PROTON: A Media Access Control Protocol for Optical Networks with Star Topology

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***Abstract*— A new multiple access protocol called PROTON (PROTocol for Optical Networks) is developed for optical local area networks based on a passive star topology. PROTON uses wavelength division multiplexing (WDM) and is highly bandwidth-efficient. One of the available wavelengths is used as a control channel. Time is divided into fixed-sized slots. The size of the slots is the same for the control and the data channels. Before transmitting a packet, a station must compete with others for a slot in a data wavelength, using a collision-free procedure. Transmitting stations and the corresponding wavelengths for their data transmissions are determined at each station by a simple arbitration scheme. The protocol is suitable for networks where the number of users can be much larger than the number of available data channels. In addition to propagation delays, it is considered that transmitter and receiver tuning times as well as the times required to process control packets are not negligible. Whenever possible, and to maximize the throughput of the network, tuning and processing times of transmitters and receivers are overlapped with each other and with data transmission times. Also, data slot requests and packet transmissions are scheduled in a pipeline fashion, thus reducing the detrimental effects on throughput and packet delay of long propagation delays. The paper includes an analysis of the maximum throughput characteristics of PROTON. An analytical model is developed, and several performance measures are obtained.**

I. Introduction

It is estimated that a single optical fiber has a capacity of at least 30 THz in its low-loss region (1.2–1.6 μm) [2], [10]. At the present time, however, it is only practical to use a fraction of this huge capacity. This is usually achieved by dividing the total bandwidth of the fiber into a number of noninterfering channels, a technique known as wavelength division multiplexing (WDM). By using this technique, channels that have a relatively lower bandwidth are obtained, but with the advantage that they operate at transmission speeds that are manageable by commercial devices, reducing the famous *opto-electronic and processing bottlenecks* [10].

PROTON, the protocol introduced in this paper, is ideal for local area networks where the number of stations exceeds the number of available wavelength channels. One wavelength is used as a control channel, while the rest are used as data channels. All stations are connected through a central passive star. Each station operates with at least a single TT/TR (tunable transmitter/tunable receiver) pair for the data channels, and one FT/FR (fixed transmitter/fixed receiver) pair for the control channel. Depending on the number of available wavelength channels, PROTON can support hundreds and up to a few thousand stations.

In the past few years, several protocols have been proposed [1], [3]–[9], [11]–[13], for optical local area networks (O-LAN's). In contrast to our protocol, in order to support a network of M stations, protocols like [3]–[6], [12] require more than M wavelengths ($M+1$ for [3]–[6], $2M$ for [12]), imposing a serious restriction on the maximum number of users in the network, typically limiting this number to no more than a hundred stations. Protocols like [11], [15]–[17] present a throughput curve that diminishes after a certain offered load is exceeded, due to collisions in the control and/or data channels. In our protocol, channel collisions do not occur. Receiver collisions (or destination conflicts [13]) can occur, but can be reduced by increasing the processing in each station. Since channel collisions do not occur, the throughput of the protocol results in a monotonically increasing function of the input load when the destination distribution is uniform.

Most of the previously proposed protocols for O-LAN's have made the assumption that tuning times are negligible. Only a handful protocols like [3], [4], [8], [13] have considered otherwise. While in [3], [4] tuning times can be assumed to be a part of packet transmission times, tuning times are considered explicitly in [8], [13]. In [8], the existence of processing times is also contemplated. As explained below, in this paper we specifically consider the existence of nonnegligible and nonequal tuning and processing times for the transmitter and the receiver. Furthermore, we also consider that propagation delays are not negligible. To our knowledge, this is the first time that a protocol captures all these factors in such a great detail.

Several attempts have been made to compare the protocols for optical LAN's that have been proposed in recent years [10], [12], [13]. Table I presents the comparison of some of the protocols proposed in recent years.

In Table I, the protocols are compared according to the following metrics.

- *Network Interface Unit:* The NIU structure description is used to determine if the protocol uses a control channel and the number of fixed and tunable transceivers required by each station. We follow the notation in [13], where $[CC]-FT^iTT^j-FR^kTR^l$ denotes that each station has i fixed transmitters, j tunable transmitters, k fixed receivers, and l tunable receivers. The prefix CC is

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TABLE I
COMPARISON OF VARIOUS MAC PROTOCOLS FOR O-LAN'S

Schemes	NIU Structure	Tuning Time	Processing Complexity	Processing Time	Propagation Delay	Throughput	Wavelengths per Network	Scalability
Habbab [13]		No	Low	No	No	Low	≥ 2	Yes
Mehravari [20]	CC-TTTR	No	Low	No	No	Low	≥ 2	Yes
Sudhakar [26]		No	Low	No	No	Low	≥ 2	Yes
Jia [16]		Yes	Moderate	No	Yes	Moderate	≥ 2	Yes
Chlamtac [7]	TTTR	No	Low	No	No	Low	≥ 1	No
Ganz [10]								
S-ALOHA	$TTFR^M$	No	Low	No	No	Low	≥ 1	Yes
Random TDMA		No	Very high	No	No	High	≥ 1	No
Lu [19]		No	High	No	Yes	High	≥ 2	Yes
Humblet[15]	CC-FTTT	No	Moderate	No	Yes	High	2M	No
PROTON	-FRTR	Yes, S	Moderate	Yes, S	Yes	High	≥ 2	No
M-S Chen[3]	$CC\text{-}FT^2$	Yes -	High	No	Yes	Moderate	M+1	No
Chipalkatli[5]	-FRTR	No	Very High	No	Yes	High	M+1	No
M-Chen[4]	CC-FTTT -FRTR	Yes -	Very High	No	Yes	High	M+1	No
Dowd [9]	CC-TT							
TDMA-C	-TTFRTR	Yes, S	High	Yes	Yes	Low	≥ 2	No

\- tuning time is part of data packet
S separate times for transmitter and receiver

optional, and when present denotes a protocol that uses a control channel.

- *Tuning Time:* This is used to indicate if the protocol accommodates or ignores transceiver tuning times. An S following a "Yes" indicates that the protocol separates transmitter tuning times from receiver tuning times.
- *Processing Complexity:* This is a measure of the processing requirements.
- *Processing Times:* This indicates if the protocol explicitly considers the time involved in processing the pertinent information before a data packet can be transmitted. In reservation-based protocols, an S following a "Yes" indicates that the protocol considers separate processing times (of the control channel) for the sender and the receiver.
- *Propagation Delay:* Whether or not the protocol considers media propagation delays.
- *Throughput:* Characterization of the maximum throughput that a protocol can achieve.
- *Wavelengths per Network:* Indicates the minimum number of wavelengths that the protocol requires to operate. M is the total number of stations in the network.
- *Scalability:* Is there an upper limit (theoretically) on the maximum number of stations that can be added given a working implementation and a fixed number of channels?

It is interesting to observe that almost none of the previous proposals have explicitly considered processing times. For very high transmission speeds, it is not inconceivable that protocols with high processing requirements and small data packets can have processing times larger than a data packet time! Yet, the potentially detrimental effects of processing times have hardly been considered before. Some protocols such as [3], [4] consider that transceiver tuning times are simply part of the data packet time. This consideration can result in gross data channel inefficiencies, especially when low-speed tunable devices are being used.

The rest of this paper is organized as follows. Section II contains a detailed description of PROTON. In Section III, we analyze several performance issues of the protocol, including its maximum achievable throughput. In addition, we provide analytical expressions for the most important performance characteristics of the network. Numerical results from several simulations of the protocol and their comparison with analytical results are presented in Section IV. A summary of our work and concluding remarks are presented in Section V.

II. THE PROTON PROTOCOL

Before we describe the protocol in detail, we introduce a number of assumptions and necessary notation.

- *Wavelength Channels:* Each fiber in the network has a total of $W + 1$ channels numbered $\lambda_0, \lambda_1, \cdots, \lambda_W$ (typically, W will not exceed 100). Wavelength λ_0 is reserved for the control channel, while the remaining W wavelengths are data channels.
- *Network Stations:* There are a total of M stations numbered $m_1, m_2, \cdots, m_M$ in the network.
- *Transmitters and Receivers per Station:* Each station in the network has at least a tunable and a fixed transmitter/receiver pair.
- *Tuning Times:* Every TT and TR of any station can tune to any data channel. The worst-case (longest) tuning time

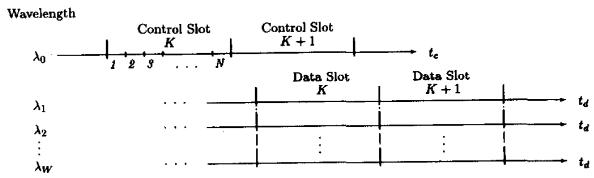

Fig. 1. Structure of the control and data channels.

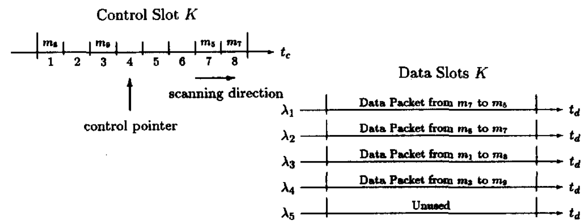

Fig. 2. Example of the data channel arbitration scheme.

of any receiver is denoted as τ_{TR}, while the worst-case tuning time of any transmitter is τ_{TT}.

- *Processing Times:* Stations in the network, whether expecting to transmit or to receive a packet, process the control slots to determine the next action to take. Processing times are considered to be the worst-case time from the reception of the first bit of a control packet until the moment when the processing of the control packet is completed. We distinguish two processing times incurred by every station: τ_{PR} is the processing time required to determine if packets are being addressed to the station, and τ_{PT} is the corresponding time required by the station to process its own packet transmission request.
- *Propagation Delays:* The round-trip propagation delay of a packet δ is not negligible.

The round trip propagation delay δ is the same for any station. Therefore, stations that are physically closer to the star coupler than others must incorporate optical delays. We assume that all stations in the network are synchronized by using a common clock.

A. Basic Protocol

The control channel is divided into control slots, and each control slot is divided into N equally-sized mini-slots, where $N \geq M$, as shown in Fig. 1. There is at least one mini-slot reserved for every station in the network, and some stations may have more than one mini-slot assigned to them. When a station needs to send a packet to another station, it must first place the address of the intended receiver in its corresponding mini-slot. Each receiver is constantly reading the control slots, waiting for its address to appear in a control mini-slot. The data channels are also divided into equally-sized slots. The size of the data slots is equal to the size of a control slot. All data slots are aligned in time with respect to each other, but data slots and control slots do not need to be aligned.

Since usually $N > W$, an arbitration scheme for the data channels is required for those cases where there are more stations in need to transmit than available data channels. The arbitration mechanism is based on that presented in [14], and is as follows. Associated with each control slot, there is a pointer (the *control pointer*) that uniquely determines the stations that can use the data slots during a given control slot. The control pointer prompts to a single control mini-slot. If the mini-slot contains the address of a receiver, then the owner of the control mini-slot may send a packet using wavelength λ_1. The rest of the control mini-slots are scanned (the direction can be chosen arbitrarily, i.e., left-to-right or right-to-left, both with wrap-around) until a nonempty mini-slot is found, and when this happens, the owner of this mini-slot is assigned wavelength λ_2 for the transmission of its data packet. The process is repeated until no more nonempty mini-slots are found, or until there are no more data channels, whatever occurs first. Accordingly, Fig. 2 shows an example of the arbitration scheme using a left-to-right scanning of a control slot. Here, stations m_7, m_8, m_1, and m_3 have won the right to send a data packet to stations m_5, m_7, m_8, and m_9, using, respectively, wavelengths λ_1, λ_2, λ_3, and λ_4. Note that since only four stations placed a receiver's address in their assigned control mini-slots, and since there are five data channels, the data slot in wavelength λ_5 will not be used. Obviously, to achieve fairness, the control pointer needs to be moved to another mini-slot with each new control slot.

The size of the data slots is dependent on the size of the control slots. The lower bound on the size of the control slots is determined by the number of stations in the network, and the transmission speed of the control slot.

For a typical network size, the normalized propagation time of packets (denoted by "a") is high. Thus, several control packets may be en-route through the network before the first transmitted packet of a batch is actually received after a round-trip propagation delay to and from the optical star coupler. In our protocol, control packets are sent in a pipelined fashion, i.e., back-to-back. Thus, a station can send several consecutive requests through the control channel even before it can actually receive the first request that sent earlier. This mechanism greatly increases the throughput of the network.

Every station has two subsystems for the processing of control slots. As explained before, if a station wants to send a data packet, it must request access to a data channel. This is achieved by first placing the intended receiver's address in a control minislot assigned to the transmitting station. Then, it must wait until the corresponding control slot returns after a round-trip propagation delay, possibly filled with requests from other stations. Then, the station aspiring to transmit a packet processes the control slot with a dedicated subsystem, the transmitting request subsystem (TRS). The processing time τ_{PT} is the worst possible time spent by any TRS processing a control slot. Depending on the processing speed of the TRS, τ_{PT} can span from more than one to several control slots.

If a station is able to send data packets using consecutive control slots or slots that are almost next to each other, it must have enough processing resources in its TRS so that two or more control slots can be processed in parallel (at the same time), overlapping its corresponding processing times τ_{PT}. Clearly, from the performance standpoint, for small values of τ_{PT} this is not essential, but can become a necessity when τ_{PT} is several control slots in length.

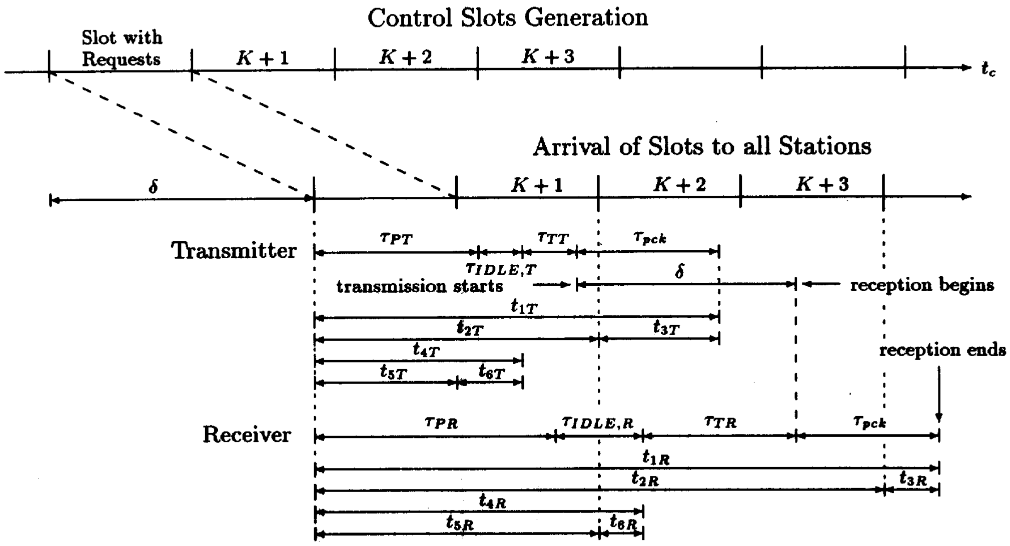

Fig. 3. Sequence of events in the transmission and reception of a data packet.

Aside from the TRS subsystem, every station is equipped with a control channel processing subsystem (CCPS). The job of the CCPS is to continually process the control slots to find if the address of its station appears, indicating that a packet from another station will soon be transmitted. Whereas for the TRS subsystem it is not essential to overlap (if necessary) the processing of consecutive control slots, the CCPS subsystem must be able to parallel-process *all* control slots with overlapping processing. This is because a receiver has no way of knowing in which control packet its address will appear.

Fig. 3 shows the sequence and timing of events that must take place for the successful coordination of the transmission and reception of a data packet. For this figure, we assume that the packet time $\tau_{pck} = 1.0$, and then we determine the different times according to $t_{1T} = \tau_{PT} + \tau_{IDLE,T} + \tau_{TT} + 1$, $t_{2T} = \lfloor t_{1T} \rfloor$, $t_{3T} = t_{1T} - t_{2T}$, $t_{4T} = \tau_{PT} + \tau_{IDLE,T}$, $t_{5T} = \lfloor t_{4T} \rfloor$, and $t_{6T} = t_{4T} - t_{5T}$. The times for the receiver are determined in a similar way.

According to Fig. 3, we observe that the entire process is initiated when a station (sender) places the address of another station (the intended receiver) in the control channel, using a mini-slot assigned to the sender. After a propagation delay δ, the control slot that carries the request is distributed to all stations in the network. The sender processes the control slot in a time no worse than τ_{PT}, and if it finds that it has won the right to use a slot in a data channel, proceeds (possibly after waiting a time $\tau_{IDLE,T}$) to tune its TT to the assigned wavelength. At the end of the tuning operation, the sender starts to transmit the data packet, which reaches the receiver δ units of time later. The receiver, on the other hand, after scanning and processing the control channel, proceeds (possibly after waiting a time $\tau_{IDLE,R}$) to tune its TR to the appropriate channel. The reception of the data packet begins after the tuning operation is completed.

To synchronize the arrival and reception of the data packet, it is likely that an idle time (between control packet processing and device tuning) will need to be inserted, either on the transmitter or on the receiver side.

The amount and frequency of packets that can be transmitted and received by a station is a function of multiple variables. These variables include the processing power of the TRS and CCPS subsystems, and tuning speed and number of TT's and TR's per station. While any station can overlap control packet processing and idle times, stations with a single TR cannot overlap tuning and data packet receiving times. Taking into consideration the times that can and cannot be overlapped, it is possible to determine the minimum number of slots required for two successive receptions of data packets by a given station. This number, denoted *pass_slots*, can be calculated according to:

if ($t_{3R} \leq t_{6R}$)
$pass\_slots = t_{2R} - t_{5R}$;
else
$pass\_slots = \lceil t_{1R} \rceil - t_{5R}$;

where $pass\_slots \geq 1$. Note that since τ_{TR} is assumed to be the worst tuning time of any TR, the value of *pass_slots* is the same for any station with a single TR. For a station with multiple TR's, the value of *pass_slots* is not constant, and depends on the number of TR's of a station that are currently in use, as well as the relative times when the TR's were called into action.

Similarly, we can determine the minimum number of slots required for two successive transmissions from a given station. We denote this number by T_{wait}. Note that T_{wait} is likely to be different from *pass_slots*, since in a transmitter, control channel processing is not as stringent as is with a receiver (a sender needs only to process the control packets where it has previously placed requests). Furthermore, tuning times can be quite different between TT's and TR's.

A station with several packets in its queue typically will not transmit at the maximum rate $1/T_{wait}$. Rather, the station will wait T_{wait} plus a random number of slots (0 to $T_{wait} - 1$) between successive transmissions. This is to prevent a cyclic condition that occurs near maximum sustainable load, where in each cycle several stations concentrate their transmission requests within a few slots, leaving the other slots practically unused (each cycle is T_{wait} slots long), resulting in a drastic reduction in throughput.

Setting aside hardware failure, there are some situations where a data packet may be lost after transmission. For example, two or more stations could try to send data packets at the same time to a station that only has a single TR (in this case, the packet to be received is the one closest to the control pointer). Also, a data packet may be sent to a station that is currently busy preparing for the reception of another data packet. Therefore, it is evident that implementing ACK's in the protocol is a desirable feature. This can be achieved by dividing the control channel in two parts, one for the addresses of the intended receivers, and the other for ACK signals generated by the receivers. After a data packet has been received successfully, the receiver sends an ACK signal. This signal must be sent an exact and predefined number of slots after the packet has been received, thus avoiding the need to explicitly indicate the identity of the receiver sending the ACK. The relative order of ACK signals within a control slot is also important. Each ACK sender must choose a position that reflects the data channel number where it previously received the data packet.

III. Performance Evaluation

In this section, we study several aspects that describe the behavior of the network under a variety of conditions. We begin by investigating the effects that the processing and tuning speeds, number of channels, propagation delay, and total number of stations have on the maximum achievable throughput of the network. Several network characteristics such as throughput, channel efficiency, rate of arrival of new packets, and the total number of packets presented to the network are related to each other. Finally, we develop a model for the delay characteristics of a network that is using this protocol.

A. Modeling Assumptions

- *Packet Generation:* Packets are generated in each station due to independent Bernoulli processes. The probability of generating a new packet in a slot time is (for any station) equal to σ, the ratio of the input load I to the number of stations M. All stations have equal probability of receiving a packet, but a station generating a packet is not allowed to send the packet to itself.
- *Buffer Size:* Each station in the network has an infinite buffer capacity for data packets.
- *Transmitters and Receivers in Each Station:* Each station has a FT/FR pair for the control channel, and a TT/TR pair for the data channels.
- *Packet Acknowledgment:* After completely receiving a data packet, the receiver sends an ACK signal to the sender in the next control slot. If a sender does not get the ACK signal in the appropriate control slot, it assumes that the packet was not received and immediately schedules a retransmission.

The input load of the network I is considered to be the average rate of data packets generated by all stations in the network. The offered load to the network G is equal to the average rate of data packets being carried by the network. The internal load G^* is the average number of stations that request permission for transmission during any slot.

B. Maximum Achievable Throughput (MAT) of the Network

The throughput S of a network is equal to the total rate of data slots that are received successfully and normalized by the network capacity. The maximum value of throughput that can be reached for a given network S_{MAT} is a function of several variables, including the number of stations, data channels, and the tuning speeds of TT's and TR's. Here we derive the MAT of a network as a function of these variables.

In order to find the MAT of a network configuration, we assume that the network is operating under heavy load, i.e., in all control slots, all data channels are being requested. Under these conditions, the probability p_{req} that a station is addressed with data packets in any slot is equal to the probability that 1 or more (up to W) data packets are being sent to this station in any slot. This is

$$p_{req} = \sum_{y=0}^{W-1} \binom{W}{y} \left(\frac{1}{M-1}\right)^{W-y} \left(1 - \frac{1}{M-1}\right)^{y}. \quad (1)$$

Since when a station is free and receives a request for reception at least $pass\_slots$ slots must pass before it can receive another data packet, we can also consider that, in order to successfully receive a data packet, a station should not have been addressed anytime during the previous $(pass\_slots - 1)$ slots. Thus, the probability for a successful reception p_{rec} becomes

$$p_{rec} = (1 - p_{req})^{(pass\_slots-1)} p_{req}. \quad (2)$$

Since this probability is the same for any of the M stations in the network, the MAT of the network is simply

$$S_{MAT} = (p_{rec})\frac{M}{W}. \quad (3)$$

Fig. 4 shows the maximum achievable throughput for networks with a) 10 data channels, and b) 50 data channels. The plots show the MAT as a function of the number of stations and the number of slots that must pass $pass\_slots$ in each station before two consecutive receptions can be achieved. In both plots, each curve represents a value of $pass\_slots = \{1, 2, 3, 4, 6, 10, 15\}$.

In Fig. 4, it is interesting to observe that the common assumption (in networks where the stations have a single TR with negligible tuning time) *number of packets lost due to a busy receiver is negligible* is hardly valid. In Fig. 4(a) and (b), this is represented by the uppermost curves. As seen in

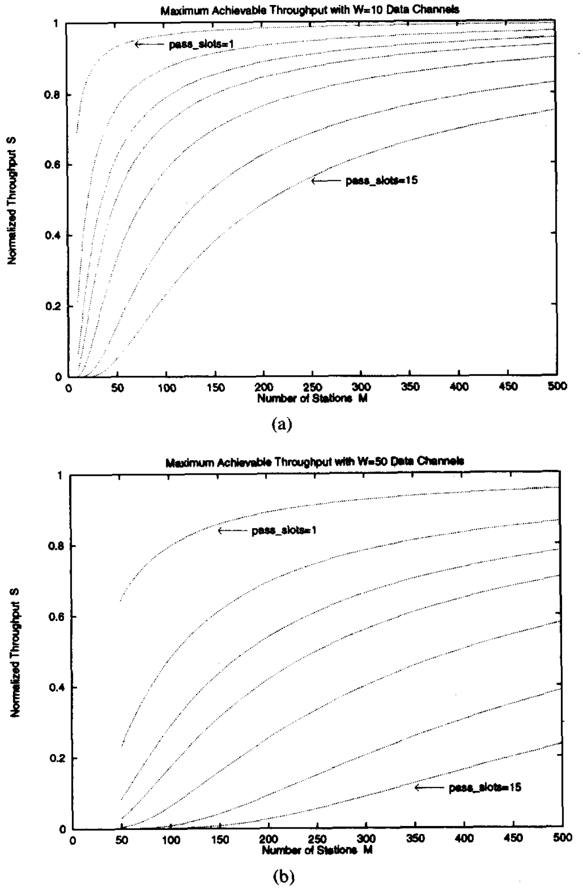

Fig. 4. MAT for pass_slots = 1,2,3,4,6,10,15 with a) W = 10, and b) W = 50.

these curves, the MAT can be as low as about 0.7 under this condition.

C. Throughput and Load Characteristics

1) Input Load versus Throughput: Since no collisions can occur in the data channels, the network throughput does not decrease as the input or offered loads increase. In fact, as long as the normalized input load I/W does not exceed the maximum achievable throughput of the network S_{MAT}, the throughput is almost directly proportional to the normalized input load. When the normalized input load is smaller than S_{MAT}, all packets being sent are eventually successful in reaching their destinations. This usually happens during the first transmission or few retransmissions. When the normalized input load reaches the value S_{MAT}, the network becomes saturated, and the throughput curve reaches a maximum. For subsequent increases of the input load, the throughput characteristic becomes constant at a value equal to S_{MAT}. Therefore, the network throughput can be described by

$$S = \begin{cases} I/W & \text{if } I \le W \cdot S_{MAT} \\ S_{MAT}. & \text{otherwise} \end{cases}. \tag{4}$$

2) Offered Load to the Network and Internal Load: Recall that during any given slot, at most W stations will be able to transmit a data packet. When there are more than W transmission requests in a control slot, some stations will not obtain a data slot. Therefore, for a given slot, internal and offered loads are equal only when the number of transmission requests is less than or equal to W. That is, in any slot j, the relation between G_j and G_j^* is

$$G_j = \begin{cases} G_j^* & \text{if } G_j^* \le W \\ W & \text{otherwise} \end{cases}. \tag{5}$$

Since in any slot, zero or up to M stations may request transmission permissions, the internal load can be modeled as a simple process with binomial distribution and with mean equal to G^*. The average offered load to the network is simply the mean of this distribution subjected to the restriction that only up to W stations may actually place a packet in a data slot. That is

$$G = \overline{G} = \sum_{x=1}^{W} x \binom{M}{x} p^x (1-p)^{M-1} + \sum_{x=W+1}^{M} W \binom{M}{x} p^x (1-p)^{M-1} \tag{6}$$

where $p = G^*/M$ and $G^* \ge G$. When the ratio M/W is large, for $G^* < W \cdot S_{MAT} < W$ (ergodic system), internal and offered loads are practically equal.

3) Offered Load versus Channel Efficiency: The channel efficiency of the network η_{NW} is the ratio of the number of data slots carrying packets that are received successfully to the total number of data slots carrying data (successful + unsuccessful). For low load conditions, and for stations with a small value of the parameter $pass\_slots$, the channel efficiency should be very high (close to unity). As the load increases, the channel efficiency tends to decrease since the stations in the network tend to use more of the available data channels in every slot, increasing the possibility that two or more stations try to send a packet to the same destination. The channel efficiency also tends to decrease as the value of $pass\_slots$ increases, since under this condition it is more likely that a sender will find the receiver of its destination busy with another packet. For a given value of $pass\_slots$, the worst-case scenario for the channel utilization occurs when the offered load is maximum, that is, $|G| = |W|$. At this point, the channel efficiency is close to the MAT of the network. From the above observations, we conclude that the channel efficiency of the network: 1) is a function of the offered load and number of slots that must pass in each station before two successful receptions can be achieved, 2) it can reach values very close to unity for light loads and small values of $pass\_slots$, and 3) it decreases as a function of increased load until it reaches a value close to S_{MAT} (for the peak offered load). We therefore approximate the relation *channel efficiency versus offered load* with a straight line with end coordinates $(0, 1)$ and $(1, S_{MAT})$, yielding the following expression:

$$\eta_{NW}(G) = \frac{S_{MAT} - 1}{W} G + 1. \tag{7}$$

4) Input Load versus Offered Load: When the stations of a network have a single TR for the data channels, some data packets will eventually be lost. Therefore, retransmissions at a future time will be needed, resulting in different values for the input and offered loads. Thus, for values of the input load smaller than the total capacity of the network, the offered load is always greater than or equal to the input load, with equality occuring only when every station has a fixed receiver for every channel in the network and the capacity to process all data channels simultaneously. During every slot, the number of packets that are sent successfully is proportional to the channel efficiency value for the current offered load. Obviously, the packets that are not delivered successfully will be retransmitted at a future time, becoming part of the internal load of a future slot. Moreover, if in a given slot the internal load is greater than W, the stations that were not successful in their request for a data channel will try again later. Finally, in subsequent slots, additional packets will be generated at the stations (with constant rate I), contributing to the total value of the internal load of the subsequent slots. With these observations in mind, we proceed to approximate this situation with the following difference equation

$$G^*_{k+j} = I + G_k(1 - \eta_{NW}(G_k)) + (G^*_h - G_h) \tag{8}$$

where k denotes a particular slot, j is the mean number of slots between retransmissions of a lost packet, $(j - h)$ is the mean number of slots between the unsuccessful request for a packet transmission and the actual transmission of the packet, and G^*_{k+j} is the internal load of the network during slot $(k + j)$. We are only interested in finding the steady state solution of the equation, i.e., when all the subindices tend to infinity. Obviously, when the network is in steady state, $\lim_{k+j\to\infty} G^*_{k+j} = \lim_{h\to\infty} G^*_h$ and $\lim_{k\to\infty} G_k = \lim_{h\to\infty} G_h$. Therefore, substituting (7), we can rewrite (8) as (note that the term G^* disappears)

$$\left(\frac{S_{MAT} - 1}{W}\right) \cdot \left(\lim_{k\to\infty} G_k^2\right) + \lim_{k\to\infty} G_k - I = 0. \tag{9}$$

Furthermore, we can take the limit of (9) with $G = \lim_{k\to\infty} G_k$ and solve the resulting equation as a simple second-order equation ((9) converges to a value independently of the value of G_0). Note, however, that the solution of (9) can take values of G greater than the maximum capacity of the network W for sufficiently large values of I. Therefore, we must restrict the value of G to the maximum network capacity. Excluding the root that results in decreasing values of G for increasing values of I, the resulting final expression is

$$G = \lim_{k\to\infty} G_k = \begin{cases} \frac{-1+\sqrt{1+4I\left(\frac{S_{MAT}-1}{W}\right)}}{2\left(\frac{S_{MAT}-1}{W}\right)} & \text{if } \frac{-1+\sqrt{1+4I\left(\frac{S_{MAT}-1}{W}\right)}}{2\left(\frac{S_{MAT}-1}{W}\right)} \le W \\ W & \text{otherwise} \end{cases}. \tag{10}$$

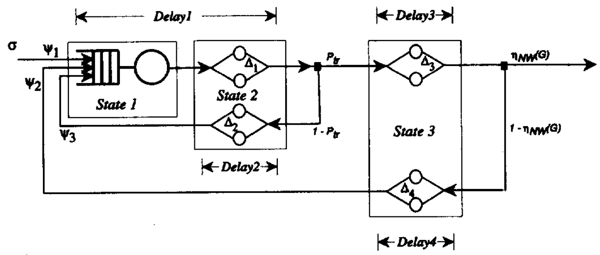

Fig. 5. Model for the average delay characteristics of data packets.

D. Delay Characteristics

In every station, outgoing data packets can be in one of three possible states: 1) new packets or packets to be retransmitted, 2) packets awaiting the outcome of a transmission request, and 3) packets already sent but awaiting confirmation of successful reception. All outgoing data packets can be stored in a single queue, provided they are marked with their generation timestamp and appropriate state. Newly generated packets are marked as being in state 1. The amount of time that a packet has to wait in this state (before a transmission request is placed) depends on the number of packages in state 1 and the value of T_{wait}. A packet is marked with state 2 right after a transmission request for that packet is sent. After being marked, the packet has to wait for a roundtrip delay and a processing time before it can be known if the packet is going to be transmitted (placing the packet in state 3), or sent back to state 1 (if no data channels are available). When a packet is in state 3, it has to wait additional time in order to find if the packet was successfully received by the destination. This time interval is equal to the: 1) time involved in tuning the TT to the adequate channel, 2) transmission delay between sender and receiver so that the packet reaches its destination, 3) time necessary by the destination to process and to send the ACK signal back to the sender, and 4) time delay necessary for the ACK signal to reach the original sender. The model that is used for computing the average packet delay is presented in Fig. 5.

When a packet is generated, it is marked with state 1, and if there are no packets to be retransmitted, the station places a request for transmission in the control channel and marks the packet as being in state 2. The probability that the request is successful is P_{tr}. This probability is calculated by considering the probability that a request is unsuccessful, which is simply the probability that W or more (up to $M - 1$) stations also requested a data slot and that the packet in question is not among the W winners. This probability is the complementary probability of P_{tr}. Therefore,

$$P_{tr} = 1 - \sum_{x=W}^{M-1} \binom{M}{x} p_{int}^x (1 - p_{int})^{M-1} \cdot \left(1 - \frac{W}{x+1}\right) \tag{11}$$

where $p_{int} = G^*/M$. The average number of transmission requests that are needed before a packet can be transmitted is simply $1/P_{tr}$. When a request is successful, the packet is marked as being in state 3, and a copy of the packet is sent to the intended receiver. The probability of success in this case is simply $\eta_{NW}(G)$, given in (7).

The approximate number of packets waiting in the queue of each station is determined by modeling each queue as a system with bulk arrivals and single departures. The total arrival rate Ψ is the sum of the external arrival rate ψ_1 and the feedback rates ψ_2 and ψ_3. Ψ is determined by

$$\Psi = \psi_1 + \psi_2 + \psi_3 = \frac{\sigma}{P_{tr} \cdot \eta_{NW}} = \frac{I}{P_{tr} \cdot \eta_{NW} \cdot M} \quad (12)$$

where

$$\psi_1 = \sigma = \frac{I}{M} \quad (13)$$

$$\psi_2 = \Psi \cdot P_{tr} \cdot (1 - \eta_{NW}) \quad (14)$$

$$\psi_3 = \Psi \cdot (1 - P_{tr}). \quad (15)$$

The arrival rates α_1, α_2 and α_3 that denote the bulk arrivals of 1, 2, or 3 packets respectively, are determined by computing the probabilities of 1, 2, or 3 arrivals α_1, α_2, and α_3.

The resulting queueing system can be solved with the following equilibrium equations

$$(\alpha_1 + \alpha_2 + \alpha_3)p_0 = \mu_1 p_1 \quad (16)$$

$$(\alpha_1 + \alpha_2 + \alpha_3 + \mu_1)p_1 = \mu p_2 + \alpha_1 p_0 \quad (17)$$

$$(\alpha_1 + \alpha_2 + \alpha_3 + \mu)p_k = \mu p_{k+1} + \sum_{i=k-3\geq 0}^{k-1} p_i \alpha_{4-(k-i)} \quad (18)$$

where $p_0, p_1, p_2, \cdots$ denote the probability of being in state $0, 1, 2, \cdots$, respectively, and μ is the departure rate. When the system is in state 1, the departure rate is approximately equal to $\mu_1 = 1$; when in any other state, this rate is equal to the reciprocal of T_{wait} and the mean of the additional random number of slots that the station waits. Therefore, when the system is in a state other than 1, $\mu = 1/(T_{wait} + \sum_{i=0}^{T_{wait}-1}(i/T_{wait}))$. The approximate average number of packets waiting in the queue (in state 1) $\overline{N_q}$ is given by

$$\overline{N_q} = \sum_{k=0}^{\infty} k \cdot p_k \quad . \quad (19)$$

Finally, the average delay $\overline{T_d}$ incurred by a packet waiting in the queue is determined by the well-known Little's Law

$$\overline{T_d} = \frac{\overline{N_q}}{\Psi}. \quad (20)$$

We now proceed to determine the overall delay incurred by a packet from the moment it is generated until it is successfully received. From Fig. 6, the delays in the trajectories traversed by the data packets awaiting for transmission are

$$\overline{D_1} = \frac{\overline{N_q}}{\Psi} + \delta + T_{PT} = \frac{\overline{N_q}}{\Psi} + \Delta_1 \quad (21)$$

$$D_2 = \lceil \delta + T_{PT} \rceil - (\delta + T_{PT}) = \Delta_2 \quad (22)$$

$$D_3 = T_{IDLE,T} + T_{TT} + \delta = \Delta_3 \quad (23)$$

$$\begin{aligned} D_4 = &\lceil 2\delta + T_{PT} + T_{IDLE,T} + T_{TT} \rceil \\ &- (2\delta + T_{PT} + T_{IDLE,T} + T_{TT}) + 1 + \lceil \delta + 1 \rceil = \Delta_4. \end{aligned} \quad (24)$$

Taken into consideration all posible routes and branching probabilities, the final result, the approximate average packet delay $\overline{\Delta_{pck}}$, becomes

$$\overline{\Delta_{pck}} \approx \frac{\left(\frac{\overline{D_1} + D_2(1-P_{tr})}{P_{tr}} + D_3\right) + D_4(1 - \eta_{NW}(G))}{\eta_{NW}(G)}. \quad (25)$$

IV. Experimental Results

In this section, we present and compare results from several simulations and from the application of the analytical models developed in the previous section. The primary network model that was simulated consisted of a total of 100 stations and 11 channels (10 data + 1 control). Each group of simulations was run under the modeling assumptions described in the previous section, and with progressive levels of input loads. Although in every simulation convergence was observed within a few thousand cycles, all simulations were run for a total of 200 000 cycles (i.e., control slots). Several system parameters of the network model were varied, including tuning and processing times, and round trip delay. The effects of these variations on the system's performance are presented in a sequence of plots. These plots show the relationships between some of the most important performance parameters (i.e., input load I, offered load G, network throughput S, channel efficiency η, and average packet delay $\overline{\Delta_{pck}}$) of the system.

Fig. 6 presents the channel efficiency versus offered load characteristics for various networks. For simplicity, we denote the system parameters of a network as a quintet of the form $(\delta, T_{PT}, T_{TT}, T_{PR}, T_{TR})$ to represent, respectively, the round-trip propagation delay, transmitter's processing time, transmitter's tuning time, receiver's processing time, and receiver's tuning time. In Fig. 6, the network parameters are: $A = (0,1,0,1,0)$, $B = (70,1,0,1,0)$, $C = (0,16,0,1,0)$, $D = (0,1,0,16,0)$. Network A has ideal system parameters that result in the best possible performance. Network B is similar to A, but has a propagation delay equal to 70 control slots. As seen in Fig. 6, propagation delays appear not to have a significant effect on channel efficiency. The reason for this is that propagation delays only shift in time actions to be taken for any packet, but other activities can continue while packets are in transit. Also, since here it is assumed that each station has enough computing resources to process in parallel as many control slots as necessary, networks C and D show no apparent negative effects on channel efficiency, even though their processing times are high. In fact, and excluding average packet delay, propagation delays as well as processing times have minimal detrimental effects on the performance parameters of a network. Here, a key observation

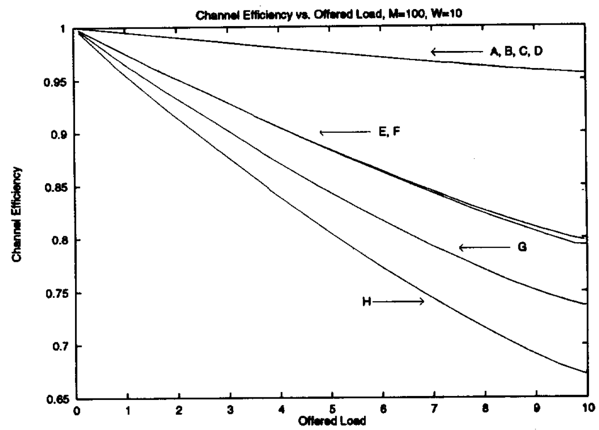

Fig. 6. Channel efficiency versus offered load characteristics for several networks with different system parameters.

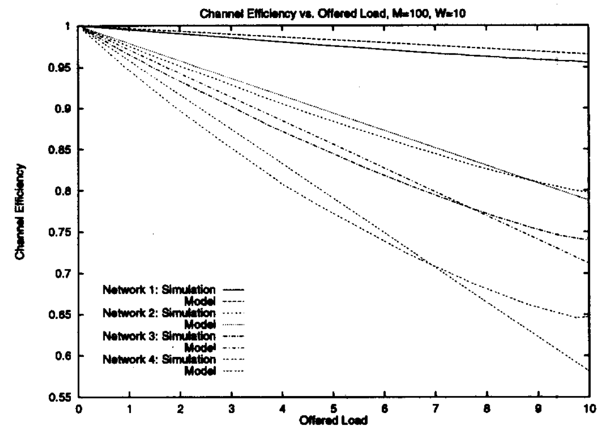

Fig. 7. Channel efficiency versus offered load characteristics.

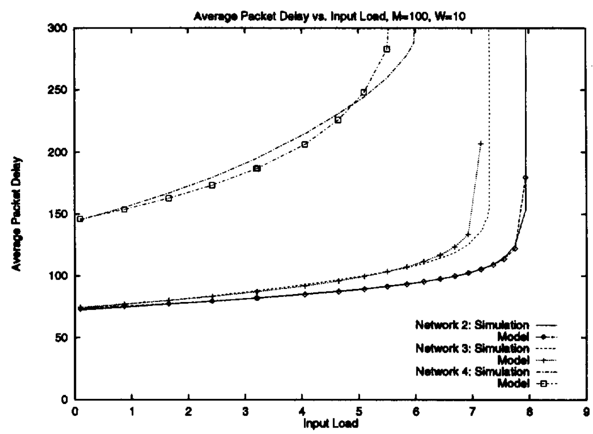

Fig. 8. Average packet delay versus input load characteristics.

is that networks A, B, C, and D have all the same value for the parameter $pass\_slots$ (in all of them it is equal to one). The parameters of the other networks are: $E = (0, 1, 0, 1, 2)$, $F = (35, 1.3, 0.7, 1.3, 1.7)$, $G = (0, 1, 0, 1, 4)$, and $H = (0, 1, 0, 1, 4)$. It can be verified that the values of the parameter $pass\_slots$ for these networks are equal to 3, 3, 4, and 5, respectively. As observed in Fig. 6, the curves for each of these networks appear to be very dependent on their respective value of $pass\_slots$. In fact, the performance of the two networks with $pass\_slots = 3$ is almost identical, and the network with worst performance is the one with the highest value of $pass\_slots$.

Figs. 7–10 present a comparison between performance plots obtained through simulation and through the application of the developed models. The system parameters for these networks are: Network 1 $= (0, 1, 0, 1, 0)$, Network 2 $= (35, 1.3, 0.7, 1.3, 1.7)$, Network 3 $= (35, 1.4, 1.6, 1.4, 2.6)$, and Network 4 $= (70, 1.4, 2.6, 1.4, 4.6)$. Network 1 is the network with ideal system parameters, while Network 4 has the highest values for each of the system parameters that were controlled, resulting in the worst possible performance amongst these networks. For networks A, B, C, and D, the values of the parameter $pass\_slots$ are $1, 3, 4, 5$, respectively. Fig. 7 presents the channel efficiency versus offered load characteristic for each network. We observe that the nonlinear simulation curves are matched with a linear analytical model, which provides a good approximation, especially for the networks with a small value for the parameter $pass\_slots$. In this and subsequent graphs, performance decreases as the value of the parameter $pass\_slots$ increases.

Next, we study the average packet delay versus input load characteristics in Fig. 8. For networks 2 and 3, the minimum $\overline{\Delta_{pck}}$ is 70, since the round trip propagation delay is 35 and at least two round trips are required before a packet can successfully be transmitted. For network 4, the minimum packet delay is 140. For these networks, as long as the input load is kept around 2/3 of the network capacity, the average packet delay is kept very close to the minimum possible. The minimum input load I required for network saturation is closely related to the value of $pass\_slots$ of the network. Higher values of $pass\_slots$ result in smaller values for the minimum I required for saturation and vice-versa.

The relation between average packet delay and throughput is presented in Fig. 9. This graph is almost identical to that shown in Fig. 8, except for the fact that throughput has taken the place of input load in the abscissas axis, and the range of this axis has been changed proportionally. This similarity was expected because throughput is equal to normalized input load as long as I is kept at a value that is less or equal to the network capacity W. It is important to observe that the curves do not show a throughput that decreases after a certain peak is reached, behavior observed in networks that use some form of an ALOHA-based protocol [11], [15], [16], [17].

Finally, Fig. 10 shows the offered load versus input load characteristics for the networks under study. For low input loads, I and G are almost identical, since at these loads most of the packets get to their destination on the first try and few retransmissions occur. As I increases, the channel efficiency of the network decreases, resulting in more retransmissions and in G becoming noticeably larger than I, approximately according

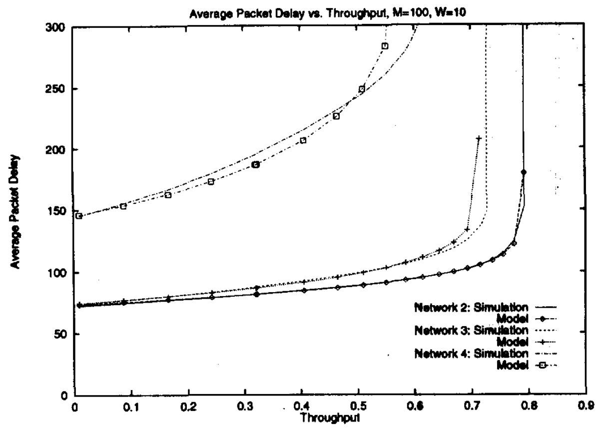

Fig. 9. Average packet delay versus throughput characteristics.

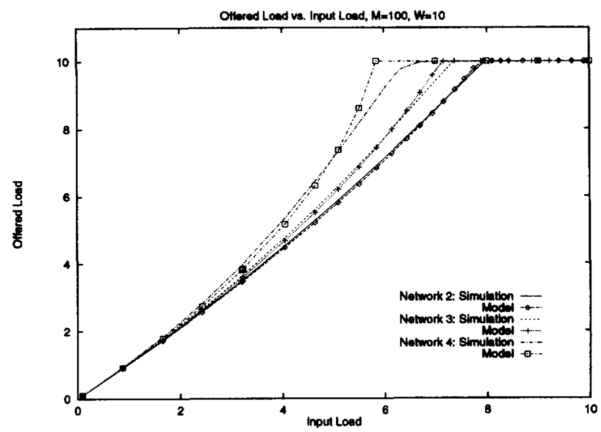

Fig. 10. Offered load versus input load characteristics.

to (10). Obviously, for a stable network operation, the value of I should be kept low enough so that the corresponding value of G does not become too close to the network capacity W.

V. Summary and Conclusions

In this paper, we proposed a new collision-free media access protocol for optical networks. PROTON can accommodate a wide range of tunable devices and processors into a pipelined and highly efficient protocol that can readily be used with present technology. For a given network, the parameter $pass\_slots$ describes the minimum number of slots that must pass before any two data packets can successfully be received by a single station. By using several examples, we showed the importance of knowing the value of $pass\_slots$ in assessing the performance of a network. PROTON is a reservation-based protocol. As such, emphasis was made in designing a high-throughput protocol. Nevertheless, we showed that the average packet delay characteristics of the protocol are still very good. Contrary to ALOHA-based reservation protocols, the throughput of the proposed scheme grows monotonically as the input and offered loads increase.

Although it was not explicitly considered, the protocol can easily be expanded to handle multiple TT/TR pairs per station. In an actual implementation, specialized servers are likely to be present in the network. It is evident that to achieve high throughput, these servers will need more than one micro-slot each (to place reservations), and will be required to have multiple transmitter/receiver pairs. With enough receivers per station, it can be proved that a network using the proposed protocol can achieve a throughput equal to unity.

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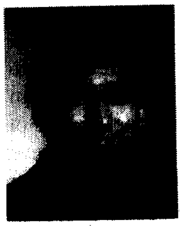

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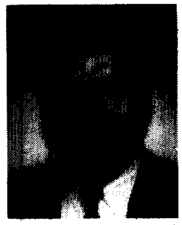

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