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Average Packet Transfer Delay Performance of an Enhanced CSMA/CD based Single Channel Fast Ethernet Optical LAN

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ABSTRACT CSMA/CD based Ethernet does not guarantee delay bound and behaves poorly under heavy load conditions because of excessive collisions and backoff algorithm. It is not suitable for supporting real-time multimedia traffic. This paper presents an enhanced CSMA/CD to improve average packet transfer delay by reducing collisions. Here, each node has a finite buffer capacity that helps to reduce collisions. To guarantee a bounded delay for multimedia applications, the maximum retransmission attempt and backoff limit are reduced from 16 to 10 and from 10 to 8 respectively. Average packet transfer delay based on proposed CSMA/CD is evaluated against several design parameters i.e. number of nodes, bus length and offered load. The delay results show significant improvement. Percentage of collisions is reduced to less than 3% and thereby average packet transfer delay is reduced to less than 2 ms compared to conventional CSMA/CD.

(CSMA/CD, Optical LAN, Fast Ethernet, collision reduction, average packet transfer delay)

INTRODUCTION

The CSMA/CD media access control protocol used in Ethernet does not guarantee bounded delay and behaves poorly under heavy network load, leading to intolerable delay as well as packet loss due to a large number of collisions. It cannot support audio/video traffic in the presence of bursty data traffic even when the network utilization is fairly low (i.e. 10 to 15 percent) [1]. For audio/video traffic, data generated at the transmitter must be delivered to the receiver within a bounded time [1, 2]. Any data delayed more than that time is useless, and considered to be lost. Such loss of information leads to quality degradation in received audio/video, and therefore, must be kept as low as possible.

In this study, to overcome the above mentioned shortcomings and to enhance CSMA/CD performance, three new concepts are added to conventional CSMA/CD. The first concept, each node in the LAN has a finite buffer capacity. A node competes to get access in the medium after its buffer is full and transmits all packets in buffer if access is permitted. To minimize waiting delay of packets in buffer prior to transmission, a time-out period is set, beyond which a node tries to transmit considering its buffer is full.

In this protocol, each node has to wait until its buffer capacity is full. So, number of nodes trying to transmit at a time is reduced and thereby collision rate, bandwidth loss and backoff delays are also reduced.

Secondly, maximum retransmission attempt limit and backoff limit are reduced from 16 to 10 and from 10 to 8 respectively to guarantee a tolerable delay for multimedia traffic.

The final one, a special-jamming signal is introduced to eliminate packet loss. It gives transmission priority to the node that already has finished its maximum retransmission attempt.

Thus it eliminates packet loss due to excessive collisions.

However, if multiple nodes send special-jamming signal at a time, a priority scheduler resolves the problem. In this case, the time-stamp of special-jamming signals is to be used to make a decision. When a node generates special-jamming, it is transmitted to all nodes on the network. If a node generates special jamming signal itself as well as receives from another node within a very short time gap, which event occurs first will get preference. This means, if a node generates special-jamming signal before receiving from another node it will transmit first.

In the worst case scenario, when multiple nodes send special-jamming signal at exactly the same time (which is a highly unlikely case, especially when the time stamp among the signals goes down to several decimal places), the node having the smallest source address (SA) will transmit first. During this period, other nodes wait until their access is permitted accordingly. To accomplish this comparison each node has a comparator unit that compares the source addresses of the nodes with special-jamming signal and finally finds out the node with smallest SA.

This paper is organized as follows: the following section deals with enhanced CSMA/CD protocol. Then, the comparator unit (CU) operation of the proposed protocol is discussed followed by assumptions and simulation parameters. Next, the results and discussions are presented. The paper ends with a conclusion.

ENHANCED CSMA/CD PROTOCOL

The proposed protocol is based on the physical bus topology where all N nodes are spaced equally along the bus. All nodes share a single fiber cable that consists of only one wavelength or channel for data transmission. Each node is equipped with a transmitter and a receiver. A finite buffer is placed at each transmitter. Each node is also equipped with a comparator unit (CU). The CU is responsible for handling the mechanism of data packets transfer in case of multiple nodes finished their maximum retransmission attempts at exactly the same time. The CU is assumed to have a very fast processing time to reduce the total delay. The structure of the protocol is shown in Figure 1. The modifications proposed are contained within the dotted boxes. The rest represents the conventional CSMA/CD protocol.

In this model, each node has a finite buffer capacity and a fixed time-out period. Whenever a node generates a new packet, it is stored in buffer if space is available and number of packets in buffer is incremented by one. This process continues until buffer is full by the pre-set maximum number of packets or time-out period is expired. Any node having either enough packets to fill its buffer or has expired the time-out limit moves on to check out the channel condition.

If the channel is busy, the node computes delay and random backoff time. Otherwise, the node waits for inter frame gap and then begins to transmit by releasing the packets in its buffer in an ordered manner. The next step is to detect collision while transmission is in progress. If a node detects a collision, it will abort transmission. A modification is introduced in this collision detection portion. In this algorithm, a node trying to transmit a packet tries maximum of 10 attempts (in an incremental manner) and after that it sends a 40 bits special-jamming signal which differs from normal 32 bits jamming signal. This special-jamming signal gives the node priority to start transmission while other stations will backoff. Thus it ensures no packet is lost or discarded. In conventional CSMA/CD, the packet is discarded after 16 attempts. The contents of common jamming signal and special-jamming signal is specified in Figure 2.

Jamming signal and special-jamming signal both consist of synchronization, starting of jamming/special-jamming, source address and jamming/special-jamming data. Special-jamming signal has additional priority byte. Synchronization serves to give nodes in the network time to detect the presence of the jamming and the special-jamming signal and being reading the signal before other information arrives. Starting of jamming/special-jamming indicates the start of the jamming/special-jamming. The Source Address identifies the node that originated the jamming and the special-jamming signal. The additional Priority in the special-jamming signal gives the transmission priority followed by the special-jamming signal. The size of jamming and special-jamming signal includes all bytes from the source address field through the jamming data field and priority field respectively. The synchronization and start frame fields are not included when quoting the size of a frame.

However, in case of multiple nodes sending special-jamming signal almost at the same time, there is also a scheduling scheme. In this case, the problem can be resolved depending on either time-stamp of special-jamming signals (as explained earlier) or by the smallest source address priority as explained in the next section.

COMPARATOR UNIT OPERATION

When a node transmits special-jamming signal the Source Address value of that node is copied and transferred to the comparator unit (CU). If there is only one input at any time in the CU then it does nothing except hold that data until the completion of current transmission corresponding to that special-jamming signal. When multiple nodes transmit special-jamming signal at exactly the same time, multiple inputs are transferred to the CU. At this condition, the CU becomes active; it compares the input values, makes a decision about which node will get the chance of transmission first depending on the smallest Source Address priority. This process takes place at all nodes and all of them are expected to make the same prioritization decision since they all receive the same information.

ASSUMPTIONS

Average packet transfer delay of proposed CSMA/CD based single channel optical LAN is evaluated within Fast Ethernet environment. Assumptions that have been made in simulation process are as follows:

- Arrivals at all nodes follow a Poisson distribution.
- All nodes generate traffic at the same rate. Packets are assumed to be generated at any node j with nominal rate R_j. For this simulation, R_j = 980 packets/sec (which corresponds to a 1% of the maximum packet transmission rate). However, 100 nodes are attached to the network, which represents a total of 100% incoming traffic.
- Packet length is fixed to 1024 bits/packet to make sure that no packet is shorter than twice the minimum frame size, and all collisions are detectable during transmission time.
- Each node has a finite buffer capacity.
- Nodes are equally spaced along the bus.
- All received packets are error-free. Errors occur due to collision only.
- No packet priority is considered.
- The system is lossless i.e. there is no packet loss.
- A new generated packet joins the tail of the queue in buffer if space is available, otherwise it is lost. It is also lost when the node is busy (i.e. during transmission or undergoing a collision).
- Each user is allowed to transmit all packets in its buffer during each transmission. Packets in buffer are assumed to be transmitted on a first-come-first-served (FCFS) basis.
- The packets are deleted from buffer immediately after the successful transmission is completed.

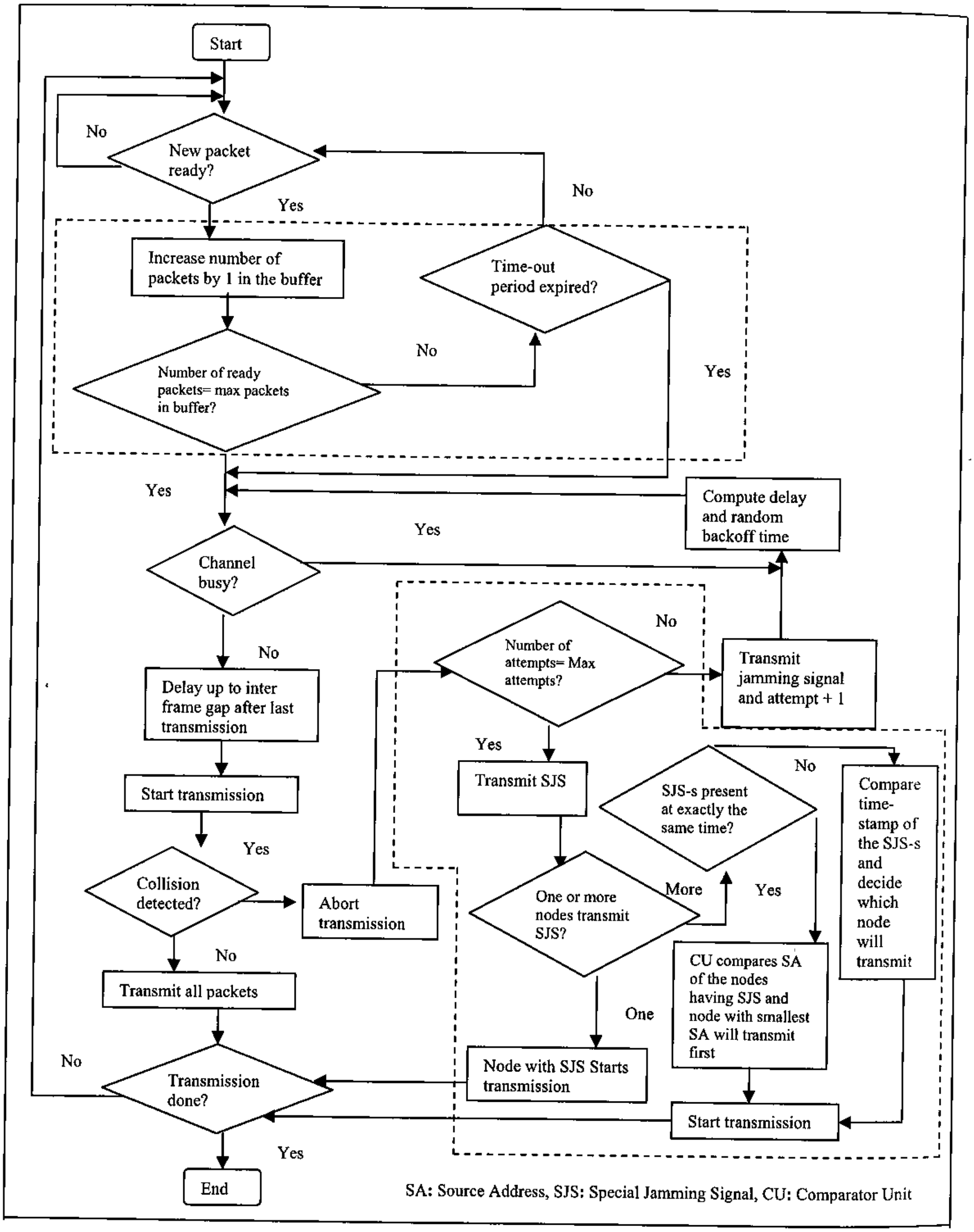

Figure 1. Flow chart of proposed CSMA/CD

Synchronization (1 byte)	Start of jamming (1 byte)	Source Address (1 byte)	Jamming Data (3 byte)

(a) Jamming signal

Synchronization (1 byte)	Start of special-jamming (1byte)	Source Address (1byte)	Special-jamming data (3 byte)	Priority (1byte)

(b) Special-jamming signal

Figure 2. Jamming signal and special-jamming signal format

SIMULATION PARAMETERS

Visual Basic 6.0 is used for simulating the protocol. The pseudo-random number generator used generates Poisson traffic. The effect of different buffer size on average packet transfer delay is observed first and then the optimum buffer size is determined by simulation. Performance of the protocol is investigated with optimum buffer capacity i.e. 10 packets/buffer.

Some initial waiting time is introduced by buffer as the first packet arriving in the buffer waits for more packets to arrive before it can be transmitted. In order to minimize the starting delay caused by buffer for lightly loaded network, the time-out period, T = 0.05 sec is chosen considering 20% network load condition. Here it is assumed that up to 20% of load, network load can be considered small enough. At 100% offered load, packet generation rate at any node is 980 packets/sec. At 20% offered load, packet generation rate is 196 packets/sec. According to this rate, 0.05 sec is required to generate 10 packets. So, time-out period is chosen 0.05 sec. T can also be chosen considering other network load condition like 10% or 15 etc. Value of T increases with decreasing network load, which means buffer fulfillment time increases with decreasing network load. Longer buffer fulfillment time is not desired. So, T is chosen at 20% network load condition.

The average packet transfer delay is influenced by number of nodes, bus length and offered load. Number of nodes varies from 5 to 100 in steps of 5. Optical fiber is used as the transmission medium so that the maximum bus length can be 2 km. According to Dutton [3], LANs span up to 2 km of length and this maximum length is taken as the maximum bus length. To observe the protocol performance influenced by bus length, it is varied from 100 meter to 2 km in steps of 100 meter. Offered load is varied from 5% to 100% in steps of 5%. To observe the proposed protocol's performance influenced by number of nodes and offered load, bus length is kept fixed at 500 meter. Most of the conventional LANs are operated within bus length of 200 meter. In order to compare performance enhancement of this protocol, a higher bus length is chosen. All parameters used in simulation are summarized in Table 1.

Table 1. Simulation parameters of the proposed CSMA/CD

DESIGN PARAMETERS	VALUES (FAST ETHERNET)
Maximum station, N	100
Transmission rate	100 Mbps
Packet length	1,024 bits
Optical fiber bus length	500m
Maximum packets	100,000
Propagation speed	$2x10^8$ m/s
Slot time	512 bits
Inter frame gap	0.96 μs
Buffer size	10 packets
Time-out period, T	0.05 sec
Attempt limit	10 times
Back off limit	8 times
Jam size	32 bits
Special-jamming size	40 bits
Minimum frame size	512 bits

RESULTS AND DISCUSSIONS

Average packet transfer delay is the delay experienced by a packet from the beginning of its first transmission attempt to the end of its successful transmission [4]. It is computed by dividing the total delay of all packets by the number of packets successfully delivered [5]. It is convenient to regard the transfer delay as consisting of three components. The first component t_w, is called *waiting time* or *access time*. It is the time elapsed from the first transmission attempt of a packet until the beginning of its transmission on the channel. The second component t_{prop}, is called *propagation time*, is the time elapsed from the beginning of the transmission of the message until the arrival of the first bit of the message at the destination. The third component is *transmission or service time* t_{tx}, which is the time, elapsed between the arrival of the first bit of the message at the destination and the arrival of the last bit. As soon as the last bit arrives at the destination, the transfer is complete. So, the total delay is a function of transmission time (t_{tx}), waiting time (t_w) and propagation delay (t_{prop}) as given by the Equation 1 below.

$$\text{Average Delay,} = \frac{TotalDelay}{PacketsSuccessfullyTransmitted}$$

$$= \frac{Transmissiontime + Waitingtime + PropagationDelay}{TotalBitsSend} \tag{1}$$

Average packet transfer delay analyzed against number of nodes, bus length and offered load are as follows.

Average Packet Transfer Delay vs. Number of Nodes

Figure 3 represents average packet transfer delay versus number of nodes with bus length of 500 meters and 100% offered load. Contention problem increases with increasing nodes in spite of buffer because larger number of nodes leads to a larger number of transmission attempts. This increases collision rate and consequently re-sensing and retransmission process increase the waiting time of packets in each node. Therefore, overall average packet transfer delay increases with increasing number of nodes.

In Figure 3, average delay raises from 0.694 ms to 0.955 ms. At 100 nodes, delay is less than 1 ms. This is much less than typical acceptable delay of 100 ms [6] and the delay shown in [7]. In [7], Kweon presented a real-time communication technique over Fast Ethernet by traffic smoothing. In his approach, the average delays were 8.2 ms and 8.4 ms in the strict traffic smoothing model and the coarse time scale traffic smoothing model respectively. This average delay improvement of the proposed protocol happens because of reduced collision rate (compared to [6]) as shown in Figure 4. Delay improvement compared to the model proposed by Rodellar [8] is depicted in Figure 5.

The delay obtained here is very close to the packet delay carrying voice and data using VoIP where packet delay without QoS is from 0.8 to 1.16 ms and with QoS from 0.58 to 0.66 ms [9]. As a whole, average packet transfer delay of this protocol is excellent and it can be used for real-time traffic applications.

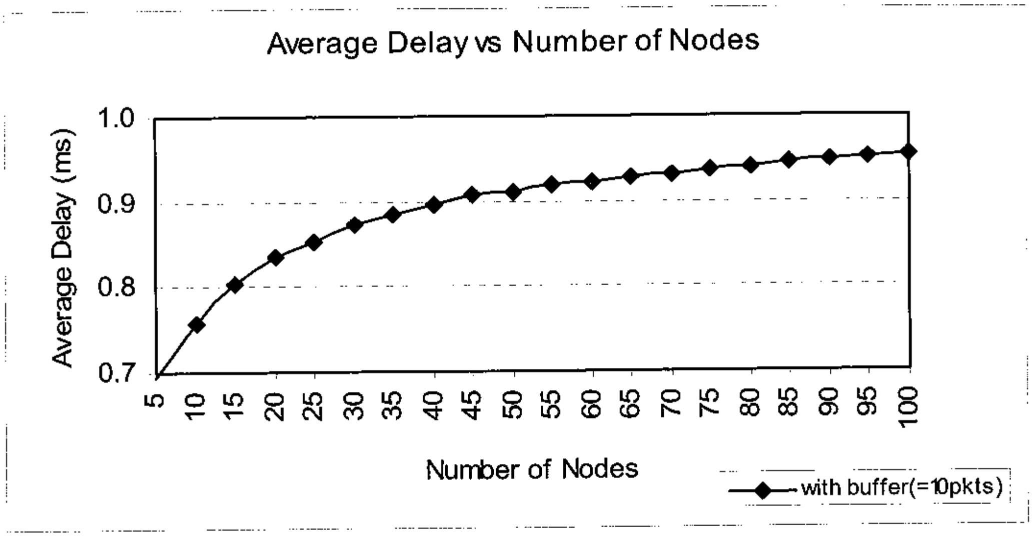

Figure 3. Average packet transfer delay versus number of nodes

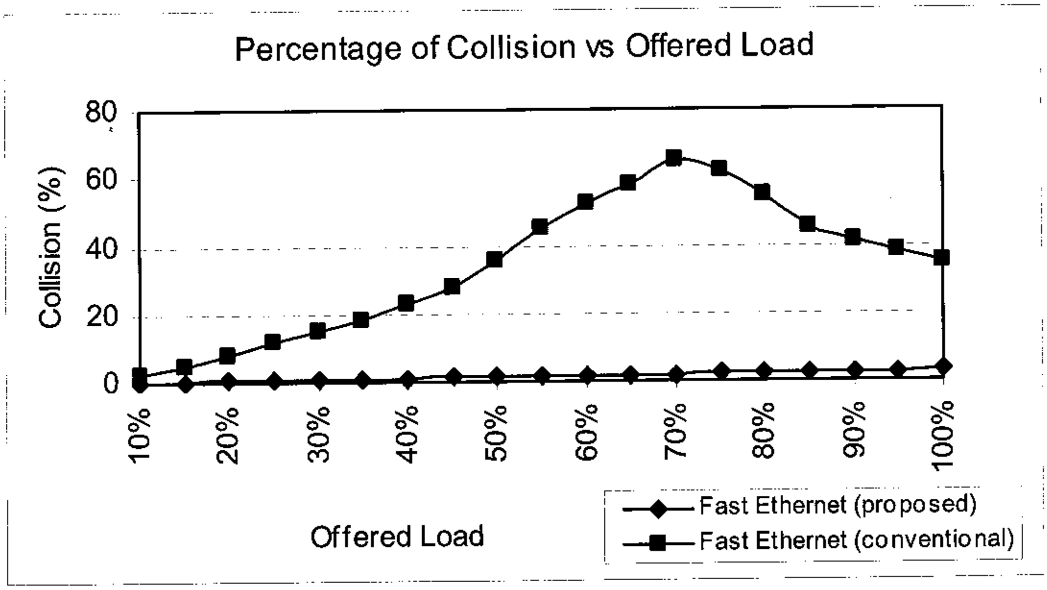

Figure 4. Comparison of percentage of collision versus offered load

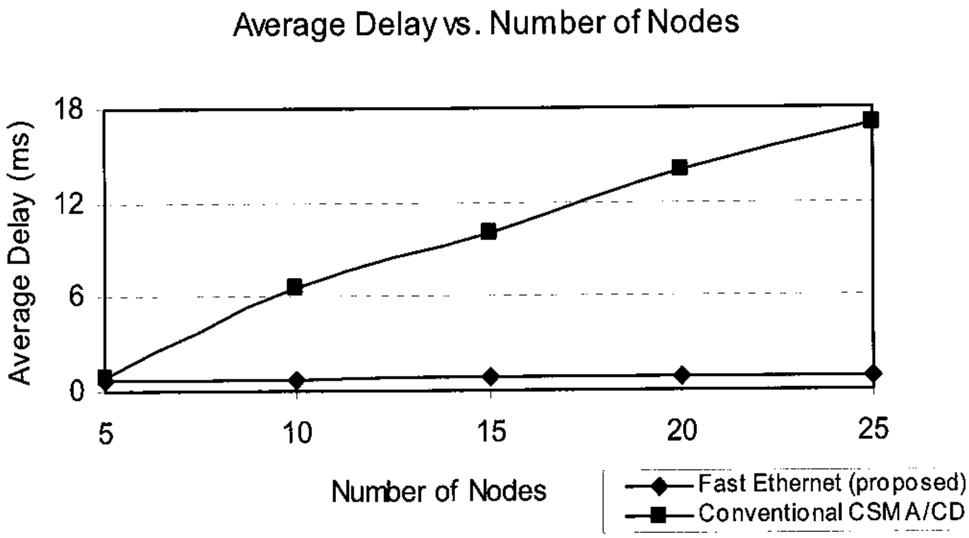

Figure 5. Comparison of average packet transfer delay versus number of nodes

Average Packet Transfer Delay vs. Bus Length

Average packet transfer delay versus bus length is depicted in Figure 6 with 100 nodes connected at 100% offered load. Average packet transfer delay consists of transmission time, propagation delay and waiting time. Only propagation delay increases with increasing bus length. But, increase in propagation delay is much less (in the range of μsec) compared to transmission time and waiting time. Hence, the overall effect of increasing bus length on average delay is very small.

Average packet delay is around 1.03 ms for a relatively long distance of 2 km (for a LAN) which is very small compared to the typical acceptable delay of 100 ms of the conventional LAN [6] and the delay shown in [10]. In [10], Wong introduced a CSMA/CD based Ethernet over passive optical network for delivery of VoIP traffic, where average packet delay is 14.03 ms for 1 km distance. Thus performance improvement of the proposed CSMA/CD is apparent.

Average Packet Transfer Delay vs. Offered Load

Effect of offered load on average packet transfer delay is presented in Figure 7. Offered load varies from 5% to 100% with 100 nodes at a bus length of 500 meter. Waiting time is affected by offered load. Due to increased offered load, collision rate increases. Each collision results in re-sensing and retransmission which finally increase waiting time of packets. As a whole, total delay increases from 0.051 ms to 1.01 ms with increasing offered load.

In Figure 7, at 100% offered load, average packet delay of proposed protocol is 1.01 ms. It is smaller compared to the average delay of conventional Fast Ethernet, which is 1.5 ms for the same offered load [11, 12]. The delay obtained here is also less than that shown in [13]. The delay improvement compared to typical acceptable 100 ms delay of conventional LAN [6] is depicted in Figure 7 too.

At light load, average delay of proposed protocol is higher than conventional protocol. This is due to longer transmission delay introduced by buffer. But at heavy load (beyond 60% offered load), average delay proposed protocol is better than conventional protocol because of less backoff delay due to less collision. So, from Figure 7, it is evident that though buffering causes increased transmission delay the overall result of average delay is not affected. It is still comparable and within acceptable limit due to less collision than the conventional system.

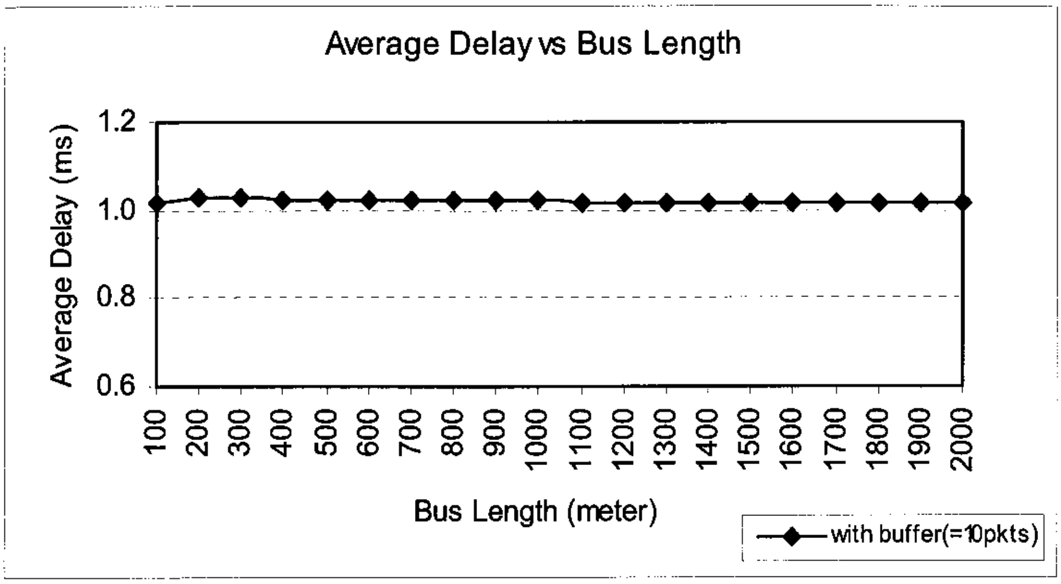

Figure 6. Average packet transfer delay versus bus length.

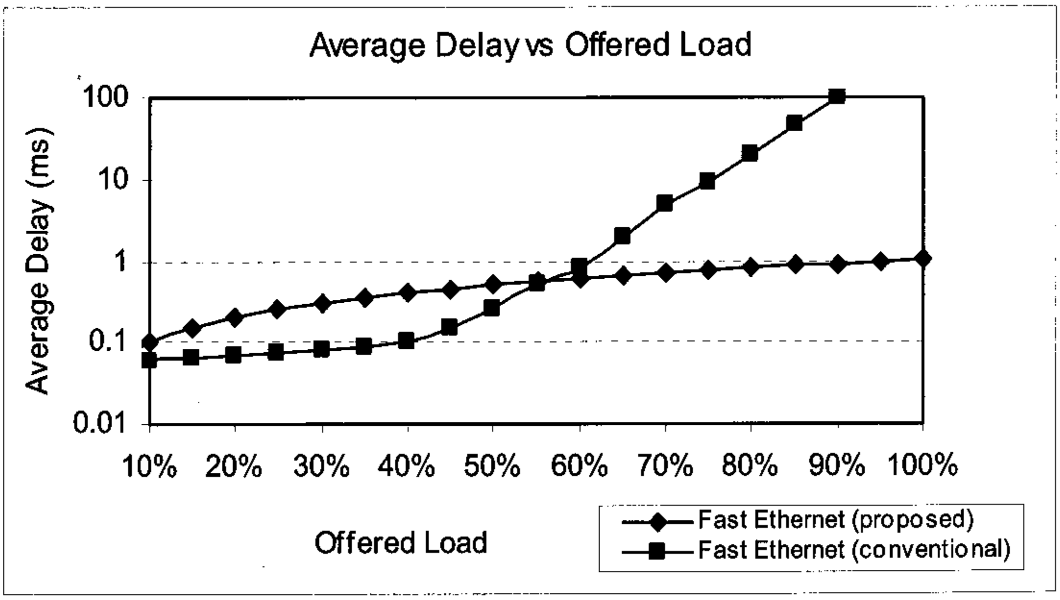

Figure 7. Comparison of average packet transfer delay versus offered load

CONCLUSIONS

This paper shows average delay performance of a single channel Fast Ethernet network based on enhanced CSMA/CD. Although transmission delay is increased a little due to buffer, it is compensated by reduced backoff delay (waiting time) as percentage of collision is minimized. Moreover, reduced maximum retransmission attempt limit and backoff limit also help to reduce exponential backoff delay. So, average delay is within acceptable range and much less than typical delay of conventional LANs. Moreover, this small average packet delay could be used to support multimedia applications.

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