i transmits the buzz signal only on bus A, deferring to transmissions from upstream. Eventually, the leftmost station will be the only one transmitting. It transmits as soon as it senses both bus A and bus B to be idle. Thereafter, transmissions follow the rule for U-Net where bus A is the forward bus.

As in CSMA-DCR, stations detect the end of a round when the channel is idle for a period of at least $2\tau + \Delta$. At this time stations revert to using CSMA as the basic access mechanism. A time-space diagram showing the activity on the channel in this scheme is given in Fig. 2.39. The capacity and delay for Buzznet are given by

$$C(M, N) = \frac{1}{1 + \omega + 2\delta + (6a + 2\delta)/N} \quad (2.53)$$

$$D(M, N) = N(1 + \omega + 2\delta) + 6a + 2\delta$$

Note that CSMA-DCR gives the same performance as Buzznet for $a < 0.5 + \omega/2 - \delta/2$ but for $a > 0.5 + \omega/2 - \delta/2$ Buzznet is superior. The capacity of Buzznet with $\omega = \delta = 0$ is plotted as a function of a in Fig. 2.38, and compared to that of CSMA-DCR.

Comments: (i) The implementation of this scheme must ensure that the buzz signal is a uniquely identifiable bit pattern or an out-of-band signal. Depending on the length of this pattern, some time is required to detect the buzz signal which effectively increases the overhead incurred when the network switches from random access mode to DAMA mode. If the buzz signal is out-of-band (either in frequency or time), some of the channel capacity is lost to allow for this signal.

(ii) Like CSMA-DCR, all stations are required to continuously monitor the channel (even when idle) so that, upon becoming backlogged, they can use the correct scheduling mode: random access or DAMA.

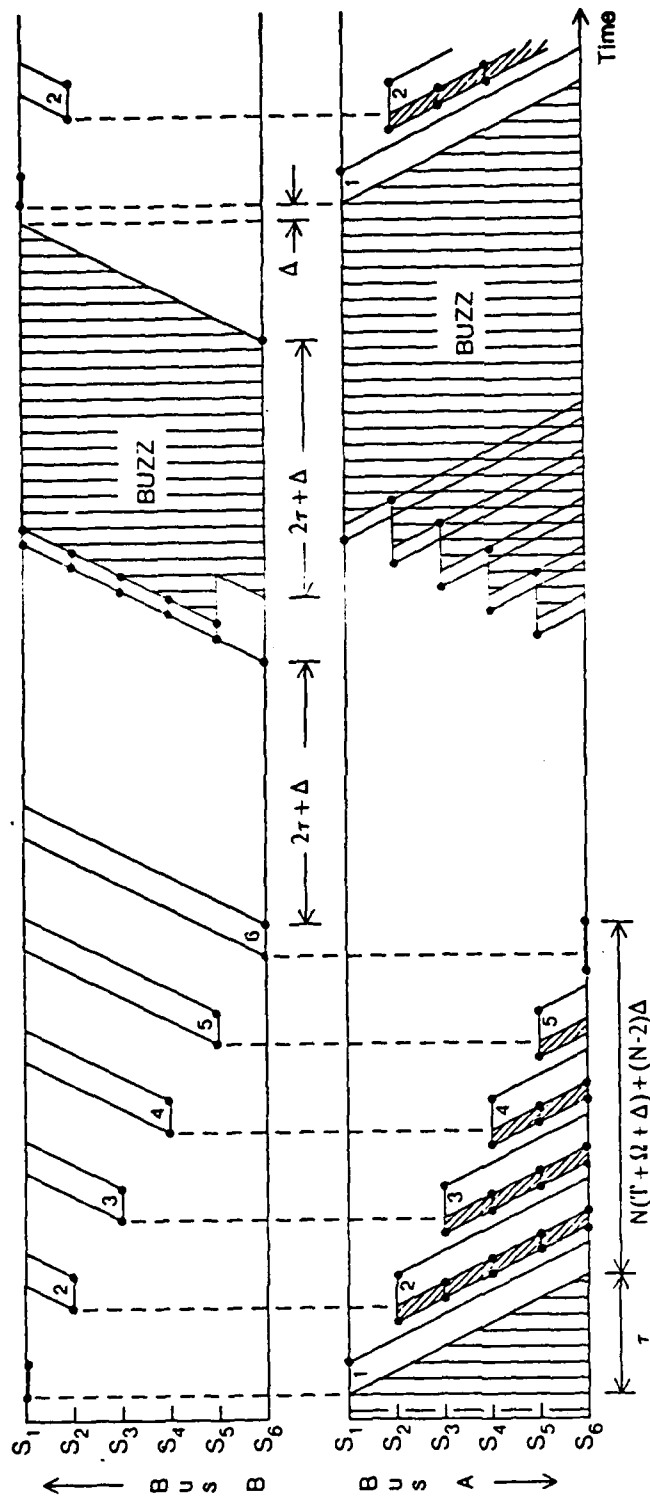


Fig. 2.39 Activity on the channel in Buzznet under the heavy traffic condition.

2.8 Summary

From the numerous implicit-token DAMA schemes that have recently been proposed, we have identified three basic access mechanisms according to which it has been possible to classify them. These are the scheduling delay access mechanism, the reservation access mechanism, and the attempt-and-defer access mechanism. In this chapter, we described these access mechanisms together with the various network configurations proposed for their implementation. Then, for each of the basic access mechanisms, we discussed those schemes belonging to that class.

With this classification, we were able to present the numerous network schemes in a unified manner and identify characteristics by which they may be similar or may differ. We have also suggested that characteristics of some schemes can be seen as variations or special cases of another, and that a feature proposed for one could, in some cases, be applied to another. These characteristics may be divided into three categories according to whether they are (i) physical attributes, (ii) logical attributes, or (iii) performance factors.

Typical physical attributes of concern to the network designer would be the number and type of channels required (unidirectional, bidirectional, control-wires), synchronous or asynchronous operation, and the type of tap required. Some schemes, notably those of the scheduling delay class, require a single tap on the bidirectional bus. Others require, in addition, a control wire with an associated tap which is an active device. The unidirectional schemes require separate receive and transmit taps and, in some cases (e.g., Fasnet, U-Net, TLP), two of each. Synchronous schemes (e.g., Fasnet) require a common clock to which all transceivers are synchronized. Asynchronous schemes, on the other hand, require a preamble preceding each transmission for receiver synchronization. All of these factors should be considered in choosing a network design.

Logical attributes of the schemes are those characteristics that describe details of the access protocol in terms of information that must be available to, and functions that must be performed by the stations. These characteristics include the correspondence (if any) between the stations' logical and physical orders, which impacts the flexibility of the scheme to topological changes; fully distributed or partially centralized access protocols; and pieces of information that must be available to the network interface so as to correctly compute timing intervals and correctly execute the access protocol. This information consists of both fixed, network dependent parameters such as τ , Δ and M , as well as information that must be extracted from activity on the channel (such as the index number of the currently transmitting station in schemes using the scheduling delay access mechanism, or the contents of the AC field in Fasnets). As compared to a fully distributed protocol, a partially centralized one may simplify the design of a station's network interface but it raises concerns about reliability. Additional logical attributes are cold start procedures, procedures to recover from abnormal events, and procedures to keep all stations in a consistent state (such as when the access method switches from random access mode to DAMA mode in CSMA-DCR and Buzznet).

Performance factors have been expressed in terms of network capacity and maximum delay. Both of these parameters depend on the overhead incurred by the access protocol between two consecutive transmissions and the overhead between two consecutive rounds. In some schemes, notably BRAM, DSMA, and UBS-RR, the inter-packet overhead increases linearly with increasing α . As a result, the maximum delay increases linearly and the capacity decreases sharply with increasing values of α and, in this respect, these schemes are unsuitable for high bandwidth applications. For the other schemes, the inter-packet overhead depends not on α but only on parameters such as δ and ω .^{*} The inter-round overhead is less significant

^{*}The preamble is not strictly independent of the bandwidth W . Depending on the implementation,

than the interpacket overhead since, for a reasonably large population size (which is to be expected in high bandwidth environments), rounds consist of many transmissions and the inter-round overhead is incurred infrequently. Nevertheless, for those schemes which use a reversing sequential service discipline, such as BID and U-Net, the implicit token propagates only once across the channel for every round. The inter-round overhead in this case is on the order of a . For a non-reversing service discipline, the implicit token propagates back and forth across the network within each round. In such schemes, as exemplified by Expressnet, the inter-round overhead is $2a$. For schemes that require some activity (or lack of it) to determine the end of a round, (e.g., the blocking signal in TLP, the buzz signal in Buzznet, and the 2τ idle period in L-Expressnet and the Control-Wire), the inter-round overhead is somewhat larger ($3a \sim 6a$).

a longer preamble may be required at high bandwidths than that required at low bandwidths.

Chapter 3

Throughput-Delay Performance Of Expressnet and Fasnet

In this chapter, we model Expressnet and Fasnet operating under their various service disciplines and, from the analysis, obtain their throughput-delay characteristics. From the descriptions of these schemes given in chapter 2, it is clear that Expressnet achieves the NGSS discipline; Fasnet achieves the HOLS discipline if an active/dormant flag is used to maintain fairness or else the GSS discipline if the method of gating is used to maintain fairness.

We note that Expressnet can be made to operate in GSS mode merely by having each station, upon generating a packet, wait for the event EOT(in) before attempting to transmit that packet. Similarly, one could operate Fasnet in NGSS mode by allowing each station, upon generating a packet in a given round, to transmit that packet in the next empty slot as long as this station has not yet seen an empty slot go by in the current round and has not yet transmitted in the current round. Otherwise it waits for the SOC. The SOC in Fasnet and the EOT(in) in Expressnet are analogous events. In HOLS on the other hand, the transmission of each packet is synchronized to an event which sweeps the entire population of

stations from the most upstream to the most downstream. Thus only Fasnet can support this discipline.

In this analysis we consider only fixed length packets. In Expressnet however the access protocol allows for packets of any length. In the most recent description of Fasnet [12], slots are required to be of a fixed length in order to simplify the hardware implementation. Nevertheless, in gated Fasnet, variable length packets can be accommodated simply by allowing a station to access a number of consecutive slots. This is feasible because downstream stations may only transmit after the current station, and will have full knowledge of the slot usage. In non-gated Fasnet only fixed length packets equal to the size of a slot can be achieved since the order of transmissions does not correspond to the physical order; therefore the station does not know how many consecutive empty slots (if any) follow the one in which it begins to transmit.

We begin this chapter by describing the model used in the analysis. Then we present the analysis for each of the three service disciplines. The performance measures derived from these analyses are the channel throughput, the expected delay incurred by a packet and the variance of this delay. Quantitative numerical results are presented after the analysis.

3.1 The Model

Consider a system of M stations. Each station has a single packet buffer and is either idle or backlogged. An idle station will generate a packet in a random time which is exponentially distributed with mean $1/\lambda$. A backlogged station does not generate any packets and becomes idle upon successful transmission of its buffer. This model corresponds to the case of interactive stations, widely used in the past to

analyze slotted ALOHA, CSMA and other access schemes. The end-to-end propagation delay of the signal travelling across the network is denoted by τ . This corresponds to the propagation delay between the extreme stations on one of the channels (e.g., the outbound channel on Expressnet or channel A on Fasnet). The time required to transmit a packet is $T = B/W$ where B is the number of bits in a packet (assumed fixed) excluding the preamble if any in Expressnet and the AC field in Fasnet, and where W is the bandwidth of the channel. The overhead before each transmission to determine which station gets access to the channel is denoted by t_o . In Fasnet t_o is given by the length of the AC field. In Expressnet t_o is given by twice the carrier detection time, i.e., 2Δ . In Fasnet, since the system is synchronous, the time required to transmit the preamble is $\Omega = 0$. In Expressnet, Ω is non-zero if the system is operated asynchronously. Thus to transmit a packet of length T requires a transmission period of $X = T + t_o + \Omega$. The time that the channel becomes idle between rounds is the inter-round overhead and is denoted by Y . In Expressnet $Y = 2\tau$.^{*} In Fasnet Y must be an integral number of slots and is taken to be $Y = \lceil 2\tau/X \rceil X + X$.

In the next sections we present the analysis of this model for each of the service disciplines.

3.2 Analysis of the Non-Gated Sequential Service Discipline

An analysis of a loop system where stations are serviced in a prescribed sequence and which fits the NGSS discipline of Expressnet is given by Kaye [47] based on the work in [48]. A summary of this analysis is presented in this section.

^{*}If the topology of Expressnet is such that the two extreme stations are collocated then the inter-round gap Y is equal to τ . See [11]

The probability that there are n packet transmissions in a train, denoted by p_n , is given by [48]

$$p_n = p_0 \binom{M}{n} \prod_{j=0}^{n-1} [e^{\lambda(jX+Y)} - 1] \quad 0 < n \leq M \quad (3.1)$$

where p_0 is determined by $\sum_{n=0}^M p_n = 1$. The probabilities p_n satisfy the following recursive formula

$$\frac{p_n}{p_{n-1}} = \frac{M - n + 1}{n} [e^{\lambda[(n-1)X+Y]} - 1] \quad (3.2)$$

Based on this distribution, Kaye derived the distribution of waiting time $\tilde{w}$, defined as the period between the moment when a packet is generated by a station and the moment when its transmission commences [47]. The expected waiting time and second moment of $\tilde{w}$ are then derived to be given by

$$E[\tilde{w}] = \frac{1}{\bar{n}} \sum_{n=1}^M np_n \frac{[(n-1)X+Y]e^{\lambda[(n-1)X+Y]} - 1}{e^{\lambda[(n-1)X+Y]} - 1} - \frac{1}{\lambda} \quad (3.3)$$

$$E[\tilde{w}^2] = \frac{1}{\bar{n}} \sum_{n=1}^M np_n \frac{[(n-1)X+Y]e^{\lambda[(n-1)X+Y]} [(n-1)X+Y - 2/\lambda]}{e^{\lambda[(n-1)X+Y]} - 1} + \frac{2}{\lambda^2} \quad (3.4)$$

where

$$\bar{n} = \sum_{n=0}^M np_n. \quad (3.5)$$

The mean and variance of packet delay are obtained by adding X to $E[\tilde{w}]$ and X^2 to $E[\tilde{w}^2] - (E[\tilde{w}])^2$ respectively.

From the distribution $\{p_n\}$, one can also easily derive the average network throughput S for a given value of λ . It is simply given by

$$S = \frac{\bar{n}T}{\bar{n}X + Y} \quad (3.6)$$

Note that as $\lambda \rightarrow \infty$, $\bar{n} \rightarrow M$ and the throughput reaches a maximum given by $MT/(MX + Y)$.

3.3 Analysis of the Gated Sequential Service Discipline

We present in this section some additional definitions and both a mean value and distribution of delay analysis for the GSS discipline.

Let $n(t)$ denote the number of backlogged stations at time t and let $t_e^{(r)}$ denote the start of the r^{th} round. Since only those stations that are backlogged at $t_e^{(r)}$ can transmit in round r , the number of transmissions in the r^{th} round is given by $n(t_e^{(r)})$. The number of backlogged stations at the start of round $r+1$ depends on the length of round r and the arrival of packets during this round. Hence, the number of backlogged stations at $t_e^{(r+1)}$, denoted by $n(t_e^{(r+1)})$, depends only on $n(t_e^{(r)})$ and the events that occur during the r^{th} round. Thus $\{n(t_e^{(r)}), r \in (-\infty, \infty)\}$ constitutes an embedded Markov process. So we can use the properties of Markov processes to derive the analytic solution for the performance of the system.

3.3.1 Mean Value Analysis

For the mean value analysis the state of the system at an embedded point is described sufficiently by the number of stations that are backlogged at this instant. Consider two consecutive embedded points $t_e^{(r)}$ and $t_e^{(r+1)}$. The time interval $[t_e^{(r)}, t_e^{(r+1)}]$ is called a cycle. Each cycle is considered to be divided into two sub-cycles. The first is that part of the cycle where packets are being transmitted (i.e. the round). The second is that period which is the inter-round overhead. Let $\mathbf{P}$ be

the transition matrix for the embedded Markov process $n(t_e^{(r)})$. The elements of P are denoted by p_{ik} and are defined as

$$p_{ik} \triangleq \Pr\{n(t_e^{(r+1)}) = k \mid n(t_e^{(r)}) = i\} \quad (3.7)$$

where $n(t_e^{(r+1)})$, the state of the system at $t_e^{(r+1)}$, is merely the number of stations that generate new packets during round r . Since those stations that transmit during the round can only generate a new packet after they have transmitted the one backlogged from the previous round, the probability of generating a new packet is not the same for all stations. Therefore, in computing the transition probabilities p_{ik} , we must account for all possible ways that k out of M stations can become ready. To do this we use a recursive approach by considering the instants of time that correspond to the end of a transmission period.

Define the function $G(n, m, s)$ as the probability that, in a round of length n , m stations have generated new packets by the end of the s^{th} transmission period of that round. We compute $G(n, m, s)$ in terms of $G(n, m', s-1)$ and this we do by computing the probability that $m - m'$ new packets are generated in the s^{th} transmission period out of a possible $M - (n - s + 1) - m'$. (Since $n - s + 1$ stations are still waiting to transmit in this round, they cannot generate new packets.) Summing over all values of m' gives $G(n, m, s)$ as follows.

$$G(n, m, s) = \sum_{m'=0}^m \binom{M - (n - s + 1) - m'}{m - m'} p(X)^{m-m'} [1 - p(X)]^{M - (n - s + 1) - m} \cdot G(n, m', s-1) \quad s \neq 0 \quad (3.8)$$

where $p(t)$ is the probability of a single station generating a packet during an interval t . Since inter-arrival times are exponentially distributed with rate λ , this is given by

$$p(t) = 1 - e^{-\lambda t}. \quad (3.9)$$

At the beginning of the round ($s = 0$) there must be with probability 1 no new packets generated and so

$$G(n, m, 0) = \begin{cases} 1 & m = 0 \\ 0 & m \neq 0 \end{cases} \quad (3.10)$$

Starting with these initial conditions, the recursion in (3.8) ends at $G(n, m, n)$, the probability that m stations have generated new packets by the end of the first sub-cycle.

Using (3.8) and considering additional arrivals during the inter-round overhead period allows us to compute the elements of the transition matrix P .

$$p_{ik} = \sum_{j=0}^k G(i, j, i) \binom{M-j}{k-j} p(Y)^{k-j} [1 - p(Y)]^{M-k} \quad (3.11)$$

Given P we can calculate the stationary distribution of the backlog at the embedded points and the average throughput and the average delay using results from the theory of regenerative processes. The stationary distribution is denoted by $\Pi = (\pi_0 \dots \pi_M)$.

Average Throughput: Since $n(t_e^{(r)})$ is a regenerative process the average channel throughput can be computed as the ratio of the expected time that the channel is busy in a cycle to the expected length of a cycle [50], [49]. Hence the expected throughput, denoted by S , is simply

$$S = \frac{\sum_{i=1}^M \pi_i T_i}{\sum_{i=0}^M \pi_i (iX + Y)} \quad (3.12)$$

Average Packet Delay: Consider each station to be a single buffer queueing system with loss and exponential inter-arrival times. The expected delay of a packet

in such a system can be computed as the difference between the expected inter-departure time and the expected inter-arrival time. Let s_i denote the expected throughput of packets from station i . The expected inter-departure time from station i is simply $1/s_i$. Hence the expected delay of a packet from station i is given by

$$d_i = \frac{1}{s_i} - \frac{1}{\lambda} \quad (3.13)$$

Averaging over all the stations gives the expected delay of a packet D as

$$\begin{aligned} D &= \sum_{i=1}^M \frac{s_i}{S} d_i \\ &= \frac{M}{S} - \frac{1}{\lambda} \end{aligned} \quad (3.14)$$

where we have used the fact that $S = \sum_{i=1}^M s_i$.

Using Little's result, the average packet delay can also be computed as the ratio of the average backlog of packets to the average channel throughput. This approach, although significantly more involved than the one above, is nevertheless presented below. We first show how to compute the average backlog B .

Let $b_e(i)$ be the expected sum of the backlog over a cycle given that $n(t_e^{(r)}) = i$. This can be expressed qualitatively as

$$b_e(i) = \sum_{j=1}^M E[\text{time in a cycle that station } j \text{ has a packet in its buffer} \mid n(t_e^{(r)}) = i]$$

Then B is computed by removing the condition on $n(t_e^{(r)})$ and dividing by the expected length of a cycle.

In order to compute the contribution to the backlog of a new arrival in an interval, say of length t seconds, we need to compute the expected time of that

arrival. Denote the time of this arrival by t_a . Then, assuming that the arrival remains in the backlog for the remainder of the interval following its arrival time, the contribution to the sum of the backlog of that arrival is $E[t - t_a]$. We define the function $u(t)$ as the value of $E[t - t_a]$ corresponding to a given generation rate λ . That is

$$u(t) = t - E[t_a | N(t) \neq 0] \quad (3.15)$$

where $N(t)$ is the number of arrivals in $[0, t]$ from a Poisson source with rate λ and t_a is the time of the first arrival. From the properties of the Poisson process it is straightforward to show that

$$E[t_a | N(t) \neq 0] = \frac{1}{\lambda} - \frac{te^{-\lambda t}}{1 - e^{-\lambda t}} \quad (3.16)$$

and so $u(t)$ becomes

$$u(t) = \frac{t}{1 - e^{-\lambda t}} - \frac{1}{\lambda} \quad (3.17)$$

The packets that contribute to the sum of the backlog can be separated into three groups. The first group are those that are transmitted in the current round. The packet that is transmitted in the j^{th} transmission period of the round contributes an amount jX to the sum of the backlog. Given that there are i backlogged packets at the beginning of the cycle the total contribution to the sum of the backlog of packets of type one is simply $\sum_{j=1}^i jX$.

Packets of the second group are those that are generated by a station that has completed a transmission in the current round. Consider the station that has transmitted in transmission period j . This station will generate a new packet in the interval $[t_e^{(r)} + jX, t_e^{(r+1)}]$ with probability $p(iX + Y - jX)$. Hence the contribution to the expected sum of the backlog of this packet is $p(iX + Y - jX)u(iX + Y - jX)$.

Packets of the third group are those that arrive to a station that was idle at the beginning of the round. Such a station can generate a packet at any time in the cycle. The expected amount that such a packet will contribute to the sum of the backlog in a round with $n(t_e^{(r)}) = i$ is $p(iX + Y)u(iX + Y)$.

Summing all the contributions from these three groups gives the expected sum of the backlog over a cycle with $n(t_e^{(r)}) = i$.

$$b_s(i) = (i+1)i/2 + \left[\sum_{k=0}^{i-1} p(kX + Y)u(kX + Y) \right] + (M-i)p(iX + Y)u(iX + Y) \quad (3.18)$$

From (3.18) one can compute the expected backlog B . Dividing this by the expression for the expected throughput given in (3.12) yields the expected packet delay D .

$$D = \frac{\sum_{i=0}^M \pi_i b_s(i)}{\sum_{i=0}^M \pi_i iT} \quad (3.19)$$

3.3.2 Distribution of Delay Analysis

We now derive the distribution of packet delay in order to compute the higher order moments of delay. In the distribution of delay analysis we select a single station and consider packets only from this station. We refer to this station as the tagged station. This approach will not only yield an expression for the distribution of delay but, by tagging different stations on the network, will enable us to compare the performance achieved by the different stations. From this we can see how a station's physical location on the network can affect the quality of the service it gets from the network.

Let N , $1 \leq N \leq M$ denote the tagged station. As in the mean value analysis we consider the beginning of a cycle to constitute an embedded point defining an embedded Markov chain. However, in order to completely describe the state of the system at the embedded point, the state descriptor must contain information about the state of the tagged station, the number of active stations upstream from the tagged station and the number of active stations downstream from the tagged station. Thus we describe the state of the system at the beginning of the current round by a vector with three elements $(\delta(t_e^{(r)}), n_u(t_e^{(r)}), n_d(t_e^{(r)}))$ where $n_u(t)$ and $n_d(t)$ are the number of active stations upstream and downstream from the tagged station at time t respectively and $\delta(t)$ indicates the state of the tagged station at time t . $\delta(t)$ can take on the values 0 and 1 denoting the tagged station to be idle or busy respectively. $n_u(t)$ and $n_d(t)$ are in the range $[0, N - 1]$ and $[0, M - N]$ respectively. To simplify the notation for the state descriptor we define $S(t) \triangleq (\delta(t), n_u(t), n_d(t))$.

We first compute the transition matrix P for the embedded Markov process $S(t_e^{(r)})$. We partition the stations into three groups. The first consists of those stations on the upstream side of the tagged station; the second consists of those stations on the downstream side of the tagged station; the third consists of the tagged station. We compute the state transition probabilities by considering new arrivals to the system from each group separately. We now present a generalized form of the recursive function that was used in the mean value analysis. We use this generalized version to compute the state transition probabilities for the upstream and downstream groups of stations.

Consider a sequence of x consecutive transmissions by stations from a single group. Define the function $G_Z(x, m, s | y)$ to be the probability that, in a transmission sequence of length x , m stations have generated new packets by the end of the

s^{th} transmission period in this sequence given that y stations had already generated new packets at the beginning of the sequence. The subscript Z denotes the size of the population of stations of this group. We can write this function as

$$G_Z(x, m, s | y) \triangleq \Pr\{n_g(t_b + sX) = x - s + m \mid n_g(t_b) = x + y\}$$

where $n_g(t)$ denotes the number of stations from group g that are in the backlogged state and t_b is the time corresponding to the beginning of the first transmission in the sequence.

For $s > 0$ we can compute G_Z recursively by considering the number of new arrivals during the s^{th} transmission period.

$$G_Z(x, m, s | y) = \sum_{m'=y}^m \binom{Z - (x - s + 1) - m'}{m - m'} p(X)^{m-m'} [1 - p(X)]^{Z - (x - s + 1) - m} \bullet G_Z(x, m', s - 1 | y) \quad s \neq 0, s \leq x \quad (3.20)$$

At the beginning of the sequence there must be exactly $x + y$ stations backlogged and so the boundary conditions on (3.20) are given by

$$G_Z(x, m, 0 | y) = \begin{cases} 1 & m = y \\ 0 & m \neq y \end{cases} \quad (3.21)$$

Since, in a given round, new arrivals to the system do not affect the order of transmissions in this round and since each station's arrival process is independent, the transition probabilities over one cycle can be computed as the product of the transition probabilities of each group of stations over the cycle. Using equation (3.20) and conditioning on the size of the backlog of the upstream and downstream stations at the beginning and end of their respective transmission sequences allows us to compute the elements of the transmission matrix $\mathbf{P}$. For $\delta(t_e^{(r)}) = 0$ we get

$$P(0, i, j)(\beta, k, l)$$

$$= \left\{ \begin{aligned} & [p((i+j)X + Y)]^\beta [1 - p((i+j)X + Y)]^{1-\beta} \\ & \cdot \left[\sum_{h=0}^k G_{N-1}(i, h, i | 0) \binom{N-1-h}{k-h} [p(jX + Y)]^{k-h} [1 - p(jX + Y)]^{N-1-k} \right] \\ & \cdot \left[\sum_{h=0}^l \sum_{y=0}^h \binom{M-N-j}{y} [p(iX)]^y [1 - p(iX)]^{M-N-j-y} G_{M-N}(j, h, j | y) \right. \\ & \quad \cdot \left. \binom{M-N-h}{l-h} [p(Y)]^{l-h} [1 - p(Y)]^{M-N-l} \right] \end{aligned} \right. \quad (3.22)$$

For $\delta(t_e^{(r)}) = 1$ we get

$$P(1, i, j)(\beta, k, l)$$

$$= \left\{ \begin{aligned} & [p(jX + Y)]^\beta [1 - p(jX + Y)]^{1-\beta} \\ & \cdot \left[\sum_{h=0}^k G_{N-1}(i, h, i | 0) \binom{N-1-h}{k-h} [p((j+1)X + Y)]^{k-h} \right. \\ & \quad \cdot \left. [1 - p((j+1)X + Y)]^{N-1-k} \right] \\ & \cdot \left[\sum_{h=0}^l \sum_{y=0}^h \binom{M-N-j}{y} [p((i+1)X)]^y [1 - p((i+1)X)]^{M-N-j-y} \right. \\ & \quad \cdot \left. G_{M-N}(j, h, j | y) \binom{M-N-h}{l-h} [p(Y)]^{l-h} [1 - p(Y)]^{M-N-l} \right] \end{aligned} \right. \quad (3.23)$$

From **P** we can compute the stationary distribution at the embedded points $\Pi = (\pi_0 \dots \pi_M)$.

Consider now an arbitrary arrival to the system in cycle r from the tagged station. The delay incurred by this packet consists of two components; the delay incurred from the instant of arrival until the end of cycle r and the delay incurred from the beginning of round $r+1$ until the end of the transmission of this packet. The distribution of delay is given by the convolution of the distributions of these two components of delay.

Since the arrival process is memoryless we recognize that the distribution of delay of the first component of delay is given by the distribution of delay of a packet over an interval $[0, t]$ given that the arrival occurs in this interval and that the packet remains backlogged for the remainder of the interval. Let t_a , $0 \leq t_a \leq t$, denote the arrival time of the packet. Then the delay incurred by the packet over the interval $[0, t]$, denoted by D , is $D = t - t_a$. Since inter-arrival times are exponentially distributed with mean $1/\lambda$ we can write

$$\begin{aligned}\Pr\{t_a \leq a \mid t_a \leq t\} &= \frac{1}{1 - e^{-\lambda t}} [1 - e^{-\lambda a}] \quad \text{or} \\ \Pr\{t_a > a \mid t_a \leq t\} &= \frac{1}{1 - e^{-\lambda t}} [e^{-\lambda a} - e^{-\lambda t}]\end{aligned}$$

The cumulative distribution function of delay is given by

$$\Pr\{D < d \mid t_a \leq t\} = \Pr\{t_a > t - d \mid t_a \leq t\} = \frac{e^{-\lambda d}}{1 - e^{-\lambda t}} [e^{\lambda d} - 1]$$

Differentiating with respect to d gives the probability density function

$$\text{pdf}_D(d) = \lambda e^{\lambda d} \frac{e^{-\lambda t}}{1 - e^{-\lambda t}}$$

From this distribution function we can compute the Laplace transform of the distribution of delay of a packet over the interval $[0, t]$ given that the packet arrives in this interval. This distribution function denoted by $\mathcal{E}^*(t, s)$ is

$$\mathcal{E}^*(t, s) = \frac{\lambda}{\lambda - s} \frac{e^{-st} - e^{-\lambda t}}{1 - e^{-\lambda t}} \quad (3.24)$$

Given that this arbitrary arrival is in a round with $S(t_e^{(r)}) = (\alpha, i, j)$ and $n_s(t_e^{(r+1)}) = k$ then the Laplace transform of the distribution of delay of the first component of packet delay is given by $\mathcal{E}^*(jX + Y, s)$ if $\alpha = 1$ or $\mathcal{E}^*((i+j)X + Y, s)$ if

$\alpha = 0$. The second component of delay is simply $(k+1)X$. The Laplace transform of the distribution of the total delay incurred by a packet arriving in such a round, denoted by $d_{(\alpha,i,j)(1,k,*)}^*(s)$, is given by the product of the transforms of the two distributions.

$$d_{(\alpha,i,j)(1,k,*)}^*(s) = \begin{cases} \mathcal{E}^*((i+j)X + Y, s)e^{-s(k+1)X} & \alpha = 0 \\ \mathcal{E}^*(jX + Y, s)e^{-s(k+1)X} & \alpha = 1 \end{cases} \quad (3.25)$$

The probability that this arbitrary packet arrives in a cycle with $S(t_e^{(r)}) = (\alpha, i, j)$ and $n_u(t_e^{(r)}) = k$ is given by

$$\zeta_{(\alpha,i,j)(1,k,*)} \triangleq \Pr\{S(t_e^{(r)}) = (\alpha, i, j), n_u(t_e^{(r+1)}) = k \mid \delta(t_e^{(r+1)}) = 1\} \quad (3.26)$$

where by conditioning on $\delta(t_e^{(r+1)}) = 1$ we have conditioned on the event of an arrival from the tagged station in cycle r . Let $\Pr\{\delta(t_e^{(r+1)}) = 1\} = 1/K$ then

$$\begin{aligned} \zeta_{(\alpha,i,j)(1,k,*)} &= \frac{\Pr\{S(t_e^{(r)}) = (\alpha, i, j), n_u(t_e^{(r+1)}) = k, \delta(t_e^{(r+1)}) = 1\}}{\Pr\{\delta(t_e^{(r+1)}) = 1\}} \\ &= K \Pr\{S(t_e^{(r)}) = (\alpha, i, j), n_u(t_e^{(r+1)}) = k, \delta(t_e^{(r+1)}) = 1\} \\ &= K \sum_{l=0}^{M-N} \Pr\{S(t_e^{(r)}) = (\alpha, i, j), S(t_e^{(r+1)}) = (1, k, l)\} \\ &= K \sum_{l=0}^{M-N} \pi_{(\alpha,i,j)} p_{(\alpha,i,j)(1,k,l)} \end{aligned} \quad (3.27)$$

and the constant K can be determined from

$$\sum_{\alpha=0}^1 \sum_{i=0}^{N-1} \sum_{j=0}^{M-N} \sum_{k=0}^{N-1} \zeta_{(\alpha,i,j)(1,k,*)} = 1 \quad (3.28)$$

Using (3.27) to remove the conditions on α, i, j and k in (3.25) we can express the Laplace transform of the distribution of delay of a packet from the tagged station

as

$$D^*(s) = \sum_{\alpha=0}^1 \sum_{i=0}^{N-1} \sum_{j=0}^{M-N} \sum_{k=0}^{N-1} \zeta_{(\alpha,i,j)(1,k,*)} d_{(\alpha,i,j)(1,k,*)}^*(s) \quad (3.29)$$

By successive differentiation of (3.29) and letting $s = 0$ one can compute the moments of delay to any order.

3.4 Analysis of the Head of Line Service Discipline

The analysis of HOLS consists of a generalization of the analysis of GSS. Here too we present a mean value analysis and a distribution of delay analysis. This analysis for HOLS is exact for the case $Y \leq X$ but becomes inexact, and in fact leads to pessimistic results, for the case $Y > X$. In the discussion of numerical results in the following section, simulation is also used for HOLS when $Y > X$.

3.4.1 Mean Value Analysis

As in the analysis for GSS, we consider two consecutive embedded points $t_e^{(r)}$ and $t_e^{(r+1)}$ and define the state of the system at the embedded points by the number of backlogged stations at that instant. Let $\mathbf{P}$ be the transition matrix for the embedded Markov process $n(t_e^{(r)})$. For the HOLS discipline, the number of transmissions in cycle r may be greater than $n(t_e^{(r)})$. In order to compute the elements of $\mathbf{P}$ we condition on the number of transmissions in the first sub-cycle.

$$p_{ik} = \sum_{l=i}^M \Pr\{n(t_e^{(r+1)}) = k \mid L = l, n(t_e^{(r)}) = i\} \Pr\{L = l \mid n(t_e^{(r)}) = i\} \quad (3.30)$$

where L is a random variable denoting the number of transmissions in the round. Note that by conditioning on L we have removed the dependency of $n(t_e^{(r+1)})$ on $n(t_e^{(r)})$, that is

$$\Pr\{n(t_e^{(r+1)}) = k \mid L = l, n(t_e^{(r)}) = i\} = \Pr\{n(t_e^{(r+1)}) = k \mid L = l\}.$$

Let $\theta_i(l) \triangleq \Pr\{L = l \mid n(t_e^{(r)}) = i\}$ and $\phi_l(k) \triangleq \Pr\{n(t_e^{(r+1)}) = k \mid L = l\}$. Instead of enumerating all possible events over the cycle, we use recursive functions in order to compute $\theta_i(l)$ and $\phi_l(k)$. Consider a round of L transmissions and the transmission period which is s transmission periods from the end of the round. Define the function $F(m, s)$ as the probability of m given stations each generating a packet in the next s transmissions. $F(m, s)$ can be computed recursively in terms of $F(m - j, s - 1)$ and the probability that there are j arrivals during transmission period s . This gives

$$F(m, s) = \sum_{j=0}^m \binom{m}{j} p(X)^j [1 - p(X)]^{m-j} F(m - j, s - 1) \quad (3.31)$$

where $p(X)$, the probability that an idle station generates a packet within a transmission period, has been defined in equation (3.9). In the case $s > 0$, the number of backlogged stations must be at least one to ensure that a packet transmission occurs in the next transmission period and, since there are exactly s transmissions left in the sub-cycle, there can therefore be no more than $s - 1$ new packets generated in transmission period s . Thus $F(m, s) = 0$ for $m \geq s$, $s \neq 0$. At the end of the sub-cycle, i.e., $s = 0$, no more packets will be generated in this sub-cycle with probability one. Hence $F(0, 0) = 1$. Starting with these boundary conditions, $\theta_i(l)$ can be computed as $\binom{M-i}{l-i} F(l-i, l)$ times the probability that $M-l$ stations do nothing in the sub-cycle. Note that in the case where all stations are idle at the beginning of the round ($n(t_e^{(r)}) = 0$) then the channel remains idle until the

slot immediately after some station generates a packet. Since there can be multiple arrivals during the last idle slot, the case $n(t_e^{(r)}) = 0$ can be computed as a convex combination of all the cases $n(t_e^{(r)}) > 0$. Thus $\theta_i(l)$ can be expressed as

$$\theta_i(l) = \begin{cases} [1 - p(lX)]^{M-l} \binom{M-i}{l-i} F(l-i, l) & 0 < i \leq M \\ \sum_{j=1}^M \frac{\binom{M}{j} p(X)^j [1 - p(X)]^{M-j}}{1 - [1 - p(X)]^M} \theta_j(l) & i = 0 \end{cases} \quad (3.32)$$

Having conditioned on the length of the round, we can compute $\phi_l(k)$ using a similar (and simpler) recursive function to the one used in the GSS analysis. Consider the s^{th} transmission period in a given round as counted from the *beginning* of the round. Define the function $G(m, s)$ as the probability that, out of the s stations that transmitted in the previous s transmission periods, m of them have generated new packets. For HOLS only those stations that have transmitted in the previous $s - 1$ transmission periods could have generated packets to transmit in the next round; we do not need to consider any arrivals from a station that has not yet transmitted in the current round. $G(m, s)$ can be computed recursively in terms of $G(j, s - 1)$ and the probability of $m - j$ new arrivals in the s^{th} transmission period. We express $G(m, s)$ as

$$G(m, s) = \sum_{j=0}^m \binom{s-1-j}{m-j} [p(X)]^{m-j} [1 - p(X)]^{s-1-m} G(j, s-1) \quad (3.33)$$

where the boundary conditions are $G(m, s) = 0$ for $m \geq s$ since a station can have at most one packet in its buffer waiting to be transmitted and the last station to transmit could not have generated a new one, and $G(0, 1) = 1$ since the number of new packets generated after the first transmission in the round is zero with probability one. $\phi_l(k)$ is given by

$$\phi_l(k) = \sum_{j=0}^{l-1} G(j, l) \binom{M-j}{k-j} [p(Y)]^{k-j} [1 - p(Y)]^{M-k} \quad (3.34)$$

where the term $p(Y)^{k-j}[1-p(Y)]^{M-k}$ accounts for the probability that $k-j$ stations generate packets during the second sub-cycle. Note that this assumes that any packets generated during the second sub-cycle remain backlogged until the next cycle. Although this assumption is exact when $Y \leq X$, it becomes inexact when $Y > X$ since, in the latter case, it may be possible for a non-dormant station to both generate and transmit a packet during the second sub-cycle. This analysis leads to pessimistic results when $Y > X$.

The elements of $\mathbf{P}$ are now computed as

$$p_{ik} = \sum_{l=\max(1,i)}^M \theta_i(l) \phi_l(k).$$

Given $\mathbf{P}$ we can calculate the stationary distribution of the backlog at the embedded points and the average throughput and average delay using results from the theory of regenerative processes as in the mean value analysis of the GSS discipline.

Average Throughput: The average channel throughput S is computed as the ratio of the expected time in a cycle that the channel is carrying packets to the expected length of a cycle. The expected number of transmissions in a cycle is given by

$$\bar{L} = \sum_{i=0}^M \pi_i \sum_{l=\max(1,i)}^M \theta_i(l) l. \quad (3.35)$$

Hence the throughput is given by

$$S = \frac{\bar{L}T}{\pi_0 X / (1 - e^{-M\lambda X}) + \bar{L}X + Y} \quad (3.36)$$

where the term $X/(1 - e^{-M\lambda X})$ is the expected length of the idle time in a cycle where $n(t_e^{(r)}) = 0$.

Average Packet Delay: As in the analysis of GSS, the average delay of a packet is given by

$$D = \frac{M}{S} - \frac{1}{\lambda} \quad (3.37)$$

where S is given by equation (3.36) above. Alternatively, the average delay can be computed from the average backlog and the average throughput by using Little's result.

The packets that contribute to the sum of the backlog can be separated into three groups. The first group is those packets that are transmitted during the current round. Packets of this group are those that are backlogged at $t_c^{(r)}$ and all the ones that subsequently arrive to idle stations that have not yet transmitted in this round. The expected sum of the backlog due to packets in this group depends on what time during the cycle these new arrivals take place. It is extremely tedious, if not impossible to enumerate every possible combination of events. Instead we again resort to a recursive approach in order to compute the expected sum of the backlog due to these packets.

Define the function $H(m, s)$ as the expected sum of the backlog over the remaining s transmission periods of the first sub-cycle given that there are m non-dormant stations currently backlogged. Note that s represents the maximum number of remaining transmission periods. If there are no more backlogged stations ($m = 0$), then the sub-cycle comes to an end. Thus we require that

$$H(0, s) = 0 \quad (3.38)$$

For the case $m > 0$ however, the sub-cycle does not end with the current transmission period and $H(m, s)$ is computed as the backlog over this transmission period

and the remaining $s - 1$ transmission periods. So

$$H(m, s) = mX + \sum_{n=m-1}^{s-1} \mu_{ms}(n) [(n - m + 1)u(X) + H(n, s - 1)] \quad (3.39)$$

where $\mu_{ms}(n)$ is the probability that $n - m + 1$ new arrivals occurred out of a possible $s - m$ and is given by

$$\mu_{ms}(n) = \binom{s-m}{n-m+1} [p(X)]^{n-m+1} [1 - p(X)]^{s-n-1} \quad (3.40)$$

The expected sum of the backlog, conditioned on $n(t_e^{(r)}) = i$, due to packets of the first group is simply given by $H(i, M)$.

Packets from the second group are those that are generated by a dormant station and hence only transmitted in the round following the current one. The contribution to $b_s(i)$ of any dormant station is the expected length of time from the instant that the new packet is generated to the end of the cycle multiplied by the probability that a packet is generated by this station. Consider the j^{th} station to transmit in a round of length L . This station will generate a new packet in the interval $[t_e^{(r)} + jX, t_e^{(r+1)}]$ with probability $p((L - j)X + Y)$. Hence the contribution to the expected sum of the backlog of this packet, conditioned on the length of a round, is $p((L - j)X + Y)u[(L - j)X + Y]$. Summing contributions from all such packets and removing the condition on the length of the round gives the expected sum of the backlog due to packets from the second group.

Packets of the third group are those that arrive to an active station and are transmitted in round following the current one. This group includes only packets generated in the inter-round overhead period. Given that the round is of length L , there are $M - L$ stations that can contribute a packet to this group. The contribution of each of these to the expected sum of the backlog is $p(Y)u(Y)$. Summing these

contributions and removing the condition on the length of the round gives the expected sum of the backlog due to packets from the third group.

For a cycle with $n(t_c^{(r)}) = 0$, the round starts at the beginning of the first slot after the first arrival. In this case $b_s(0)$ can be expressed in terms of $b_s(i)$, $i > 0$, by adding the additional contribution to the backlog of those arrivals in the last idle slot. Hence $b_s(i)$ is given by

$$b_s(i) = \begin{cases} H(i, M) + \sum_{l=i}^M \theta_i(l) \sum_{j=0}^{l-1} p(jX + Y) u(jX + Y) + \sum_{l=1}^M \theta_i(l) (M - l) p(Y) u(Y) & 1 \leq i \leq M \\ \sum_{j=1}^M \frac{\binom{M}{j} p(X)^j [1 - p(X)]^{M-j}}{1 - [1 - p(X)]^M} [ju(X) + b_s(j)] & i = 0 \end{cases} \quad (3.41)$$

From equation (3.41) we can calculate the expected backlog as

$$B = \frac{\sum_{i=0}^M \pi_i b_s(i)}{\pi_0 X / (1 - e^{-M\lambda X}) + \bar{L}X + Y} \quad (3.42)$$

From B and the expression for the throughput given in (3.36), the average packet delay D is expressed as

$$D = \frac{\sum_{i=0}^M \pi_i b_s(i)}{\bar{L}T}. \quad (3.43)$$

3.4.2 Distribution of Delay Analysis

In the GSS analysis we considered a single station to be tagged and then considered packets from this station. To compute the transition probabilities of the embedded Markov process, we divided the stations into three groups and calculated

the transition probabilities of each group independently. This was possible since the transmissions of the stations in each group are contiguous due to the predetermined order in which the stations are allowed to transmit in GSS. For HOLS, the order of transmissions varies according to the arrival pattern of packets to the stations. Therefore we are unable to use this approach to compute the distribution of delay in the general case without resorting to an extremely cumbersome state descriptor. However, if we consider that the tagged station is either the most upstream station (station 1) or the most downstream station (station M), the size of one of the three groups is always zero and the analysis becomes more easily manageable. In this section we present the analysis for the case where one of the end stations is the tagged one.

The Markov process that we are now required to analyze is sufficiently described by the number of stations in each group at the embedded points. We define the state descriptor as $S(t_e^{(r)}) \triangleq (\delta(t_e^{(r)}), n(t_e^{(r)}))$ where $\delta(t_e^{(r)})$ denotes the state of the tagged station and $n(t_e^{(r)})$ is the number of backlogged stations excluding the tagged one. In order to compute the transition matrix $\mathbf{P}$, we condition on both the number of transmissions in a round and the transmission period in which the tagged station transmits. The elements of $\mathbf{P}$ are given by

$$P_{(\alpha, i)(\gamma, k)} = \sum_{l=\alpha+i}^M \sum_{y=0}^l \Pr\{S(t_e^{(r+1)}) = (\gamma, k) \mid L = l, TP_t = y, S(t_e^{(r)}) = (\alpha, i)\} \\ \bullet \Pr\{L = l, TP_t = y \mid S(t_e^{(r)}) = (\alpha, i)\} \quad (3.44)$$

where L is a random variable denoting the number of transmissions in the current round and TP_t is a random variable denoting the transmission period, counted from the beginning of the round, in which the tagged station transmits a packet. We adopt the convention $TP_t = 0$ to indicate that the tagged station does not transmit in the current round. Note that conditioning on L and TP_t removes the dependency

of $S(t_e^{(r+1)})$ on $S(t_e^{(r)})$. Thus

$$\begin{aligned} \Pr\{S(t_e^{(r+1)}) = (\gamma, k) \mid L = l, TP_t = y, S(t_e^{(r)}) = (\alpha, i)\} \\ = \Pr\{S(t_e^{(r+1)}) = (\gamma, k) \mid L = l, TP_t = y\} \end{aligned} \quad (3.45)$$

Let $\theta_{(\alpha, i)}(l, y) \triangleq \Pr\{L = l, TP_t = y \mid S(t_e^{(r)}) = (\alpha, i)\}$ and $\phi_{(l, y)}(\gamma, k) \triangleq \Pr\{S(t_e^{(r+1)}) = (\gamma, k) \mid L = l, TP_t = y\}$. θ and ϕ are now computed using recursive functions in a similar way to that used in the mean value analysis of HOLS. In order to compute θ we must distinguish which station is tagged.

Most upstream station tagged: Consider a round of length L . For the transmission period that is s transmission periods away from the end of the round where $0 \leq s \leq L$, define the function $f(x, m, s)$ as the joint probability of the tagged station transmitting in the x^{th} transmission period *from the end of the round* (i.e. arriving in transmission period $x + 1$) and m given stations *excluding* the tagged station each generating a packet in the next s transmissions. Since x is counted from the end of the round and TP_t is counted from the beginning of the round we have the relationship

$$TP_t = L - x + 1. \quad (3.46)$$

For the case $s = 0$ we must have

$$\begin{aligned} f(x, 0, 0) &= 1 \quad \text{and} \\ f(x, m, 0) &= 0 \quad \text{for } m > 0 \end{aligned} \quad (3.47)$$

since, when the sub-cycle has ended, there must be with probability one no backlogged stations. For the case $s \neq 0$ we already have at least one backlogged station (otherwise the sub-cycle would end which implies that $s = 0$) and thus cannot allow more than $s - 1$ of the remaining stations to generate new packets. This requires

that

$$f(x, m, s) = 0 \quad \begin{array}{l} \text{for } m \geq s, s \leq x, s \neq 0 \\ \text{or } m \geq s - 1, s > x \end{array} \quad (3.48)$$

The distinction between the case $s \leq x$ and $s > x$ is made because the number of given stations m does not include the tagged one. For the case where $m < s, s \leq x, s \neq 0$ or $m < s - 1, s > x$, $f(x, m, s)$ can be computed recursively in terms of $f(x, m', s - 1)$ by considering the number of new packets that are generated in the next transmission period. Ignoring the tagged station we can have up to m new arrivals.

$$f(x, m, s) = \begin{cases} \sum_{j=0}^m \binom{m}{j} [p(X)]^j [1 - p(X)]^{m-j} f(x, m - j, s - 1) & m < s, s \leq x, s \neq 0 \text{ or } m < s - 1, s > x \\ 0 & \text{otherwise} \end{cases} \quad (3.49)$$

$\theta_{(\alpha, i)}(l, y)$ is computed as the product of $f(l - y + 1, l - i - \alpha, l)$, the number of ways of choosing $l - i - 1$ stations out of a possible $M - i - 1$ (since i stations and the tagged station are already given) and the probability that the remaining $M - l$ stations do not generate any packets for the duration of the sub-cycle. In addition there are special cases that must be accounted for. In the case $\alpha = 1$ and $y \neq 1$, $\theta_{(1, i)}(l, y) = 0$ since, if station 1 is backlogged at $t_e^{(r)}$, it will transmit in the first transmission period of the round with probability one. Similarly $\theta_{(0, i)}(l, 1) = 0, i \neq 0$ since station 1 cannot transmit in the first transmission period if it is idle and other stations are active at the beginning of the round. In the case where all stations are idle at the beginning of the round ($S(t_e^{(r)}) = (0, 0)$) then the channel remains idle until some station generates a packet. Transmissions then begin at the beginning of the next slot. In this case we can consider the round to be one with $S(t_e^{(r)}) = (\alpha, i)$ with probability $\frac{\binom{M-1}{i} p(X)^{i+1} [1 - p(X)]^{M-i-1}}{1 - [1 - p(X)]^M}$ if $\alpha = 1$ or $\frac{\binom{M-1}{i} p(X)^i [1 - p(X)]^{M-i}}{1 - [1 - p(X)]^M}$ if $\alpha = 0$.

That is, the case $\alpha = i = 0$ is a convex combination of all the cases $\alpha + i \neq 0$. Thus $\theta_{(\alpha,i)}(l, y)$ is given by

$$\theta_{(\alpha,i)}(l, y) = \begin{cases} 0 & y = 1, \alpha = 0, i \neq 0 \\ & \text{or } y \neq 1, \alpha = 1, \forall i \\ [1 - p(lX)]^{M-l} \binom{M-i-1}{l-i} f(l+1, l-i, l) & y = 0, \alpha = 0, i \neq 0 \\ [1 - p(lX)]^{M-l} \binom{M-i-1}{l-i-1} f(l-y+1, l-i-1, l) & y = 1, \alpha = 1, \forall i \\ [1 - p(lX)]^{M-l} \binom{M-i-1}{l-i-1} [1 - p((y-2)X)] p(X) & y > 1, \alpha = 0, i \neq 0 \\ \quad \bullet f(l-y+1, l-i-1, l) & \\ \sum_{m=0}^{M-1} \frac{\binom{M-1}{m} p(X)^{m+1} [1 - p(X)]^{M-m-1}}{1 - [1 - p(X)]^M} \theta_{(1,m)}(l, y) & \forall y, \alpha = 0, i = 0 \\ + \sum_{m=1}^{M-1} \frac{\binom{M-1}{m} p(X)^m [1 - p(X)]^{M-m}}{1 - [1 - p(X)]^M} \theta_{(0,m)}(l, y) & \end{cases}$$

$$\alpha + i \leq l \leq M$$

$$0 \leq y \leq l$$
(3.50)

Most downstream station tagged: Consider a sub-cycle of length $L = l$ where station M transmits in transmission period $TP_t = y > 0$. That part of the sub-cycle up to and including transmission period y is of length $L_1 = y$ and that part which is after transmission period y is of length $L_2 = l - y$. With these definitions $\theta_{(\alpha,i)}(l, y)$ can be expressed as

$$\begin{aligned} \theta_{(\alpha,i)}(l, y) &= \Pr\{L_1 = y, L_2 = l - y \mid S(t_c^{(r)}) = (\alpha, i)\} \\ &= \Pr\{L_1 = y \mid S(t_c^{(r)}) = (\alpha, i)\} \Pr\{L_2 = l - y \mid L_1 = y, S(t_c^{(r)}) = (\alpha, i)\}. \end{aligned}$$
(3.51)

We note that since station M , when active and backlogged, defers its transmission to all other active backlogged stations, it will transmit only when no other station

is waiting to do so. Therefore we can compute $\Pr\{L_1 = y \mid S(t_e^{(r)}) = (\alpha, i)\}$ from $F(y - 1 - i, y - 1)$ where $F(m, s)$ is the recursive function given in equation (3.31), by considering only the $M - 1$ non-tagged stations. Hence

$$\begin{aligned} \Pr\{L_1 = y \mid S(t_e^{(r)}) = (\alpha, i)\} \\ = [p((y - 1)X)]^{1-\alpha} [1 - p((y - 1)X)]^{M-y} \binom{M-i-1}{y-i-1} F(y-i-1, y-1) \end{aligned} \quad (3.52)$$

where the term $[p((y - 1)X)]^{1-\alpha}$ is the probability that station M is ready with a packet to transmit at the beginning of transmission period y and the term $[1 - p((y - 1)X)]^{M-y}$ is the probability that none of the $M - y$ other stations generate any packets in the first $y - 1$ transmission periods in the round.

Obviously any station that transmits after station M has had its turn must have generated its packet either during or after the transmission of station M . Thus by conditioning on the number of stations that become backlogged during this transmission period we can use the same recursive function to compute $\Pr\{L_2 = L - y \mid L_1 = y, S(t_e^{(r)}) = (\alpha, i)\}$ as

$$\begin{aligned} \Pr\{L_2 = l - y \mid L_1 = y, S(t_e^{(r)}) = (\alpha, i)\} &= [1 - p((l - y + 1)X)]^{M-l} \\ &\quad \sum_{m=0}^{l-y} \binom{M-y}{l-y} \binom{l-y}{m} p(X)^m [1 - p(X)]^{l-y-m} F(l-y-m, l-y) \end{aligned} \quad (3.53)$$

By substituting equations (3.52) and (3.53) into (3.51) we can compute $\theta_{(\alpha, i)}(l, y)$

as

$$\theta_{(\alpha,i)}(l,y) = \begin{cases} 0 & 0 < y \leq i, \forall \alpha, i > 0 \\ & y = 0, \alpha = 1, \forall i \\ [1 - p(lX)]^{M-l} \binom{M-i-1}{l-i} F(l-i, l) & y = 0, \alpha = 0, i > 0 \\ [1 - p(lX)]^{M-l} [1 - p((y-1)X)]^{l-y} \binom{M-i-1}{y-i-1} \binom{M-y}{l-y} F(y-i-1, y-1) \\ \quad \cdot [p((y-1)X)]^{1-\alpha} \sum_{m=0}^{l-y} \binom{l-y}{m} p(X)^m [1 - p(X)]^{l-y-m} F(l-y-m, l-y) & y > i, \alpha + i \neq 0 \\ \sum_{i=0}^{M-1} \frac{\binom{M-1}{i} p(X)^{i+1} [1 - p(X)]^{M-i-1}}{1 - [1 - p(X)]^M} \theta_{(1,i)}(l, y) \\ \quad + \sum_{i=1}^{M-1} \frac{\binom{M-1}{i} p(X)^i [1 - p(X)]^{M-i}}{1 - [1 - p(X)]^M} \theta_{(0,i)}(l, y) & \forall y, \alpha = 0, i = 0 \end{cases} \quad (3.54)$$

A recursive approach is also used to compute ϕ . Note that, although $\phi_{(l,y)}(\gamma, k)$ depends on the length of the round and the transmission period in which the tagged station transmits, it does not depend on which station is tagged. Consider a round of length L and the s^{th} transmission period in this round where $1 \leq s \leq L$. Define the function $g(x, m, s)$ as the probability that out of s stations that transmitted in the previous s transmission periods, m of them, *excluding the tagged station*, generated new packets during these s transmissions given that the tagged station transmitted in the x^{th} transmission period in the round. Note that for this function, s and x are measured from the *beginning* of the round.

For any $s > 0$ there are s dormant stations and at most $s - 1$ of them can have new packets in their buffer since the station that has just completed its transmission could not have generated a new packet instantaneously. If the tagged station has transmitted in transmission period y where $y < s$ then at most $s - 2$ of the

non-tagged stations could have new packets in each of their single buffers. Hence $g(x, m, s)$ must satisfy

$$g(x, m, s) = 0 \quad \text{if } m \geq s, \forall s \text{ or if } m \geq s - 1, s > x \quad (3.55)$$

For $s = 1$ the number of dormant stations that are backlogged must be zero with probability one by the same argument. Thus

$$g(x, 0, 1) = 1 \quad \forall x. \quad (3.56)$$

For $s > 1$ with $m < s$ or $m < s - 1, s > x$, $g(x, m, s)$ can be computed recursively in terms of $g(x, m', s - 1)$ by considering the new arrivals to dormant stations in the transmission period s . We distinguish two cases: Either $s \leq x$ in which case there are $m - m'$ dormant stations out of a possible $s - 1 - m'$ that generate new packets in transmission period s or $s > x$, in which case there are $m - m'$ dormant stations out of a possible $s - 2 - m'$ that generate new packets in transmission period s . This gives

$$g(x, m, s) = \begin{cases} \sum_{j=0}^m \binom{s-1-j}{m-j} p(1)^{m-j} [1 - p(1)]^{s-1-m} g(x, j, s-1) & s \leq x \\ \sum_{j=0}^m \binom{s-2-j}{m-j} p(1)^{m-j} [1 - p(1)]^{s-2-m} g(x, j, s-1) & s > x \end{cases} \quad (3.57)$$

$\phi(l, y)(\gamma, k)$ is now computed from $g(x, m, s)$ and the probability of some arrivals during the inter-round gap

$$\phi(l, y)(\gamma, k) = \begin{cases} \sum_{j=0}^k g(l+1, j, l) \binom{M-1-j}{k-j} [p(Y)]^{k-j+\gamma} [1 - p(Y)]^{M-k-\gamma} & y = 0 \\ [p((l-y)X + Y)]^\gamma [1 - p((l-y)X + Y)]^{1-\gamma} \\ \quad \cdot \sum_{j=0}^k g(y, j, l) \binom{M-1-j}{k-j} [p(Y)]^{k-j} [1 - p(Y)]^{M-1-k} & y \neq 0 \end{cases} \quad (3.58)$$

The elements of $\mathbf{P}$ can be computed directly from θ and ϕ . From $\mathbf{P}$ we can calculate the stationary distribution Π .

In order to compute the distribution of delay for a packet from the tagged station we consider an arbitrary arrival to the system from this station in a given cycle. This arrival can be categorized into one of four types, each with a relatively simple distribution of delay. The Laplace transform of the distribution of delay of a packet of type v , $1 \leq v \leq 4$, which is generated in round r where $S(t_c^{(r)}) = (\alpha, i)$, $S(t_c^{(r+1)}) = (\gamma, k)$, $L = l$ and $TP_i = y$, is denoted by $d_{(\alpha, i)(\gamma, k)l, y}^{*(v)}(s)$. This is derived for each packet type.

Packets of type 1 are those that are generated by the tagged station during the first sub-cycle in a cycle where station 1 has not yet transmitted. Suppose that the tagged station transmits in transmission period TP_i . In the case where station 1 is the tagged station this packet must have been generated in transmission period $TP_i - 1$. In the case where station M is the tagged station, this packet must have been generated at some time during the previous $TP_i - 1$ transmission periods. The distribution of delay of such a packet is given by

$$d_{(\alpha, i)(\gamma, k)l, y}^{*(1)}(s) = \begin{cases} \mathcal{E}^*(X, s)e^{-sX} & \text{station 1 tagged} \\ \mathcal{E}^*((y-1)X, s)e^{-sX} & \text{station } M \text{ tagged} \end{cases} \quad (3.59)$$

Packets of type 2 are those that arrive during sub-cycle 2 (i.e., the inter-round gap). The delay incurred by this packet consists of two parts. The first part is the delay incurred until the beginning of the next cycle, the distribution of which is given by $\mathcal{E}^*(Y, s)$; the second part is the delay incurred in the next round while the tagged station waits for a turn to transmit. The distribution of this delay, conditioned on $S(t_c^{(r+1)}) = (1, k)$ is given by conditioning on L and TP_i in the

$(r+1)^{\text{th}}$ round, as

$$\sum_{l'=k+1}^M \sum_{y'=1}^{l'} \theta_{(1,k)}(l', y') e^{-y' \cdot X}. \quad (3.60)$$

Since the total delay of a packet of this type is the sum of two independent random variables, the distribution of delay in the Laplace domain is given by the product of the distributions of each part. Hence

$$d_{(\alpha,i)(\gamma,k)l,y}^{*(2)}(s) = \mathcal{E}^*(Y, s) \sum_{l'=k+1}^M \sum_{y'=1}^{l'} \theta_{(1,k)}(l', y') e^{-y' \cdot X} \quad (3.61)$$

Note that for the case where station 1 is the tagged station, $\theta_{(1,k)}(l', y') = 0$ for $y' \neq 1$ and so equation (3.61) reduces to $d_{(\alpha,i)(\gamma,k)l,y}^{*(2)}(s) = \mathcal{E}^*(Y, s) e^{-X}$.

Packets of type 3 are the packets that are generated when the tagged station is dormant, that is, after the tagged station has already transmitted in the current round. Given that the packet is generated in round r then the delay incurred in this round, conditioned on $L = l$ and $TP_l = y$ has a distribution given by $\mathcal{E}^*((l-y)X + Y, s)$ and the delay incurred in round $(r+1)$ conditioned on $n_*(t_e^{(r+1)}) = k$ has a distribution given by the expression in (3.60). Hence the distribution of delay for a packet of type 3, given by the product of these distributions, is

$$d_{(\alpha,i)(\gamma,k)l,y}^{*(3)}(s) = \mathcal{E}^*((l-y)X + Y, s) \sum_{l'=k+1}^M \sum_{y'=1}^{l'} \theta_{(1,k)}(l', y') e^{-y' \cdot X}. \quad (3.62)$$

Packets of type 4 distinguish the special case where the system is idle at the beginning of the round. Packets of this type are those that are generated by the tagged station in a round where $\mathcal{S}(t_e^{(r)}) = (0, 0)$ and are transmitted in the first sub-cycle of this cycle. In the case where station 1 is the tagged station, this packet will be transmitted in the slot immediately following the one in which it was generated

and therefore is indistinguishable from a packet of type 1; however, in the case where station M is tagged, the distribution of delay of this packet must take into account the idle slot immediately preceding the first transmission of the round, in which the packet could have been generated. The distribution of delay of a packet of type 4 is

$$d_{(\alpha,i)(\gamma,k)l,y}^{*(4)}(s) = \begin{cases} \mathcal{E}^*(X, s)e^{-sX} & \text{station 1 tagged} \\ \mathcal{E}^*(yX, s)e^{-sX} & \text{station } M \text{ tagged} \end{cases} \quad (3.63)$$

Let $\zeta_{(\alpha,i)(\gamma,k)l,y}^{(v)}$ denote the probability that an arbitrary arrival from the tagged station is in a cycle with $S(t_e^{(r)}) = (\alpha, i)$, $S(t_e^{(r+1)}) = (\gamma, k)$, $L = l$, $TP_l = y$ and is of type v . $\zeta_{(\alpha,i)(\gamma,k)l,y}^{(v)}$ is also the fraction of packets which are of type v and arrive in such a round. We now show how to derive ζ and then use this to compute the distribution of delay for a packet from the tagged station by removing the conditions on the expressions given in equations (3.59) through (3.63) for the distribution of delay.

In order to compute $\zeta_{(\alpha,i)(\gamma,k)l,y}^{(v)}$ we first derive $\Pr\{S(t_e^{(r)}) = (\alpha, i), S(t_e^{(r+1)}) = (\gamma, k), L = l, TP_l = y\}$ as follows

$$\begin{aligned} & \Pr\{S(t_e^{(r)}) = (\alpha, i), S(t_e^{(r+1)}) = (\gamma, k), L = l, TP_l = y\} \\ &= \Pr\{S(t_e^{(r)}) = (\alpha, i)\} \Pr\{S(t_e^{(r+1)}) = (\gamma, k), L = l, TP_l = y \mid S(t_e^{(r)}) = (\alpha, i)\} \\ &= \Pr\{S(t_e^{(r)}) = (\alpha, i)\} \Pr\{L = l, TP_l = y \mid S(t_e^{(r)}) = (\alpha, i)\} \\ & \quad \bullet \Pr\{S(t_e^{(r+1)}) = (\gamma, k) \mid L = l, TP_l = y\} \\ &= \pi_{(\alpha,i)} \theta_{(\alpha,i)}(l, y) \phi_{(l,y)}(\gamma, k) \end{aligned} \quad (3.64)$$

In an arbitrary cycle, a packet of type v , $1 \leq v \leq 4$, is generated by the tagged station if and only if this cycle meets certain conditions. A packet of type 1 is

generated if $\delta(t_e^{(r)}) = 0$, $n(t_e^{(r)}) > 0$ and $TP_t > 0$. A packet of type 2 is generated if $\delta(t_e^{(r)}) = 0$, $n(t_e^{(r)}) > 0$, $\delta(t_e^{(r+1)}) = 1$ and $TP_t = 0$. A packet of type 3 is generated if $TP_t > 0$ and $\delta(t_e^{(r+1)}) = 1$. A packet of type 4 is generated if $S(t_e^{(r)}) = (0, 0)$ and $TP_t > 0$. For any other conditions no packets are generated by the tagged station. Consider now a large number of cycles with $S(t_e^{(r)}) = (\alpha, i)$, $S(t_e^{(r+1)}) = (\gamma, k)$, $L = l$, $TP_t = y$. Suppose that there are Z such cycles. In the limit as $Z \rightarrow \infty$, the number of packets of type v is given by Z times the probability that the tagged station generates a packet of type v in a cycle. This probability is given by

$$\Pr\{\text{Tagged station generates a packet of type } v\} = \begin{cases} \pi_{(0,i)}\theta_{(0,i)}(l, y)\phi_{(l,y)}(\gamma, k) & v = 1, i > 0, y > 0 \\ \pi_{(0,i)}\theta_{(0,i)}(l, 0)\phi_{(l,0)}(1, k) & v = 2, i > 0 \\ \pi_{(\alpha,i)}\theta_{(\alpha,i)}(l, y)\phi_{(l,y)}(1, k) & v = 3, y > 0, \\ \pi_{(0,0)}\theta_{(0,0)}(l, y)\phi_{(l,y)}(\gamma, k) & v = 4, y > 0 \end{cases} \quad (3.65)$$

The probability that an arbitrary arrival is of type v is given by the ratio of the number of packets generated by the tagged station which are of type v divided by the total number of packets generated by the tagged station. Hence ζ is given by

$$\zeta_{(\alpha,i)(\gamma,k),l,y}^{(v)} = \begin{cases} K\pi_{(0,i)}\theta_{(0,i)}(l, y)\phi_{(l,y)}(\gamma, k) & v = 1, \alpha = 0, i > 0, y > 0 \\ K\pi_{(0,i)}\theta_{(0,i)}(l, 0)\phi_{(l,0)}(1, k) & v = 2, \alpha = 0, \gamma = 1, i > 0, y = 0 \\ K\pi_{(\alpha,i)}\theta_{(\alpha,i)}(l, y)\phi_{(l,y)}(1, k) & v = 3, y > 0, \gamma = 1 \\ K\pi_{(0,0)}\theta_{(0,0)}(l, y)\phi_{(l,y)}(\gamma, k) & v = 4, \alpha = 0, i = 0, y > 0 \\ 0 & \text{otherwise} \end{cases} \quad (3.66)$$

where K is a constant such that

$$\sum_{v=1}^4 \sum_{\alpha=0}^1 \sum_{i=0}^{M-1} \sum_{\gamma=0}^1 \sum_{k=0}^{M-1} \sum_{l=\max(1, \alpha+i)}^M \sum_{y=0}^l \zeta_{(\alpha,i)(\gamma,k),l,y}^{(v)} = 1 \quad (3.67)$$

From ζ and the equations for the delay given in (3.59) through (3.63), it is straightforward to compute the Laplace transform of the distribution of delay for a

packet generated by the tagged station. Denoting this distribution by $D^*(s)$ we get

$$D^*(s) = \sum_{v=1}^4 \sum_{\alpha=0}^1 \sum_{i=0}^{M-1} \sum_{\gamma=0}^1 \sum_{k=0}^{M-1} \sum_{l=\alpha+i}^M \sum_{y=0}^l \zeta_{(\alpha,i)(\gamma,k)l,y}^{(v)} d_{(\alpha,i)(\gamma,k)l,y}^{*(v)}(s). \quad (3.68)$$

By successive differentiation of (3.68) and letting $s = 0$, one can compute the moments of delay to any order.

3.5 Numerical Results

We discuss in this section numerical results showing the performance of Expressnet and Fasnet operating under the three service disciplines. Although the emphasis is to show Expressnet operating under the NGSS discipline and Fasnet operating under the GSS and HOLS disciplines, we also present some representative figures showing Expressnet operating under GSS and Fasnet operating under NGSS. Recall that only Fasnet can support the HOLS service discipline. Let $a \triangleq \tau/T$. The unit of time is taken to be the transmission time of a packet (i.e., $T = 1$). In both Expressnet and Fasnet we neglect the inter-packet overhead t_o since this is assumed to be small compared to the length of the packet. The inter-round overhead Y is then taken to be $2a$ for Expressnet and $[2a] + 1$ for Fasnet. The performance of these networks for various values of a and M is presented in terms of the throughput as a function of the generation rate of packets and the throughput-delay trade-off. These results show that all three service disciplines exhibit similar performance characteristics. This is to be expected since they are merely variations of a basic round robin algorithm. However there are interesting differences which we will highlight in the discussion. All numerical results are obtained from analysis with the exception of HOLS when $Y > 1$, in which case simulation is used. The reason is that, as pointed

out in section 4, the analysis for HOLS gives pessimistic results when $Y > 1$. In most of the results shown below, the preamble in Expressnet has been assumed to be negligible except for certain figures where its effect is explicitly shown.

In Figs. 3.1, 3.2 and 3.3 we show the behavior of the throughput S as a function of the aggregate generation rate $M\lambda$ for $a = 1.0$ and 10 , and $M = 20, 30$ and 50 , for NGSS in Expressnet, and GSS and HOLS in Fasnet. $M\lambda$ is the rate at which packets would be presented to the system if all stations were in the idle state. The curves show that S increases steadily as $M\lambda$ increases from zero until some finite value of $M\lambda$ (in the vicinity of one), and remains practically constant as $M\lambda$ increases further. This shows that the system remains stable as the load increases to infinity. (Contrast this to CSMA/CD where stability can only be achieved by using some form of dynamic control or a long rescheduling delay leading to a high packet delay [51].) For HOLS with $a = 10$, the curves, which were obtained by simulation, exhibit a slight hump before S levels off to its constant value. This occurs since, at the generation rate corresponding to the hump in S , all stations are on the average transmitting in every round, but some stations happen to generate and transmit their packets during the inter-round overhead; this results in a *lower* effective overhead and hence higher throughput than expected. As mentioned at the beginning of this chapter, it is possible to operate Expressnet under the GSS discipline and Fasnet under the NGSS discipline.

In order to show the effect of the service discipline on the channel throughput, we plot in Fig. 3.4 for Expressnet and in Fig. 3.5 for Fasnet S vs. $M\lambda$ for each of these two service disciplines. Note how, as a result of gating (i.e., the delaying of packets until the round following the one in which they were generated), the throughput achieved by GSS is always less than or equal to that achieved by NGSS.

In Fig. 3.6 we show on a single sheet for comparison purposes, S vs. $M\lambda$ for

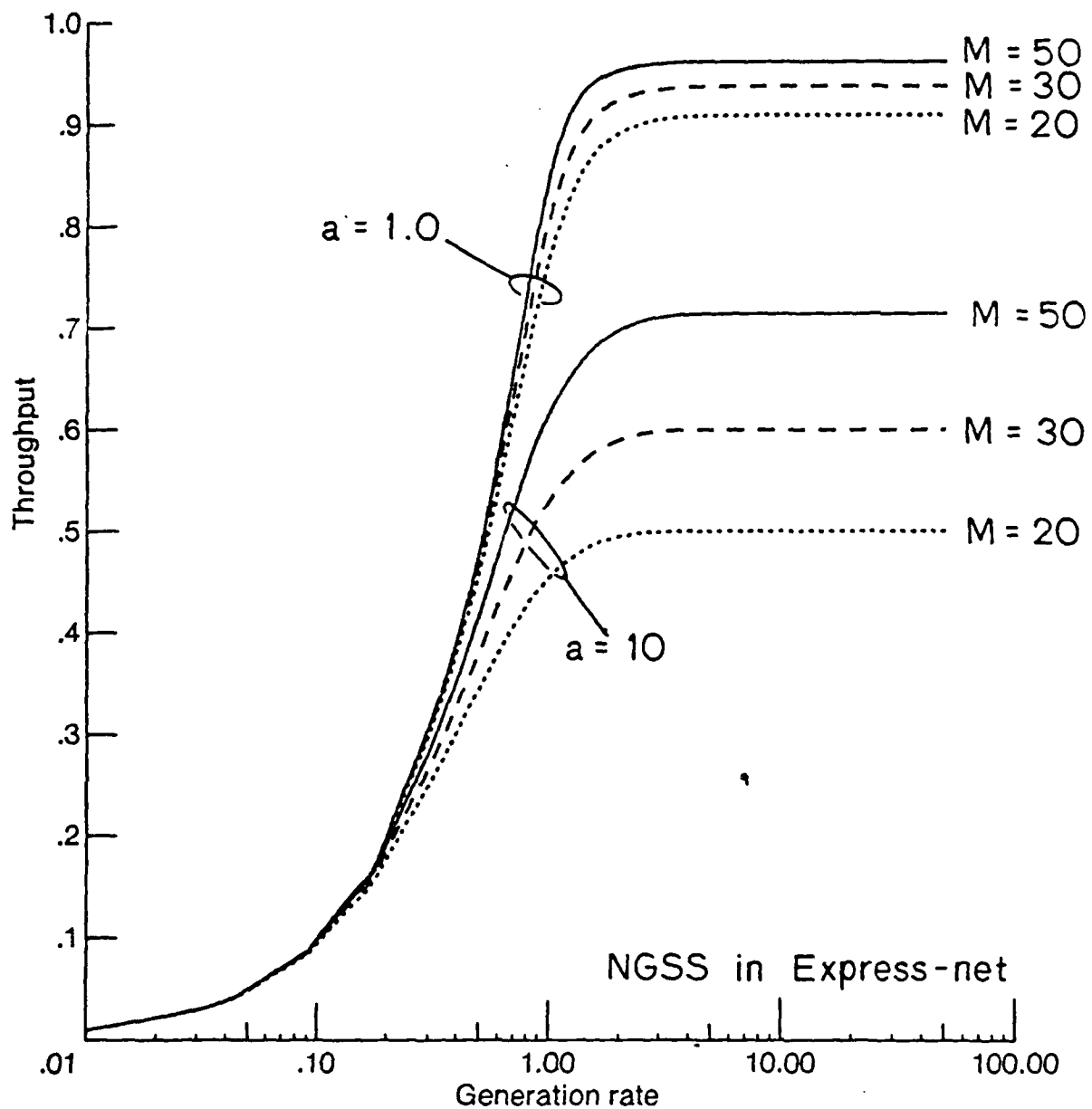


Fig. 3.1 Channel throughput as a function of the generation rate $M\lambda T$ for NGSS operating in Expressnet.

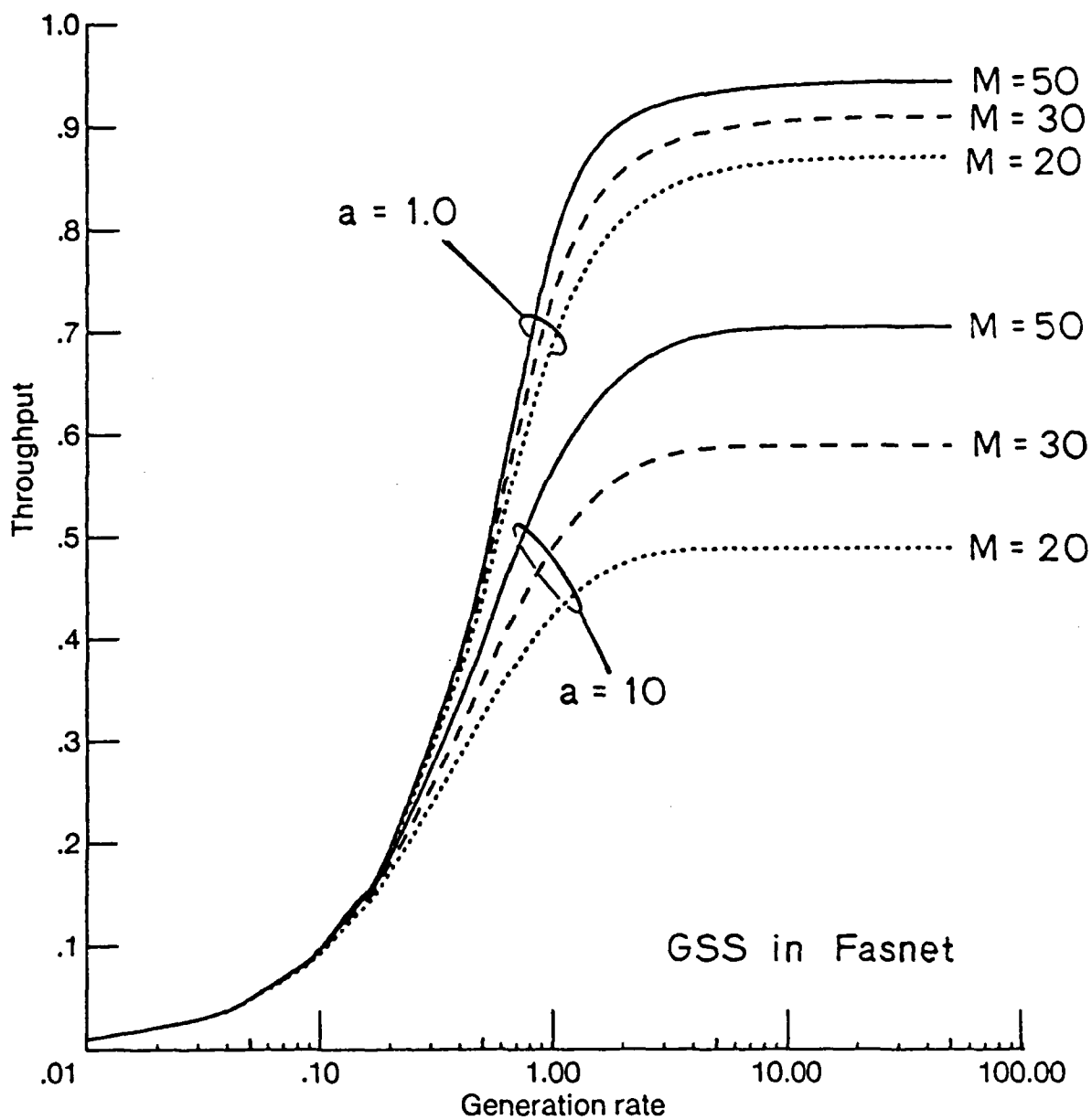


Fig. 3.2 Channel throughput as a function of the generation rate $M\lambda T$ for GSS operating in Fasnet.

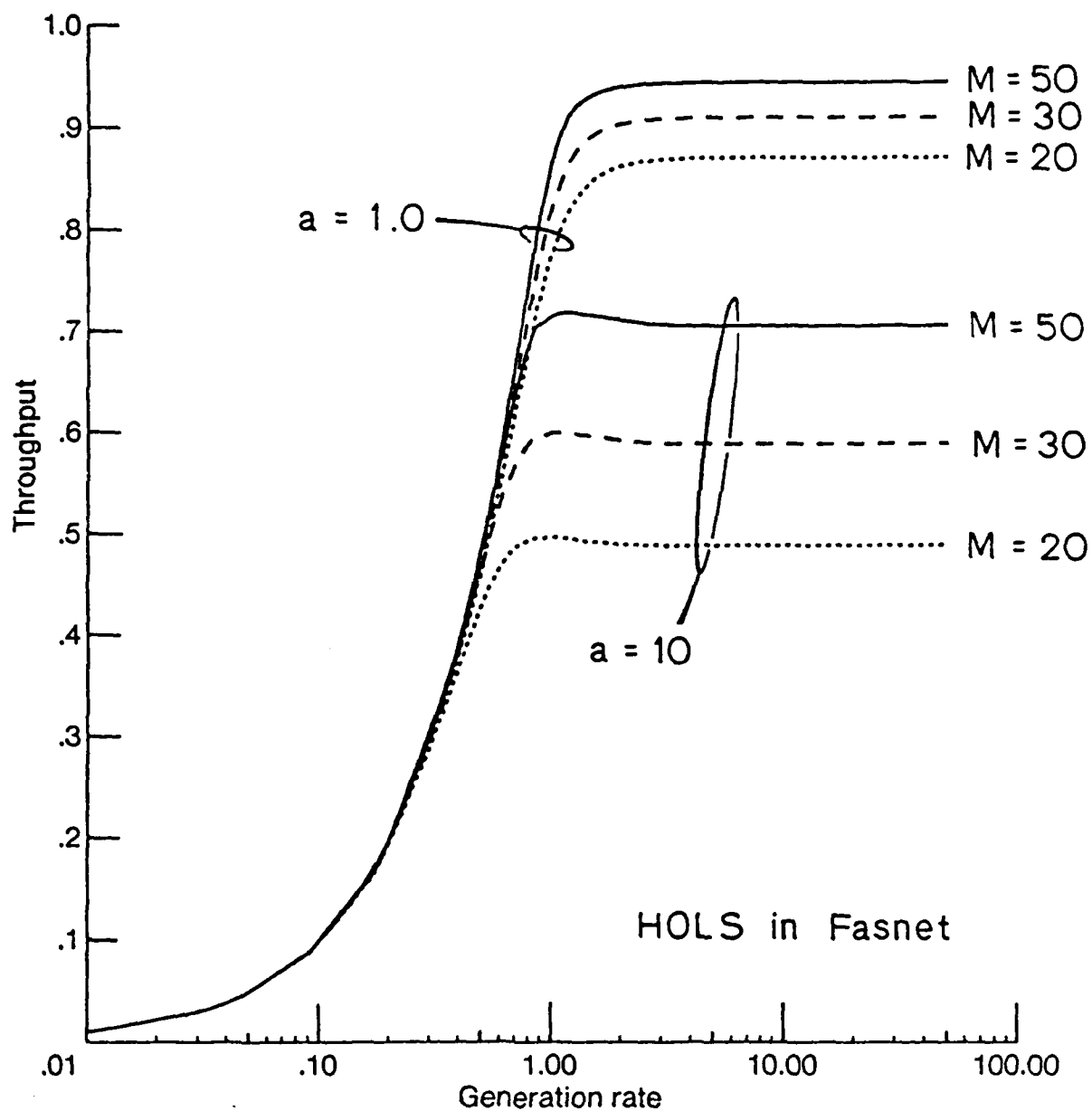


Fig. 3.3 Channel throughput as a function of the generation rate $M\lambda T$ for HOLS operating in Fasnet. The curves for $a = 10$ were obtained by simulation.

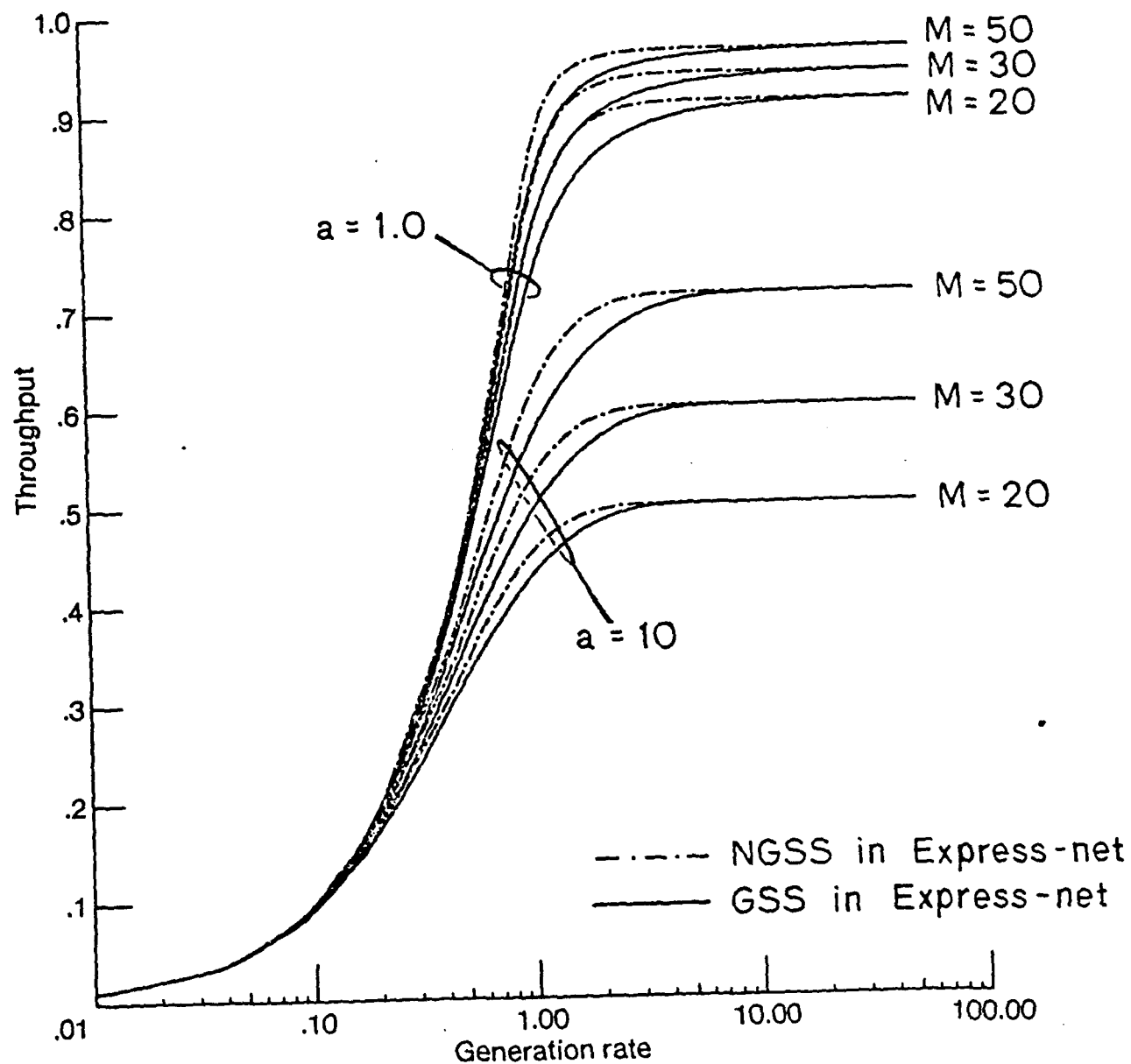


Fig. 3.4 Channel throughput as a function of the generation rate $M\lambda T$ showing the difference between NGSS and GSS operating in Expressnet.

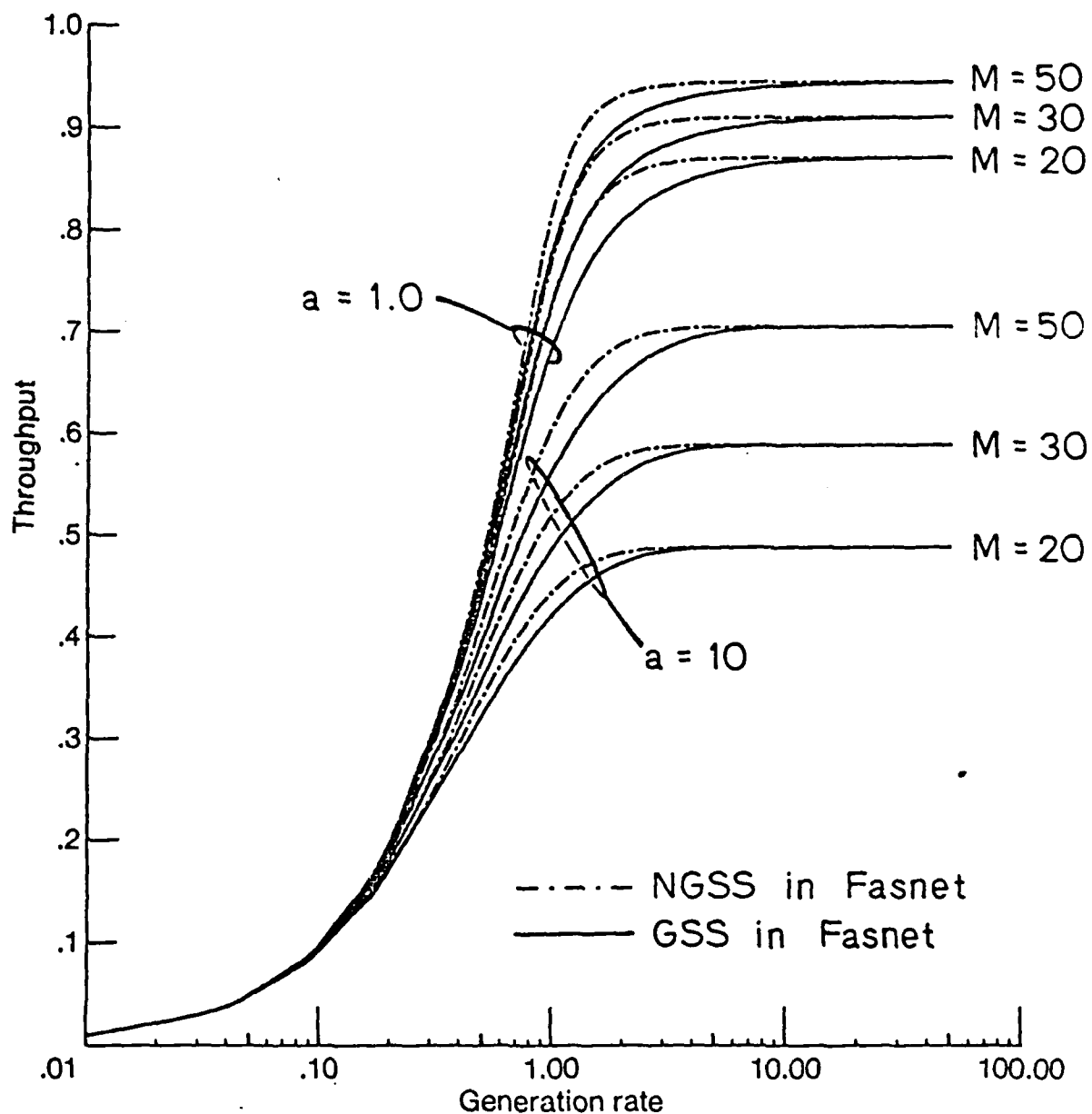


Fig. 3.5 Channel throughput as a function of the generation rate $M\lambda T$ showing the difference between NGSS and GSS operating in Fasnet.

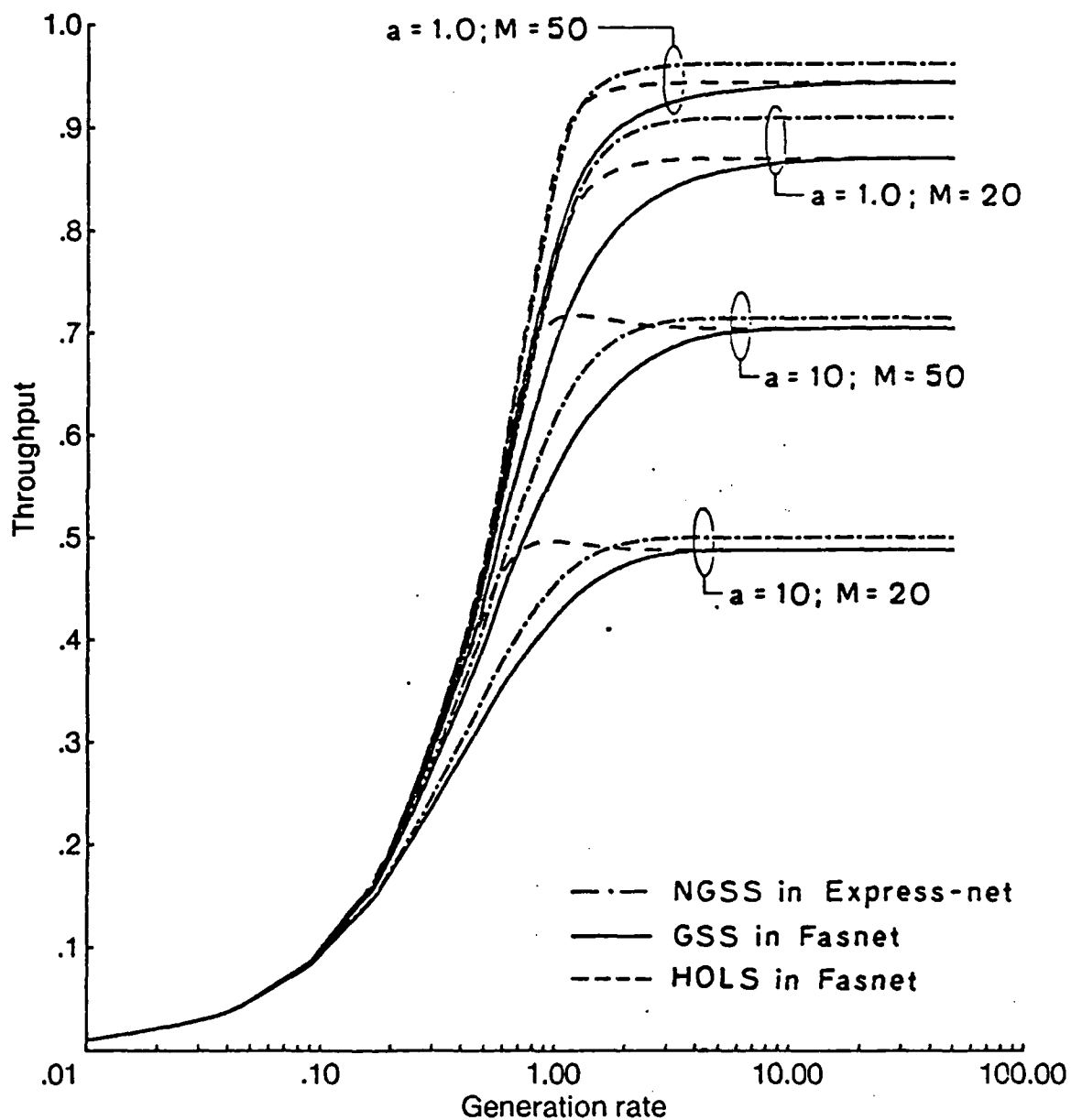


Fig. 3.6 Channel throughput as a function of the generation rate $M\lambda T$ comparing NGSS in Expressnet, GSS in Fasnet and HOLS in Fasnet. The curves showing HOLS with $a = 10$ were obtained by simulation.

the three service disciplines. As the network reaches saturation, S approaches a finite value, given by $MT/MX + Y$ independent of the service discipline, which corresponds to the value of the channel utilization $C(M, N)$ computed in chapter 2. We call this value of the channel utilization the network capacity. For NGSS and GSS the network capacity represents the maximum channel utilization. For HOLS, the maximum channel utilization is slightly higher than the network capacity for the reasons discussed above. The difference between the network capacity for NGSS and that for GSS and HOLS seen in Fig. 3.6 is a result of the different values of Y in Expressnet and Fasnet for the same value of a ($2a$ and $\lceil 2a \rceil + 1$, respectively); recall that the preamble Ω is assumed here to be zero.

The relationship between S and average delay D normalized to T for $a = 0.1$, 1.0 and 10, and $M = 20, 30$ and 50 is shown in Fig. 3.7 for NGSS in Expressnet, Fig. 3.8 for GSS in Fasnet, and in Fig. 3.9 for HOLS in Fasnet. We see that, for a given S , D is fairly insensitive to M as long as M is large enough so that this value of S can be achieved. We also see that the *normalized* average delay increases as a gets larger. However if a ($= \tau W/B$) has become larger because the channel bandwidth W has increased or the packet size B has decreased, meaning that T ($= B/W$) has decreased, then the *actual* delay is smaller than that obtained with small a ; the packet transmission time has decreased thus reducing the size of a slot and the length of a round.* On the other hand, if a has become larger because the size of the network has increased, meaning that T has not changed, then the actual delay will have increased as represented by the normalized delay. Although

*This can be exemplified with a special case. Consider a saturated system ($\lambda \rightarrow \infty$) with M stations and inter-round overhead Y . Suppose that, for a given bandwidth, the transmission time is T . The overhead $t_o + \Omega$ is assumed to be negligible. In this case the actual delay is $D = MT + Y$ and the normalized delay $(D/T) = M + Y/T$. Suppose now that the bandwidth of the system is increased meaning that a increases and T decreases. Let this new value of the transmission time be T' where $T' < T$. In this case the normalized packet delay (D'/T') is given by $(D'/T') = M + Y/T' > (D/T)$; i.e., the normalized delay has *increased*. The actual delay, however, is $D' = MT' + Y < D$; i.e., the actual delay has *decreased*.

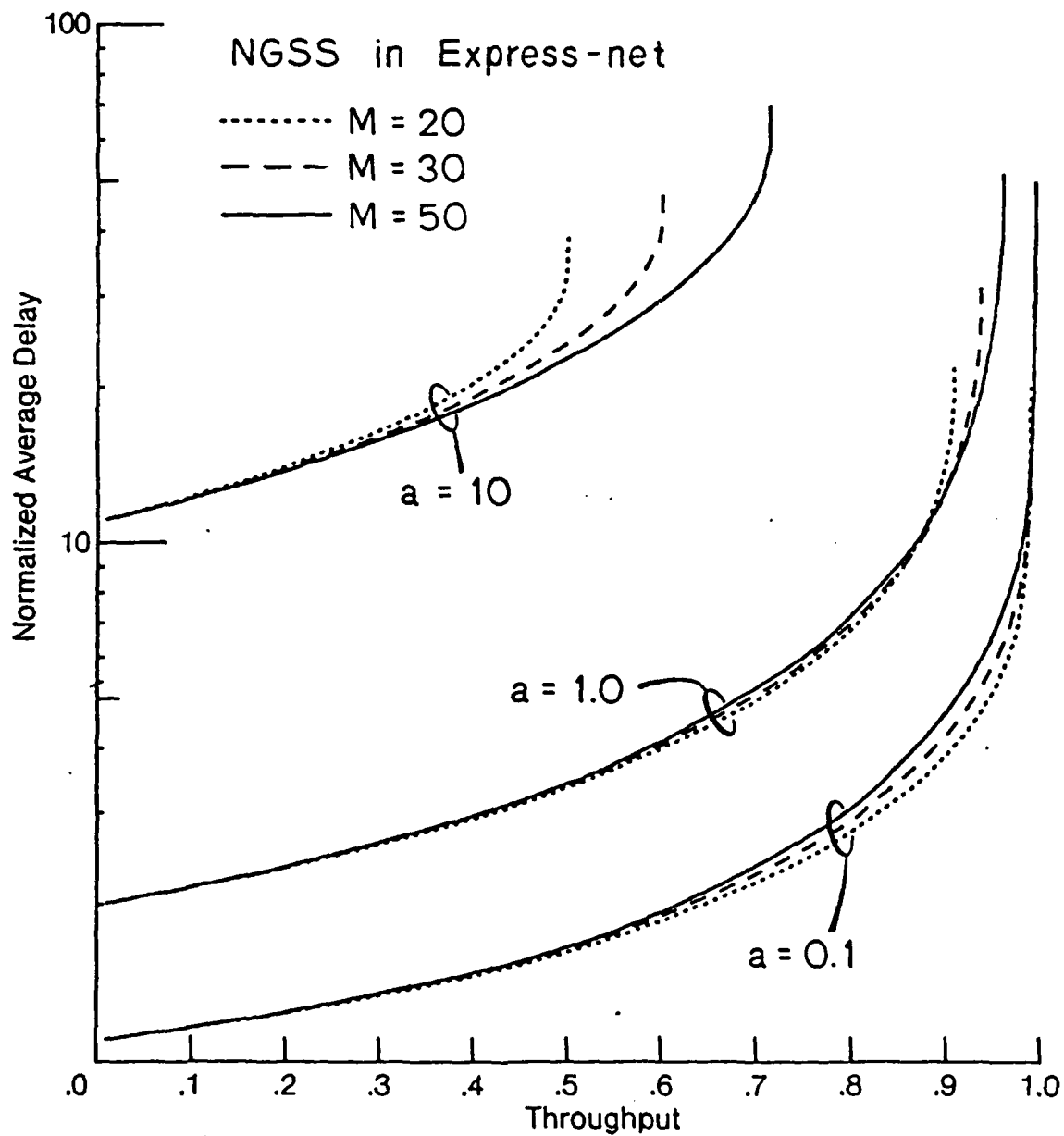


Fig. 3.7 Average delay normalized by the packet transmission time T versus the channel throughput for NGSS in Expressnet.

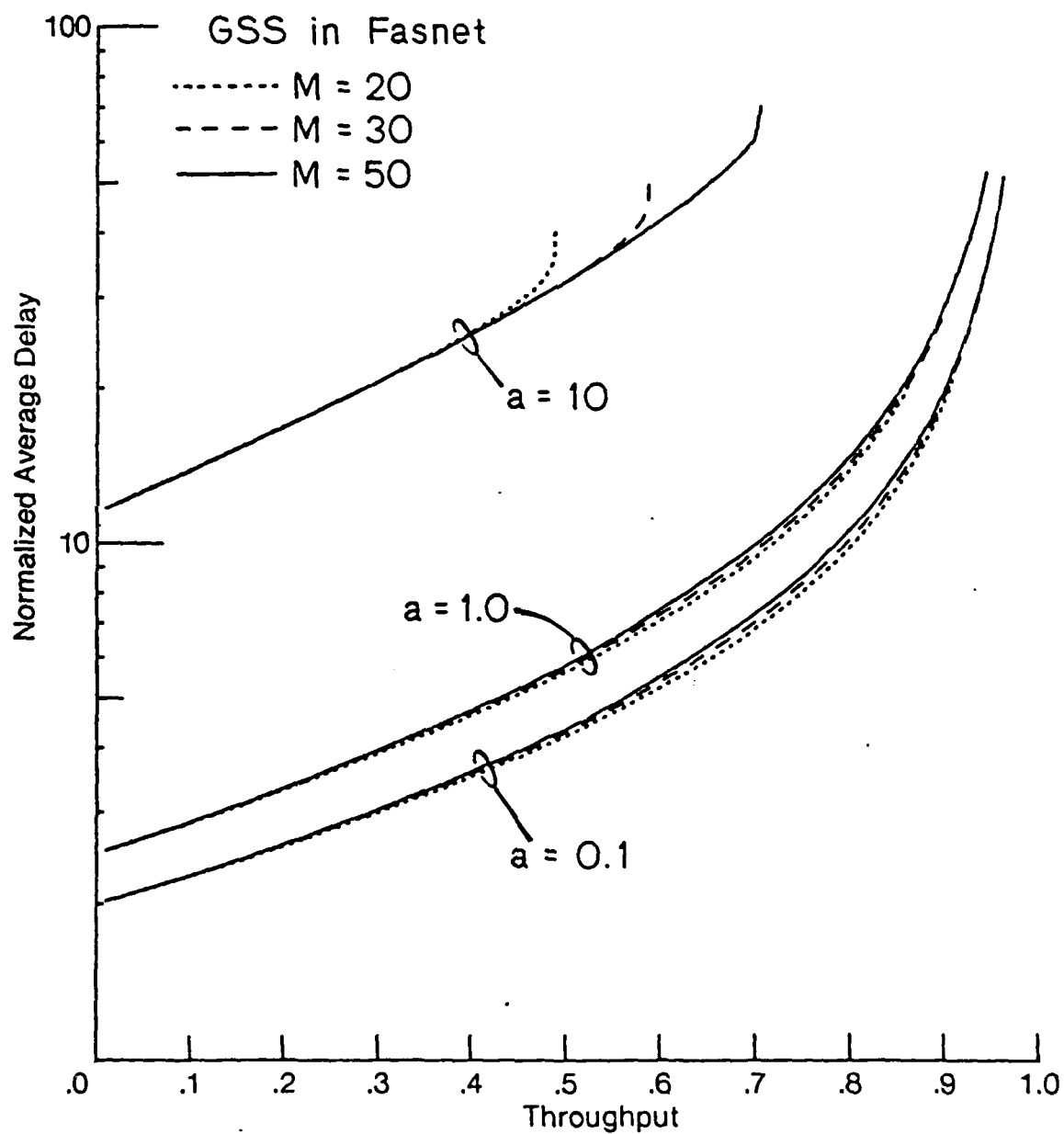


Fig. 3.8 Average delay normalized by the packet transmission time T versus the channel throughput for GSS in Fasnet.

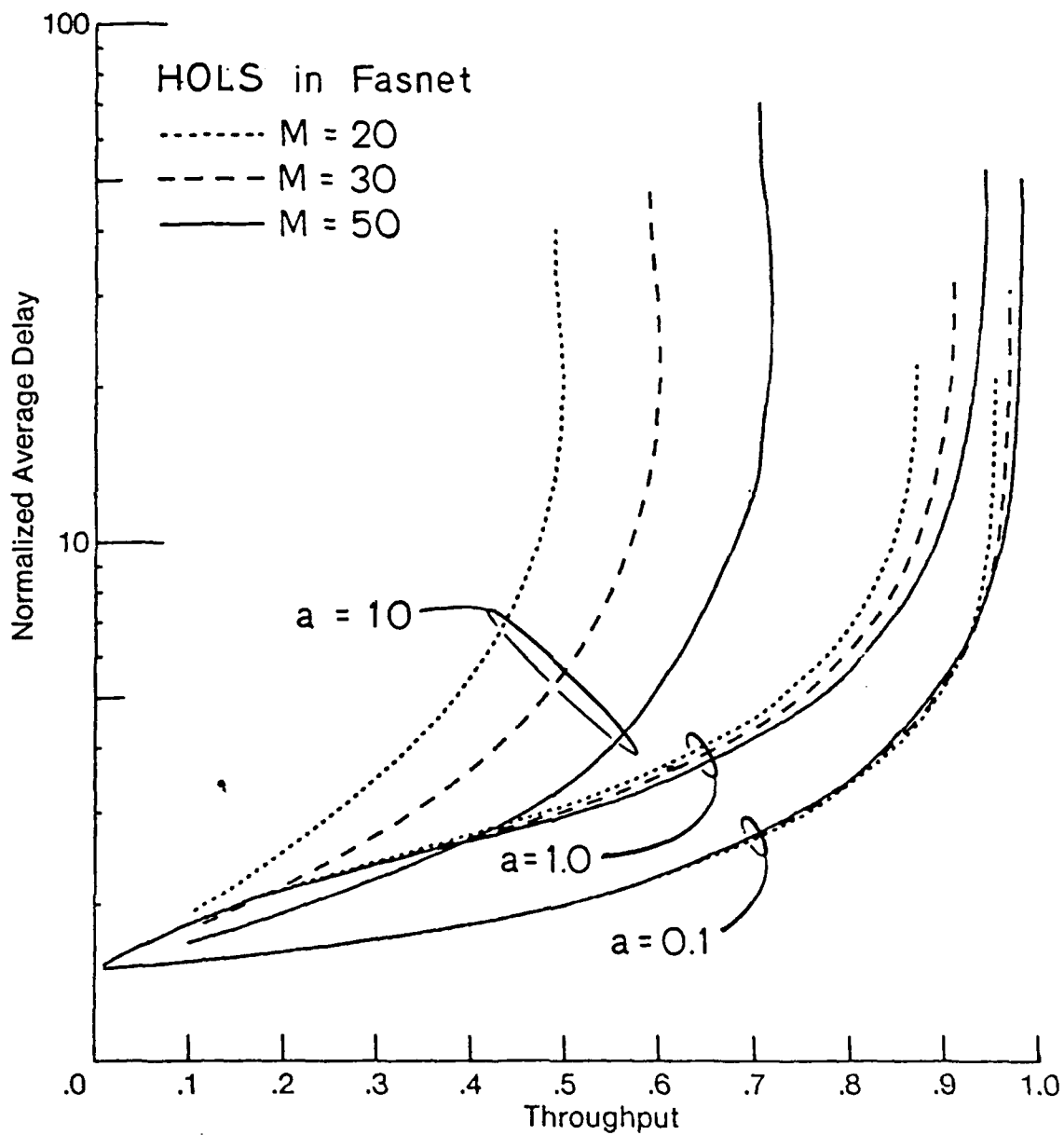


Fig. 3.9 Average delay normalized by the packet transmission time T versus the channel throughput for HOLS in Fasnet.

the performance trends for all three disciplines are similar, the results do show some differences. These are highlighted in Figs. 3.10 and 3.11. We plot on a single sheet the throughput-delay trade-off for each of the three disciplines for $a = 0.1$ in Fig. 3.10, and for $a = 1.0$ and 10 in Fig. 3.11. In particular one should note that, for large a ($a = 10$), HOLS achieves substantially lower delay than the other two schemes as long as the throughput is not close to saturation. This is due to the fact that in HOLS, having generated a packet, a non-dormant station transmits this packet in the next available slot regardless of when the start of cycle appears. In particular, at $S = 0$, D will be equal to $1.5T$ since a station can transmit its packet in the slot immediately following the one in which it was generated, instead of incurring on the average a delay of aT while waiting for the SOC as in GSS, or the locomotive as in NGSS. If a is made large because W is increased then T decreases and so, in absolute terms, the delay aT remains constant and equal to τ . Also in Fig. 3.10 is plotted the relationship between S and D for CSMA/CD.* As with Expressnet, we assume that the preamble for CSMA/CD is negligible. This figure shows how favorably the delay performance of the round robin schemes compares to that of CSMA/CD. No throughput-delay curves are plotted for CSMA in Fig. 3.11 since for $a = 1.0$ and 10 this access scheme achieves a very small network capacity.

Operating Expressnet in GSS mode results in a throughput-delay trade-off similar to that of GSS in Fasnet. However, for GSS in Expressnet, there is a small improvement in the throughput-delay characteristics as compared to GSS in Fasnet, since in Expressnet the inter-round overhead is smaller than that of Fasnet for a given a . We show in Fig. 3.12 the average packet delay as a function of the throughput for both Expressnet and Fasnet operating under the GSS discipline. On the other hand, one may operate Fasnet under the NGSS discipline and achieve a throughput-delay trade-off similar to that shown in Fig. 3.7 for NGSS in Expressnet.

*The CSMA/CD scheme considered here is the slotted non-persistent version analysed in [9].

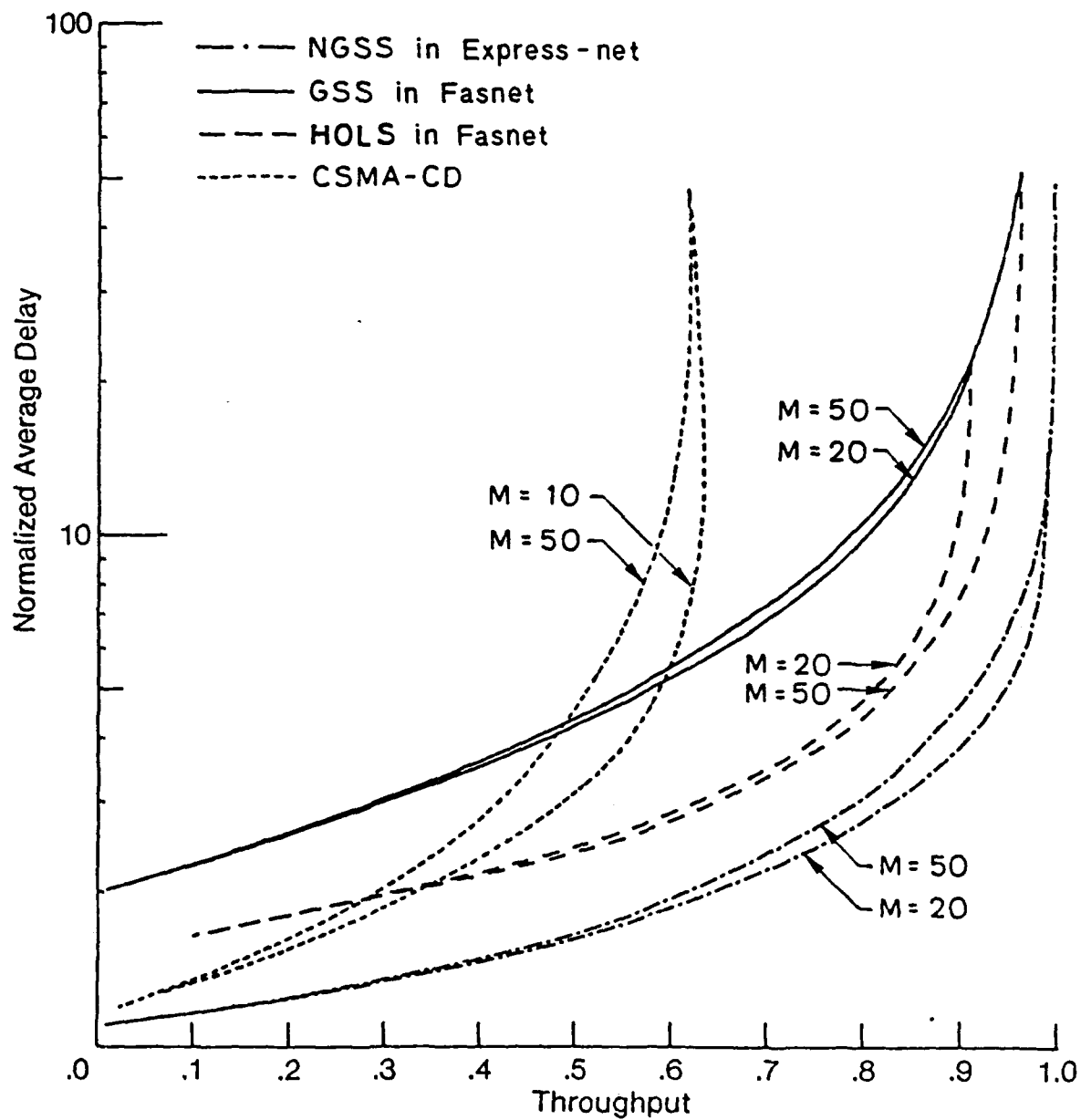


Fig. 3.10 Average delay normalized by the packet transmission time T versus the channel throughput with $a \approx 0.1$, comparing NGSS, GSS, HOLs and CSMA/CD.

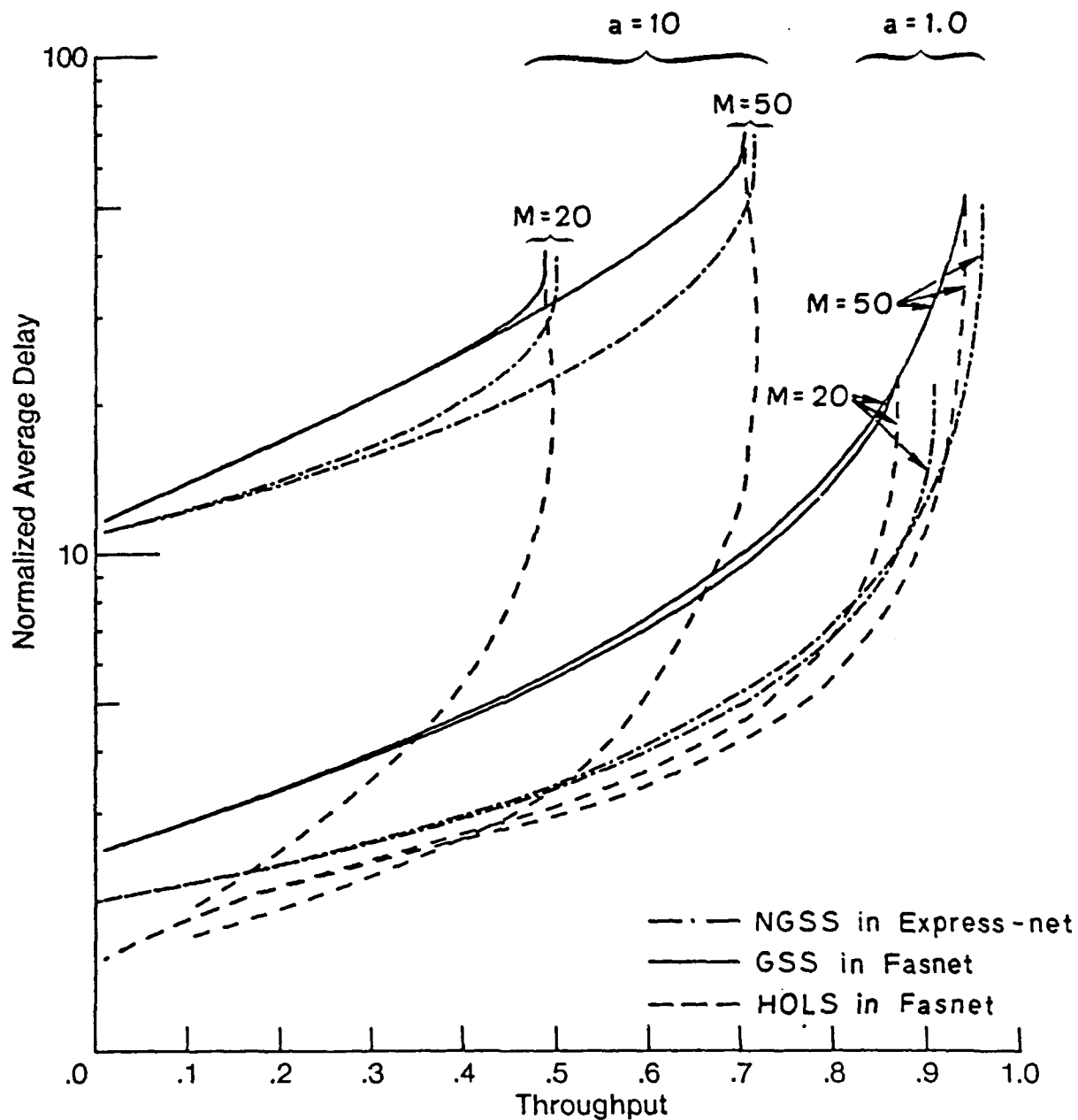


Fig. 3.11 Average delay normalized by the packet transmission time T versus the channel throughput with $a = 1.0$ and 10 comparing NGSS, GSS and HOLS. The curves for HOLS with $a = 10$ were obtained by simulation. All the other curves were obtained from the analysis.

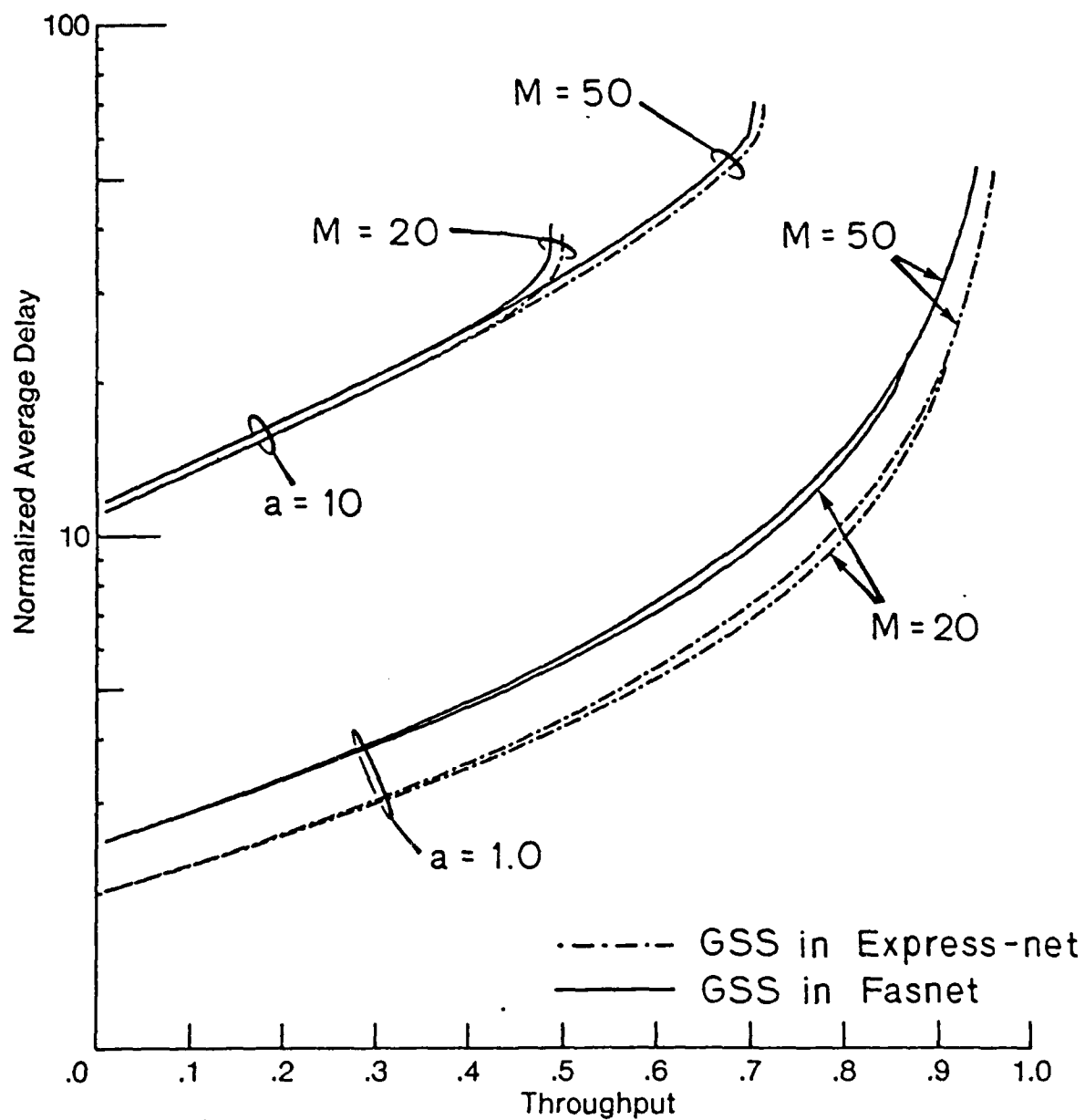


Fig. 3.12 Average delay normalized by the packet transmission time T versus the channel throughput for GSS in Expressnet as compared to GSS in Fasnet.

In this case however, there is a slight degradation in the throughput-delay characteristics as compared with NGSS in Expressnet as shown in Fig. 3.13 where we plot the average packet delay as a function of the channel throughput for both Expressnet and Fasnet operating under the NGSS discipline.

The average packet delay (normalized to T) versus S for NGSS in Expressnet, in the case where the preamble is not negligible, is plotted in Fig. 3.14 for $\alpha = 1$ and 10 and $M = 50$. As expected, the delay for a given S increases as the length of the preamble increases.

The results presented above were obtained from the mean value analysis and represent the average performance over all stations. For Expressnet with the NGSS discipline, service is offered to each station when it sees the EOT (either on the outbound or inbound channel). Therefore the EOT can be viewed as an implicit token passed from each station to the next in sequence. Due to the symmetry of this organization, the system is fair and all stations achieve the same performance. In GSS and HOLS on the other hand, the synchronizing event is the beginning of a slot which always sweeps the channel from the most upstream station to the most downstream station. As will be seen in the results discussed below, this mode of operation favors the upstream stations by giving channel access to the most upstream of all the stations contending for a given slot. In the distribution of delay analysis of GSS and HOLS we derived the performance achieved by each station. This enables us to determine the extent to which this performance is affected by the station's location on the network.

First we consider GSS. In Figs. 3.15, 3.16 and 3.17 we show M times the throughput achieved by the most upstream station and the most downstream station as a function of $M\lambda$, for various values of M and α . We refer to these two stations as station 1 and station M respectively. The curves show that initially, S increases as

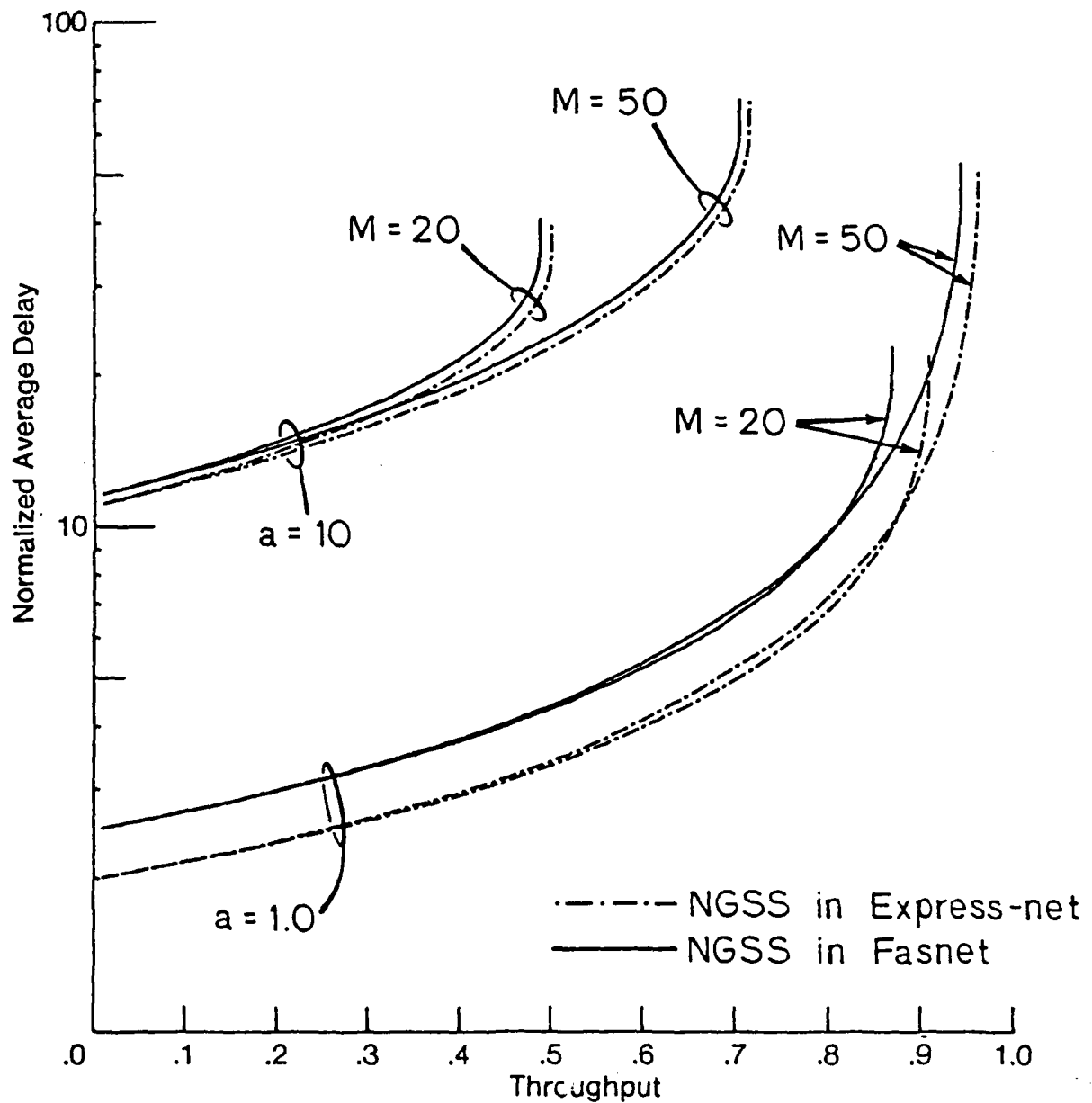


Fig. 3.13 Average delay normalized by the packet transmission time T versus the channel throughput for NGSS in Fasnet as compared to NGSS in Expressnet.

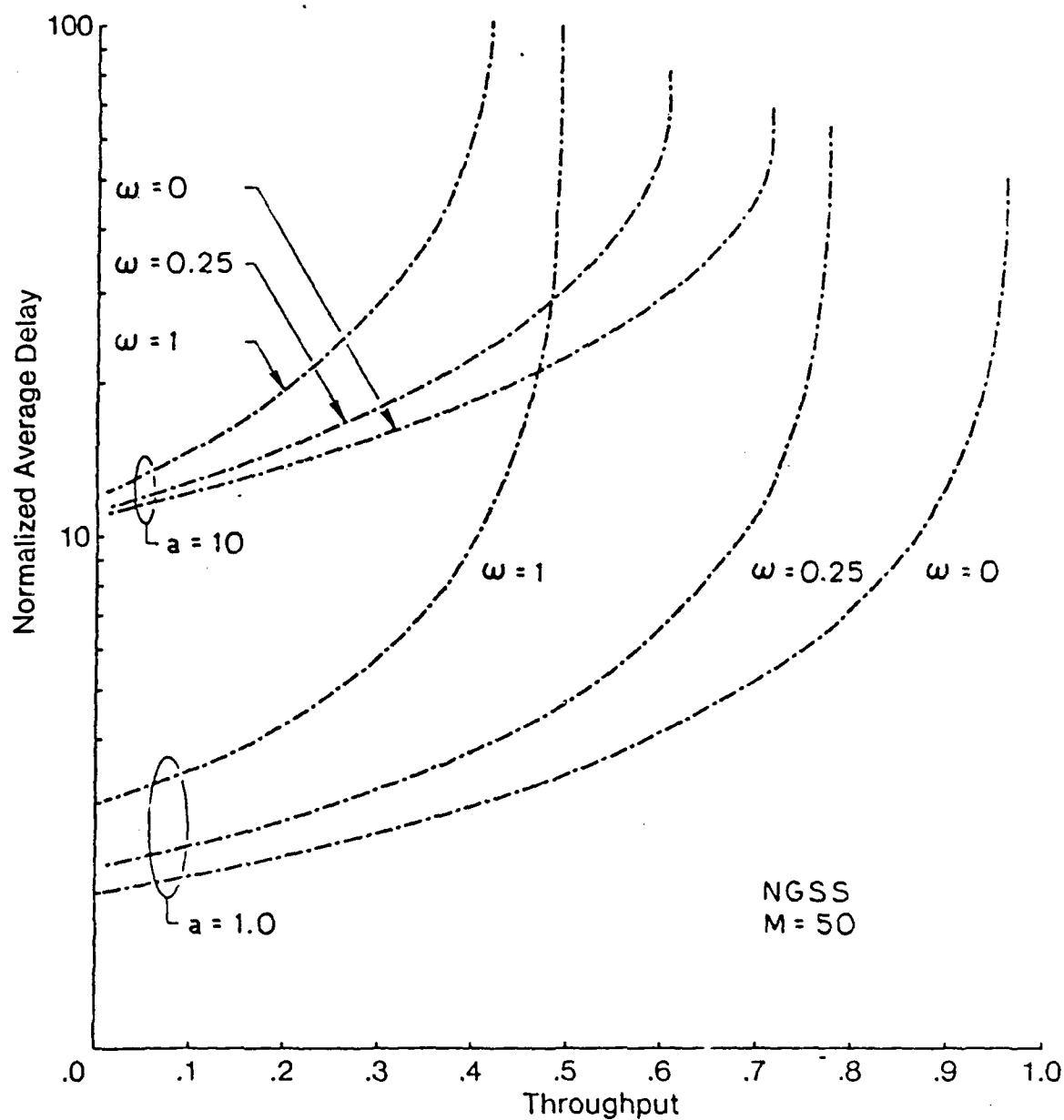


Fig. 3.14 Average delay normalized by T versus the channel throughput showing the effect of the preamble on the throughput-delay performance of NGSS in Expressnet.

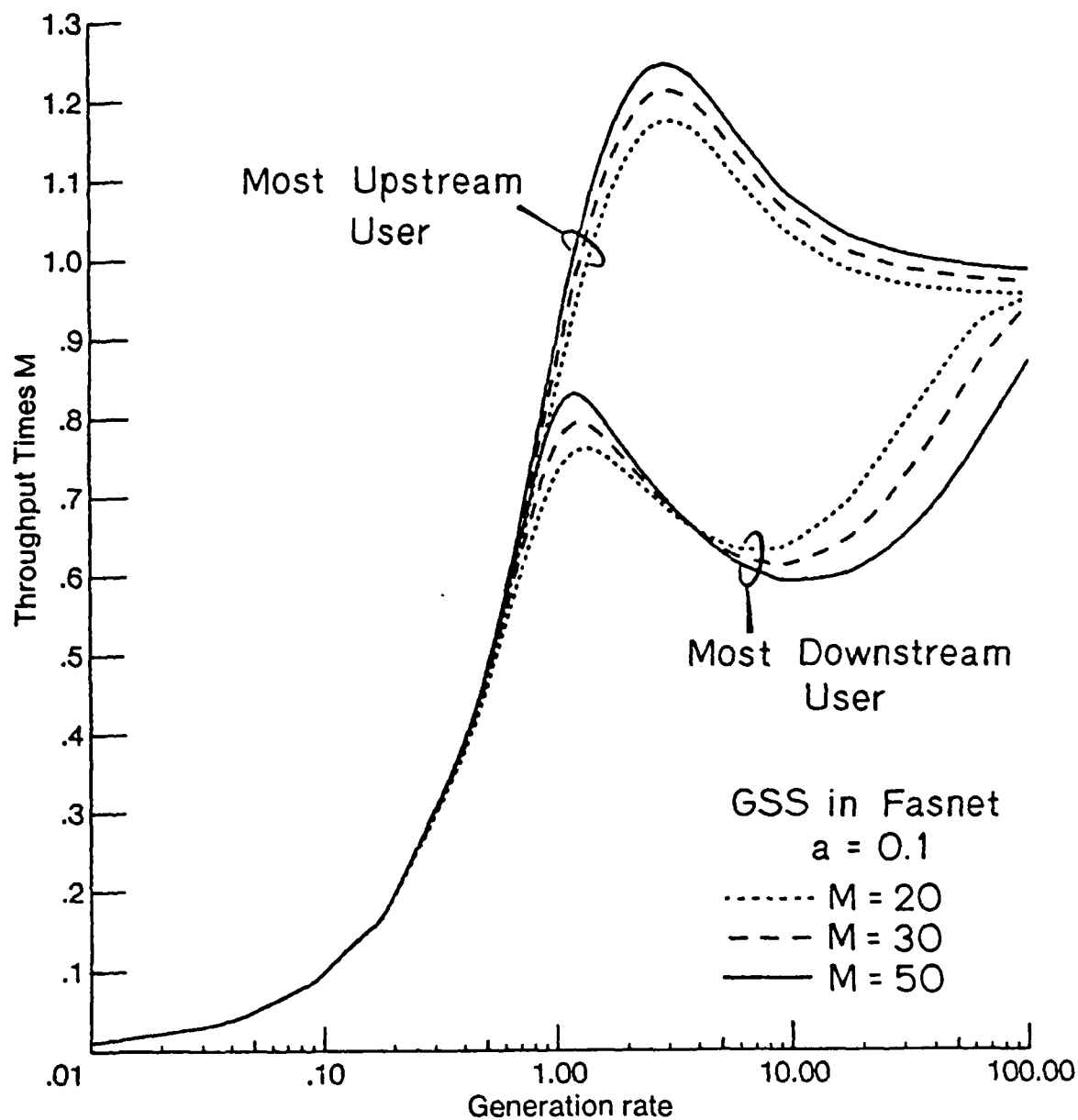


Fig. 3.15 Throughput multiplied by M versus the generation rate $M\lambda T$ for GSS in Fasnet as achieved by the most upstream station and the most downstream station for $a = 0.1$.

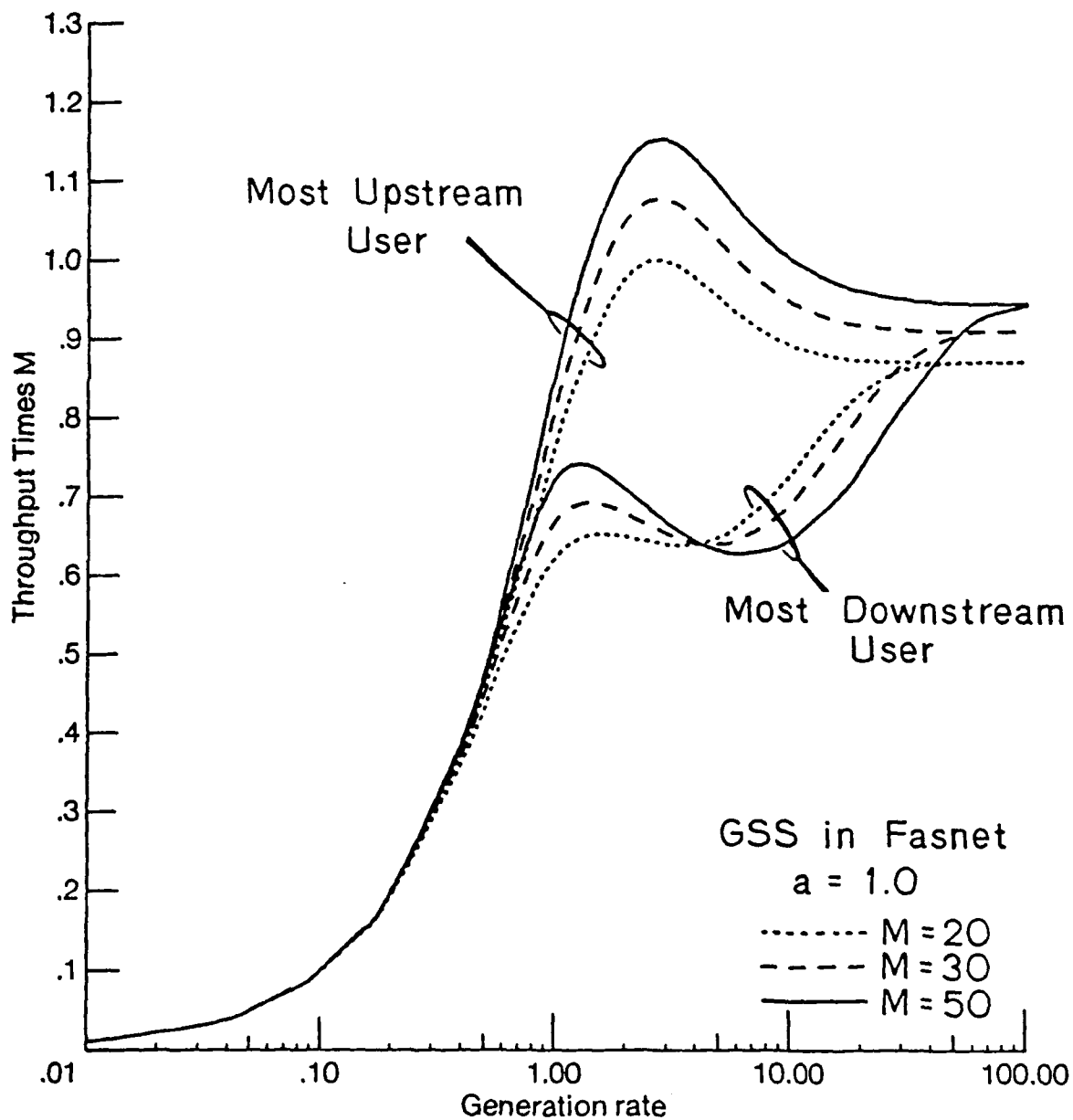


Fig. 3.16 Throughput multiplied by M versus the generation rate $M\lambda T$ for GSS in Fasnet as achieved by the most upstream station and the most downstream station for $a = 1.0$.

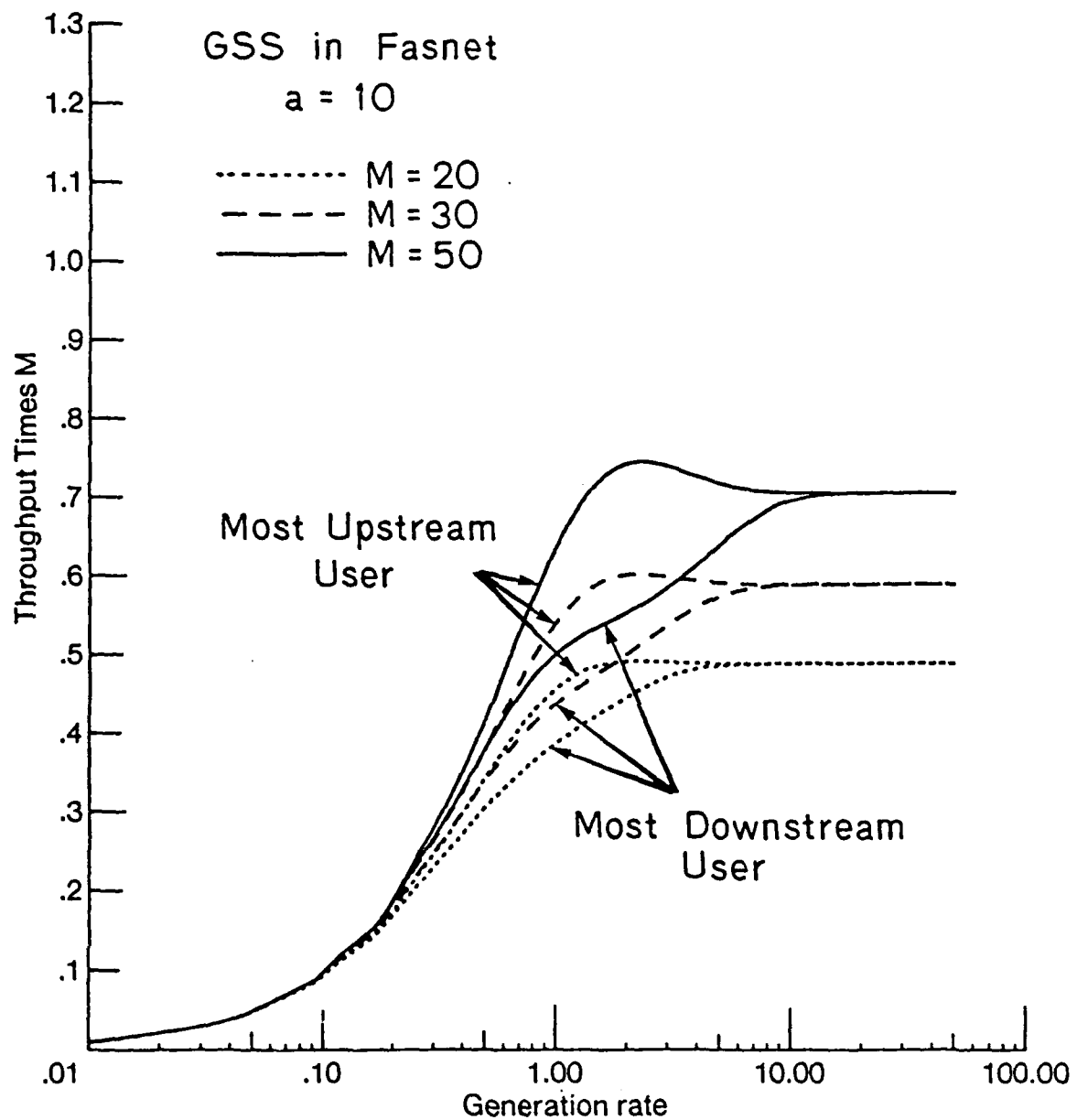


Fig. 3.17 Throughput multiplied by M versus the generation rate $M\lambda T$ for GSS in Fasnet as achieved by the most upstream station and the most downstream station for $a = 10$.

$M\lambda$ increases from zero. At low loads there are long idle periods between packet generations, rounds are short, and the throughput achieved by each station is not sensitive to its position on the network. As the network capacity is approached, we see that station 1 achieves a significantly higher throughput than station M . This occurs since in any given round, station 1, having transmitted its packet at the beginning of the round, has the remainder of that round in which to generate a new packet before the next SOC. Station M on the other hand, having transmitted at the end of the round, has only the inter-round overhead period before the next SOC in which to generate its new packet and thus is less likely to be ready at the beginning of the next cycle. As M increases the difference in S between these two stations increases. For large values of a this difference is not as pronounced as for small a due to the fact that the inter-round overhead becomes the dominant factor affecting the performance results and its effect is the same on all stations. Finally, as $M\lambda \rightarrow \infty$, station M will generate a new packet during the inter-round gap with probability 1 assuming that $a > 0$; hence station 1 and station M will achieve the same throughput which is given by the network capacity divided by M . In the limiting case where $a = 0$, station M , having transmitted at the end of a given round, will be ready at the beginning of the next round with probability 0; in particular, at $\lambda = \infty$, station M will transmit once in every two rounds and achieve a throughput of only half that achieved by the other stations. The throughput achieved by any of the other stations lies within the bounds of station 1 and station M . In fact, any given station will achieve a throughput which is greater than any station downstream from it and less than any station upstream from it. The throughput times M for each station in a network with $a = 1.0$ and $M = 10$ is shown in Fig. 3.18. Recall that each station has only a single buffer. If however, a multiple packet buffer is provided then a station could generate additional packets for transmission before transmitting the one at the front of the queue. This would

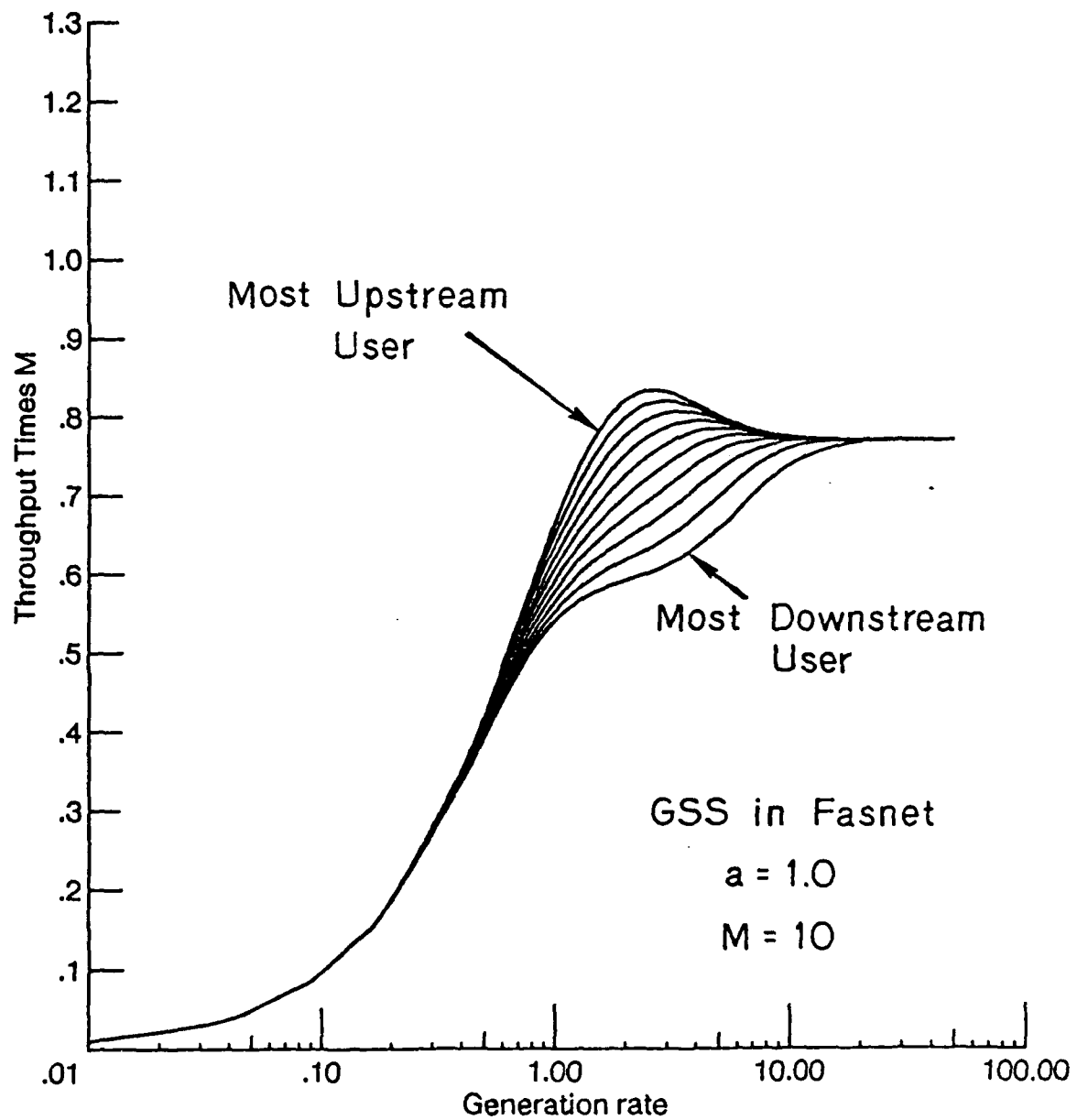


Fig. 3.18 Throughput multiplied by M versus the generation rate $M\lambda T$ for GSS in Fasnet with $a = 1$ and $M = 10$ as achieved by each station on the network.

reduce the extent of the unfairness suffered by those stations on the downstream side of the network. In the limiting case where each station had an infinite buffer, all of the stations would achieve the same throughput assuming that they were all generating packets at the same rate.

For the HOLS discipline, the throughput is plotted as a function of $M\lambda$ for various values of α and M in Figs. 3.19, 3.20 and 3.21, and exhibits the same characteristics as for GSS. Note that for HOLS there is not as much of a discrepancy in the service achieved by the individual stations.

It is important to point out that for Fasnet, there are two separate channels on which stations can transmit data. In the analysis we consider only one of these and, in addition, we assume that all stations are generating packets for this channel at the same rate. In actual fact the downstream stations on a given channel will most probably require a lower throughput on this channel than the upstream ones since they will be transmitting mostly on the other channel. In fact, the most downstream station on a given channel will not transmit any packets on that channel since there is nobody further downstream to receive it.

The difference in average delay between the most upstream and the most downstream stations in Fasnet is shown for GSS in Fig. 3.22, and for HOLS in Fig. 3.23. Since in a given round station 1 is serviced before station M , it achieves a lower delay for a given S . It is interesting to note that in GSS the delay of station 1 is bounded from above by the maximum length of a cycle which is $MX + Y$. For station M the delay is bounded by twice the maximum length of a round plus an inter-round overhead period, that is $(2M - 1)X + Y$ (the value given in chapter 2 for the bound on the packet delay in Fasnet), even though at saturation ($\lambda \rightarrow \infty$) the delay will be $MX + Y$. In HOLS and also NGSS the delay of a packet from any station is always bounded by $MT + Y$. If Expressnet were to be operated under

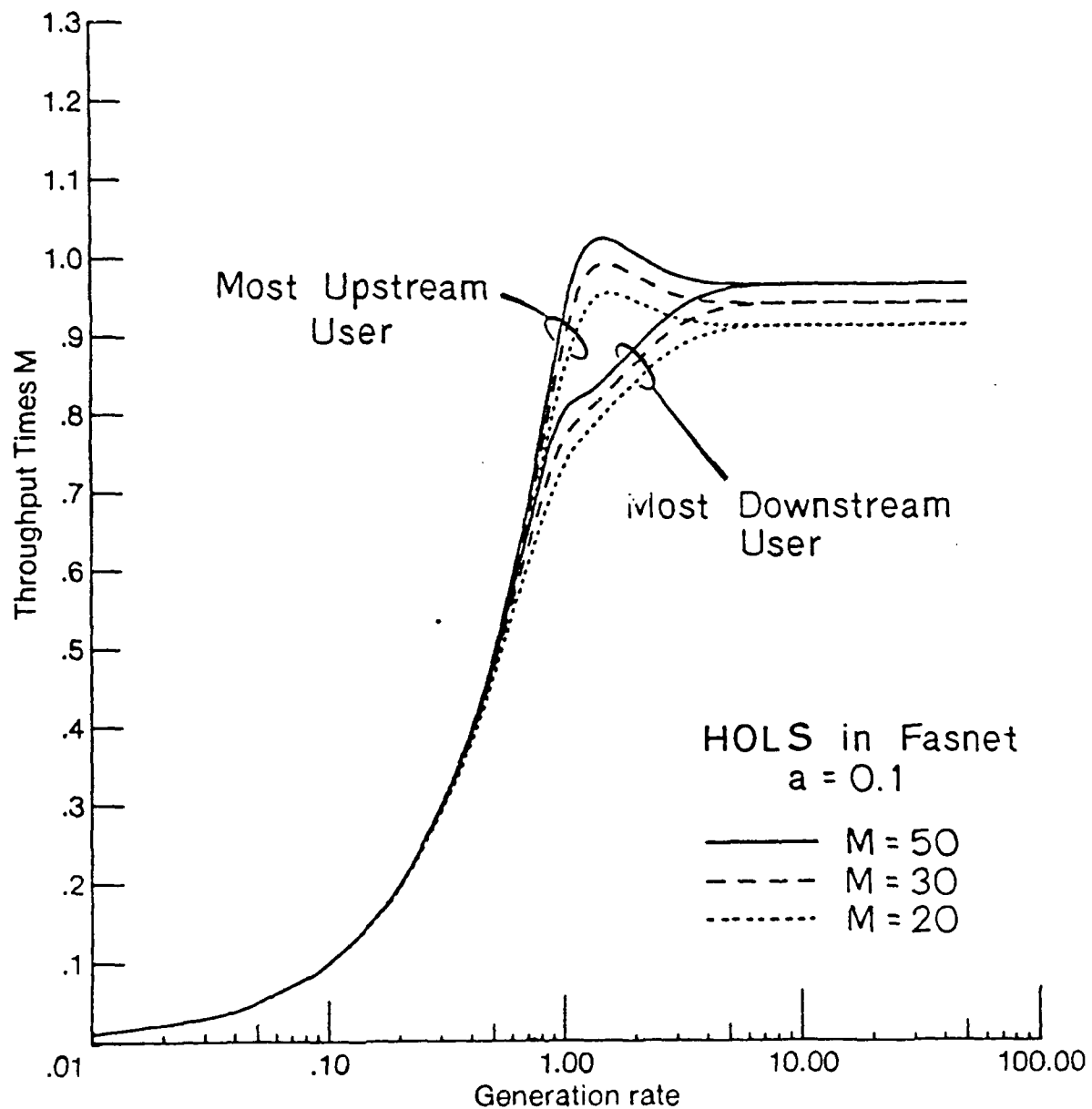


Fig. 3.19 Throughput multiplied by M versus the generation rate $M\lambda T$ for HOLS in Fasnet as achieved by the most upstream station and the most downstream station for $a = 0.1$.

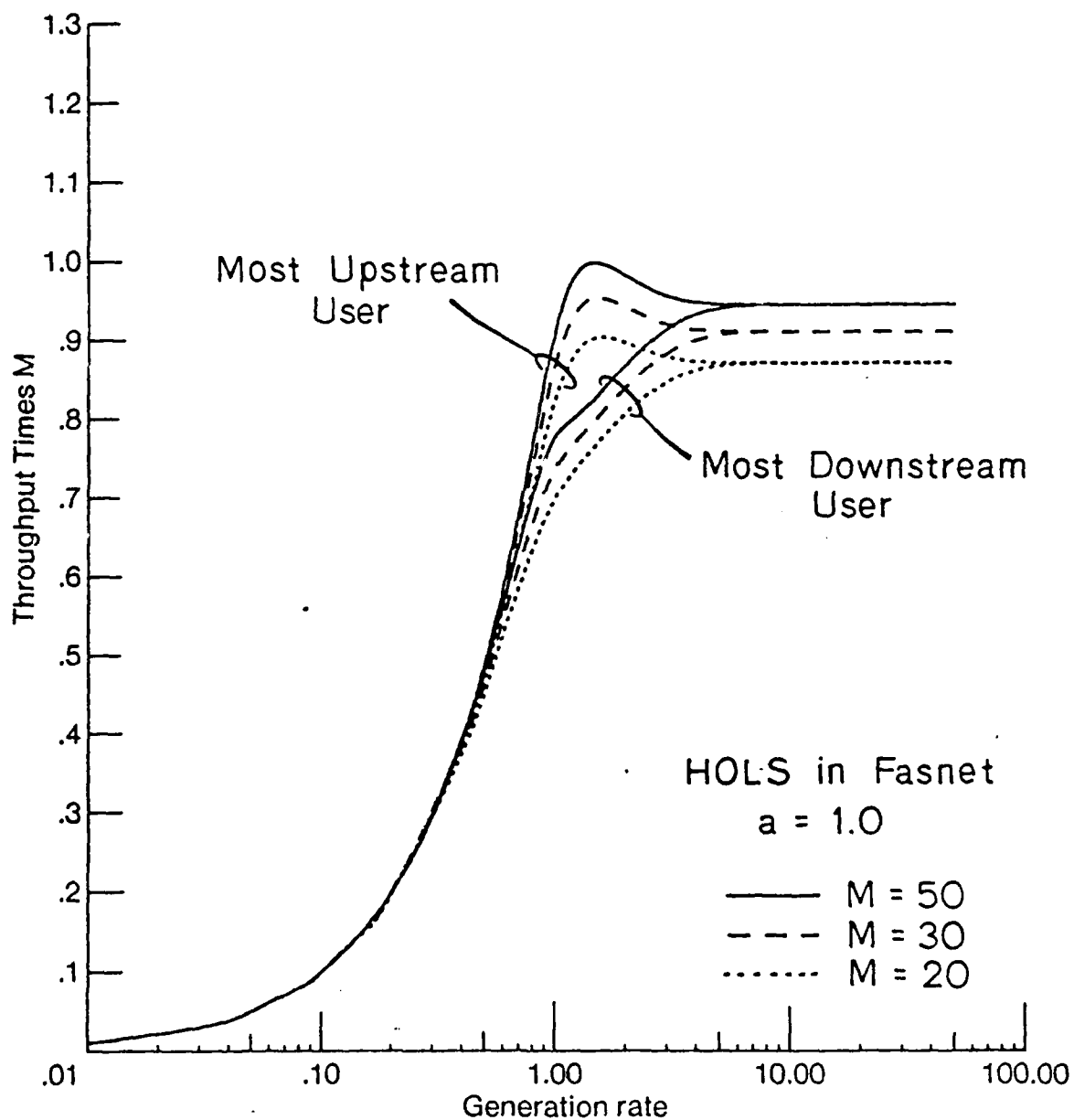


Fig. 3.20 Throughput multiplied by M versus the generation rate $M\lambda T$ for HOLS in Fasnet as achieved by the most upstream station and the most downstream station for $a = 1.0$.

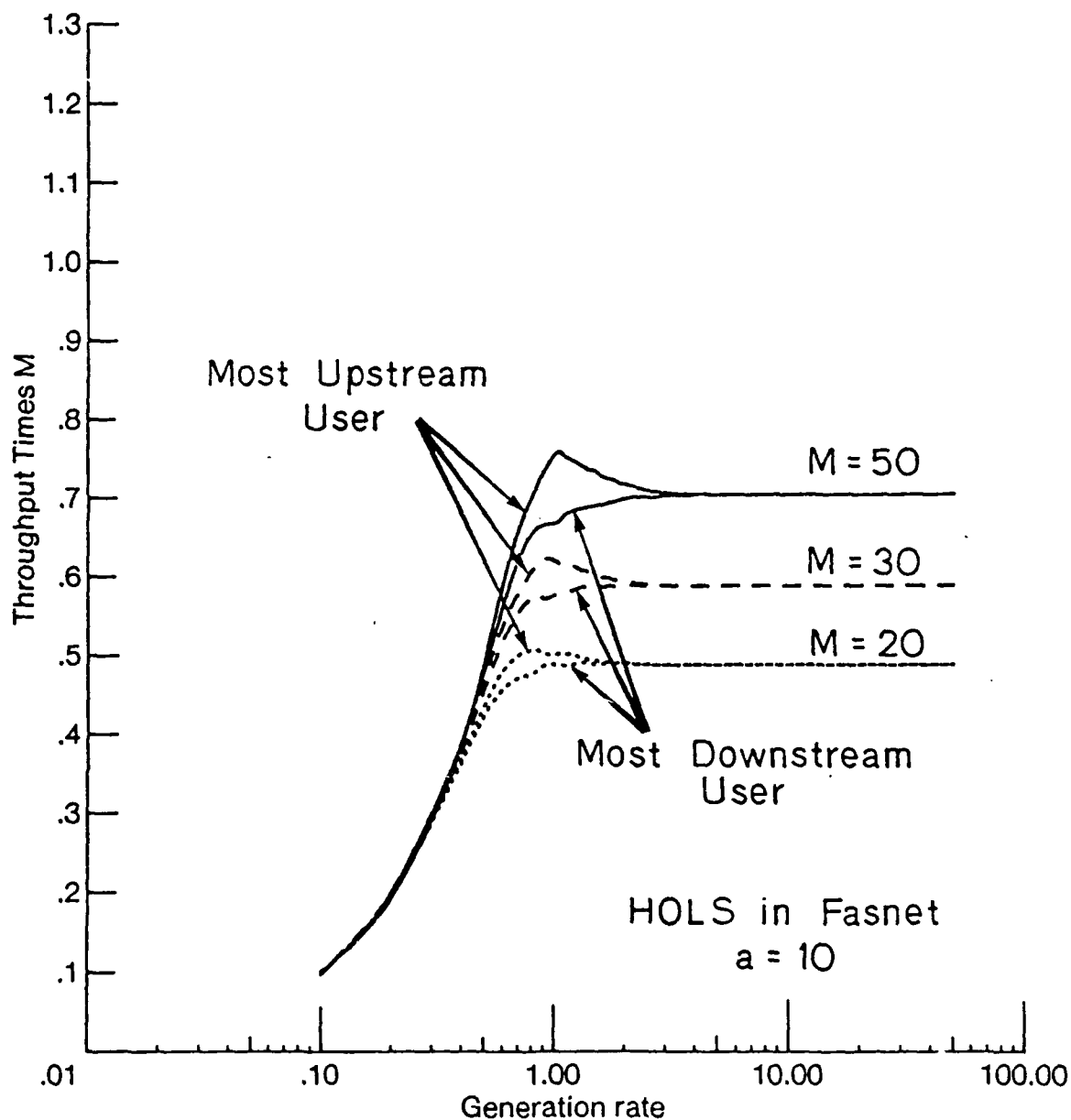


Fig. 3.21 Throughput multiplied by M versus the generation rate $M\lambda T$ for HOLS in Fasnet as achieved by the most upstream station and the most downstream station for $a = 10$. These curves were obtained by simulation.

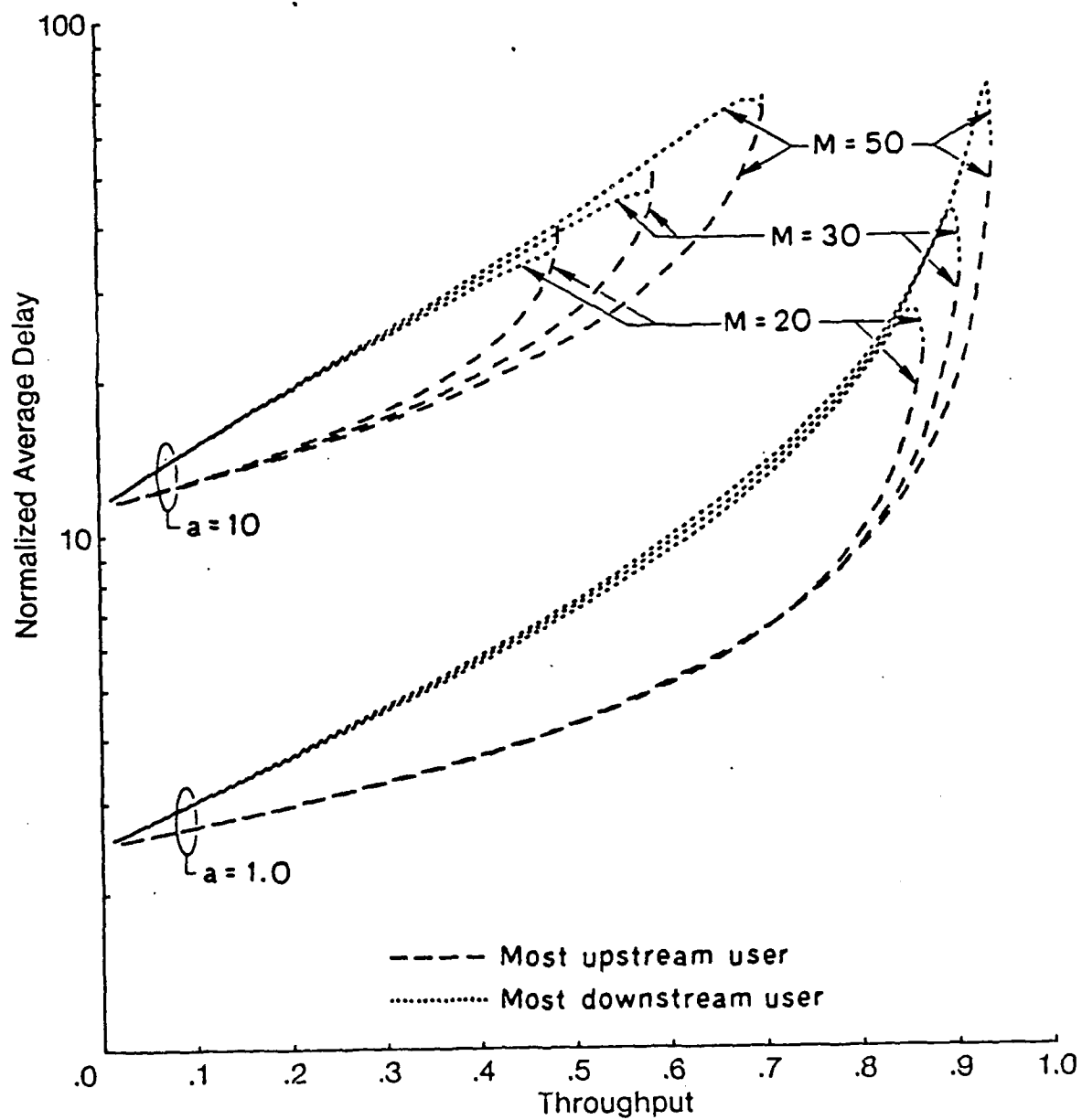


Fig. 3.22 Normalized average delay versus the throughput for GSS in Fasnet as achieved by the most upstream station and the most downstream station.

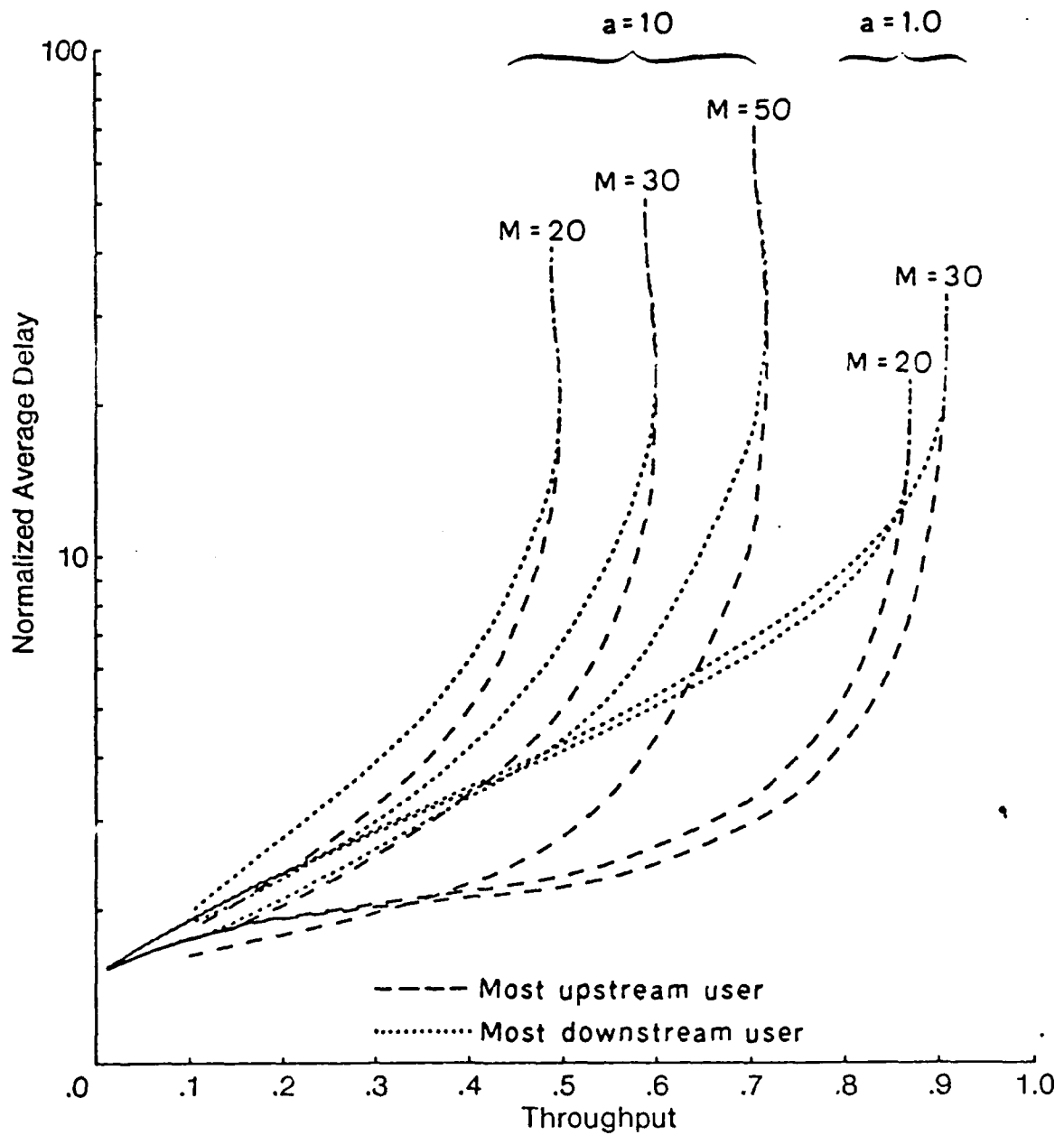


Fig. 3.23 Normalized average delay versus throughput for HOLS in Fasnet as is achieved by the most upstream station and the most downstream station.

the GSS discipline then, in this system too, the average delay achieved by any station would be dependent on that station's location on the network. There would however be a slight improvement in the overall throughput-delay characteristics as compared to the throughput-delay characteristics achieved in Fasnet under GSS, due to the smaller inter-round overhead for a given α in Expressnet. In Fig. 3.24 we plot the throughput-delay trade-off as achieved by station 1 and station M in both Expressnet and Fasnet operating under the GSS discipline, which shows the similarity of their characteristics.

Finally, we examine the variance of delay. The relationship between the variance and the throughput for each of the three service disciplines is shown in Figs. 3.25, 3.26, and 3.27 for $\alpha = 1.0$ and 10 and for various values of M . For GSS and HOLS we show the variance of delay versus S as achieved by station 1 and station M . Since for NGSS all stations achieve the same performance, we show the variance versus S as achieved by any station on the network. For $S = 0$ the variance is non-zero due to the randomness between the time of arrival of a packet and the time at which the station may transmit this packet. Depending on the service discipline, the time at which a station may transmit may be after the next locomotive or after the next SOC in the case of NGSS or GSS, or at the beginning of the next slot in the case of HOLS. This implies that, for large α , the variance for HOLS at low S is lower than for GSS and for NGSS since the randomness in the packet delay in this case is associated with the time of arrival taking place within a slot which is shorter than the period separating two consecutive locomotives or SOCs. As $\lambda \rightarrow \infty$, the variance drops to zero since at each station, a new packet is generated as soon as the previous one is transmitted, all rounds are of full length and the packet delay is deterministic and equal to $MX + Y$. It is interesting to note that the variance incurred is highest for S close to the network capacity while the variance is zero at the network capacity. The coefficient of variation of delay as a function of the

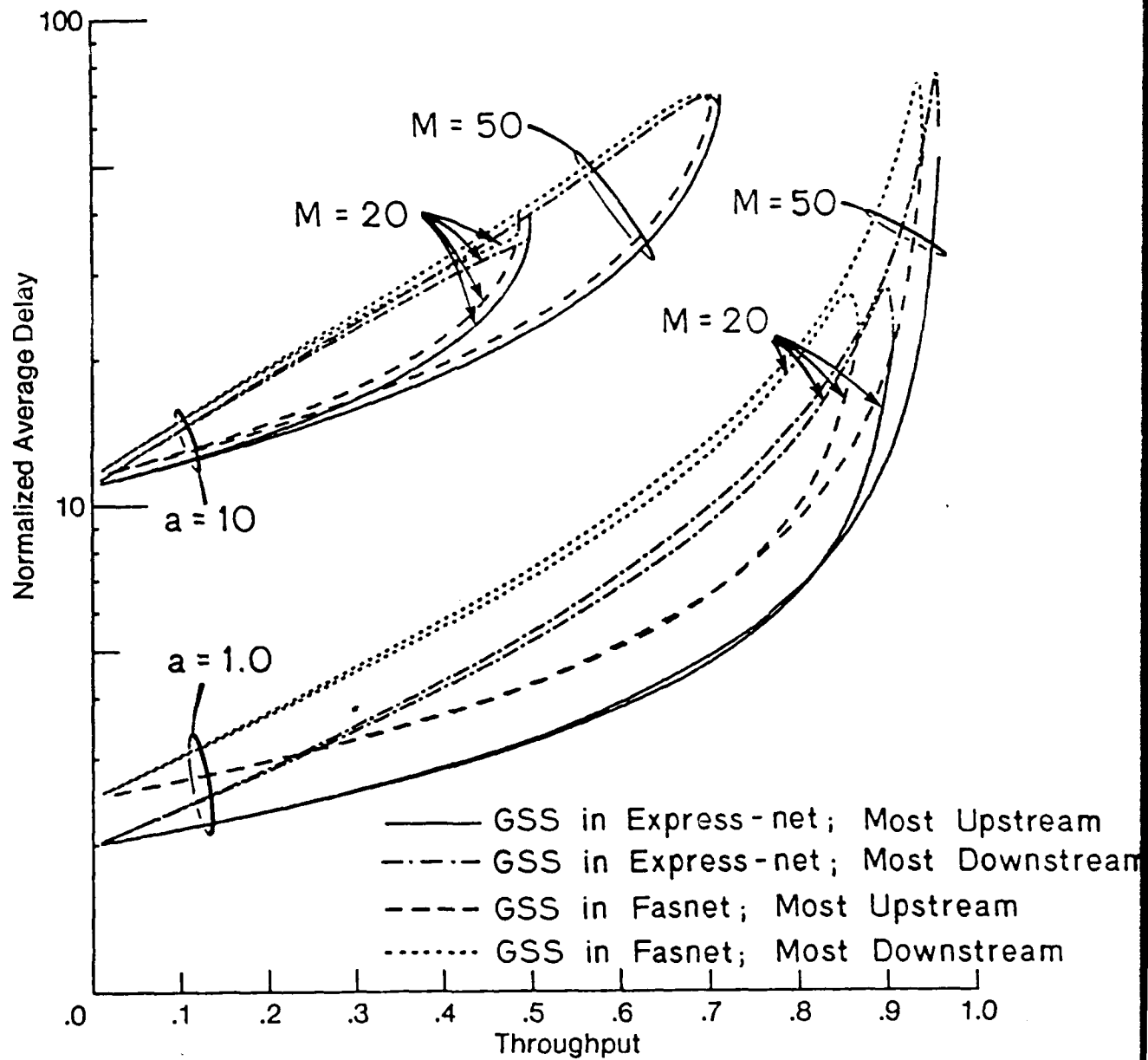


Fig. 3.24 Comparison of the throughput-delay characteristics as achieved by the most upstream station and the most downstream station in both Expressnet and Fasnet operating under the GSS discipline.

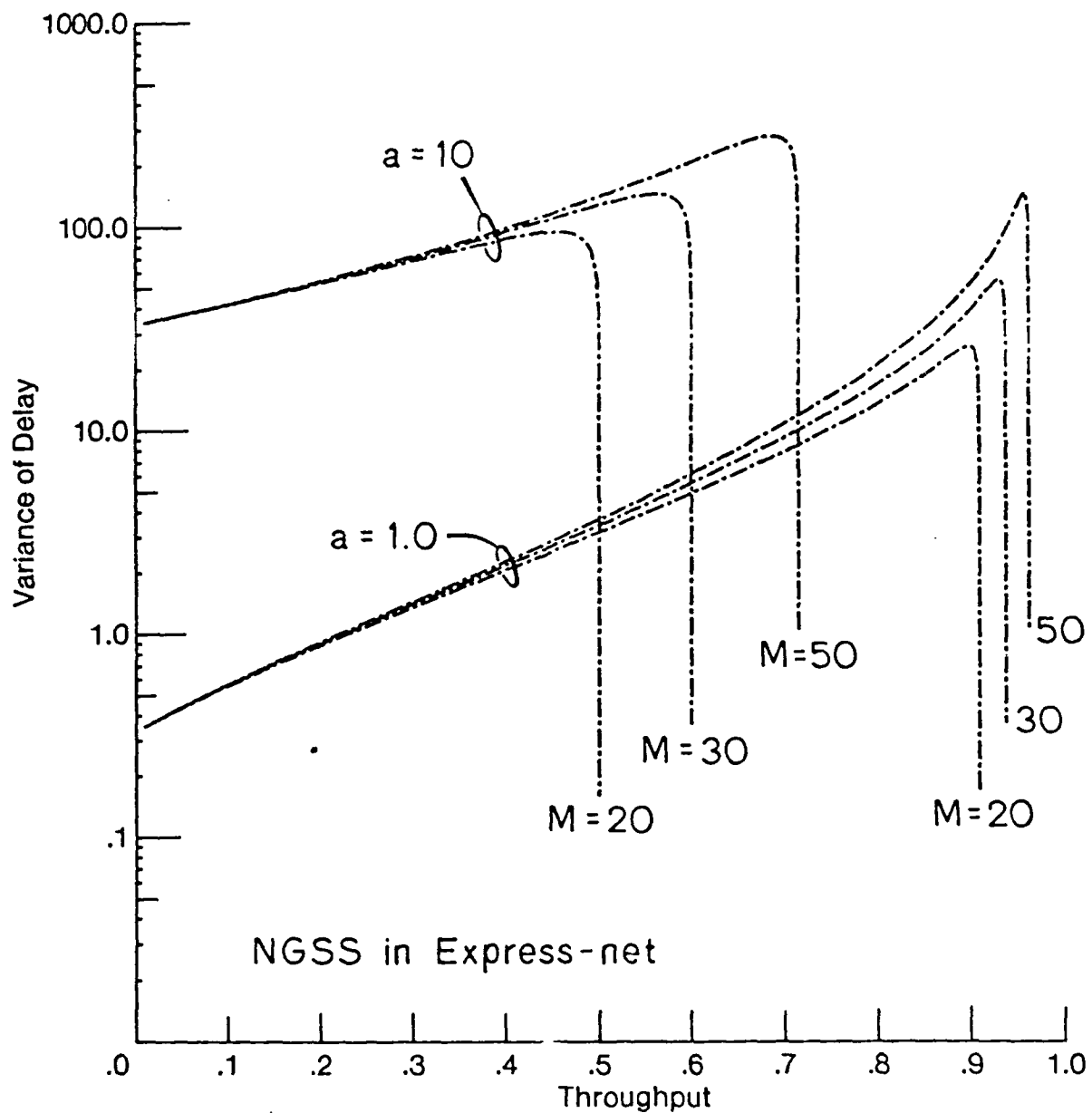


Fig. 3.25 Variance of delay versus throughput for NGSS in Expressnet. The curves shown are for the variance as achieved by any station on the network.

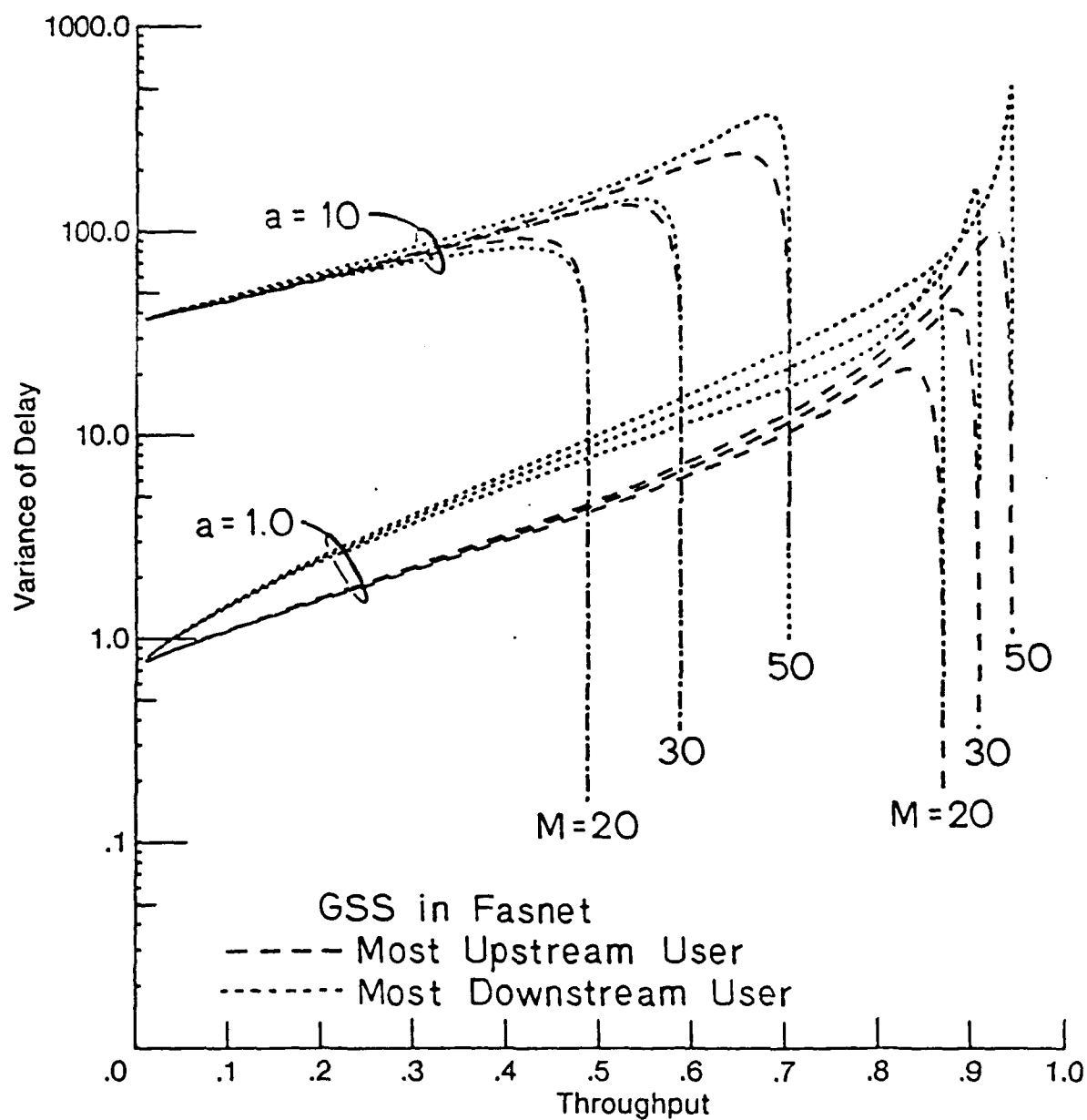


Fig. 3.26 Variance of delay versus throughput for GSS in Fasnet. The variance as achieved by the most upstream station and the most downstream station is shown for each of the values of a and M .

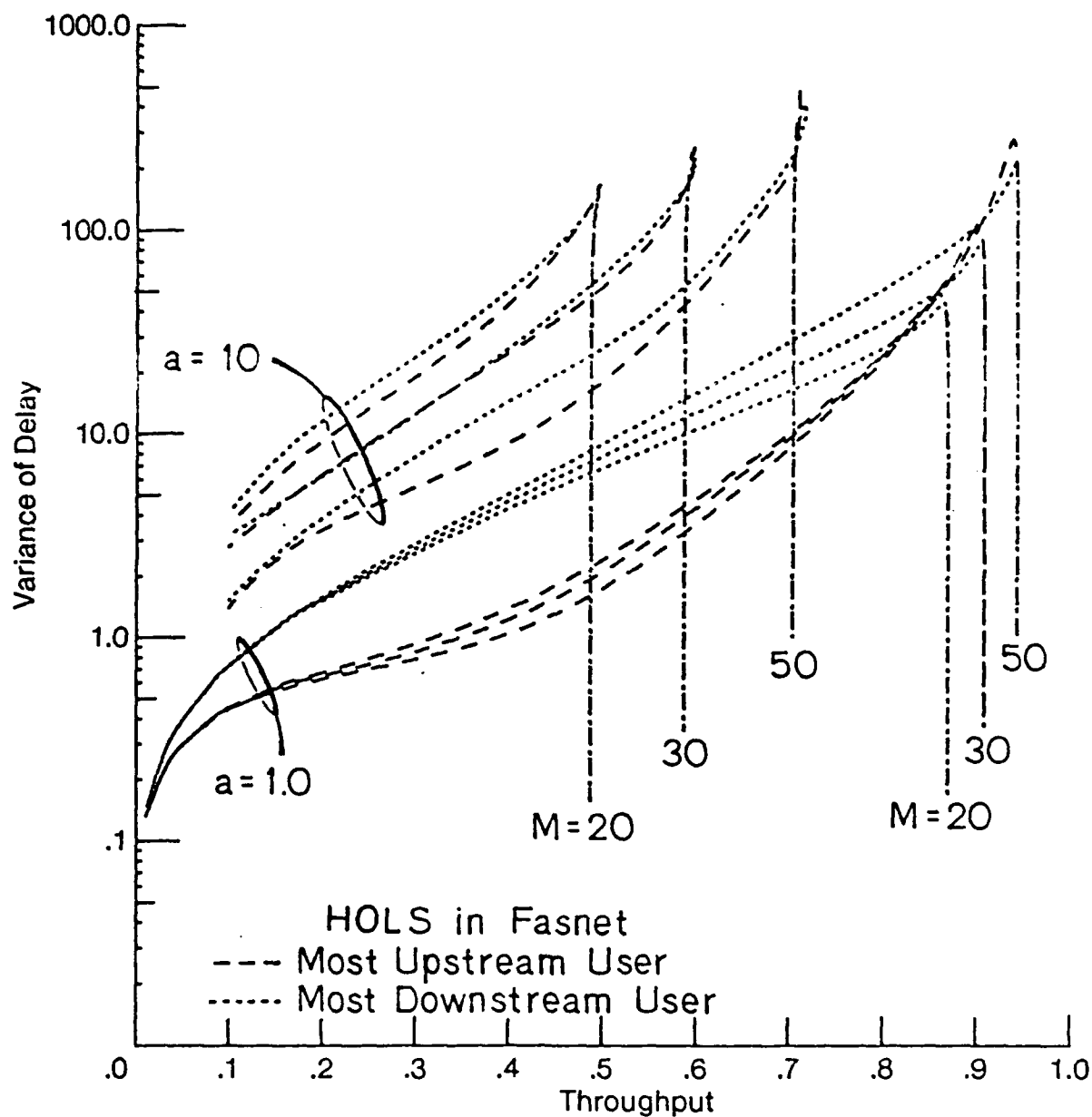


Fig. 3.27 Variance of delay versus throughput for HOLS in Fasnet. The variance as achieved by the most upstream station and the most downstream station is shown for each of the values of a and M . The curves shown for $a = 10$ were obtained by simulation.

throughput is shown for each of the three service disciplines in Figs. 3.28, 3.29, and 3.30 for $a = 1.0$ and various values of M . We noted that, for all the cases examined, the coefficient of variation for GSS and NGSS is always less than one while for HOLS it may be greater than 1.

3.6 Summary

The three service disciplines that were identified in chapter 2, Non-Gated Sequential Service, Gated Sequential Service and Head Of Line Service, were analyzed for two schemes, Expressnet and Fasnet. From the analyses of these service disciplines numerical results were computed. We showed that these systems, unlike random access techniques, can achieve a channel utilization close to 100 percent even when the channel bandwidth is high or the propagation delay of the signal over the network is large. In addition, the network remains stable as the load increases to infinity without the need for any dynamic control of the access protocol. The throughput delay characteristics are excellent and the maximum delay is bounded from above by a finite value which is easily computed. As the throughput approaches the network capacity the variance of delay reaches a peak and then drops to zero. At network capacity the system becomes deterministic with all stations transmitting in every round.

Finally, we noted that all three service disciplines exhibit similar performance characteristics. However, in GSS and HOLS there is an element of unfairness which favors some stations over others depending on their location on the network, while for NGSS the access protocol is completely fair with all stations achieving the same performance.

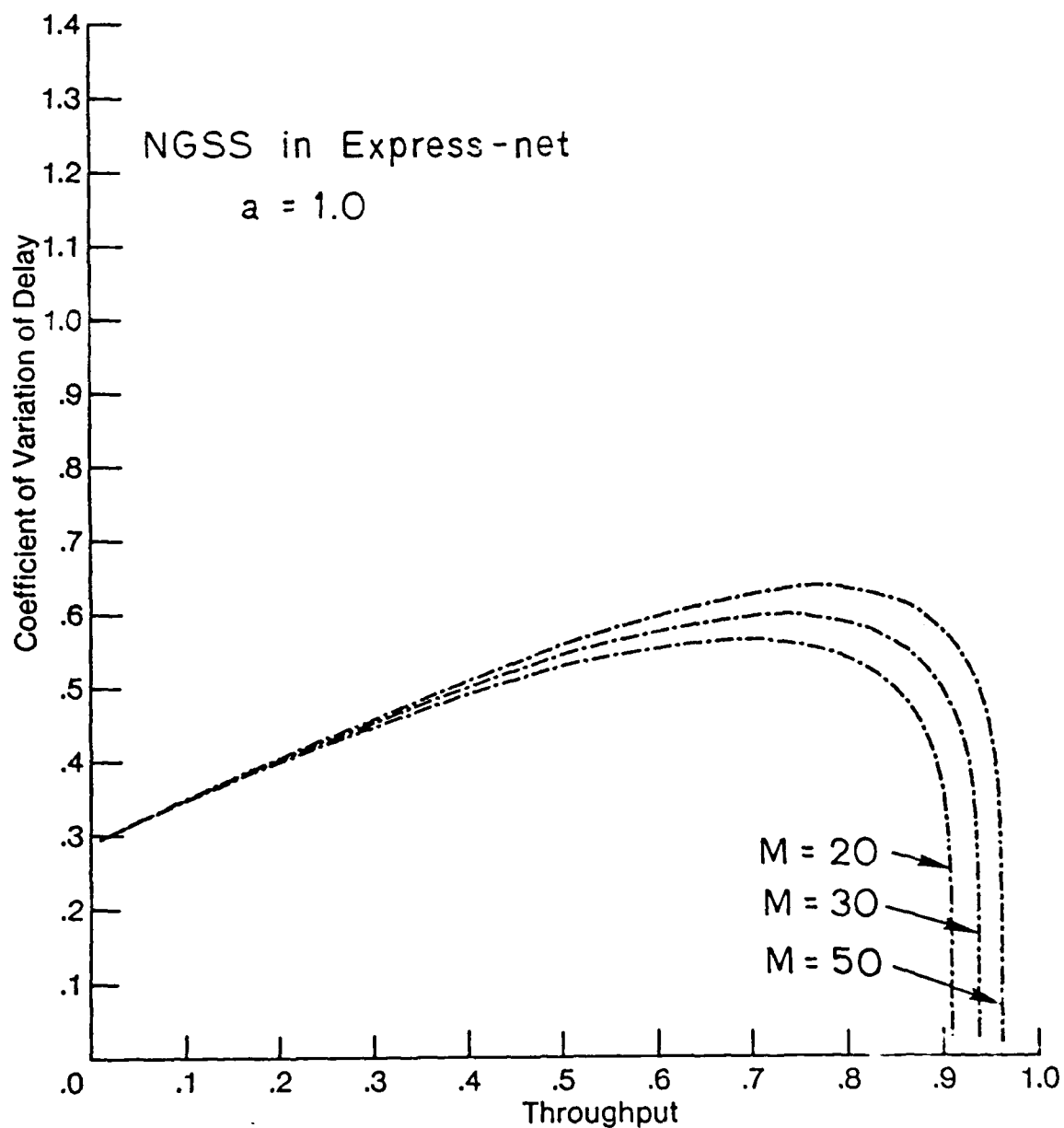


Fig. 3.28 Coefficient of variation of delay versus throughput for NGSS in Express-net with $a = 1.0$. The curves shown are for the variance as achieved by any station on the network.

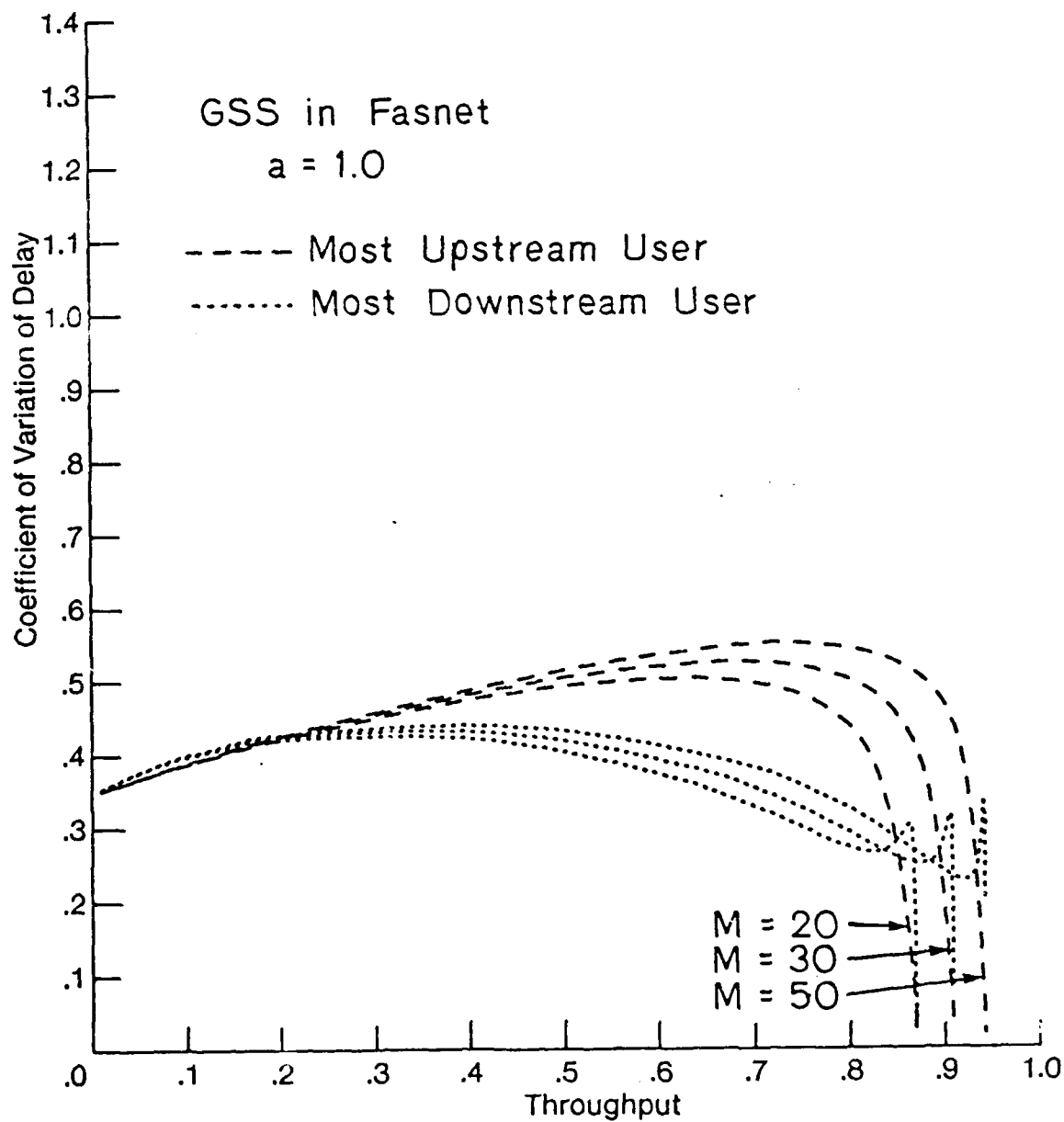


Fig. 3.29 Coefficient of variation of delay versus throughput for GSS in Fasnet with $a = 1.0$. The variance as achieved by the most upstream station and the most downstream station is shown for each of the values of M .

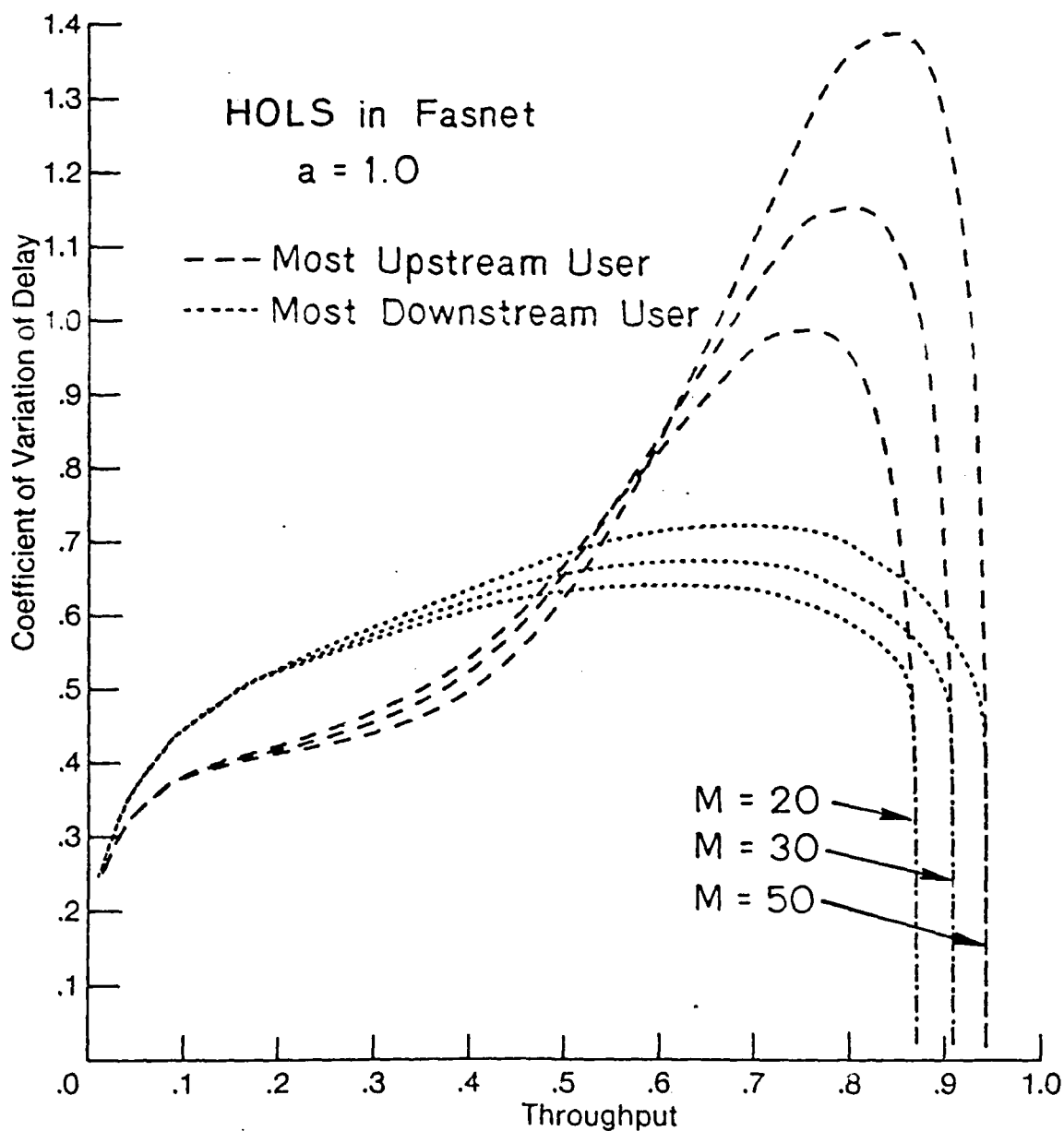


Fig. 3.30 Coefficient of variation of delay versus throughput for HOLs in Fasnet with $a = 1.0$. The variance as achieved by the most upstream station and the most downstream station is shown for each of the values of M .

Chapter 4

Integrated Voice and Data Networks

In this chapter we explore the characteristics of an integrated voice and data access protocol for Expressnet. We first describe in sections 4.1 to 4.3 the method by which voice and data are combined on Expressnet. In section 4.4 we give some notations and definitions followed in sections 4.5 and 4.6 by the analysis. The analysis of the deterministic system, i.e., one with no silence suppression, is given in section 4.5 and the analysis of the stochastic system is given in section 4.6. Some numerical results are presented in section 4.7. We also give in the appendix a proof of the stability of the deterministic system.

4.1 Integrating Voice and Data on Expressnet

Two constraints must be satisfied when voice and data traffic are to be integrated on Expressnet. The first is the delay constraint for voice packets — real time speech must be delivered to the receiver within some finite time after being generated. The second is the minimum bandwidth requirement for data. In [11],

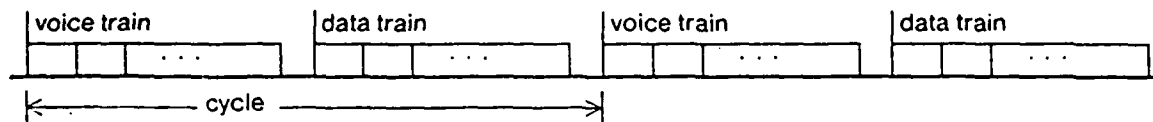


Fig. 4.1 Activity on the inbound channel showing consecutive voice and data trains constituting cycles.

an access mechanism for Expressnet is described that satisfies these constraints and dynamically allocates the channel bandwidth among voice and data stations according to the network load. To accomplish this, two types of trains are considered — a voice train type and a data train type. Trains alternate between these two types and stations transmit their packets on the train corresponding to the type of the packet. We define a *cycle* to consist of a voice train followed by a data train, together with the associated overhead (Fig. 4.1).

To satisfy the delay constraint, the access protocol in [11] limits the number of active voice sources (i.e., voice sources that are participating in a conversation) to some maximum. We denote this maximum by $N_{\max}$. In addition, the length of the data train is restricted to some maximum. The ratio of the maximum allowable length of the data train and maximum length of a cycle determines the minimum bandwidth available to data stations.

The details by which the length of a cycle is limited to the prescribed maximum are described in [11]. Suffice it to say that some method exists to limit the length of a data train to its predetermined maximum and to block a new voice source from accessing the network if the maximum number of active voice sources allowable has already been reached. The essence of this voice/data protocol is the DAMA nature of access method. Any network providing round robin service could, hypothetically, integrate voice and data in the way just described.

Since stations are serviced in a round robin fashion, a given station will periodically be able to access the channel and transmit successfully if it has a packet. We refer to the instants in time when the station in question is able to transmit as the *service instants* associated with that station. The time between two consecutive service instants corresponding to a given station is a random variable and depends on the number of active voice sources and the load placed on the network by data stations.

In [11], only fixed size packets were considered and an analysis of the protocol was made by considering the maximum load condition. In the next section we describe a new method whereby stations generate packets. With this method, stations adapt to the random nature of the inter-service times by automatically adjusting the length of a packet to match the network load. In addition, an overload condition (such as when there are more than $N_{\max}$ voice sources using the network simultaneously) can be accommodated gracefully; such an event was not considered in [11].

4.2 Operation of a Voice Station

Assume that vocoders[†] digitize speech at a constant rate. At each sampling instant, the bits that are the output of the vocoder are stored in a buffer, appended to the existing contents of the buffer. Simultaneously, the network interface listens to the activity on the network and attempts to transmit according to the Express-net access protocol described above. Once the station has determined that it has access to the channel, it transmits the packet preamble and header, followed by the

[†]We use the term "vocoder" to describe the unit that converts the analog signal to digital form. This may be accomplished by simple PCM coding or by some more efficient method based on the structure of speech.

contents of the buffer. Thus, the packet formation time, i.e., the time spent forming a packet, and the network access time, i.e., the time spent waiting for one's service instant, occur concurrently.*

While speech has traditionally been characterized as a continuous stream, in fact, its nature consists of alternating *talkspurts* and *silences* which are of random lengths [52]. To reduce the channel bandwidth required by a single voice source, a voice station can exploit this structure of the speech signal by discarding those bits that correspond to the silence periods, storing in the buffer only those speech samples corresponding to the talkspurt periods. This technique is referred to as *silence suppression*. Depending somewhat on how the hardware differentiates between silences and talkspurts, silence suppression can reduce the average bit rate of a voice station by about 60% of the vocoder rate [52].

If silence suppression is in effect, we expect that it is feasible to allow more than $N_{\max}$ active voice sources at any given time while still maintaining, on average, fewer than $N_{\max}$ transmissions in the voice trains. In this case, the voice delay and data bandwidth requirements will be preserved. However, there is a non-zero probability that more than $N_{\max}$ stations are in talkspurt at the same time, meaning that some packets may incur delays which are greater than the allowable maximum. To accommodate such an event, we assume that speech samples that have been delayed for longer than the specified maximum have become worthless and the station is required to discard them. When speech is discarded in this manner, we say that *clipping* has occurred. A clipped segment of the speech is referred to as a *glitch*. Subjective listening tests have shown that there is little or no qualitative

*Gonsalves used a similar approach for an experimental study of voice communication on an Ethernet local area network [22]. In that work, a station waits until the packet has reached some minimum length and only then attempts to access the network using the CSMA access protocol. However, if the station is not immediately successful in accessing the network, the packet is allowed to grow while the station is trying to gain access to the network.

loss in the intelligibility of the received speech signal for small amounts of clipping. Nevertheless, by any quantitative measure, clipping does degrade the quality of the received speech signal.

4.3 Network Activity

To understand more clearly the operation of a voice station and its interaction with events on the channel, we describe the activity on the channel for some hypothetical situations. For simplicity, we consider the case where there is no silence suppression and where the length of the data train is zero.

Consider the case corresponding to some number of active stations each transmitting in every round. The length of a transmission of any one of these stations, say S_i , depends on the number of bits in its buffer at the service instant corresponding to the transmission in question. The number of bits in the buffer depends in turn on the length of time separating the current service instant at S_i from the previous one at S_i . The time separating two consecutive service instants at a given station is the cumulative sum of all the transmissions between these two service instants plus all the overhead associated with the protocol (plus the length of the data train if there were one). In steady state, the inter-service times are the same for all stations and the packet transmission times are all equal. (We prove the correctness of these claims in the ensuing analysis.)

Now consider the scenario depicted in Fig. 4.2 which corresponds to the activity one might see on the inbound channel over a sequence of rounds. The first two rounds, i.e., the ones labelled *a* and *b*, show transmissions by three stations S_2 , S_3 , and S_4 . The system is in steady state at this time and so transmissions are all of

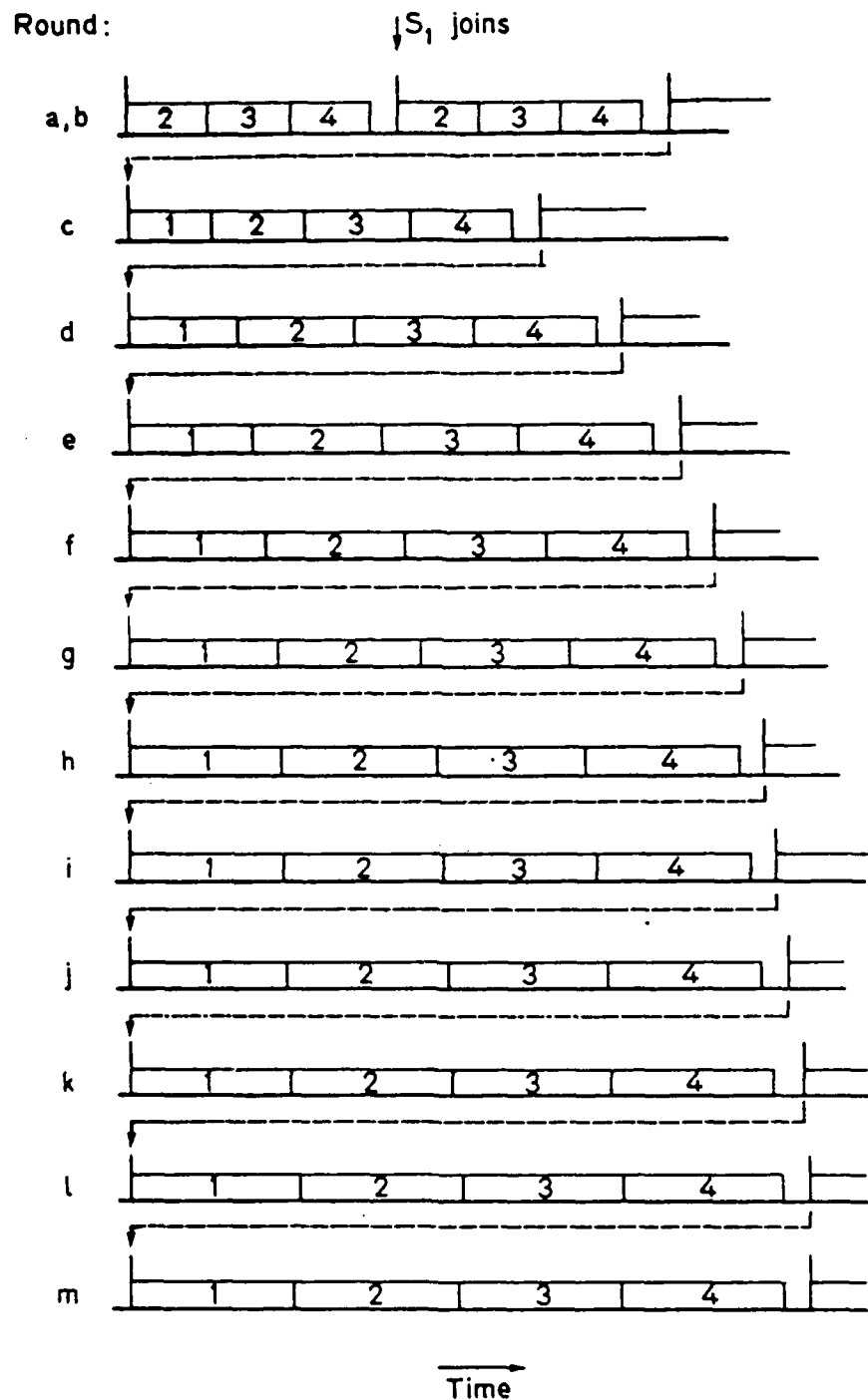


Fig. 4.2 Hypothetical scenario showing how the protocol automatically adjusts the packet transmission time to accommodate a fourth station joining a network which is already servicing three other stations. After a few rounds, a new steady state condition is reached.

equal length and the time between service in round a and service in round b is the same for each of the three stations.

Assume that some other station, S_1 , becomes active at time t_1 which in Fig. 4.2 corresponds to the beginning of round b . Since S_1 has nothing to transmit at this time, it misses its turn in round b . However, when its turn comes up in round c , S_1 transmits the contents of its buffer in accordance with the access protocol. As a result of the transmission by S_1 , S_2 must wait a longer time in round c for its turn to transmit than it had to wait in round b . The extra time that it must wait is exactly the length of the transmission by S_1 . Consequently, the number of bits in S_2 's buffer is greater in round c than in round b . Hence, the transmission time of S_2 in round c is greater than that in round b .

Similarly, S_3 must wait a longer time for service in round c than it did in round b . In this case, the extra time is the length of the transmission by S_1 plus the increase in the transmission time of S_2 . Clearly, the transmission of S_3 in round c is longer than the transmission by S_3 in round b and is also longer than the transmission of S_2 in round c .

We can continue this argument for the transmissions in round d and so on. Eventually, the lengths of the transmissions reach a value that corresponds to the steady state with four stations. Fig. 4.2 shows this progression. (The transmission time of a packet is a function of the number of bits in that packet. Since this is a discrete quantity, the new steady state will be reached in a finite number of cycles.) To a high degree of accuracy, the new steady state is reached in round l . Consequently, we have shown round m to be identical to round l .

In summary, when a new station becomes active and joins the network, the inter-service times increase causing the packet transmission times to steadily increase from transmission to transmission until the new steady state is reached.

In Fig. 4.3, we show a similar scenario. However, we now begin in steady state with four active stations and depict the situation where one of these, S_4 , leaves the network immediately after completing its transmission in round α . In this case, the transmission times of the remaining three sources become progressively shorter until the steady state for a network with three stations is reached.

4.4 Notation and Definitions

Assume that there are N active voice sources which, without loss of generality, are numbered $1, 2, \dots, N$.^{*} Consider two consecutive cycles, labeled $r-1$ and r as shown in Fig. 4.4. Let $t_i^{(r)}$ denote the service instant of S_i in round r . Let $L_i^{(r)} \triangleq t_i^{(r)} - t_i^{(r-1)}$. Let $D_{\max}$ denote the maximum delay allowable. Let $B_i^{(r)}$ denote the number of bits of speech that have been accumulated by S_i at time $t_i^{(r)}$. Denoting the vocoder rate by V , $B_i^{(r)}$ is in the range $\lfloor VL_i^{(r)} \rfloor \leq B_i^{(r)} \leq \lfloor VL_i^{(r)} \rfloor + 1$ and is taken to be

$$B_i^{(r)} = \lfloor VL_i^{(r)} \rfloor. \quad (4.1)$$

The maximum number of bits that may be transmitted by any station in a single packet is denoted by $B_{\max} = \lfloor VD_{\max} \rfloor$. Obviously, if there is no delay restriction, $D_{\max} = \infty$ and hence $B_{\max} = \infty$.

Denote the length of the preamble by Ω , the number of bits in the header by H , the bandwidth of the channel by W and the bandwidth guaranteed to data stations by W_d . We define a *transmission period* as the time required by a station to transmit a packet (including the protocol overhead 2Δ , the preamble and packet

^{*}A single two-way conversation requires two active voice sources.

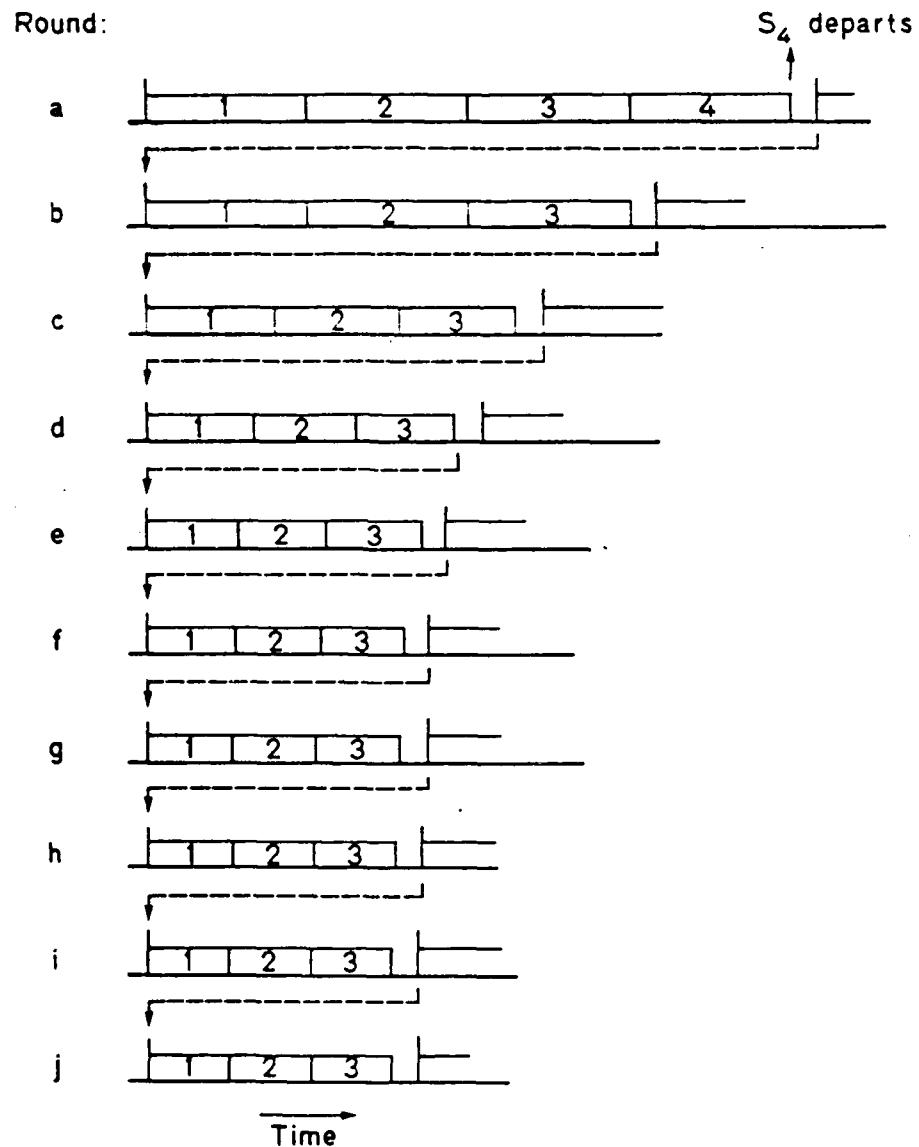


Fig. 4.3 A second hypothetical scenario showing how the protocol automatically adjusts the packet transmission time to accommodate a station leaving a network. Again, after a few rounds, a new steady state condition is reached.

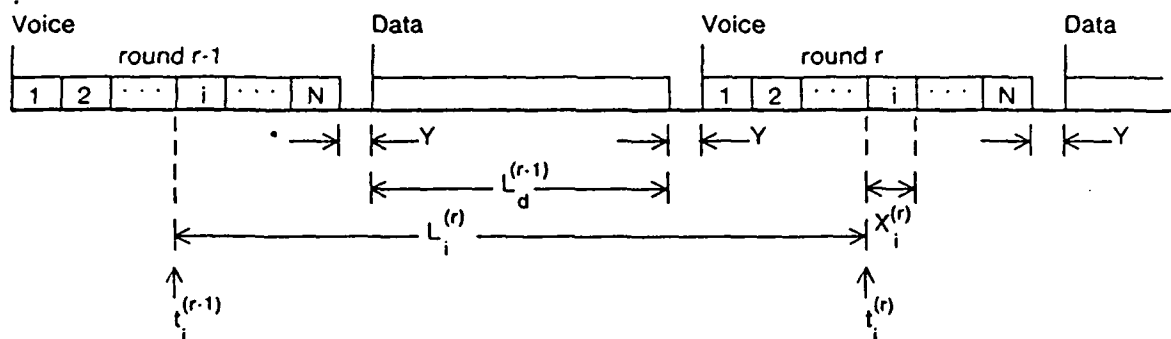


Fig. 4.4 Activity on the inbound channel showing the events that constitute a cycle and the notation used to describe these events.

header, and the speech samples). Expressed as a function of the number of bits accumulated in the station's buffer, the length of a transmission period is given by

$$X(b) = \begin{cases} 0 & b = 0 \\ 2\Delta + \Omega + H/W + \min(b, B_{\max})/W & b > 0 \end{cases} \quad (4.2)$$

Letting $L_d^{(r)}$ denote the length of the data train in the r^{th} cycle and Y the overhead between two consecutive trains, $L_i^{(r)}$ can be expressed as

$$L_i^{(r)} = \sum_{j=i}^N X(B_j^{(r-1)}) + \sum_{j=1}^{i-1} X(B_j^{(r)}) + 2Y + L_d^{(r-1)}. \quad (4.3)$$

The delay of a voice bit is defined to be the period of time from the instant at which it is generated at the vocoder until the instant at which it is transmitted. Clearly, of all the bits in a given voice packet, the first bit is the one incurring the largest delay. We measure the delay of a packet by the delay incurred by this bit.*

Not including the delay incurred due to the transmission of the preamble and the

*In measuring the delay of a packet in this way we have made the following assumptions concerning the operation of the receiver: The receiving station recognizes that a packet is destined for it as soon as it has received the header information. The speech samples may be passed on to the receiver's decoder as soon as they arrive to the receiver. That is, the receiver does not have to buffer the entire packet before delivering the speech to the decoder.

header, the packet delay can be expressed as

$$D_i^{(r)} = \min(D_{\max}, L_i^{(r)}) \quad (4.4)$$

The transmission time of the preamble and header and the 2Δ protocol overhead are not counted in $D_i^{(r)}$ since this conveniently makes the arithmetic a little simpler. To include these amounts, one merely need add $(2\Delta + \Omega + H/W)$ to all delay results.

In the next two sections, we use the above notation and definitions to provide an analysis of the system. Thereafter, we present some performance results.

4.5 No Silence Suppression: The Deterministic System

4.5.1 Analysis

Assume that there are N active voice sources and that there is no silence suppression. Assume also that data trains are of a constant length denoted by L_d . Under these assumptions, all rounds are of equal length and packets are of equal size. (See proof below.) That is, $L_i^{(r)}$, $B_i^{(r)}$, $X(B_i^{(r)})$, and $D_i^{(r)}$ are constants which we denote by L , B , X , and D , respectively. Under these conditions, (4.1), (4.2) and (4.3) reduce to

$$B = \lfloor LV \rfloor, \quad (4.5)$$

$$X = 2\Delta + \Omega + H/W + \min(B, B_{\max})/W, \quad (4.6)$$

and

$$L = NX + 2Y + L_d. \quad (4.7)$$

Since $B_{\max} = \lfloor V D_{\max} \rfloor$, we can rewrite X as*

$$X = 2\Delta + \Omega + H/W + \min(L, D_{\max})V/W. \quad (4.8)$$

Obviously, if N is small then $L < D_{\max}$, while if N is large then $L > D_{\max}$. $N_{\max}$ is now formally defined as that number of stations such that $L \leq D_{\max}$ for $N \leq N_{\max}$ and also such that $L > D_{\max}$ for $N > N_{\max}$. Thus, (4.8) can be expressed as

$$X = \begin{cases} 2\Delta + \Omega + H/W + LV/W & N \leq N_{\max} \\ 2\Delta + \Omega + H/W + D_{\max}V/W & N > N_{\max} \end{cases} \quad (4.9)$$

Substituting (4.9) into (4.7) gives

$$L = \begin{cases} \frac{NH + NW(2\Delta + \Omega) + W(2Y + L_d)}{W - NV} & N \leq N_{\max} \\ N(2\Delta + \Omega) + N(H + D_{\max}V)/W + 2Y + L_d & N > N_{\max} \end{cases} \quad (4.10)$$

and hence,

$$D = \begin{cases} \frac{NH + NW(2\Delta + \Omega) + W(2Y + L_d)}{W - NV} & N \leq N_{\max} \\ D_{\max} & N > N_{\max} \end{cases} \quad (4.11)$$

Equation (4.11) gives the relationship between the delay of a voice packet, the number of voice sources and the constant length of the data train.

Suppose now that we require the network to guarantee both a maximum delay to voice traffic of $D_{\max}$ and a minimum bandwidth to data users of W_d . To guarantee $D \leq D_{\max}$, we limit the number of voice sources to $N_{\max}$ and set the length of the data cycle to the maximum allowable value which we denote by $L_{d\max}$. To ensure

*We ignore the slight error introduced by truncation as we convert from discrete units of bits in (4.6) to continuous units of time in (8). For packets which are on the order of 500 bits, the error is on the order of 0.1%.

a minimum bandwidth of W_d to data users at the maximum voice load, we require that

$$L_{d\max}/D_{\max} = W_d/W. \quad (4.12)$$

Replacing L_d in (4.10) with $L_{d\max}$ as given by (4.12) gives

$$L = \frac{NW(2\Delta + \Omega) + NH + D_{\max}W_d + 2WY}{W - NV} \quad N \leq N_{\max} \quad (4.13)$$

and since we require that $N \leq N_{\max}$, we have that $L \leq D_{\max}$ and so the packet delay is simply $D = L$. Letting $L = D_{\max}$ in (4.13), we can compute the maximum number of voice sources such that we can guarantee that $D \leq D_{\max}$. This maximum is given by

$$N_{\max} = \left\lfloor \frac{(W - W_d)D_{\max} - 2WY}{H + VD_{\max} + W(2\Delta + \Omega)} \right\rfloor. \quad (4.14)$$

For completeness, consider the case where $L_d = L_{d\max}$ but, for some reason, $N > N_{\max}$. In this case, $L > D_{\max}$ and is given by

$$L = N(2\Delta + \Omega + H/W) + (NV + W_d)D_{\max}/W + 2Y \quad N > N_{\max} \quad (4.15)$$

As required by the protocol, those speech samples that are older than $D_{\max}$ are discarded, i.e., they are not transmitted. Hence the packet delay for the case where $N > N_{\max}$ is simply $D = D_{\max}$.

To provide a quantitative measure of the effect of overloading the network in this way (i.e., allowing more than $N_{\max}$ voice stations to be active at any one time) we calculate the fraction of speech that is lost under this condition, expressed as a function of the number of active voice sources N . Denote the fraction of speech lost by G . Clearly, $G = (L - D_{\max})/L$ and, using the value of L given by (4.15),

can be expressed as

$$G = \begin{cases} 0 & N \leq N_{\max} \\ \frac{NW(2\Delta + \Omega) + N(H + VD_{\max}) + (W_d - W)D_{\max} + 2WY}{NW(2\Delta + \Omega) + N(H + VD_{\max}) + W_d D_{\max} + 2WY} & N > N_{\max} \end{cases} \quad (4.16)$$

Finally, we calculate the *actual* bandwidth achieved by the data stations, denoted by W_A , when $L_d = L_{d\max}$. W_A can be computed from the ratio

$$\frac{W_A}{W} = \frac{L_{d\max}}{L}. \quad (4.17)$$

Substituting for $L_{d\max}$ from (4.12) and for L from (4.13) and (4.15) gives

$$W_A = \begin{cases} \frac{W_d D_{\max}(W - NV)}{NW(2\Delta + \Omega) + NH + D_{\max}W_d + 2WY} & N \leq N_{\max} \\ \frac{W_d D_{\max}W}{NW(2\Delta + \Omega) + NH + (NV + W_d)D_{\max} + 2WY} & N > N_{\max} \end{cases} \quad (4.18)$$

Obviously, $W_A \geq W_d$ for $N \leq N_{\max}$ but $W_A < W_d$ for $N > N_{\max}$. This implies that one of the effects of allowing greater than $N_{\max}$ voice sources on the network at a given time is to reduce the bandwidth available to data stations to a value which is less than the required minimum.

The steady state analysis has assumed that N is fixed. Given that N varies as new calls are set up or existing ones are terminated, one may ask whether the steady state is ever reached. We now show that, in fact, the system reaches steady state in a time period that is short as compared with the average time between changes in N .

Let $W_d = 0$. Assume that voice sources become active according to a Poisson process with rate a pairs of stations per second, and that a pair of stations remains

active for an exponentially distributed period of time with mean $1/d$ s. According to this model, $N/2$ represents a simple birth-death process where the expected holding time in state n is $h_n = 1/(a + nd)$. One can easily show that the expected number of active calls at any time is $N/2 = a/d$ and that $h_{N/2} = 1/Nd$ s.

To examine how the system evolves as N is increased or decreased from a given value, we monitored the length of successive rounds using equations (4.2) and (4.3). For all values of W and N considered, the system moves from the old steady state to within 1% of the new steady state within 1ms. Assuming that the average length of a call is 3 minutes, i.e., $1/d = 180$ s, and that $N = 1000$ (which is close to $N_{\max}$ for $W = 100$ Mb/s) the expected time between changes in N is 180ms — two orders of magnitude larger than the time for the system to reach its steady state. Thus, it is reasonable to assume that the system is quasi-stationary with respect to changes in N .

4.5.2 Stability of the Deterministic System

We show that the deterministic system converges to a steady state as was assumed in the analysis above. As in the analysis, we assume that there are N active voice sources and that $L_d^{(r)}$ is a constant given by L_d .

Consider two consecutive voice rounds $r-1$ and r . Define $T_i^{(r)} \triangleq X(B_i^{(r)})$ where $X(b)$ is defined by (4.2). $T_i^{(r)}$ can be expressed as

$$T_i^{(r)} = 2\Delta + \Omega + H/W + V L_i^{(r)}/W \quad (4.19)$$

where, from (4.3), $L_i^{(r)}$ is given by

$$L_i^{(r)} = \sum_{j=i}^N T_j^{(r-1)} + \sum_{j=1}^{i-1} T_j^{(r)} + 2Y + L_d. \quad (4.20)$$

Hence,

$$T_i^{(r)} = \frac{V}{W} \left[\sum_{j=1}^{i-1} T_j^{(r)} + \sum_{j=i}^N T_j^{(r-1)} \right] + (2\Delta + \Omega + H/W) + \frac{V}{W} [2Y + L_d]. \quad (4.21)$$

Let $T_i^{(r)} \triangleq T + P_i^{(r)}$ where T is a constant and $P_i^{(r)}$ is the perturbation around this constant T . Hence, (4.21) can be written as

$$\begin{aligned} T + P_i^{(r)} &= \frac{V}{W} NT + \frac{V}{W} \left[\sum_{j=1}^{i-1} P_j^{(r)} + \sum_{j=i}^N P_j^{(r-1)} \right] \\ &\quad + (2\Delta + \Omega + H/W) + \frac{V}{W} [2Y + L_d] \end{aligned} \quad (4.22)$$

If we choose T in such a way as to satisfy (4.13) and (4.9), i.e., $L = NT + 2Y + L_d$ and $T = (2\Delta + \Omega + H/W) + LV/W$, we get

$$\begin{aligned} T &= (2\Delta + \Omega + H/W) + \frac{V}{W} [NT + 2Y + L_d] \\ &= \frac{V}{W} NT + (2\Delta + \Omega + H/W) + \frac{V}{W} [2Y + L_d]. \end{aligned} \quad (4.23)$$

Clearly, we have equivalent terms on the left and right hand sides of (4.22). Denoting V/W by x , (4.22) in its simplified form can be expressed as

$$P_i^{(r)} = x \left[\sum_{j=1}^{i-1} P_j^{(r)} + \sum_{j=i}^N P_j^{(r-1)} \right] \quad (4.24)$$

In vector notation (4.24) becomes $\mathbf{P}^{(r)} = \mathbf{A}\mathbf{P}^{(r)} + \mathbf{B}\mathbf{P}^{(r-1)}$ where

$$\begin{aligned} [\mathbf{P}^{(r)}]_i &= P_i^{(r)} \\ [\mathbf{A}]_{ij} &= \begin{cases} x & j < i \\ 0 & j \geq i \end{cases} \\ [\mathbf{B}]_{ij} &= \begin{cases} 0 & j < i \\ x & j \geq i. \end{cases} \end{aligned} \quad (4.25)$$

By simple algebra $P^{(r)} = [I - A]^{-1} B P^{(r-1)}$. Letting $C = [I - A]^{-1}$ and $D = CB$, (4.24) reduces to $P^{(r)} = D P^{(r-1)}$. It remains now to compute C and hence D .

Lemma:

$$[C]_{ij} = \begin{cases} 0 & j > i \\ 1 & j = i \\ x(1+x)^{i-j-1} & j < i \end{cases} \quad (4.26)$$

Proof: Consider $C[I - A]$. By simple matrix multiplication one can show that $C[I - A] = I$. Hence $C = [I - A]^{-1}$.

Since all of the elements of B are either 0 or x and $D = CB$,

$$[D]_{ij} = \sum_{k=1}^N C_{ik} B_{jk} = x \sum_{k=1}^{\min(i,j)} C_{ik}. \quad (4.27)$$

Consider the cases $i \leq j$ and $i > j$. For $i \leq j$

$$\begin{aligned} [D]_{ij} &= x \sum_{k=1}^i C_{ik} \\ &= x \sum_{k=1}^{i-1} x(1+x)^{i-k-1} + x \\ &= x(1+x)^{i-1} \end{aligned} \quad (4.28)$$

For $i > j$

$$\begin{aligned} [D]_{ij} &= x \sum_{k=1}^j C_{ik} \\ &= x \sum_{k=1}^j x(1+x)^{i-k-1} \\ &= x(1+x)^{i-j-1} [(1+x)^j - 1] \end{aligned} \quad (4.29)$$

Hence

$$[D]_{ij} = \begin{cases} x(1+x)^{i-j-1} [(1+x)^j - 1] & j < i \\ x(1+x)^{i-1} & j \geq i \end{cases} \quad (4.30)$$

To prove stability of the steady state equations we show that the row sums of $\mathbf{D}$ are less than unity; hence $P_i^{(r)}$ is less than the convex combination of the $P_i^{(r-1)}$. Consider $\sum_j [\mathbf{D}]_{ij}$.

$$\begin{aligned}
 \sum_j [\mathbf{D}]_{ij} &= \sum_{j=1}^{i-1} x(1+x)^{i-j-1} [(1+x)^j - 1] + \sum_{j=i}^N x(1+x)^{i-1} \\
 &= Nx(1+x)^{i-1} - \sum_{j=1}^{i-1} x(1+x)^{i-j-1} \\
 &= 1 + Nx(1+x)^{i-1} - (1+x)^{i-1} \\
 &= 1 - (1 - Nx)(1+x)^{i-1}
 \end{aligned} \tag{4.31}$$

Recall from (4.23) that for the steady state to exist we require that $\frac{V}{W}NT = T - [2\Delta + \Omega + H/W + V(2Y + L_d)/W]$ and since $V/W = x$,

$$\begin{aligned}
 Nx &= 1 - \frac{1}{T}[2\Delta + \Omega + H/W + V(2Y + L_d)/W] \\
 &< 1.
 \end{aligned} \tag{4.32}$$

Since $Nx < 1$, then $1 - Nx > 0$ and hence $\sum_j \mathbf{D}_{ij} < 1$. Recall that $\mathbf{P}^{(r)} = \mathbf{D}\mathbf{P}^{(r-1)}$. So

$$\begin{aligned}
 P_i^{(r)} &= \sum_j [\mathbf{D}]_{ij} P_j^{(r-1)} \\
 &< \max_j \{P_j^{(r-1)}\} \sum_j \mathbf{D}_{ij} \\
 &< \max_j \{P_j^{(r-1)}\} \quad \forall i
 \end{aligned} \tag{4.33}$$

Hence $\max_i \{P_i^{(r)}\} < \max_j \{P_j^{(r-1)}\}$ and so $\max_i \{P_i^{(r)}\} \searrow$ as $r \nearrow$. Thus, in the limit, $P_i^{(r)} \rightarrow 0$ as $r \rightarrow \infty$ for all i and the system converges to a steady state.

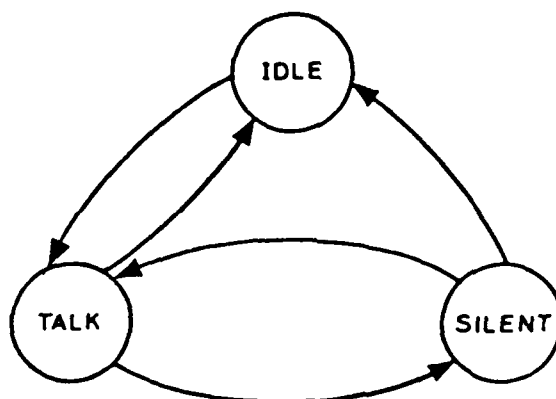


Fig. 4.5 Three state Markov chain describing the activity of a voice source.

4.6 With Silence Suppression: The Stochastic System

Assume that there are N active voice sources. Assume also that none of these voice sources becomes inactive and that no inactive voice sources become active. Silence suppression is in effect. The data train may be arbitrarily long up to some maximum length denoted by $L_{d_{\max}}$. We now present a model of a voice source followed by the analysis.

4.6.1 The Model of a Voice Source

The dynamics of a voice source have generally been modeled as a three state Markov chain as shown in Fig. 4.5 [55]. If the source is *idle*, no conversation is taking place. An idle source becomes active when a conversation begins. While active, the source alternates between the *talk* and *silent* states as it moves back and forth between talkspurt and silence. For this work, we assume that there are a fixed number of active voice sources; there are no transitions between the idle state and either the talk state or the silent state. The mean holding time in the talk state

is denoted by $1/\lambda$ (around 1.3s) and the mean holding time in the silent state is denoted by $1/\mu$ (around 1.7s). The probability of being in the talk state is denoted by p_t and is given by $p_t = \mu/(\lambda + \mu)$ (around 0.4).

In this analysis, we modify the model and assume that a station can make at most one state transition between two service instants. We make this assumption for two reasons. Firstly, it is a convenient restriction to the model, making it easier to describe (mathematically) events between two service instants at a given station; secondly, it approximates fairly closely a real speech detector which, if in the silent state, will ignore short talkspurts and, if in the talk state, will fill in short silence periods. For example, in [53], talkspurts shorter than 15ms are ignored and silences which are less than 200ms are filled in. Thus, in fact, it would be unrealistic for a given station to make more than one state transition, either from talk to silent or from silent to talk, between two consecutive service instants — a period of time which, in this work, is on the order of 1ms to 100ms. Naturally, if the system is designed such that a longer delay is tolerated, say 150ms or greater, then this assumption becomes a poor one.

4.6.2 A Mathematical Formulation

Consider the instant of time $t_i^{(r)}$ corresponding to the beginning of a transmission or potential transmission from station S_i . Define $\beta_i^{(r)} = 0$ if S_i is in the silent state at $t_i^{(r)}$ and $\beta_i^{(r)} = 1$ if S_i is in the talk state at $t_i^{(r)}$. The state of the system at $t_i^{(r)}$ is completely described by: a set of values

$$B_{i,j}^{(r)} = \begin{cases} B_j^{(r)} & j \leq i \\ B_j^{(r-1)} & j > i \end{cases} \quad 1 \leq i \leq N \quad (4.34)$$

denoting the size of the most recent packet from each of the of the N stations; a set of indicators

$$\beta_{i,j}^{(r)} = \begin{cases} \beta_j^{(r)} & j \leq i \\ \beta_j^{(r-1)} & j > i \end{cases} \quad 1 \leq i \leq N \quad (4.35)$$

denoting whether or not each of the stations was in the talk or silent state at the time that the station in question was last offered service;* and $L_d^{(r-1)}$, the length of the most recent data train. Thus, the state of the system at $t_i^{(r)}$ is completely described by the vector

$$Z_i^{(r)} = (\{B_{i,j}^{(r)}, \beta_{i,j}^{(r)}\} j = 1, 2, \dots, N; L_d^{(r-1)}) \quad (4.36)$$

which can be expressed more simply as

$$Z_i^{(r)} = (B_{i+1}^{(r-1)}, \beta_{i+1}^{(r-1)}; \dots, B_N^{(r-1)}, \beta_N^{(r-1)}; B_1^{(r)}, \beta_1^{(r)}; \dots, B_i^{(r)}, \beta_i^{(r)}; L_d^{(r-1)}). \quad (4.37)$$

The state of the system at $t_{i+1}^{(r)}$, denoted by

$$Z_{i+1}^{(r)} = (B_{i+2}^{(r-1)}, \beta_{i+2}^{(r-1)}; \dots, B_N^{(r-1)}, \beta_N^{(r-1)}; B_1^{(r)}, \beta_1^{(r)}; \dots, B_{i+1}^{(r)}, \beta_{i+1}^{(r)}; L_d^{(r-1)})$$

is determined from $Z_i^{(r)}$ and from $B_{i+1}^{(r)}$ and $\beta_{i+1}^{(r)}$. Using the model of a voice source described above, $B_{i+1}^{(r)}$ and $\beta_{i+1}^{(r)}$ can be determined from the length of time $t_{i+1}^{(r)} - t_{i+1}^{(r-1)}$ and the value of $\beta_{i+1}^{(r-1)}$. This information is contained in the state descriptor $Z_i^{(r)}$. Thus, $Z_{i+1}^{(r)}$ depends only on $Z_i^{(r)}$ and hence the process $\{Z_i^{(r)}\}_{i=1}^N$ constitutes a discrete time Markov chain. Note that the transition from $Z_N^{(r-1)}$ to $Z_1^{(r)}$ requires that the distribution on the length of the data train be introduced into

*In order to decrease the complexity of the state descriptor at the expense of accuracy, one could assume that, if S_i is in talkspurt at $t_i^{(r-1)}$ but in silence at $t_i^{(r)}$, all voice bits collected between these two times are discarded. In this case, the condition $B_i^{(r)} = 0$ is sufficient to describe the case where S_i is in silence at time $t_i^{(r)}$, thus eliminating the need for the set of indicators.

the transition probabilities while transitions from $Z_i^{(r)}$ to $Z_{i+1}^{(r)}$, $i \neq N$ do not. As a result, the Markov chain described by $\{\{Z_i^{(r)}\}_{i=1}^N\}_{r=1}^\infty$ is non-homogeneous. Nevertheless, we define two transition matrices, $\mathbf{R}_0$ and $\mathbf{R}_1$, which specify this chain completely.* These are

$$\begin{aligned} [\mathbf{R}_0]_{xy} &= \Pr\{Z_i^{(r)} = y \mid Z_{i-1}^{(r)} = x\} \quad i \neq 1 \\ [\mathbf{R}_1]_{xy} &= \Pr\{Z_1^{(r)} = y \mid Z_N^{(r-1)} = x\} \end{aligned} \quad (4.38)$$

where

$$\begin{aligned} \mathbf{x} &\triangleq (x_1, \alpha_1; \dots, x_N, \alpha_N; l_x) \quad x_i, l_x \geq 0 \\ \mathbf{y} &\triangleq (y_1, \gamma_1; \dots, y_N, \gamma_N; l_y) \quad y_i, l_y \geq 0 \end{aligned} \quad (4.39)$$

Unfortunately, solutions for non-homogeneous chains are not readily available. If, however, we consider the instants of time at which some arbitrary station, say S_i , is offered service, i.e., $t_i^{(r)}$, $t_i^{(r+1)}$, ..., then we can define an embedded Markov chain $\{Z_i^{(r)}\}_{r=1}^\infty$ which is homogeneous. Moreover, its transition matrix $\mathbf{P}_i$ can be computed from the transition matrices $\mathbf{R}_0$ and $\mathbf{R}_1$ as

$$\mathbf{P}_i = \mathbf{R}_0^{N-i} \mathbf{R}_1 \mathbf{R}_0^{i-1}. \quad (4.40)$$

The Markov chain specified by $\mathbf{P}_i$ has finite state space and is irreducible. Hence the chain is ergodic and the steady state exists. To facilitate the computation of the stationary distribution, we write the equations for the transition matrices $\mathbf{R}_0$ and $\mathbf{R}_1$.

It is convenient to first define the notation $L(\mathbf{x})$ to denote the length of a cycle as a function of the state of the system $\mathbf{x}$. $L(\mathbf{x})$ is given by

$$L(\mathbf{x}) = \sum_{j=1}^N X(x_j) + 2Y + l_x \quad (4.41)$$

*If we assume that the data train is a constant fixed length, $\mathbf{R}_1 = \mathbf{R}_0$ and the Markov chain $\{\{Z_i^{(r)}\}_{i=1}^N\}_{r=1}^\infty$ is homogeneous. One can compute the steady state distribution on this chain from the transition matrix $\mathbf{R}_0$ thus somewhat simplifying the analysis.

where $X(b)$ has been defined in (4.2). Clearly, $L(Z_i^{(r)}) = L_{i+1}^{(r)}$ where $L_i^{(r)}$ has been defined in (4.3).

As a result of the way that we have defined the state descriptor $Z_i^{(r)}$, $[R_0]_{xy}$ and $[R_1]_{xy}$ are zero if $y_j \neq x_{j+1}$ or $\gamma_j \neq \alpha_{j+1}$, $j \in \{1, 2, \dots, N-1\}$. In addition, $[R_0]_{xy} = 0$ if $l_x \neq l_y$. We first compute the elements of R_0 . To compute $[R_0]_{xy}$ for the case where it is non-zero, we must compute for all x the probability that $B_i^{(r)} = y_N$ and $\beta_i^{(r)} = \gamma_N$ for all possible values of y_N and γ_N . We identify four cases that can conveniently be treated separately.

The first of these cases is where $Z_{i-1}^{(r)} = x$ is such that $\beta_i^{(r-1)} = 0$ and $\beta_i^{(r)} = 0$, meaning that S_i was in silence at $t_i^{(r-1)}$ and has remained in this state. Given that $\beta_i^{(r-1)} = 0$, this case occurs with probability $e^{-\mu L(x)}$ where $L(x)$ is given by (4.41). $B_i^{(r)}$ can only take on one value, namely $B_i^{(r)} = 0$.

The second case is where $Z_{i-1}^{(r)} = x$ is such that $\beta_i^{(r-1)} = 1$ and $\beta_i^{(r)} = 1$ meaning that S_i was in talkspurt at $t_i^{(r-1)}$ and has remained in talkspurt. Given that $\beta_i^{(r-1)} = 1$, then $\beta_i^{(r)} = 1$ with probability $e^{-\lambda L(x)}$. In this case too, $B_i^{(r)}$ can take on only one value which is $B_i^{(r)} = \lfloor L(x)V \rfloor$.

The third case corresponds to the situation where $Z_{i-1}^{(r)} = x$ is such that $\beta_i^{(r-1)} = 1$ and $\beta_i^{(r)} = 0$. That is, S_i moves from talkspurt to silence between the two service instants $t_i^{(r-1)}$ and $t_i^{(r)}$. In this case, $B_i^{(r)}$ can take on the values in the range $0 \leq B_i^{(r)} \leq \lfloor L(x)V \rfloor$. The distribution on $B_i^{(r)}$ is computed as follows. Consider the time interval $(t_i^{(r-1)}, t_i^{(r)})$. Let t_c denote the time, as measured from $t_i^{(r-1)}$, at which the station in question made the transition from talkspurt to silence. Clearly, the

number of bits accumulated by S_i at time $t_i^{(r)}$ is $\lfloor t_c V \rfloor$. Hence,

$$\begin{aligned}
\Pr\{B_i^{(r)} = b \mid L(Z_{i-1}^{(r)}) = l, \beta_i^{(r-1)} = 1, \beta_i^{(r)} = 0\} \\
&= \Pr\{\lfloor t_c V \rfloor = b \mid L(Z_{i-1}^{(r)}) = l, \beta_i^{(r-1)} = 1, \beta_i^{(r)} = 0\} \\
&= \Pr\{b \leq t_c V < b+1 \mid L(Z_{i-1}^{(r)}) = l, \beta_i^{(r-1)} = 1, \beta_i^{(r)} = 0\} \\
&= \Pr\{b/V \leq t_c < (b+1)/V \mid L(Z_{i-1}^{(r)}) = l, \beta_i^{(r-1)} = 1, \beta_i^{(r)} = 0\}
\end{aligned} \tag{4.42}$$

Since the two-state model describing the activity of a speech source can make only a single transition between consecutive service instants, and the holding time in the talkspurt state is exponentially distributed with rate λ , (4.42) can be expressed as

$$\begin{aligned}
\Pr\{B_i^{(r)} = b \mid L(Z_{i-1}^{(r)}) = l, \beta_i^{(r-1)} = 1, \beta_i^{(r)} = 0\} &= \frac{e^{-\lambda b/V} - e^{-\lambda(b+1)/V}}{1 - e^{-\lambda l}} \\
&= \frac{e^{-\lambda b/V} (1 - e^{-\lambda/V})}{1 - e^{-\lambda l}} \\
&0 \leq b \leq \lfloor lV \rfloor
\end{aligned} \tag{4.43}$$

Given that $\beta_i^{(r-1)} = 1$, the probability that $\beta_i^{(r)} = 0$ is $1 - e^{-\lambda L(x)}$.

The last of the four cases is similar to the third except that, for this case, S_i makes a transition from silence to talkspurt, i.e., $\beta_i^{(r-1)} = 0$ and $\beta_i^{(r)} = 1$. Letting t_c denote the time at which S_i makes the transition from silence to talkspurt and using the same argument as for the third case, we have that

$$\begin{aligned}
\Pr\{B_i^{(r)} = b \mid L(Z_{i-1}^{(r)}) = l, \beta_i^{(r-1)} = 0, \beta_i^{(r)} = 1\} \\
&= \Pr\{l - (b+1)/V < t_c \leq l - b/V \mid L(Z_{i-1}^{(r)}) = l, \beta_i^{(r-1)} = 0, \beta_i^{(r)} = 1\} \\
&= \frac{e^{-\mu(l-b/V)} (e^{-\mu/V} - 1)}{1 - e^{-\mu l}}
\end{aligned} \tag{4.44}$$

Given that $\beta_i^{(r-1)} = 0$, the probability that $\beta_i^{(r)} = 1$ is $1 - e^{-\mu L(x)}$.

From these four cases, the elements of the transition matrix $\mathbf{R}_0$ can be expressed

as

$$[\mathbf{R}_0]_{xy} = \begin{cases} e^{-\mu L(x)} & \alpha_1 = 0, \beta_N = 0, y_N = 0 \\ e^{-\mu(L(x)-y_N/V)}(e^{\mu/V} - 1) & \alpha_1 = 0, \beta_N = 1, 0 \leq y_N \leq \lfloor L(x)V \rfloor \\ e^{-\lambda L(x)} & \alpha_1 = 1, \beta_N = 1, y_N = \overset{\circ}{\lfloor L(x)V \rfloor} \\ e^{-\lambda y_N/V}(1 - e^{-\lambda/V}) & \alpha_1 = 1, \beta_N = 0, 0 \leq y_N \leq \lfloor L(x)V \rfloor \\ 0 & \text{otherwise} \end{cases} \quad (4.45)$$

For the transition matrix $\mathbf{R}_1$, one must, in addition to the above, also account for the length of the data train $L_d^{(r-1)}$ which appears in the state descriptor $Z_1^{(r)}$ but not in the state descriptor $Z_N^{(r-1)}$. Assume that the length of the data train is an independent random variable drawn from some discrete distribution function which we denote by $F_L(l) \triangleq \Pr\{L_d^{(r)} = l\}$.^{*} It is then possible to compute the transition probabilities $[\mathbf{R}_1]_{xy}$ by conditioning on $L_d^{(r-1)}$ and removing this condition using the distribution specified by $F_L(l)$. Following the procedure used above, the elements of $\mathbf{R}_1$ are given by

$$[\mathbf{R}_1]_{xy} = \begin{cases} e^{-\mu(L(x)-l_x+l_y)} F_L(l_y) & \alpha_1 = 0, \beta_N = 0, y_N = 0 \\ e^{-\mu(L(x)-l_x+l_y-y_N/V)}(e^{-\mu/V} - 1) F_L(l_y) & \begin{cases} \alpha_1 = 0, \beta_N = 1 \\ 0 \leq y_N \leq \lfloor V(L(x) - l_x + l_y) \rfloor \end{cases} \\ e^{-\lambda(L(x)-l_x+l_y)} F_L(l_y) & \begin{cases} \alpha_1 = 1, \beta_N = 1 \\ y_N = \lfloor V(L(x) - l_x + l_y) \rfloor \end{cases} \\ e^{-\lambda y_N/V}(1 - e^{-\lambda/V}) F_L(l_y) & \begin{cases} \alpha_1 = 1, \beta_N = 0 \\ 0 \leq y_N \leq \lfloor V(L(x) - l_x + l_y) \rfloor \end{cases} \\ 0 & \text{otherwise} \end{cases} \quad (4.46)$$

^{*}This assumption does have its limitations. The length of the data train is not an independent random variable but depends on the past history of events on the channel such as the lengths of the previous data and voice trains.

Given the transition matrices $\mathbf{R}_0$ and $\mathbf{R}_1$, one can compute the transition matrix $\mathbf{P}_i$ which describes the state transition over the period between two consecutive service instants of S_i . Given $\mathbf{P}_i$, we can compute the stationary distribution Π_x of the Markov chain $\{Z_i^{(r)}\}$, $r \geq 1$. From Π_x , one can compute the marginal distribution of $B_i^{(r)}$ and hence the expected packet delay, as well as the expected fraction of lost speech for $B_i^{(r)} > B_{\max}$.

The marginal distribution of $B_i^{(r)}$ is denoted by $F_B(b)$ and can be computed from

$$\begin{aligned} F_B(b) &= \Pr\{B_i^{(r)} = b\} \\ &= \sum_{x \ni x_n = b} \Pi_x \end{aligned} \quad (4.47)$$

a) *Expected length of a round:* Given that S_i has accumulated b bits at time $t_i^{(r)}$, then the time interval $t_i^{(r)} - t_i^{(r-1)}$ is clearly b/V . Therefore, the expected length of a round is simply

$$\bar{L} = \sum_b F_B(b) b / V \quad (4.48)$$

b) *Expected Delay:* Since S_i transmits at most $B_{\max}$ bits, the expected delay of a voice bit is less than or equal to $\bar{L}$ and is given by

$$D = \sum_b F_B(b) \min(b, B_{\max}) / V \quad (4.49)$$

c) *Fraction of speech lost:* The fraction of speech from S_i that is discarded can be computed as the ratio of the expected amount of speech lost with each packet transmission to the expected size of a packet were there to be no loss. This ratio, denoted by G , is given by

$$G = \frac{\sum_b F_B(b) \max(0, b - B_{\max})}{\sum_b F_B(b) b} \quad (4.50)$$

d) *Actual Data Bandwidth*: The actual bandwidth achieved by data traffic, W_A , is computed as the ratio of the expected length of a data train to the expected length of a cycle. The expected length of a data train is given by $\sum_l F_L(l)l$ and the expected length of a cycle is given by (4.48). Hence,

$$W_A = \frac{V \sum_l F_L(l)l}{\sum_b F_B(b)b} \quad (4.51)$$

4.6.3 Computational Considerations

We provide some estimates of the storage and time requirements were one to use the above analysis to compute numerical values of the performance metrics. Basically, to compute any numerical results, one would have to compute the transition matrices $\mathbf{R}_0$, $\mathbf{R}_1$, and $\mathbf{P}_i$ and then (probably iteratively) compute the stationary distribution on $Z_i^{(r)}$. The storage and time requirements of these computations are directly related to the size of the state space which we crudely estimate as follows.

Consider the state descriptor $Z_i^{(r)} = (B_{i+1}^{(r-1)}, \beta_{i+1}^{(r-1)}; \dots, B_i^{(r)}, \beta_i^{(r)}; L_d^{(r-1)})$. We ignore the space requirements of the N indicators β . Assume that we do not consider the possibility of an overload and hence, we conservatively estimate the maximum value of $B_i^{(r)}$ to be $B_{\max}$. In addition, we assume that the data train is a constant fixed length. Under these assumptions, the state descriptor is reduced to a vector of size N where each element can take on values in the range $(0, B_{\max})$. With such a state descriptor, the size of the state space is given by

$$S = (B_{\max} + 1)^N. \quad (4.52)$$

We would like that the maximum number of active users on the network be close to the maximum that can be accommodated for a given channel bandwidth.

Consider for this crude estimate values of N in the range $(1, W/V)$ which, assuming that half of the speech constitutes silence, allows approximately 50 percent of the bandwidth for data stations. Since $B_{\max}$ is given by $\lfloor V D_{\max} \rfloor$, (4.52) becomes

$$S \approx (V D_{\max})^{W/V} \quad (4.53)$$

Consider some favorable numerical values. We believe that it would not be worthwhile to consider values of W less than 1Mb/s or values of $D_{\max}$ less than 1ms. In fact, it is questionable whether the values of 1Mb/s and 1ms for W and $D_{\max}$, respectively, are not themselves too low for realistic applications. Nevertheless, we consider as a numerical example $V = 64\text{kb/s}$, $W = 1\text{Mb/s}$, and $D_{\max} = 1\text{ms}$. With these values, the size of the state space is on the order of 2×10^{23} . Clearly, any results requiring this order of numerical computation are unattainable.

Thus, unfortunately, the analysis is not useful for providing numerical results. However, it serves as a formal description of the events on the network.

4.7 Numerical Results

In this section, we present and discuss numerical results showing the performance of the voice/data access protocol. Since detailed performance results for data traffic have been presented in chapter 3, we consider here only the performance achieved by voice traffic and its effect on the data bandwidth. For the case where there is no silence suppression, results were obtained from the analysis of the deterministic system presented in section 4.5. For the case where silence suppression is in effect, the numerical results were obtained by simulating the model described and analyzed in section 4.6.*

*The model of a voice source used in the simulation corresponds to the Markov chain model described

All the numerical results were obtained under the assumption that the data train is of a fixed length which is computed from (4.12) for each combination of the values of W , W_d , and $D_{\max}$ considered below. The vocoder rate is the same for all stations and is taken to be 64Kb/s. The end-to-end propagation time τ is taken to be $20\mu\text{s}$. The carrier detection time is ignored, i.e., $\Delta = 0$, and the preamble and header are taken to be a combined total of 60 bits. For $D_{\max}$, W and W_d , a range of values are considered: $D_{\max}$ is taken to be one of 1ms, 10ms, or 100ms; W is taken to be 1Mb/s, 10Mb/s or 100Mb/s; W_d takes on values in the range 0 to $0.7W$. For the case where silence suppression is in effect, holding times in the talk and silent states are taken from [52] to be $1/\lambda = 1.34\text{s}$ and $1/\mu = 1.67\text{s}$, respectively. The length of the simulation runs were varied depending on the bandwidth of the network being simulated. It was found that, at 100Mb/s, a 40s segment of conversation was sufficient to give stable results. At 1Mb/s, fewer stations can be accommodated on the network and so a longer segment of conversation was needed so as to collect a sufficient number of samples. In this case, a 100s segment of conversation was simulated.

The first performance metric that we discuss is the delay of a voice packet. We plot in Fig. 4.6 the delay as a function of the number of active voice sources for the case where there is no data traffic, i.e., $W_d = 0$. Consider first the curves corresponding to a system with no silence suppression. For $N = 1$, D is only slightly larger than the round-trip propagation time plus the time to transmit the preamble and header. As N increases from $N = 1$, the delay increases, slowly at first but rising sharply as N approaches $N_{\max}$. For $N > N_{\max}$, the delay is $D_{\max}$ and

in section 4.6 without the modification restricting the number of transitions between consecutive service instants at a given station. Nevertheless, we found that, for small values of $D_{\max}$ (1ms and 10ms), there were in fact no cases where the talker made more than one transition between consecutive service instants. For larger values of $D_{\max}$ (100ms), there were a few cases where a talker made two or three transitions between consecutive service instants but the numbers were negligible compared to the number of packets simulated.

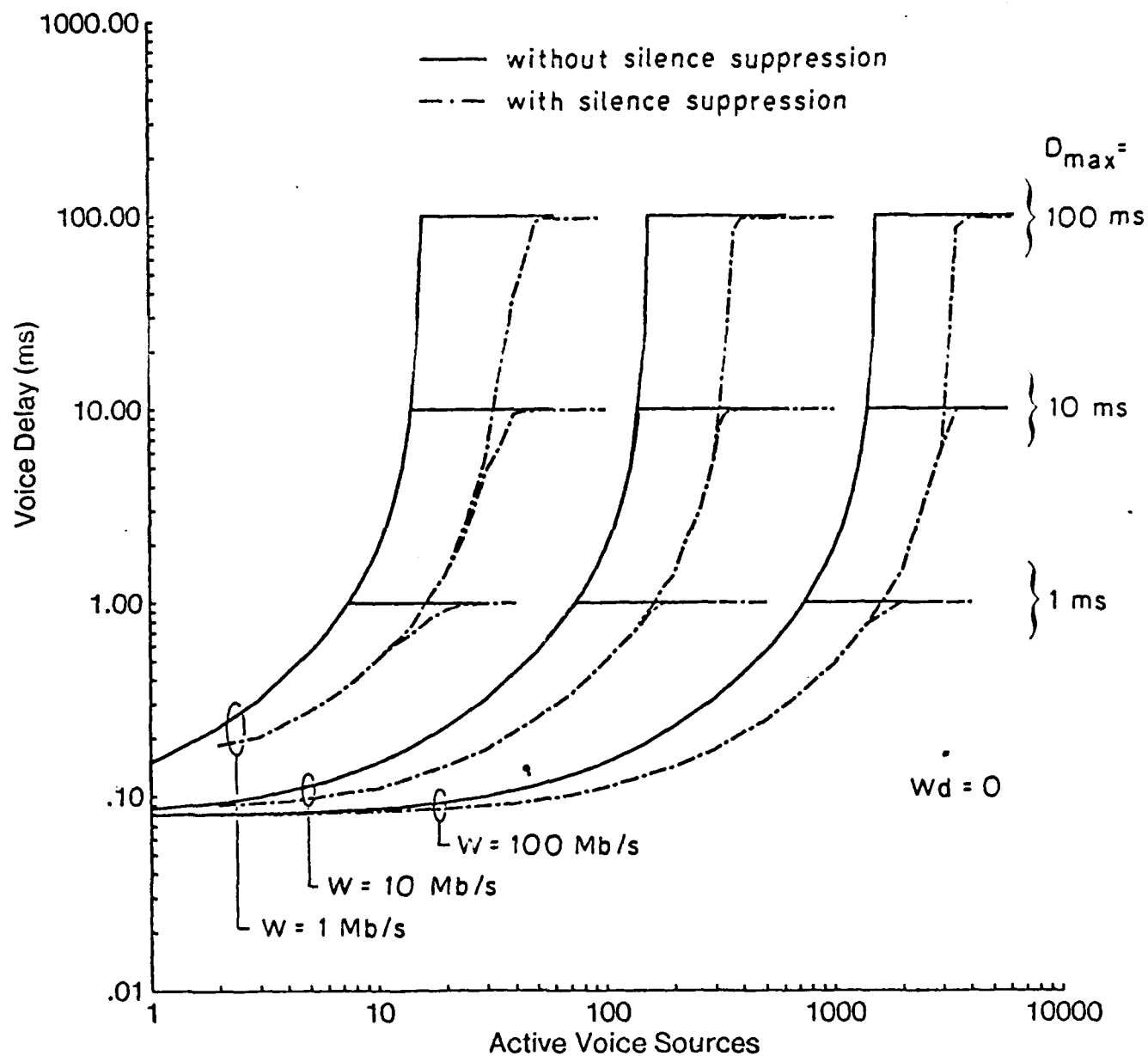


Fig. 4.6 Voice delay as a function of the number of active voice sources N with various values of W and $D_{\max}$, and $W_d = 0$. Curves corresponding to both a system with silence suppression and one without silence suppression are shown.

	$D_{\max} =$ 1ms	$D_{\max} =$ 10ms	$D_{\max} =$ 100ms	Theoretical Limit ($\lfloor W/V \rfloor$)
$W=1\text{Mb/s}$	7	14	15	15
$W=10\text{Mb/s}$	74	141	154	156
$W=100\text{Mb/s}$	741	1417	1546	1562

Table 4.1. $N_{\max}$ for various values of W and $D_{\max}$. $W_d = 0$. The theoretical maximum value of N , given by the bandwidth constraint $\lfloor W/V \rfloor$, is also indicated.

each curve becomes a horizontal line, hence the discontinuity at $N = N_{\max}$. The curves show that, for surprisingly low delay (on the order of 10ms), the network can support close to the theoretical maximum number of voice sources possible given the bandwidth constraints, i.e., the maximum number of voice sources given by $\lfloor W/V \rfloor$. Table 4.1 provides some values of $N_{\max}$ for various values of W and $D_{\max}$ and compares these to the theoretical maximum. We see that, for $D_{\max} = 10\text{ms}$, $N_{\max}$ is about 90% of $\lfloor W/V \rfloor$ and for $D_{\max} = 100\text{ms}$, $N_{\max}$ is almost 100% of $\lfloor W/V \rfloor$. The value of $N_{\max}$ grows proportionally to W . (The reader should bear in mind that these results depend on the particular values of τ , H and Ω that have been used. If these constants are increased, these percentages would decrease.) Thus, for practical purposes, there would be no advantage to a value of $D_{\max}$ that was much greater than 10ms. This also means that realistic values of $D_{\max}$ correspond to the ones for which the modification made to the talker model, i.e., only a single transition between consecutive service instants at a given station, is justified.

A comparison of the curves in Fig. 4.6 of delay versus N with and without silence suppression shows the advantages of silence suppression. For a given value of D , the value of N when silence suppression is in effect is greater than that when silence suppression is not in effect. In Table 4.2, we compare some representative values of N that lead to a given delay. We see that, when silence suppression is in effect, the number of sources that leads to a given delay is consistently greater than

W Mb/s	D ms	N with no suppression	N with suppression	Factor Increase	TASI Advantage
1	0.559	5	11	2.20	1.73
1	1.46	9	20	2.20	
10	0.559	50	108	2.16	2.07
10	1.46	90	202	2.20	
100	0.559	500	1106	2.20	2.09
100	1.46	900	2006	2.20	

$$D_{\max} = 100\text{ms} \quad W_d = 0$$

Table 4.2. Comparison of silence suppression and no silence suppression showing the number of active sources required for a given packet delay. The factor increase in N for TASI is also shown.

when there is no silence suppression by a factor of 2.2 which, as expected, is the value of $1/p_t$. (Recall that p_t is the probability of a talker being in the talk state.) For these values of D and W , there is no loss of speech due to clipping. If larger values of D are used such that clipping does occur for the silence suppression case, the factor increase in N is greater than $1/p_t$. The last column in Table 4.2 shows the factor increase in N that can be expected in a TASI system with $G < 1\%$.* Clearly, silence suppression in the packet voice system provides more efficient multiplexing of the voice channels than does TASI.

In Fig. 4.7, we plot D versus N for the same values of W and $D_{\max}$ as in Fig. 4.6. However, Fig. 4.7 shows results for the case where $W_d = 0.3W$. We see in this figure that the delay is never less than $0.3D_{\max}$ since there is a data train of this duration between consecutive packets from a given station. As a result, the curves in Fig. 4.7 are higher than those in Fig. 4.6. In addition, for $N \ll N_{\max}$, the curve is flatter with $W_d = 0.3W$ as compared to the curves with $W_d = 0$ since, for low values of N , the dominant component of the delay is the length of the data train which is a fixed

*In order to achieve a value of G that is less than 1%, practical TASI should satisfy the condition $Np_t + \sqrt{Np_t} < c$ where c denotes the number of available channels [56]. For Table 4.2, c is computed from $c = \lfloor W/V \rfloor$.

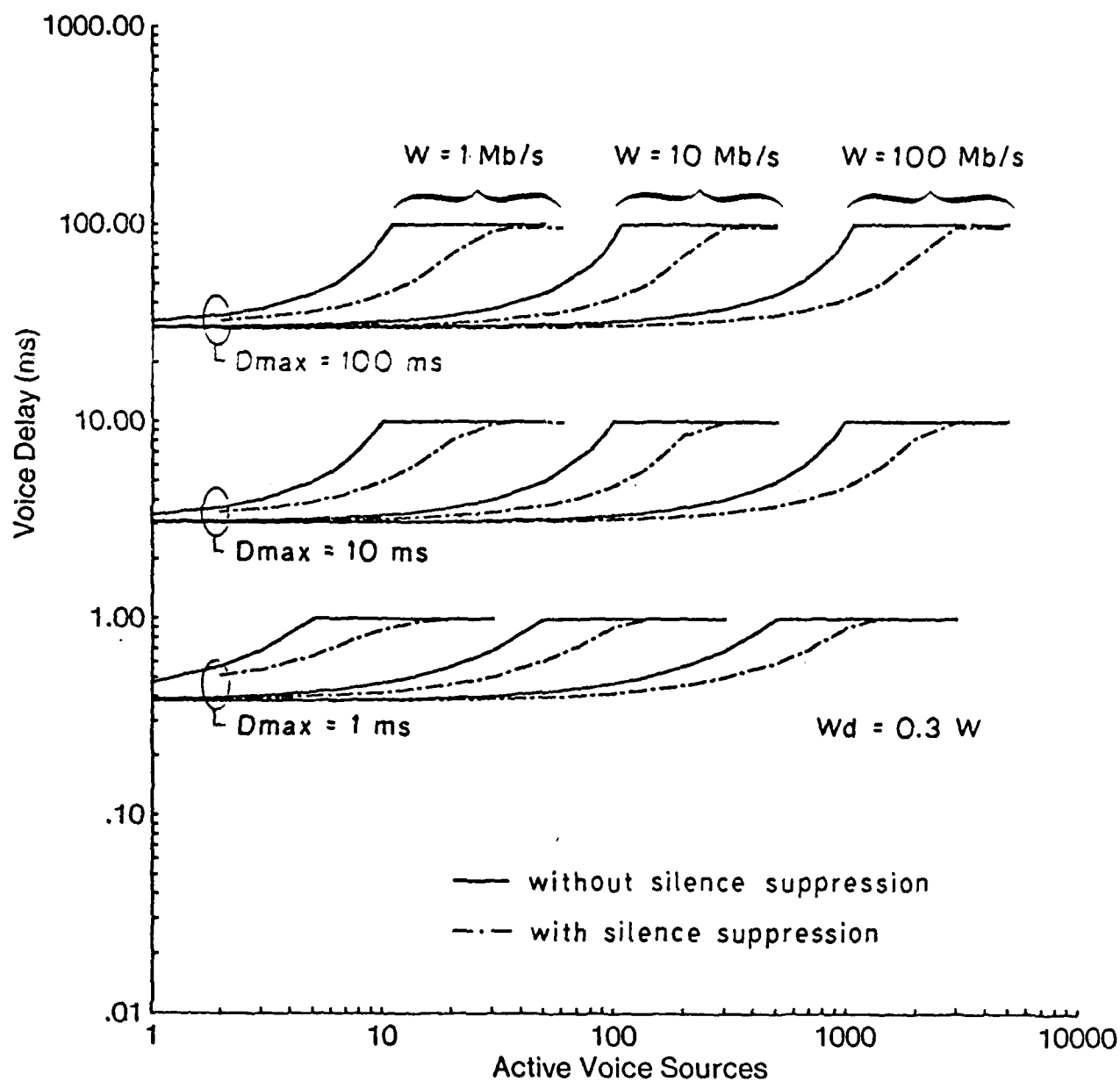


Fig. 4.7 Voice delay as a function of N with various values of $D_{\max}$ and W , and with $W_d = 0.3W$. Curves corresponding to both a system with silence suppression and one without are shown.

value. Furthermore, since 30 percent of the bandwidth is being used by data traffic, $D_{\max}$ is reached with fewer active stations than when all the bandwidth is available to voice traffic. The effect of silence suppression with $W_d = 0.3W$ is similar to that when $W_d = 0$. That is, the curves appear to be shifted towards the right and there is no particular value of N that results in a discontinuity in D .

To see more clearly the effect of data traffic on the voice delay, we show in Fig. 4.8 D versus N for various values of W_d . Only the case where $W = 100\text{Mb/s}$ and $D_{\max} = 10\text{ms}$ is shown. For a given N , D increases as W_d increases because the length of the data train is increased. Also, the value of $N_{\max}$ that is approached as $D \rightarrow D_{\max}$ becomes smaller as W_d is increased since a greater fraction of the bandwidth is being consumed by data traffic. (For $D_{\max} = 10\text{ms}$, $W = 100\text{Mb/s}$, and the case where there is no silence suppression, $N_{\max}$ falls from 1417 at $W_d = 0$ to 417 at $W_d = 70\text{Mb/s}$.)

The above results have described the characteristics of the average packet delay. It is also interesting to examine the variance of the packet delay. When no silence suppression is in effect, the voice delay is deterministic and hence, the variance of delay is zero. With silence suppression, the delay becomes stochastic. We plot in Fig. 4.9 the variance of delay of a voice packet as a function of N with $W_d = 0$ for the case where silence suppression is in effect. As can be seen in the figure, the variance is small when N is small, reaches a peak as the network approaches its maximum load carrying capacity, and then drops of to a small but constant value as $D \rightarrow D_{\max}$. This is explained as follows. As N increases, the fluctuations in the number of transmissions in a round increase, the length of a packet becomes more random, and the length of a round has greater variance. Hence the variance of delay increases. As the network becomes overloaded and clipping occurs, the delay of the voice samples that are transmitted becomes equal to $D_{\max}$ with probability 1. As

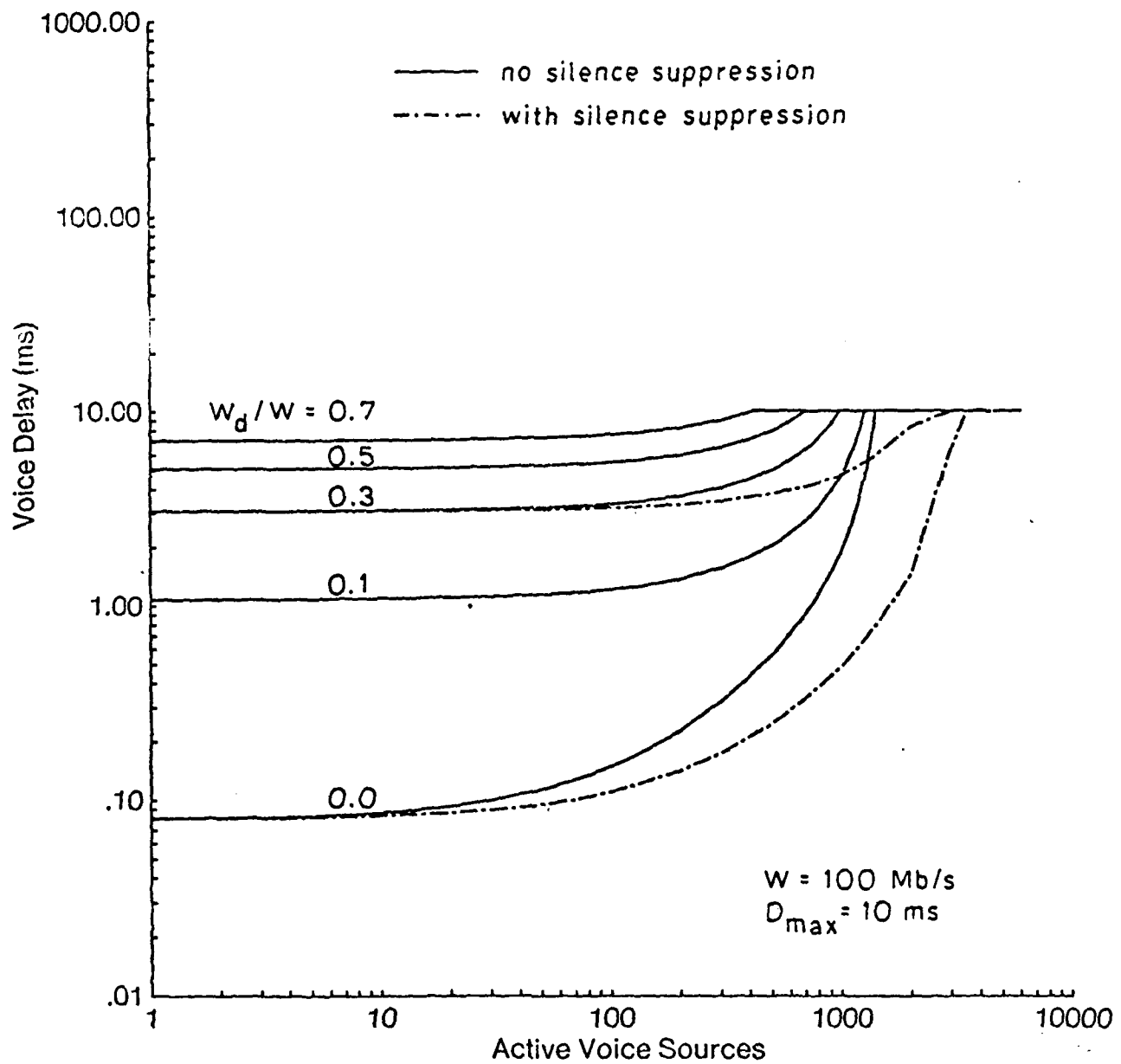


Fig. 4.8 Voice delay versus N with $W = 100\text{Mb/s}$, $D_{\max} = 10\text{ms}$, and various values of W_d showing the effect of data traffic on the voice delay.

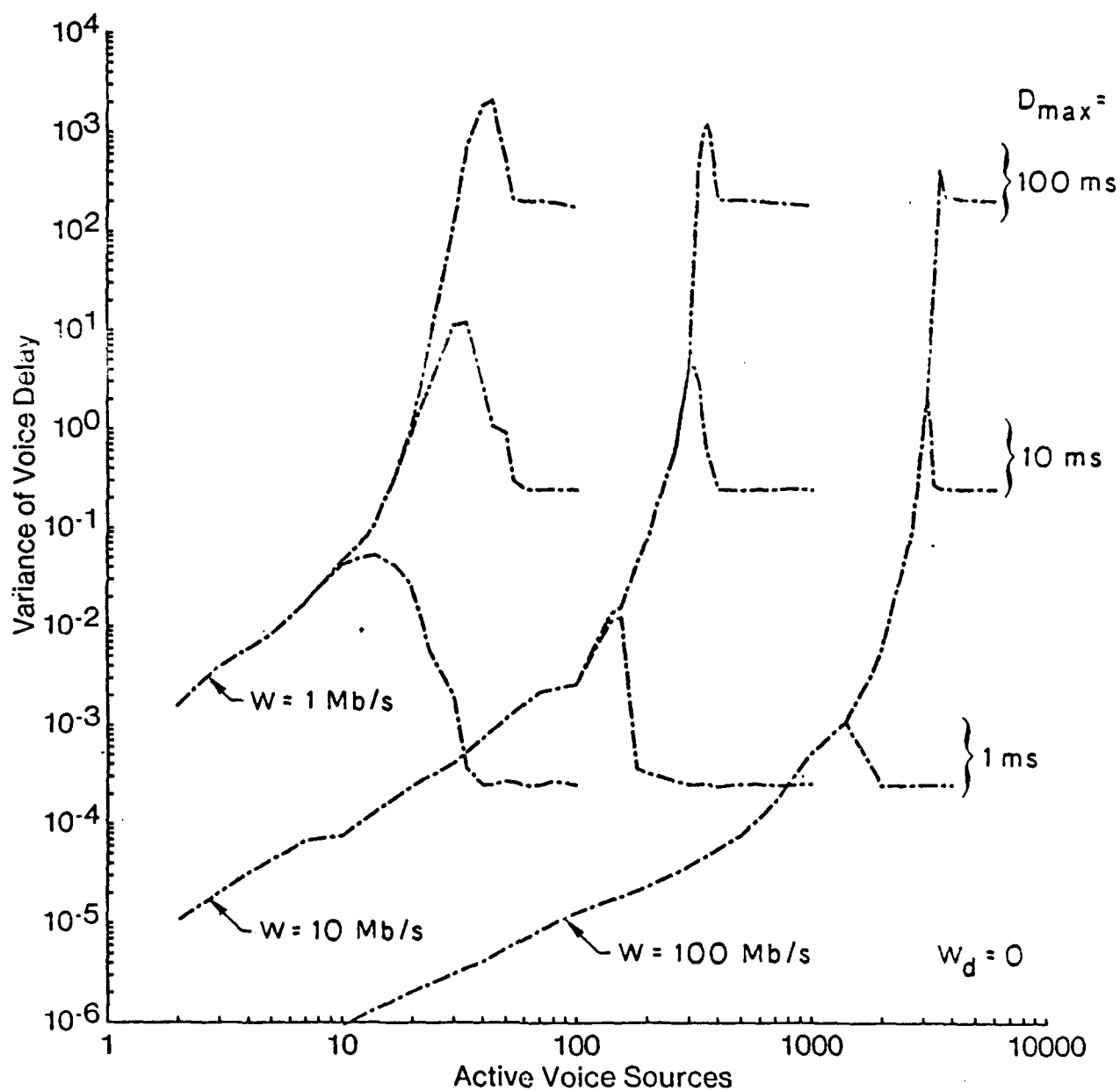


Fig. 4.9 Variance of delay as a function of N with $W_d = 0$. The curves show the case where silence suppression is in effect. If there is no silence suppression, the variance is zero.

	$N_{\max}$	N			% Increase Over $N_{\max}$		
		1% Loss	2% Loss	5% Loss	1% Loss	2% Loss	5% Loss
$W = 10, D_{\max} = 1$	74	75	76	78	1.4	2.7	5.4
$W = 10, D_{\max} = 10$	141	143	145	149	1.4	2.8	5.7
$W = 10, D_{\max} = 100$	154	156	158	163	1.3	2.6	5.8
$W = 100, D_{\max} = 1$	741	750	758	784	1.2	2.3	5.8
$W = 100, D_{\max} = 10$	1417	1432	1446	1492	1.1	2.0	5.3
$W = 100, D_{\max} = 100$	1546	1563	1578	1628	1.1	2.1	5.3

Table 4.3. Comparison of increase in N for varying degrees of clipping and various values of W and $D_{\max}$.

a result the variance should drop to zero. However, there remains some variance of delay which accounts for the random delay of packets comprising the beginnings and ends of talkspurts.

We now examine the effect of overloading the network which can occur by allowing too many voice sources to be active. The overload is measured by the fraction of speech lost due to clipping. Most previous work in this regard has been applied to TASI systems [56], where a speech loss of 0.5 percent is regarded as acceptably small [57]. We note that, with TASI, all speech loss occurs at the beginning of a talkspurt. In a packet speech system such as the one investigated in this work, we expect that multiple packets from the the same talkspurt would be clipped. Hence we expect that, as compared to a TASI system, a packet speech system could tolerate a greater fraction of speech loss for the same subjective degradation in the quality of the received speech signal. Jayant and Christensen claim that, depending on the coding scheme used, losses around 2 percent are acceptable [54].*

*Another measure of the degradation of the speech signal which has been used in the literature is the size of a glitch. In [57] it is reported that, with respect to glitches, "high quality" speech can be maintained if individual glitches are kept below 50ms. A ballpark measure of the glitch size can be obtained from the fraction of speech lost. For losses of say 2% and packets comprising 100ms of speech, the glitch size is on the order of 2ms. As a result, the size of a glitch is limited to a negligible value, for practical purposes, by the tolerable level of lost speech. This is because we have assumed that each clipped talkspurt contains multiple small glitches as opposed to a single large one. Thus in this work, the size of a glitch is not considered to be a useful measure of speech quality.

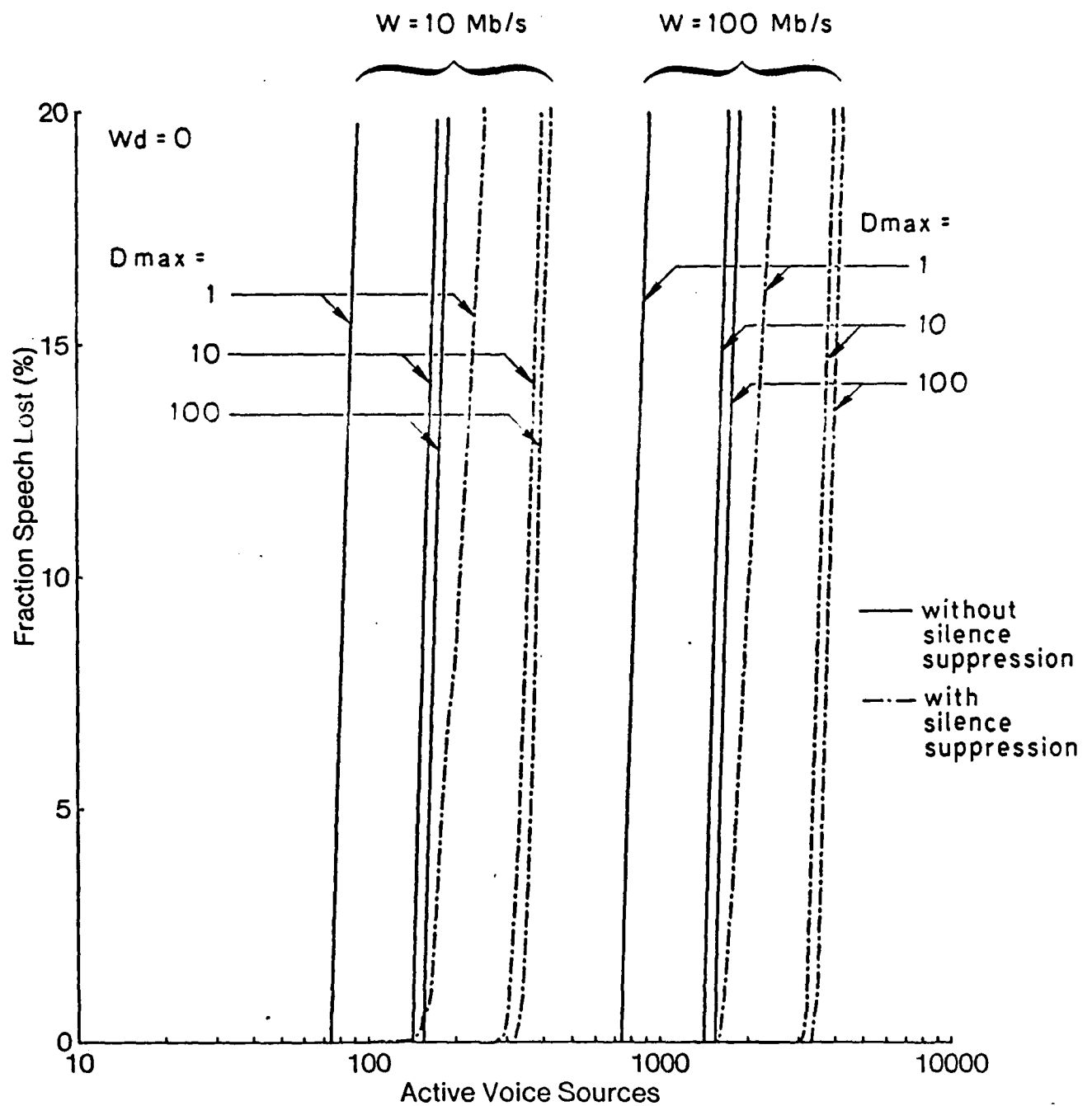


Fig. 4.10 Fraction of speech lost due to clipping versus N with $W_d = 0$.

In Fig. 4.10, we plot the fraction of speech lost G as a function of N for $W_d = 0$, $W = 10\text{Mb/s}$ and 100Mb/s , and $D_{\max} = 1\text{ms}$, 10ms and 100ms . For the case where there is no silence suppression, the curves show that there is a sharp rise in G from zero as N increases from $N = N_{\max}$, which indicates that even a few excess voice sources will cause unacceptable speech loss. Table 4.3 shows that an excess of voice sources which is on the order of 1, 2, or 5 percent of $N_{\max}$ leads to a speech loss which is on the order of 1, 2, or 5 percent, respectively. Thus, there is very little gain in N with acceptable amounts of speech loss. However, a few voice sources in excess of $N_{\max}$ can be tolerated. When silence suppression is in effect, it is possible to have more than $N_{\max}$ sources on the network without incurring any speech loss. As N is increased (beyond $N_{\max}$) so that clipping begins to occur, G increases from zero, gradually at first but then as steeply as in the case where there is no silence suppression. We note that, even with silence suppression, there is a non-zero probability that clipping will occur for $N > N_{\max}$. Thus to guarantee zero speech loss, we require that $N \leq N_{\max}$ regardless of whether silence suppression is used or not. However, Fig. 4.10 shows that, when silence suppression is in effect, there is negligible speech loss for values of N up to about $2N_{\max}$. An expanded view of the critical region of Fig. 4.10 is presented in Fig. 4.11.

Fig. 4.10 also shows the effect of $D_{\max}$ on G . As $D_{\max}$ is increased, a larger value of N can be achieved for the same loss rate. But as we noted before, increasing $D_{\max}$ above about 10ms results in a diminishing gain in N . The effect of W_d on G is shown in Fig. 4.12. Increasing W_d causes a shift of the curve to the left, meaning that, as expected, fewer voice sources can be accommodated.

Lastly, we examine the effect of voice sources on the bandwidth available to data traffic under the assumption that the length of the data train is always the maximum permissible. In Fig. 4.13, we plot W_A versus N with $W_d = 0.3W$ and

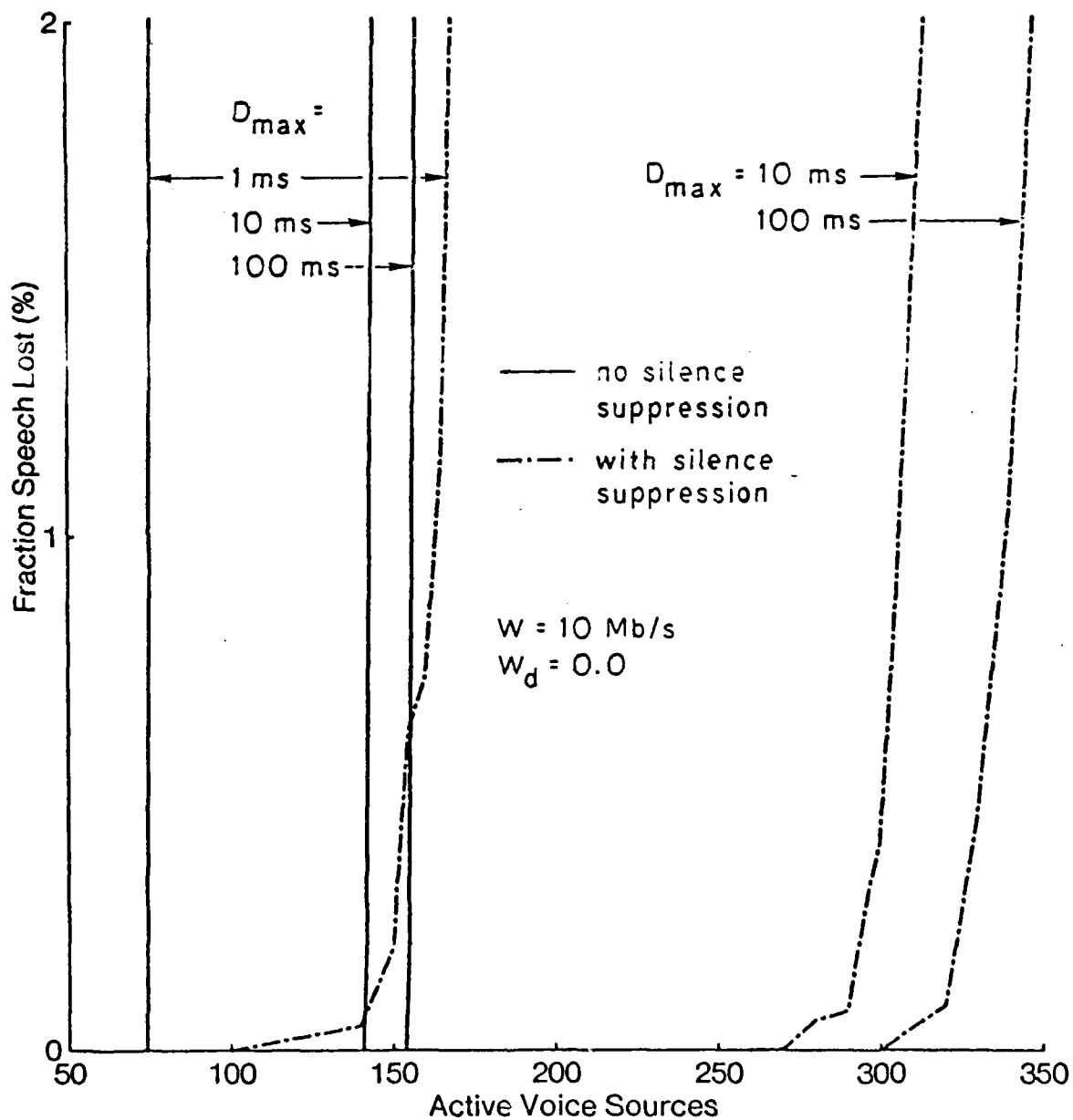


Fig. 4.11 Fraction of speech lost due to clipping versus N with $W = 10\text{Mb/s}$ and $W_d = 0$ showing the critical region where clipping begins to occur.

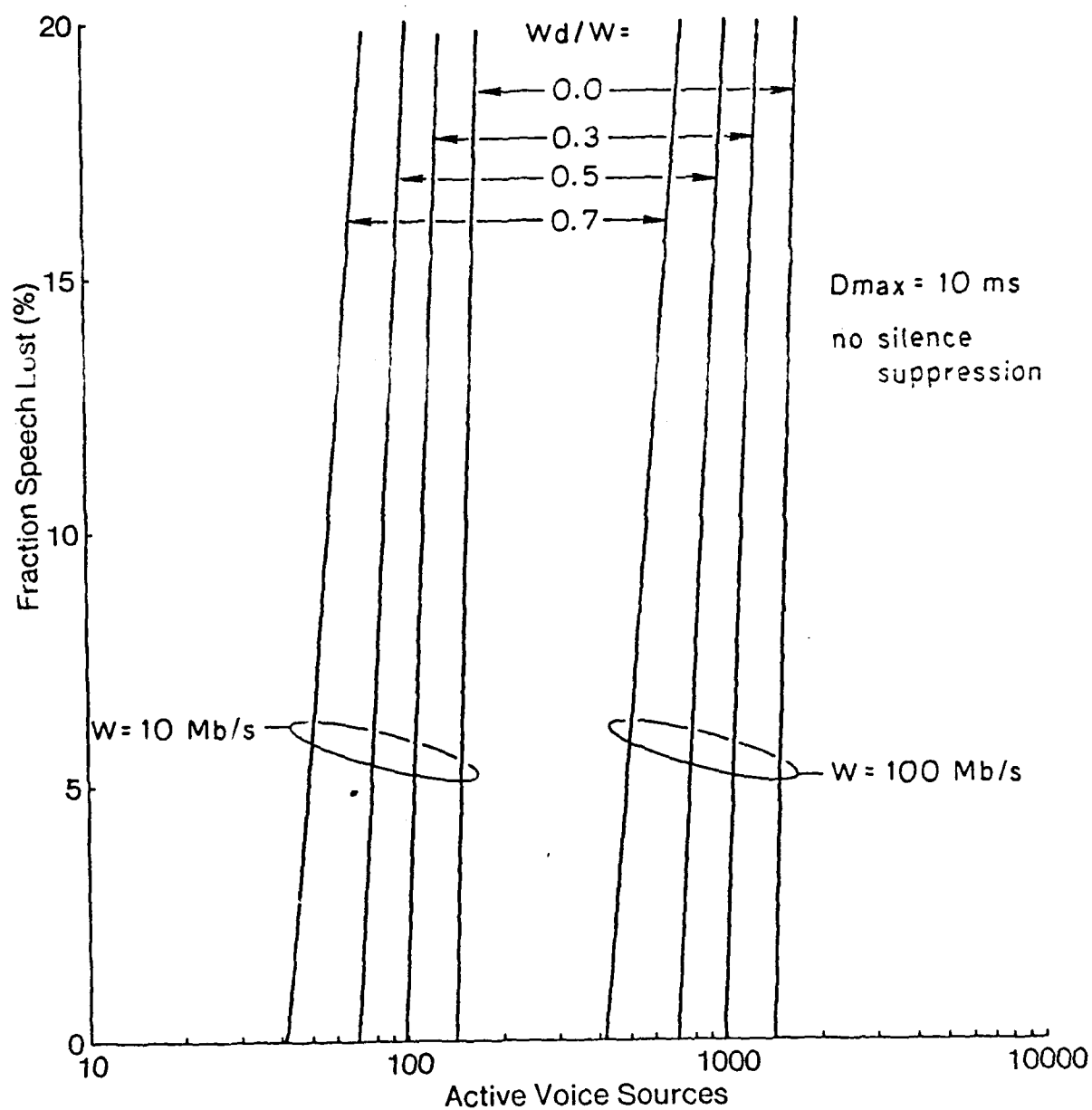


Fig. 4.12 Fraction of speech lost due to clipping versus N with $D_{\max} = 10\text{ms}$. The curves show only the case where there is no silence suppression.

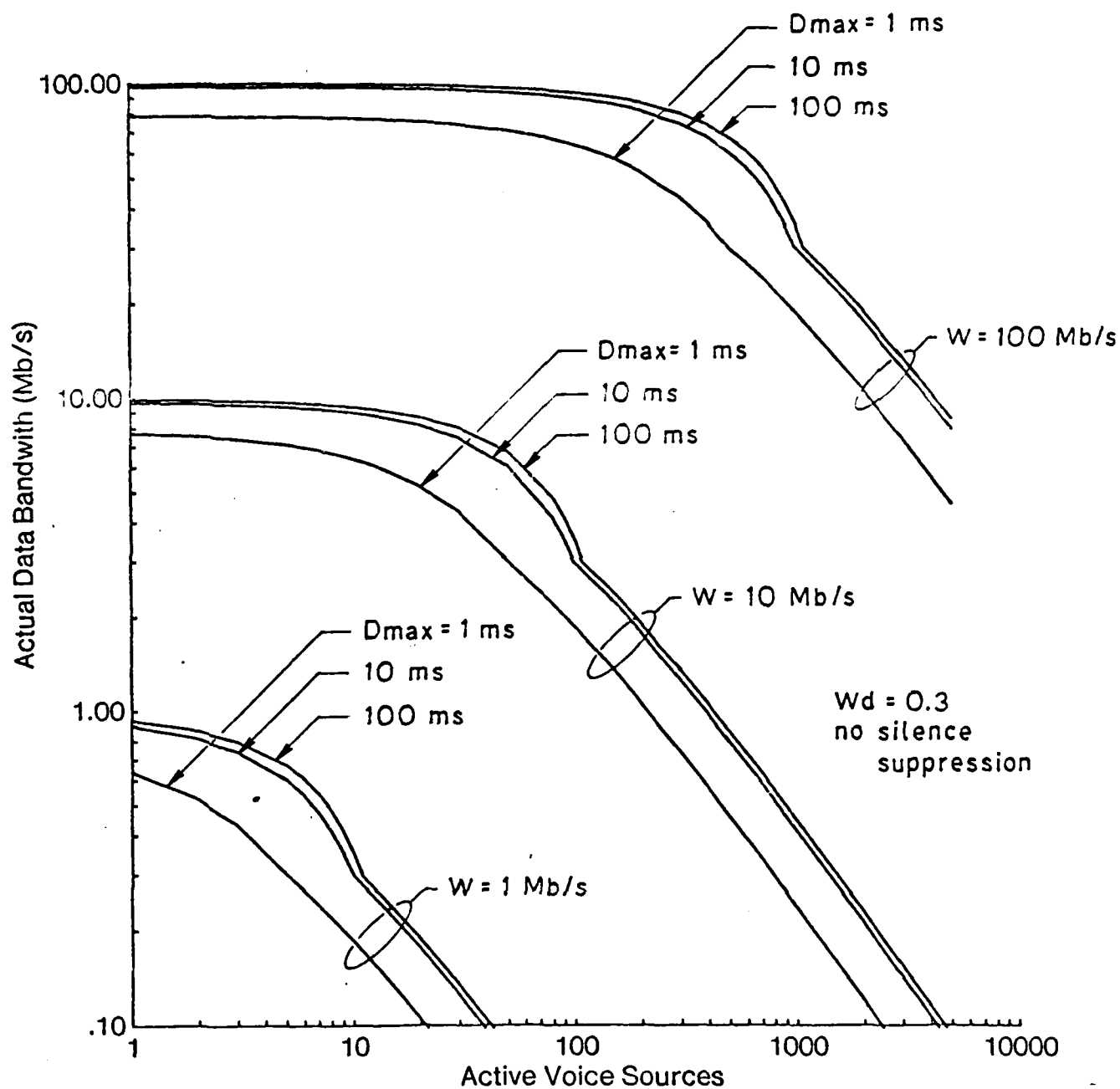


Fig. 4.13 Actual bandwidth achieved by data traffic as a function of N with $W_d = 0.3W$ and various values of W and $D_{\max}$. We only show curves for the case where no silence suppression is in effect.

various values of W and $D_{\max}$. We only show curves for the case where there is no silence suppression. The curves for the case where silence suppression is in effect are similar. As expected, Fig. 4.13 shows that, if N is small, W_A is close to W and the data traffic is able to make use of the total bandwidth. As N increases, some of the bandwidth is consumed by voice traffic and W_A decreases. At $N = N_{\max}$, W_A is equal to the minimum bandwidth required, W_d . Increasing N beyond $N_{\max}$ causes W_A to decrease below the specified minimum. Thus, allowing too many voice sources on the network degrades the performance achieved by data stations.

4.8 Summary

In this chapter, we described an access protocol for integrating voice and data on a local area network with round robin service. The protocol was designed to satisfy the minimum delay requirements of voice traffic and, at the same time, to guarantee a minimum bandwidth to data stations. In addition, the available bandwidth in excess of the minimum allocated is automatically and dynamically made available to data traffic when it is not being used by voice sources. The method by which a voice station forms a packet is unique in that the time spent in forming a packet is incurred concurrently with the time waiting to access the network. This is achieved by allowing variable size packets to be transmitted. The packet size is automatically adjusted to the load on the network, eliminating concerns regarding the optimal packet size.

A mathematical model of the system was developed and analyzed to provide a quantitative evaluation of the performance that voice sources could expect from the network. We showed that the network is able to multiplex a large number of voice sources onto the channel efficiently. Delay constraints can be not only realistic

but desirably low. In addition, we showed that data traffic was able to capture any bandwidth not being used by the voice sources. To guarantee the delay constraint and minimum bandwidth to data traffic in the deterministic system, the number of active voice sources at any time must be limited to $N_{\max}$. Even a few voice stations in excess of $N_{\max}$ (a one or two percent increase in N over $N_{\max}$ or more) causes unacceptable speech loss due to clipping and, in addition, results in data stations achieving less of the bandwidth than the specified minimum.

A comparison of simulation results showing the performance of the stochastic system with analytical results showing the performance of the deterministic system indicates that, for a given packet delay and channel bandwidth, a larger number of voice sources can be accommodated if silence suppression is in effect since each voice source has a lower bandwidth requirement on the average. For the stochastic system, there is no definitive value that N must be limited to. However, the results indicate that approximately $2N_{\max}$ voice sources can be accommodated with negligible speech loss.

Chapter 5

Conclusions and Future Research

5.1 Conclusions

In this work, we investigated and evaluated a new class of local area network proposals, suitable for broadcast bus systems that have been recently proposed in the literature. We refer to these schemes as Demand Assignment Multiple Access (DAMA) Schemes. In chapter 2 we have identified three basic access mechanisms according to which it has been possible to classify these schemes. These are the scheduling delay access mechanism, the reservation access mechanism, and the attempt-and-defer access mechanism. We described these access mechanisms together with the various network configurations proposed for their implementation. Then, for each of the basic access mechanisms, we discussed those schemes belonging to that class.

With this classification, we were able to present the numerous network schemes in a unified manner and identify characteristics by which they may be similar or may differ. We have also suggested that characteristics of some schemes can be seen

as variations or special cases of another, and that a feature proposed for one could, in some cases, be applied to another.

We also showed in chapter 2 how new network schemes evolved from the techniques used in existing schemes or schemes that had been previously proposed. This leads us to believe that the classification will be useful to facilitate the development of improved access protocols based on the concepts, and understanding of the pertinent issues, which can be gleaned from this unified presentation of the recent network proposals.

To evaluate in detail the performance of DAMA schemes, we analysed in chapter 3 the three service disciplines NGSS, GSS, and HOLS operating in two schemes, Expressnet and Fasnet. The analysis incorporated some novel recursive techniques for calculating the probabilities of various events which could potentially be applied to the analysis of other semi-Markov chains. Numerical results showed that these systems can achieve a channel utilization close to 100 percent even when the channel bandwidth is high or the propagation delay of the signal over the network is large. The throughput-delay characteristics are excellent and the maximum delay is bounded

Lastly, we applied these DAMA schemes to the communication needs of real-time speech by investigating an integrated voice and data access protocol for Expressnet. The access protocol that was proposed is unique in that it automatically and dynamically adjusts the size of a voice packet to match the load on the network thereby forcing the packet formation delay and the network access delay to be incurred concurrently. The system was mathematically modelled and analyzed. Numerical results showed that the network is able to multiplex a large number of voice sources onto the channel efficiently, delay constraints can be realistic and desirably low, and data traffic is able to capture any bandwidth not being used by

voice sources.

In summary, many of the implicit token DAMA schemes that have been proposed overcome the performance limitations of random access schemes and therefore are well suited to high bandwidth environments. Due to the round robin nature of the service disciplines, they guarantee an upper bound on the packet delay. This attribute, together with the capability to achieve high utilization of the bandwidth, makes them suitable for the integration of a large number of traffic types, including those with real time constraints.

5.2 Suggestions for Additional Research

Several areas of related research have been untouched in this work. In the performance analysis presented in chapter 3, we made the assumption that each station had only a single-packet buffer. It would be interesting to extend the analysis and numerical results of chapter 3 to accommodate the case where stations have multiple-packet buffers. With this change to the model, the specification of the state space requires a vector of size M (where M denotes the number of stations on the network) in order to fully describe the state of the system at the embedded points of the embedded Markov chain. The i^{th} element of this vector denotes the number of packets in the buffer of station S_i .

Although this approach to a solution appears to be feasible, the mathematics would be extremely cumbersome. In addition, since the state space is large for reasonable values of M ,* this method will not be useful for numerical results. A

*The size of the state space is given by M^B where B is the buffer size. Even with $B = 2$ and $M = 10$, the size of the state space is 100. In computing numerical results from the analysis in chapter 3, we were limited to $M = 50$ which corresponds to a state space of size 50.

simulation, on the other hand, appears to be a viable alternative. With simulation, one could also investigate different distributions on the generation rate of packets, as well as different packet generation rates for each of the stations.

We can identify two issues associated with voice communication on a local area network that have not been addressed. The first is peculiar to Expressnet and concerns the voice admission policy, i.e., the method whereby a new station is prevented from placing a call if the network is already supporting the maximum allowable number of active voice stations. In any implementation that is in keeping with the fully distributed spirit of Expressnet, a distributed algorithm must be devised, perhaps requiring the explicit exchange of information, that enables a voice station to determine the number of active voice sources at any time. Alternatively, a voice station could estimate the number of active voice sources from the activity on the channel. If silence suppression is not in effect, a station can make an accurate estimate by merely counting the number of packets in a voice train. If silence suppression is in effect, various techniques for estimating the number of active voice sources from the length of the voice train and the model of a talker's activity could be investigated to determine their accuracy, and to evaluate the consequences of the error in the estimates.

The second issue pertaining to voice communication on packet switched networks concerns the method by which packets are decoded at the receiver to recreate the speech signal. Recall that the inter-arrival times of packets to the receiver are random. The receiver must buffer packets as they arrive and play them back in such a way so as to best preserve the continuous nature of the original speech while minimizing the buffering delay at the receiver. Some issues to be addressed concern the required size of the buffer, the action to be taken if the buffer is emptied before a new packet arrives, and the compensation, if any, to be made to accommodate

clipped packets. Some researchers have already addressed the issues pertaining to signal reconstruction techniques [58], [59], [60]. However, additional effort in this area is required, especially with regard to the peculiarities of the voice/data access protocol described in chapter 4.

Looking beyond the integration of voice and data traffic, we expect the network to satisfy the communication needs of other services as well. It would be useful to investigate the ability of DAMA networks to carry multiple types of traffic. Questions that one might ask include: what other traffic types might one expect the network to carry; what are the delay constraints and bandwidth requirements of these; is it sufficient to describe a traffic type by its delay constraint and bandwidth requirement or are additional attributes required? Having decided on the traffic types, a method whereby these can be most efficiently integrated must be devised. A set of rules to configure the access protocol so as to accommodate an arbitrary number of traffic types would be extremely useful in the design of a local area network.

Bibliography

- [1] L. G. Roberts, "Data by the packet," *IEEE Spectrum*, Feb. 1974.
- [2] A. S. Tanenbaum, *Computer Networks*, Prentice Hall, 1981.
- [3] D. D. Clark, K. T. Pogran, D. P. Reed, "An introduction to local area networks," *Proceedings of the IEEE*, vol. 66, no. 11, Nov. 1978.
- [4] W. Stallings, "Local Network Performance," *IEEE Communications magazine*, vol. 22 no. 2, February 1984.
- [5] K. Kummerle and M. Reiser, "Local area communication networks—an overview," *Journal of Telecommunication Networks*, vol. 1, no. 4, Winter 1982.
- [6] F. A. Tobagi, "Multiaccess protocols in packet communication systems," *IEEE Transactions on Communications*, vol. COM-28, no. 4, April 1980.
- [7] L. Kleinrock and F. A. Tobagi, "Packet switching in radio channels: Part I—Carrier sense multiple access modes and their throughput delay characteristics," *IEEE Trans. Commun.*, vol. COM-23, Dec. 1975.

- [8] R. M. Metcalfe and D. R. Boggs, "Ethernet: Distributed packet switching for local computer networks," *Communications of the ACM*, vol. 19, no. 7, pp. 395-403 (1976).
- [9] F. Tobagi and V. Bruce Hunt, "Performance analysis of carrier sense multiple access with collision detection," in *Proc. Local Area Commun. Network Symp.*, Boston, MA, May 1979.
- [10] L. Fratta, F. Borgonovo and F. A. Tobagi, "The Express-net: A local area communication network integrating voice and data", in *Proceedings of the International Conference on the Performance of Data Communication Systems and Their Applications*, Paris France, 14-16 September 1981.
- [11] F. Tobagi, F. Borgonovo, L. Fratta, "Express-net: A high-performance integrated-services local area network," *IEEE Journal on Selected Areas in Communications*, vol. SAC-1 Nov. 1983.
- [12] John O. Limb, "Fasnet: A proposal for a high speed local network," in *Proc. of Office Inform. Sys. Workshop*, St. Maximin, France, Oct. 1981.
- [13] J. O. Limb and C. Flores, "Description of Fasnet, a unidirectional local area communications network," *Bell Systems Technical Journal*, September 1982.
- [14] J. O. Limb, "High speed operation of broadcast local networks," in *Proc. of the International Conf. on Communications*, Philadelphia, June 13-17, 1982.
- [15] *IEEE Project 802 Local Area Network Standards*, Draft D 802.4, Token-Passing Bus Access Method and Physical Layer Specification, IEEE Computer Soc., Silver Spring, MD, 1983.

- [16] D. Scavezze, "Nodes sound off to control access to local network," *Electronics*, June 16 1981.
- [17] J. W. Forgie, "Speech transmission in packet-switched store-and-forward networks," *Proceedings National Computer Conference*, 1975.
- [18] D. Cohen, "Issues in transnet packetized voice communication," *Proceedings Fifth Data Communications Symposium*, Snowbird, Utah, Sept. 1977.
- [19] W. E. Naylor, "Stream traffic communication in packet switched networks," *University of California, Los Angeles, Computer Systems Modelling and Analysis Group Report UCLA-ENG-7760*, August 1977.
- [20] G. E. O'Leary, "Local access facilities for packet voice," *Proceedings Fifth International Conference on Computer Communication*, 1980.
- [21] D. C. Swinehart, L. C. Steward, S. M. Ornstein, "Adding voice to an office computer network," *Proceedings Globecom* 1983.
- [22] T. A. Gonsalves, "Packet-voice communication on an ethernet local computer network: an experimental study," *Proceedings SIGCOMM 1983 Symposium on Communications Architectures and Protocols*, Austin TX, March 1983.
- [23] F. A. Tobagi and N. Gonzalez-Cawley, "On CSMA-CD networks and voice communication," *Proceedings Infocom* 1982.
- [24] J. O. Limb and L. E. Flamm, "A distributed local area network packet protocol for combined voice and data transmission," *IEEE Journal on Selected Areas in Communications*, Nov. 1983.

- [25] L. Kleinrock, *Queuing Systems, Vol. II, Computer Applications*. New York: Wiley-Interscience, 1976.
- [26] I. Chlamtac, W. Franta and K. D. Levin, "BRAM: The broadcast recognizing access method," *IEEE Trans. Commun.*, vol. COM-27, No. 8, August 1979, pp. 1183-1190.
- [27] L. Kleinrock and M. Scholl, "Packet switching in radio channels: New conflict-free multiple access schemes," *IEEE Trans. Commun.*, vol. COM-28, no. 7, July 1980.
- [28] L. Kleinrock and M. Scholl, "Packet switching in radio channels: New conflict-free multiple access schemes for a small number of data users," in *Proc. of the International Conf. on Communications*, Chicago, IL, June 1977.
- [29] Y. I. Gold and W. R. Franta, "An efficient collision-free protocol for prioritized access-control of cable or radio channels," *Computer Networks*, 1983.
- [30] Y. I. Gold and W. R. Franta, "An efficient scheduling function for distributed multiplexing of a communication bus shared by a large number of users," in *Proc. of the International Conf. on Communications*, Philadelphia, June 13-17, 1982.
- [31] M. E. Ulug, G. M. White, W. J. Adams, "Bidirectional token flow system," in *Proceedings of the 7th Data Communications Symposium*, Mexico City, October 1981.
- [32] E. D. Jensen, M. Tokoro, L. Sha, "Buss allocation scheme for distributed real time systems," *Carnegie-Mellon University Report*, December, 1980.

- [33] F. Borgonovo, L. Fratta, F. Tarini, P. Zini, "L-Express-net: A communication protocol for local area networks," in *Proceedings INFOCOM '83* San Diego, April 1983.
- [34] J. W. Mark, "Distributed scheduling conflict-free multiple access for local area communications networks," *IEEE Trans. Commun.*, vol. COM-28, pp.1968-1976, Dec. 1980.
- [35] T. D. Todd and J. W. Mark, "Waterloo experimental local network (Wlnet) physical level design," *Proc. of NTC'80*, pp. 41.4.1-41.4.5, Houston Nov/Dec 1980.
- [36] K. P. Eswaran, V. C. Hamacher, G. S. Shedler, "Collision-free access control for computer communication bus networks," *IEEE Transactions on Software Engineering*, vol. SE-7, no. 6, Nov. 1981.
- [37] K. P. Eswaran, V. C. Hamacher, G. S. Shedler, "Asynchronous collision-free distributed control for local bus networks," IBM Research Report RJ2482, San Jose, Ca, 1979.
- [38] L. Fratta, "An improved protocol for data communication bus networks with control wire," in *Proceedings SigComm*, 1983.
- [39] F.A. Tobagi and R. Rom, "Efficient round-robin and priority schemes for uni-directional broadcast systems," *Proceedings of the IFIP 6.4 Zurich Workshop on Local Area Networks*, Zurich, Switzerland, August 27-29 1980.

- [40] R. Rom and F.A. Tobagi, "Message-based priority functions in local multiaccess communications systems," *Computer Networks*, vol. 5, no. 4, July 1981, pp. 273-286.
- [41] C. Tseng and B. Chen, "D-Net, a new scheme for high data rate optical local area networks," *IEEE Journal on Selected Areas in Communications*, vol. SAC-1, no. 3, April 1983.
- [42] M. Gerla, C. Yeh, P. Rodrigues, "A token protocol for high speed fiber optics local networks," in *Proceedings Optical Fiber Communication Conference*, New Orleans, Louisiana, February 1983.
- [43] P. Rodrigues, L. Fratta, M. Gerla, "Token-less protocols for fiber optics local area networks," submitted to *ICC '84*.
- [44] M. A. Marsan and G. Albertengo, "Integrated voice and data network," *Computer Communications*, vol. 5, no. 3, June 1982.
- [45] A. Takagi, S. Yamada, S. Sugawara, "CSMA/CD with deterministic contention resolution," *IEEE Journal on Selected Areas in Communications*, vol. SAC-1, no. 5, Nov. 1983.
- [46] M. Gerla, P. Rodrigues, C. Yeh, "BUZZ-NET: A hybrid random access/virtual token local network," in *Proceedings Globecom '83*, San Diego, CA, December 1983.
- [47] A. Roger Kaye, "Analysis of a distributed control loop for data transmission," *Proc. Symp. Computer Commun. Networks and Teletraffic*, Polytechnic Institute of Brooklyn, April 1972.

- [48] C. Mack, T. Murphy and N. L. Webb, "The efficiency of N machines unidirectionally patrolled by one operative when walking time and repair times are constants," *J. Royal Stat. Soc., Ser. B*, 19, 166-172, 1957.
- [49] F. Tobagi et al., "Modeling and measurement techniques in packet communication networks," *IEEE Proceedings*, pp. 1423-1447, Nov. 1978.
- [50] F. Tobagi and L. Kleinrock "Packet switching in radio channels: Part IV—Stability considerations and dynamic control in carrier sense multiple access," *IEEE Trans. Commun.*, vol. COM-25, pp. 1103-1120, Oct. 1977.
- [51] M. Fine and F. A. Tobagi, "Performance of round robin schemes in unidirectional broadcast local networks," in *Proc. of the International Conf. on Communications*, Philadelphia, June 13-17, 1982, pp. 1C.5.1-1C.5.6.
- [52] P. T. Brady, "A technique for investigating on-off patterns of speech," *Bell System Technical Journal*, January 1965.
- [53] P. T. Brady, "A statistical analysis of on-off patterns in 16 conversations," *Bell System Technical Journal*, January, 1968.
- [54] N. S. Jayant and S. W. Christensen, "Effects of packet losses in waveform coded speech and improvements due to an odd-even sample-interpolation procedure," *IEEE Transactions on Communications*, Feb. 1981.
- [55] H. B. Kekre and C. L. Saxena, "Three-state Markov model of speech on telephone lines," *Comput. Elec. Eng.*, vol. 4, no. 3, 1977.
- [56] K. Bullington and J. M. Frazer, "Engineering aspects of TASI," *Bell System Technical Journal*, vol. 38, March 1959.

- [57] S. J. Campanella, "Digital speech interpolation," *COMSAT Technical Review*, vol. 6, Spring 1976.
- [58] B. Gold, "Digital speech networks," *Proceedings of the IEEE*, vol. 65, no. 12, Dec. 1977.
- [59] P. M. Gopal, J. W. Wong, J. C. Majithia, "An evaluation of playout strategies for voice transmission in packet networks," in *Proc. Computer Networking Symp.*, Dec. 1981.
- [60] M. Suzuki, *et. al.*, "An Adaptive control on length of jitter absorbing buffer in packet speech system," in *Proc. IEEE Int. Conf. Commun. ICC'81*, Denver, June 1981.