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Voice/Data Integration on Ring Network:**

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ABSTRACT

In this paper two access protocols for voice/data integration on ring networks are proposed. The first protocol is a hybrid scheme based on the slotted ring and the register insertion ring protocols. The second protocol is based on the standard token ring network. The central idea to both protocols is to provide decentralized priority to voice stations. Quick access to the transmission medium is guaranteed to all voice stations while still providing data stations with reasonable delay-throughput performance. Furthermore, both protocols can dynamically adapt to the voice traffic load. We develop analytical models for the proposed protocols to evaluate their performance. The analytical results have been validated through extensive simulation experiments. The result show that these two protocols perform better than some of the existing protocols.

1 Introduction

Packetized voice transmissions on a local area network have gained a considerable interest among computer and communication researchers. Extensive studies have been reported recently in developing packetized voice communication systems on local area networks (LANs) which were originally designed for data communications between a number of distributed computers located in a small area [1,2,3,4,5,6,7]. Traffic characteristics and performance requirements of voice packets differ in many ways from that of data packets. First of all, the voice transmission requires a real-time delivery in order to have an acceptable quality of sound. The primary concerns of data transmission, on the other hand, are the low average delay, high throughput and good error control and recovery procedures. Secondly, in a voice conversation, a speaker alternates between active and silence states. Voice packets are delivered to the network only when the speaker is in an active state. Typically, a voice call lasts for a few minutes, and a speaker is active for less than half of the time. The network bandwidth available during the silence period can be exploited to serve other types of traffic such as data. In this way, effective sharing of transmission, multiplexing and switching resources among a large user population can lead to the achievement of economies of scale as well as enhance service capabilities.

Despite its promising future, there are a number of issues relating to integrated voice/data transmission that need to be addressed. The critical issue is the time



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constraint transmission of voice packets that limits the maximum number of voice stations that can be connected to a network. Due to contention and queueing delay at the network, a voice packet may experience a random amount of delay to reach its destination. If this delay exceeds a threshold, the packet is deemed lost leaving a gap in the reconstructed sound. In order to deliver a speech with acceptable quality, the percentage of such lost packets should be typically less than 1 percent [8]. Based on CSMA/CD and ring networks, a number of approaches have been proposed to reduce the number of lost packets while increasing the number of connected voice stations [1,2,3,4,5,6,9,7]. Ring networks are very popular due to the bounded delay feature. However, a number of studies have shown that only a small number of voice stations can be accommodated on a token ring network to ensure an acceptable performance [9,10]. In this paper, two new access schemes are proposed based on conventional slotted/token ring and buffer insertion ring protocols. Our first access scheme, referred to as the hybrid protocol, takes the advantages from both slotted ring and buffer insertion ring protocols. A high priority for voice traffic is guaranteed by allowing voice stations to dynamically insert buffers as needed. The second scheme is a decentralized token ring protocol in which a data station polls all voice stations immediately after it obtains a data token. Data transmission, if any, follows the completion of the voice poll.

In order to study the performance of these two new protocols, analytical models have been developed in this paper. Approximate probabilistic and queueing analysis are used to solve the models. Results obtained from the analytical models are in a good agreement with simulation results over a wide range of network load. Our results show that the new protocols significantly improve the service capability of a network for both voice and data transmissions.

The paper is organized as follows. In the next section, the access protocols are presented. Section 3 presents our model assumptions and performance parameters used in our analysis. The analysis for both protocols are presented in Section 4 followed by numerical results and discussions in Section 5. Finally, Section 6 concludes the paper. For readability, a list of important notations used in the paper is included in the Appendix.

2 The Protocols

The first protocol is based on a hybrid approach of conventional slotted ring and register insertion ring access schemes [11]. The primary goal of this protocol is to give voice stations high priority by reducing ring access times of voice packets. The ring access time is defined as the time between the moment when a packet reaches the head of the buffer queue and the moment when the first bit of the packet has been put on the ring. We assume that the total ring round trip delay is larger than the length of a packet. This implies that one or more equal length slots can circulate on the ring. At the beginning of each slot, there is a special bit, called flag, indicating whether the following slot is empty or full. Examples of LAN's based on the slotted ring access scheme are the Pierce ring with destination removal [12,13] and the Cambridge ring [14] with source removal. Access schemes adopted by the voice and data stations in the hybrid protocol are described below.

A data packet ready for transmission is first put in a packet buffer. There may be a number of packets in the buffer waiting for transmission. When a slot on the ring arrives at the station, the first bit is tested to see if the slot is empty or full. If it is empty, then the first packet in the queue will be placed on the slot for transmission. The destination station is responsible to remove the packet from the ring. It appears that the destination removal policy requires additional delay at the intermediate stations compared to source removal policy in order to recognize destination address. This is because, in case of source removal, since the source station knows exactly when the transmitted packet is coming back (if either the total number of slots in the ring or the total ring delay is known), the additional delay introduced by intermediate stations to do destination address matching is avoided. However, since our protocol allows dynamic change of the ring length, as explained later, the advantage of source removal can not be utilized.

Voice signals generated by a voice station are assembled at the network interface, in equal length packets. A buffer is provided at the interface to hold voice packets. The access protocol of these voice packets is based on the register insertion protocol. There are two basic control registers at the interface of each node as shown in Figure 1. One is receive-shift-register that is connected to the incoming line of the ring and the other is transmit-shift-register. Packet transmission and reception are controlled by a switch with 3 positions as shown in Figure 1. If the station does not have an

outstanding packet on the ring, the switch is connected to position 1. All information coming on the ring is shifted in parallel into the receive-shift-register. If the packet contained in the register is destined at the station the packet is passed to the host.

When a packet is ready to be transmitted it is placed in the transmit-shift-register. The interface then keeps monitoring the ring. As soon as a slot boundary is detected, the following sequence of events starts. The switch that was originally at position 1 is moved to position 3 so that the contents of the transmit-shift-register are shifted on to the ring. Mean time, incoming information is saved in the receive-shift-register. Immediately after the last bit of the transmit-shift-register is shifted out to the ring, the switch is set at position 2 to relay the incoming information stored in the receive register. When the transmitted packet returns to the receive register the packet is removed from the ring by setting the switch to position 1. A station can have only one outstanding packet at a time since the next packet can not be sent until the current packet is removed from the ring.

With this protocol a voice station obtains higher priority over a data station in a distributed manner. The access time of a voice packet is significantly reduced because it does not have to wait for an empty slot. Consider a situation where the ring has 4 slots with length x bits each and the ring utilization is U . The average access time of a packet (for both voice and data) for a pure slotted ring case would be $x/2 + xU/(1-U)$ [15]. On the other hand, the mean access time of a voice packet in the above protocol is only $x/2$. For U equal to 0.5, the reduction of the access time is x bits. However, this reduction comes at the cost of longer propagation delay due to inserted registers, though the maximum propagation delay is bounded. Our analyses in a subsequent section show that this protocol gives performance improvement over the conventional token ring protocol.

The second protocol, referred to as the distributed token ring protocol, is based on the standard token ring access method described in IEEE 802.5 [16]. Voice data integration on a token ring network was studied by Wong and Gopal in [17]. In their research, they proposed a central controller which provides priority to voice stations. The protocol proposed in this paper provides decentralized priority to voice stations using a priority scheme similar to that proposed in the IEEE 802.5 token ring access method. There are two tokens, namely, the data token and the voice token. Only one of the two tokens is active on the ring at any time. The voice token can only be used by voice stations and the data token can be used only by data stations.

Consider the data token that circulates around the ring. Each data station monitors the ring to obtain the data token. When a data station gets the data token, irrespective of whether it has a packet to transmit or not, it polls all the voice stations by sending a voice token. When the initiating data station receives the voice token, it transmits a packet (if it has one) and/or releases the data token. The voice stations do not recognize the data token. On receiving the voice token, a voice station transmits one voice packet and sends the voice token to the next voice station.

With respect to the above description of the protocol, there are three important points to be noted. First, similar to the hybrid protocol, there is no central controller to provide priority to the voice stations. The protocol provides a "floating supervisor" scheme. This is important from the point of view of providing fault-tolerance in the event of station failure. Secondly, the above protocol is easily implemented using the priority scheme provided in the standard token ring access scheme. Finally, based on the above description it is clear that large waste of bandwidth is expected due to the polling mechanism, especially when there are no active voice stations on the ring. However, this scheme can be easily extended to dynamically adapt to the voice traffic load on the network. The main idea of such an adaptive approach is described as follows.

We incorporate control fields in both the voice and data tokens to dynamically modify the polling scheme based on the number of active voice stations. The control field in the voice token will be used to count the number of active voice stations in a voice poll. The control field in the data token will specify the number of voice polls that will occur during one data token rotation time. The data token control field is set by the data station depending on the number of active voice stations (i.e., voice control field) at the end of a voice poll. In this scheme, when the voice traffic is low not all data stations will poll voice stations thereby allocating more bandwidth to data stations. Note that the distributed token ring protocol described before is a special case of the above protocol where each data station must initiate a voice poll.

3 The Model, Assumptions, and Parameters

3.1 Model Assumptions

Let M_v and M_d denote the total number of voice and data stations, respectively. It is assumed, without loss of generality, that both voice and data stations are alternately

distributed on the ring. The arrival streams of data packets to all data stations are assumed to have identical Poisson distribution with the same rate and all packets have the same size. Each voice station, on the other hand, alternates between talkspurts and silences. In a talkspurt, the voice station is involved in some conversation with one or more stations. The speaker's voice signals are digitized and assembled into packets to be transmitted to a destination station. During a silence period the voice station does not generate any packet. The durations of the talkspurts and silence periods are assumed to be exponentially distributed with mean values of 0.17 and 0.41 seconds, respectively. Let x_v be the voice packet length and V be the digitization rate of the encoder in a voice station. This implies that time between the generations of two subsequent voice packets during a talkspurt, T_g , is give by

$$T_g = \frac{x_v}{V}$$

We assume that packet generation cycles are independent of packet transmission processes. It is also assumed that data stations and voice stations are independent from each other. This is an approximation since in reality stations are correlated. However, as will be shown later, this approximation gives reasonably accurate results.

Let λ and μ denote the mean length of a talkspurt and a silence period, respectively. A simple two state Markov chain can be used to model the behavior of a speaker [17]. Using states 0 and 1 to represent silence and talkspurt states, respectively, the transition matrix of the Markov chain is given by

$$P = \begin{pmatrix} p_{00} & p_{01} \\ p_{10} & p_{11} \end{pmatrix} = \begin{pmatrix} e^{-t/\mu} & 1 - e^{-t/\mu} \\ 1 - e^{-t/\lambda} & e^{-t/\lambda} \end{pmatrix}$$

Since all p_{ij} 's are positive, P is ergodic and therefore, the steady state probabilities, π_t and π_s , of a speaker being either in talkspurt or silence states exist and they are given by

$$\begin{aligned} \pi_t &= \frac{p_{01}}{p_{01} + p_{10}} \simeq \frac{\lambda}{\lambda + \mu} \\ \pi_s &= 1 - \pi_t \simeq \frac{\mu}{\lambda + \mu} \end{aligned}$$

We will not consider the effect of termination and generation of voice calls in our analysis and simulation. Every voice station on the ring is assumed to be involved in a conversation throughout the time considered. This assumption is reasonable because the fluctuations in the generation and termination of the voice calls are much slower than the changes in generation and transmission of voice and data packets. Therefore, the effect of the former upon the latter is negligible.

3.2 Performance Measures

Due to the primary importance of real-time delivery of human speech, performance measures showing how a local ring network preserves the integrity of the conversation are of interest. Consider, for example, the time interval between consecutive departures of packets from the ring to a destination station. This time is defined as the interdeparture time. Large values of interdeparture time may affect the playout performance of a speech at the receiver. The mean and variance of interdeparture times can be used to characterize the performance of a network. Another important measure is the voice packet transmission delay which is the total time a voice packet experiences from the beginning of a packet digitization to the reception at the destination station. Let D_v denote the voice packet transmission delay. It is given by

$$D_v = T_g + W_q^v + \frac{x_v}{C} + W_p,$$

where T_g is the generation period of a voice packet, which is fixed for a given length of voice packets as well as the coding rate of the encoder; W_q^v is the mean queueing delay faced by a voice packet at a station before it accesses the ring; $\frac{x_v}{C}$ is the voice packet transmission time; and W_p is the propagation delay. Note that W_q^v and W_p are both random variables that are derived in the subsequent analyses. It is desired that D_v be kept small since a large value of D_v can have disruptive effects on speech transmission. Another performance measure of interest is the probability that a voice packet is lost due to excessive delays. The parameter, p_{loss} , specifies the proportion of lost packets to the total number of packets and is defined as

$$p_{loss} = Pr(D_v > \tau) = 1 - Pr(D_v \leq \tau),$$

where τ is the specified time bound. In order to maintain an acceptable quality of speech, the proportion of lost packets to the total number of packets transmitted should be small. For data transmission, the important performance measure is the average transfer delay, D_d , which is given by

$$D_d = W_q^d + \frac{x_d}{C} + W_p,$$

where W_q^d and $\frac{x_d}{C}$ are the queueing delay of a data packet at the source station and its transmission time, respectively and W_p is the propagation delay.

4 The Analysis

In this section, analytical models are presented for the two protocols based on the assumptions discussed in the previous section. The analysis of the two protocols follow the same approach. For clarity, however, we have presented them separately in subsections 4.2 and 4.3. The main idea of analysis is as follows. First, an approximation technique for modeling the voice packet arrival process is discussed in subsection 4.1. In this approximation, the arrival process of voice packets is modeled by a batch arrival process. Subsequently, the distribution of the voice packet transmission delay is derived. This is carried out based on the assumption that all the voice packets that arrive during a talkspurt are completely transmitted by the end of the talkspurt period. Since this assumption is not realistic at high load, we adopt a heuristic function to approximately modify the length of the talkspurt and silence periods to incorporate this behavior.

4.1 An Approximate Model for Voice Arrival Process

As indicated in the previous section, the arrival stream of voice messages at a voice station is considered as having alternate talkspurt and silence periods which are exponentially distributed with mean length λ and μ , respectively. When a station is in a talkspurt, voice packets are generated periodically with a constant interval T_g . While during a silence period no packets are generated. Due to the lack of tractable analysis for the arrival process of voice packets, we use a Poisson batch arrival model, similar to Karvelas and Leon-Garcia's approach [4], as an approximation to the real voice arrival process. The basic idea in this approximation is to match the first two moments of voice packet interarrival times to that of a batch arrival process by selecting the mean batch interarrival time and the mean batch size. The derivation is as follows.

Let $\bar{n} = \frac{\lambda}{T_g}$ be the average number of voice packets in a talkspurt. As soon as the last packet arrives at the station, the current talkspurt terminates and an exponentially distributed silence period follows. The time between the arrival of the last packet in a talkspurt and the arrival of the first packet of the next talkspurt is equal to the silence period plus the packet generation period. This time, on average, will be $\mu + T_g$. Let x denote the packet interarrival time in this process. Then the first two moments of x are

$$E[x] = \frac{\bar{n} - 1}{\bar{n}} T_g + \frac{1}{\bar{n}} (\mu + T_g) = \frac{\lambda + \mu}{\lambda} T_g$$

$$E[x^2] = \frac{\bar{n} - 1}{\bar{n}} T_g^2 + \frac{1}{\bar{n}} (\mu + T_g)^2 = \frac{(\lambda T_g + \mu^2 + 2\mu T_g) T_g}{\lambda}$$

Now consider a batch arrival model [18]. Let $E[N_B]$ and λ_B be the mean batch size and mean batch interarrival time, and let y denote the packet interarrival time in this model. In the same way as for x , the first two moments of y are given by

$$E[y] = \frac{\lambda_B}{E[N_B]}$$

$$E[y^2] = \frac{\lambda_B^2}{E[N_B]}$$

By setting $E[x] = E[y]$ and $E[x^2] = E[y^2]$ the expressions for $E[N_B]$ and λ_B can be derived and shown to be

$$E[N_B] = \frac{\lambda^2 + 2\lambda\mu + \lambda\mu^2/T_g}{(\lambda + \mu)^2}$$

and

$$\lambda_B = \frac{(\lambda + 2\mu)T_g + \mu^2}{\lambda + \mu}$$

The total network delay experienced by the first packet of each batch is used to estimate the average voice packet delay of the real voice arrival process in the network [4].

4.2 Analysis of the Hybrid Ring Protocol

In the hybrid protocol voice packets are transmitted using a register insertion scheme and data packets are transmitted using a conventional slotted ring protocol. Based on the assumptions stated in Section 3, the steady state probabilities that a voice station is in talkspurt and silence states are π_t and π_s , respectively. Assuming that all voice stations are independent of each other, the number of active voice stations is a binomially distributed random variable with parameters M_v and π_t . Let q_i be the probability of i stations being active. Clearly, q_i is given by

$$q_i = \binom{M_v}{i} \pi_t^i \pi_s^{M_v-i} \quad (1)$$

Note that an active voice station may have a register inserted onto the ring, which contributes an additional delay of x_v/C .

For simplicity, we assume that a voice station is involved in voice transmission only when it is in an active state. This is an approximation because, due to various delays, some transmissions may still be going on even after a voice station's talkspurt is over.

However, considering the fact that the length of a talkspurt is usually much longer than a packet generation period, this approximation is reasonable for low to medium traffic load. Later in this section, we will discuss a method that is used in our analysis to reduce the error attributable to this assumption at high traffic loads.

Let T_r denote the packet rotation time, i.e., the time required by a packet to travel around the ring. Clearly, T_r depends on the number of registers inserted in the ring which depends on the number of stations being in the active state during a packet rotation time. We need to obtain the distribution of the number of active stations within one packet rotation period. Assume that there are i active stations in a particular packet rotation time. Then the probability that k stations are active in the following packet rotation time can be written as

$$P(k/i) = \sum_{j=\xi}^{\varphi} \binom{i}{j} p_{10}^j p_{11}^{i-j} \binom{M_v - i}{k - i + j} p_{01}^{k-i+j} p_{00}^{M_v - j - k} \quad (2)$$

where p_{ij} 's depend on T_r . The first binomial term corresponds to i currently active stations and the second one to the rest $M_v - i$ stations. The limits of the summation, ξ and φ , are determined by

$$\xi = \begin{cases} 0 & , k \geq i \\ i - k & , k < i \end{cases}$$

and

$$\varphi = \begin{cases} i & , M_v - i \geq k \\ M_v - k & , M_v - i < k \end{cases}$$

From Eqs.(1) and (2), we can estimate the distribution of the number of active voice stations during a rotation period. Let P_k denote the probability that k stations are active during a rotation period. Then P_k can be expressed as

$$P_k = \sum_{i=0}^{M_v} q_i P(k/i) \quad (3)$$

Suppose that l of k active stations get their registers inserted. Then the packet rotation time, T_r , is given by $(lx_v + R)/C$, where R is the length of the ring without any insertion and C the channel speed. Consider now a single station out of these k stations. Due to the fact that each packet will be generated in a period of T_g , the probability of a packet being inserted is given by

$$p_{in} = \begin{cases} \frac{T_r}{T_g} & , T_r < T_g \\ 1 & , T_r \geq T_g \end{cases}$$

Now, based on the assumption that all insertions at different stations are independent of each other, the number of insertions can be treated as a random variable with a binomial distribution. Therefore, the probability that l stations have their register inserted, given that k stations are active, can be written as

$$P(l/k) = \binom{k}{l} p_{in}^l (1 - p_{in})^{k-l} \quad (4)$$

Thus, the distribution of the number of voice stations involved in voice transmission (i.e., register inserted) during one packet rotation period is

$$P_l = \sum_{k=l}^{M_v} P_k P(l/k) \quad (5)$$

By denoting the first two moments of T_r as $\bar{T}_r$ and $\bar{T}_r^2$, we get

$$\bar{T}_r = \sum_{l=1}^{M_v} P_l \frac{l x_v + R}{C}$$

and

$$\bar{T}_r^2 = \sum_{l=1}^{M_v} P_l \left(\frac{l x_v + R}{C} \right)^2$$

Using these results we can derive the mean voice packet transmission delay D_v . This delay consists of the voice packet generation time T_g , the mean queueing time $\bar{W}_q^v$, the transmission time x_v/C , and the propagation delay from the source station to the destination station which, on average, is equal to $\bar{T}_r/2$, as described earlier.

The mean queueing time $\bar{W}_q^v$ is derived as follows. We define $\rho_v = (E[N_B]/\lambda_B)\bar{T}_r$ as the traffic intensity at each voice station and $U = M_v \rho_v x_v / (M_v \rho_v x_v + R)$ the ring utilization due to the voice traffic [15]. As mentioned in the beginning of this section, the mean waiting time seen by the first packet in each batch is used to estimate the mean waiting time of the original voice packet arrivals. The waiting time of an arriving packet has two components: (1) the residual waiting time of the first packet in the queue, $\bar{T}_{r,r}$, and (2) the time required to transmit all packets ahead in the queue. Since the average number of packets seen by the first packet in the batch is identical to the mean queue length $\bar{Q}$, $\bar{W}_q^v$ is given by

$$\bar{W}_q^v = \bar{Q} \bar{T}_r + \bar{T}_{r,r}$$

The expression for $\bar{Q}$ can be found in [19]. It is given by

$$\bar{Q} = \frac{E[N_B]}{\lambda_B} (\bar{W}_Q + \bar{T}_r)$$

where $\bar{W}_Q$ is the average queueing time over all packets in the batch arrival model. Thus, $\bar{W}_Q$ is obtained from the relation

$$\bar{W}_Q = \frac{\bar{T}_r(E[N_B^2] - E[N_B])/2E[N_B] + E[N_B]\bar{T}_r^2/2\lambda_B}{1 - E[N_B]\bar{T}_r/\lambda_B} \quad (6)$$

where $E[N_B^2]$ is the second moment of the batch size.

When a new batch of voice packets arrive at a station, it can find the station either with or without register inserted. In the former case $\bar{T}_r$, equals to the mean residual life of voice packet rotation time which is equal to $\bar{T}_r^2/2\bar{T}_r$. In the later case, $\bar{T}_r$, would be identical to the mean residual life of a packet passing by the station and is equal to $x_v/2C$ [15]. Furthermore, a new arriving batch will find the register inserted with probability ρ_v and the register not inserted with probability $1 - \rho_v$. Thus

$$\bar{W}_q^v = \rho_v \left(\bar{Q}\bar{T}_r + \frac{\bar{T}_r^2}{2\bar{T}_r} \right) + (1 - \rho_v) \left(\bar{Q}\bar{T}_r + \frac{Ux_v}{2C} \right) \quad (7)$$

Using Eqs. (6) and (7), we can obtain the mean voice packet transmission delay D_v .

Based on these results, the performance parameters such as the mean and variance of the voice packet delay, the percentage of lost voice packets and delay-throughput characteristics can be derived. As an example, let us consider the expressions for mean and variance of the interdeparture time of a voice packet for two different loading conditions. At low load, the number of registers inserted is not large enough to result in a packet rotation time longer than the packet generation period. In other words, a packet transmission will be finished and this packet will return to its source station before the next packet is generated at the same source station. In this case, the interdeparture time is approximately equal to the packet generation period at the source station. On the other hand, at heavy traffic load, the packet rotation time may be longer than the packet generation period. Thus, the interdeparture time will be the same as the packet rotation time. By denoting interdeparture time as W_s , our reasoning leads to

$$E[W_s] = \sum_{l=0}^{\eta-1} P_l T_s + \sum_{l=\eta}^{M_s} P_l \frac{l x_v + R}{C} \quad (8)$$

where $\eta = \lceil \frac{CT_s - R}{x_v} \rceil$ ($\lceil y \rceil$ means the smallest integer $\geq y$), and

$$Var[W_s] = E[W_s^2] - E^2[W_s] \quad (9)$$

The average transmission delay of a voice packet and a data packet can be derived by assuming that a packet from any specified source station is directed, with equal probability, to any of other $M_V - 1$ stations. On average, a packet travels halfway around the ring to reach its destination. Thus, the propagation delay for a voice packet is given by

$$E[T_p] = \frac{1}{2}E[T_r] = \frac{1}{2} \sum_{i=0}^{M_v} P_i \frac{l x_v + R}{C} \quad (10)$$

The analysis for data transmission can be done based on the same idea presented in [15]. In a pure slotted ring network, the mean waiting time, $\bar{d}$, for a packet at the head of a queue to find an empty slot on the ring is given by [15]

$$\bar{d} = \frac{x}{2} + \frac{xU}{1-U}$$

In case of the hybrid protocol presented here, the waiting time, $\bar{d}$, should be modified to incorporate the effect of inserted voice packets on the ring. Thus, it is given by

$$\bar{d} = \frac{l x_v + R}{l + R/x_d} + \frac{U(x_d + T_{si})}{1-U}, \quad (11)$$

where x_d is the data packet length and T_{si} is the average interval between two successive data slots and is given by

$$T_{si} = \frac{l x_v}{R/x_d},$$

if l registers are inserted. The propagation delay of a data packet is the same as that of voice packet.

The above analysis for voice packet delay is based on the assumption that a voice station transmits voice packets only when it is active. This is an approximation because, even if a talkspurt terminates, some packets generated during the talkspurt may remain in the network. This implies that some voice transmissions will be carried out during the subsequent silence period. Ignoring these transmissions in the analysis would result in underestimation of the voice packet transfer delay. When the network operates in light traffic load, the error caused by the assumption is very small because most voice stations can finish their transmission within their talkspurt periods. However, when the network is under heavy load, a large number of voice packets will remain in the station due to the longer queueing delay, and these will be transmitted during the following silence period. In this case, ignoring these packet transmissions may introduce an unacceptable error. A precise derivation of the distribution of voice transmissions during silence periods appears intractable. An alternate

method is adopted in this study. The basic idea is to treat a station as an active station in part of its silence periods so that the transmissions occurring in the silence periods can be included in an equivalent "active period" and be taken into account in our analysis. This is realized by adjusting the values of mean talkspurt and silence periods. Let λ_0 and μ_0 denote the real values of mean talkspurt and silence periods, respectively, and λ and μ denote the modified values of mean talkspurt and silence periods, respectively. We obtain the following heuristic relation to obtain the modified parameters.

$$\lambda = \lambda_0 + \rho_v \mu_0$$

and

$$\mu = \mu_0(1 - \rho_v)$$

Note that the modified parameters incorporate the voice traffic load. Numerical results have shown that this heuristic works very well for a wide range of voice traffic loads.

4.3 Analysis of the Distributed Token Ring Protocol

Let T_V and T_D denote the voice token rotation time and the data token rotation time, respectively. For a data packet, the mean transfer time $\bar{D}_d$ is given by

$$\bar{D}_d = \bar{W} + \frac{x_d}{C} + \frac{R/2}{C}$$

where $\bar{W}$ is the mean queueing time of a data packet, $\frac{x_d}{C}$ is the transmission time and $\frac{R/2}{C}$ is the mean propagation delay. Now the mean waiting time of the data packet is given by

$$\bar{W}_q^d = \bar{T}_{RD} + Q \bar{T}_D \quad (12)$$

where $\bar{T}_{RD}$ is the mean residual life of the data token rotation time. It can be shown [18] that

$$\bar{W}_q^d = (1 + C_b^2) \frac{\bar{T}_D}{2(1 - \rho)} \quad (13)$$

where $\rho = \lambda_D \bar{T}_D$ and C_b is the co-efficient of variation of the data token rotation time.

Now, T_D , can be expressed as

$$T_D = N_D x_D + M_D T_V + R \quad (14)$$

In the above expression, N_D is a random variable which denotes the number of active data stations in one data token rotation time, and T_V is the voice token rotation time,

i.e., the time required by a data station to poll all the voice stations. Note that T_V is not the same as the voice token interarrival time at a voice station. Now, if we assume the stations to be independent then the number of active data stations in one data token rotation time can be characterized by a binomial distribution, i.e.,

$$P_r[N_D = n] = \binom{M_D}{n} \rho^n (1 - \rho)^{M_D - n}$$

Based on the above distribution, we can find the mean and variance of N_D .

Now, the voice token rotation time depends on the number of voice stations involved in voice transmission which in turn depends on the voice token rotation. Using the same technique as applied in the case of the hybrid network, we can find the distribution of the number of voice stations involved in voice transmission during one voice token rotation time as

$$P_l = \sum_{i=1}^{M_v} P_k P(l/k) \quad (15)$$

where $P(l/k)$ is same as in Eq.(4). Based on the above we can obtain the distribution of the voice token rotation time.

Next we need to find the interdeparture time of the voice packets. As in the case of the hybrid protocol, we assume that the voice and data stations are alternately placed on the ring. Now, the interdeparture time of voice packets from the ring to a destination station is the same as the interarrival of the voice token at that station. This is because the voice packet transmission time is fixed. Note that, the interarrival time of a voice token at a station depends on its position with respect to the data station which is currently holding the data token. Consider a particular voice station v_i . Let k denote the number of data stations between the voice station and the current controlling data station d_i . Clearly, k can take values from 0 to $M_D - 1$. It can be argued that for values of k equal to 1 to $M_D - 1$, the voice token interarrival time, T_{iauv} , can be written as

$$T_{iauv,k} = N_v x_v + \rho_d x_d + R, \quad ; k = 1, \dots, M_D - 1 \quad (16)$$

where N_v is a random variable denoting the number of voice stations involved in voice transmission during the token interarrival time, ρ_d is the probability that the controlling data station is active and R is the propagation delay. For $k = 0$, the situation is somewhat different as the controlling data station changes "sides" with respect to the particular voice station d_i . This will result in

$$T_{iauv,0} = 2T_{iauv,k} - r \quad k = 1, \dots, M_D - 1, \quad (17)$$

where r is the delay in each station. Now, the distribution of N_v can be found in the same manner as was used to derive P_i to obtain the delay characteristics of the data packets. To integrate $T_{iav,k}$ and $T_{iav,0}$ we use the following approximation

$$T_{iav} = \frac{M_D - 1}{M_D} T_{iav,k} + \frac{1}{M_D} T_{iav,0}. \quad (18)$$

This approximation assigns linear weights to the two possible cases based on the number of times they occur.

The mean voice packet transfer time D_v is composed of the voice packet generation time T_g , mean queueing delay W_q^v , transmission time x_v/C , and mean propagation delay $R/2C$. The mean voice queueing delay W_q^v consists of the mean residual life of the voice token interarrival time and the time required to transmit all voice packets found upon arrival. Similar to the analysis for hybrid protocol, W_q^v is given by

$$W_q^v = \frac{\overline{T_{iav}^2}}{2\overline{T_{iav}}} + \frac{E[N_B]}{\lambda_B} \overline{W_Q} \overline{T_r} \quad (19)$$

where $\overline{W_Q}$ is the mean queueing time for all packets in batches and is given in equation (6).

5 Numerical Results and Discussions

In this section, we first validate our analytical models presented in the last section through simulations. For this purpose, we have developed a discrete event-driven simulator in PASCAL that had run on Sun workstations. The simulator simulates the actual behaviors of voice calls and data transmissions without such assumptions as independence between stations that are connected to the ring, complete transmission of voice packets within a talkspurt, and batch arrivals etc.. Packet generation processes, talkspurt and silence period lengths are kept same as analytical models. After validation of the models, we will use the models to study the performance of the newly developed protocols under variety of system parameters. We will also compare our protocols with existing protocols in the later of this section.

In the following discussions, we assume that the ring has the length of 300 bits, channel speed of $C=1$ Mbits/sec. Data packets are assumed to be of length 300 bits. The talkspurt and silence periods of a voice call are assumed to be exponentially distributed with the average lengths of 170 msec and 410 msec, respectively. The voice

encoder at each station has a rate of 64 Kbits/second. It should be noted, however, that our analysis is good for any set of system parameters.

Figure 2 shows the comparison of the analytical and simulation results for the voice packet transfer delay, D_v , as a function of number of voice calls in the network based on the hybrid protocol. The solid curves represent analytical results and the dot (+, *, o) lines correspond to simulation results. It can be seen from the figure that our analysis matches well with the simulations over a wide range of traffic loads and packet sizes. Note that D_v increases with the increase in the number of voice calls because a voice packet encounters more inserted registers when it travels around the ring.

Figure 3 shows the total transfer delay of a voice packet, including voice packet generation time, queueing delay, ring access time and propagation delay, versus the number of voice calls in the network. Different voice packet lengths were selected in order to observe their effects on the performance of the hybrid network. Comparison of the results of different packet lengths show that shorter packets have longer queuing delays than that of longer packets. This is because that small packet sizes result in more packets for the same amount of information and consequently more header overheads were introduced during a talkspurt. When the network operates under heavy traffic load (i.e., corresponding to high utilization), the queuing delay may increase significantly and become the main part of the total transmission delay. It is shown in the figure that the saturation points corresponding to $x_v = 960$ and $x_v = 1600$ bits are both around 35 stations. However, the average delay with packet length of $x_v = 960$ bits is much smaller than that of $x_v = 1600$ bits under low load.

Figure 4 shows the loss probability of a voice packet for the hybrid protocol. The loss probability is also the proportion of voice packets that experience transmission delay of more than the prespecified time bound τ ($\tau = 150$ ms in Figure 4). It can be seen in the figure that small packet sizes have higher loss probability than larger sizes, which is consistent with the observations described above. If the total transmission delay of a voice packet is longer than the time bound, the packet is considered lost. According to the assumption that the human speech can be reconstructed at the destination with an acceptable quality when the loss probability is less than 1%, we can derive the maximum number of voice users allowed on a network for different packet lengths as shown in Figure 5. Two curves are plotted for two different values of τ . It is observed from Figure 5 that for the time bound of 200 msec, the packet length of 1600 bits gives the maximum number of allowable voice calls which is about 38. For

the time bound of 150 ms, the maximum allowable voice calls (about 32) occurs at the packet length of about 600 bits. In both cases, for packet lengths of $640 < x_v < 2240$ the network can have the maximum capacity of simultaneously active voice calls of at least 28, as well as the same number of active data users. Since in an actual network the total number of voice users is 5 to 10 times as many as active voice calls at a time, the network using our protocol will be able to support 140 to 280 voice lines, along with same number of data users.

Figure 6 shows the comparison of the simulation and the analytical results for the distributed token ring protocol. In the figure, the voice packet delay is plotted as a function of the number of voice stations for different voice packet lengths. Similar to the case of the hybrid network it is observed that the analytical result matches well with the simulation for wide range of traffic loads and different packet sizes. Error is introduced for heavy loads where our analysis underestimates the delay. This error is primarily due to the fact that the assumption of independence among stations is not really valid and the effect of that becomes dominant at higher load. The comparison of analytical and simulation results for the data packet delay also showed similar behavior.

Figure 7a plots the voice packet delay as a function of the voice packet length for different number of voice stations. The figure shows an interesting trade-off that exists between the voice packet delay and the voice packet length. As the packet length increases the overhead reduces in terms of the packet header and control bits. On the other hand, the packet generation time increases as packet length increases. This trade-off can also be observed from the results of the hybrid protocol as shown in Figure 7b. Similar results were obtained for the token ring protocol studied in [9].

The curves in Figure 8a show the results of data packet delays as functions of the number of voice calls for the hybrid protocol. The large amount of delay corresponding to high utilization levels was mainly contributed by queueing due to the long rotation time of a slot traveling around the ring. The more voice buffers are inserted, the longer the rotation time of a slot will be. Therefore, a data packet will have to wait for a long time in the queue. For this reason, even though the selection of a longer packet length may result in smaller voice queueing delay, the disadvantageous effect on data transmission appears. Similar observation are obtained for the distributed token rings as shown in Figure 8b.

Finally we give a performance comparison among our hybrid protocol, the dis-

tributed token protocol and the conventional token passing protocol proposed in [9]. In order to make a fair comparison, the data traffic load is selected in such a way that offered load from data stations and the data transmission delay are same in all the three networks. Under this consideration we compared the performances of the three protocols in terms of their voice transmission delay as well as their loading capability for data and voice users, as depicted in Figure 9 and Figure 10, respectively. The result indicates that both our hybrid and distributed token protocols present a better behavior on supporting voice users in a ring LAN while keeping the same performance for data traffic. This means that the network can be utilized more effectively under the new protocols.

6 Conclusions

In this paper we have proposed two protocols for voice data integration on ring networks. The first protocol adopts a hybrid scheme and uses the slotted ring and the register insertion ring protocols. The second protocol is based on the standard token ring access scheme. The underlying feature of these two protocols is that they provide decentralized priority to voice stations. We argue that providing decentralized priority to voice stations is important for voice data integration in two respects. First, it allows voice station to exchange voice packets within the required time constraints, and second it has better fault-tolerance capability. It is in this respect that this work is different from the other studies reported in the literature. Our comparative studies based on analytical and simulation models of various protocols show these two protocols perform better than some of the existing ones. Our hybrid protocol provides significant improvement over the conventional token ring protocol

Approximate analytical models have been developed for the new protocols to evaluate the performance of the networks. More rigorous analytical treatment of the protocols can be undertaken specifically in order to incorporate the correlation among the stations. However, the results and conclusions of this study would still be valid since this approximation is made in all the analyses. Furthermore, our simulation experiments show that our analytical results are reasonably accurate over a wide range of loads.

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APPENDIX

A LIST OF NOTATIONS

M_d : Number of data stations on the ring.

M_v : Number of voice stations on the ring.

λ : Mean length of a talkspurt period.

μ : Mean length of a silence period.

x_v : Voice packet length in bits.

x_d : Data packet length in bits.

V : Digitization rate of the voice encoder.

T_g : Time between generation of two consecutive voice packets.

π_s : Steady state probability of voice station being in the silence state.

π_t : Steady state probability of voice packet being in the silence state.

C : Channel bandwidth in bits per second.

D_v : Voice packet transmission delay.

W_q^v : Queueing delay of a voice packet.

W_p : Propagation delay of a voice packet on the ring.

W_s : Interdeparture time of voice packets from the ring to a destination station.

P_{loss} : Probability of a packet being lost (i.e. not reaching the destination within a prespecified time τ).

D_d : Average data packet transfer delay.

T_r : Packet rotation time.

T_{RS} : Residual life of the packet rotation time.

$\bar{Q}$: Mean voice station queue length.

U : Ring utilization.

T_D : Data token rotation time in the distributed token ring protocol

T_V : Voice token rotation time in the distributed token ring protocol.

T_{iav} : Voice token interarrival time at a voice station.

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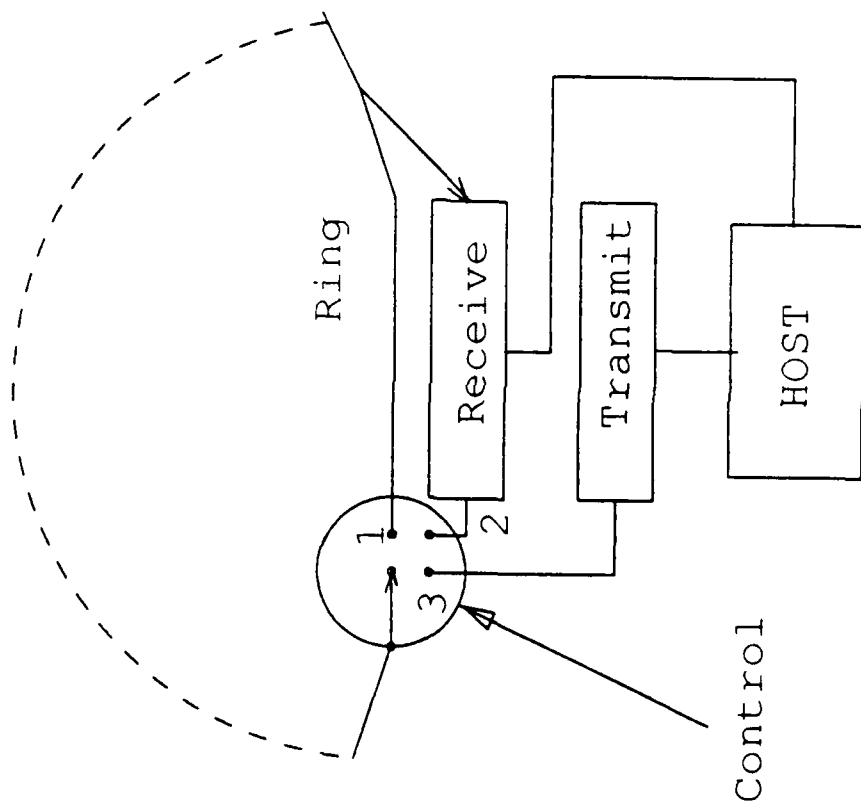


Figure 1. The logic diagram of register insertion protocol.

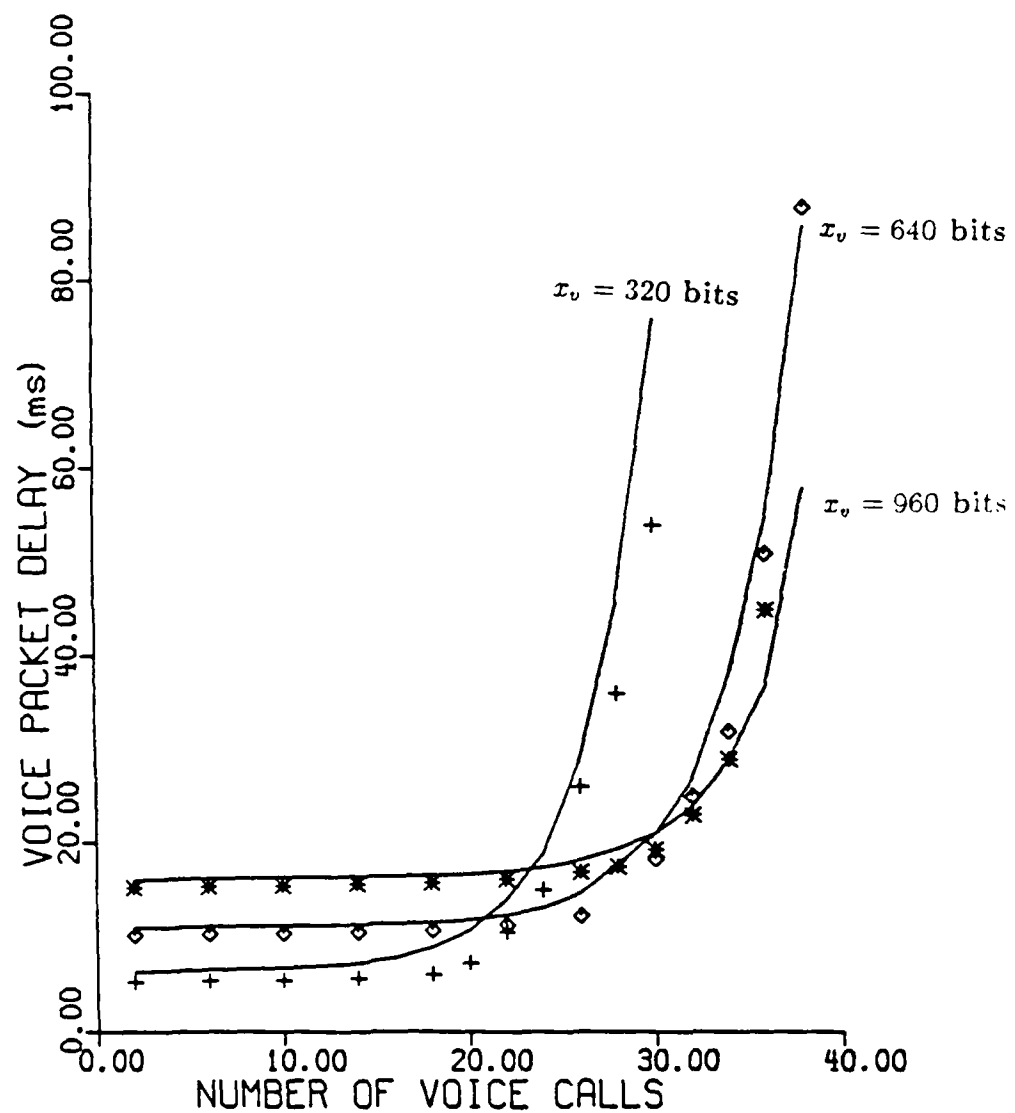


Figure 2. Comparison between analyses and simulations for hybrid protocol.

(analytical results: solid lines;
simulation results: +: $x_v = 320$ bits; o: $x_v = 640$ bits; *: $x_v = 960$ bits)

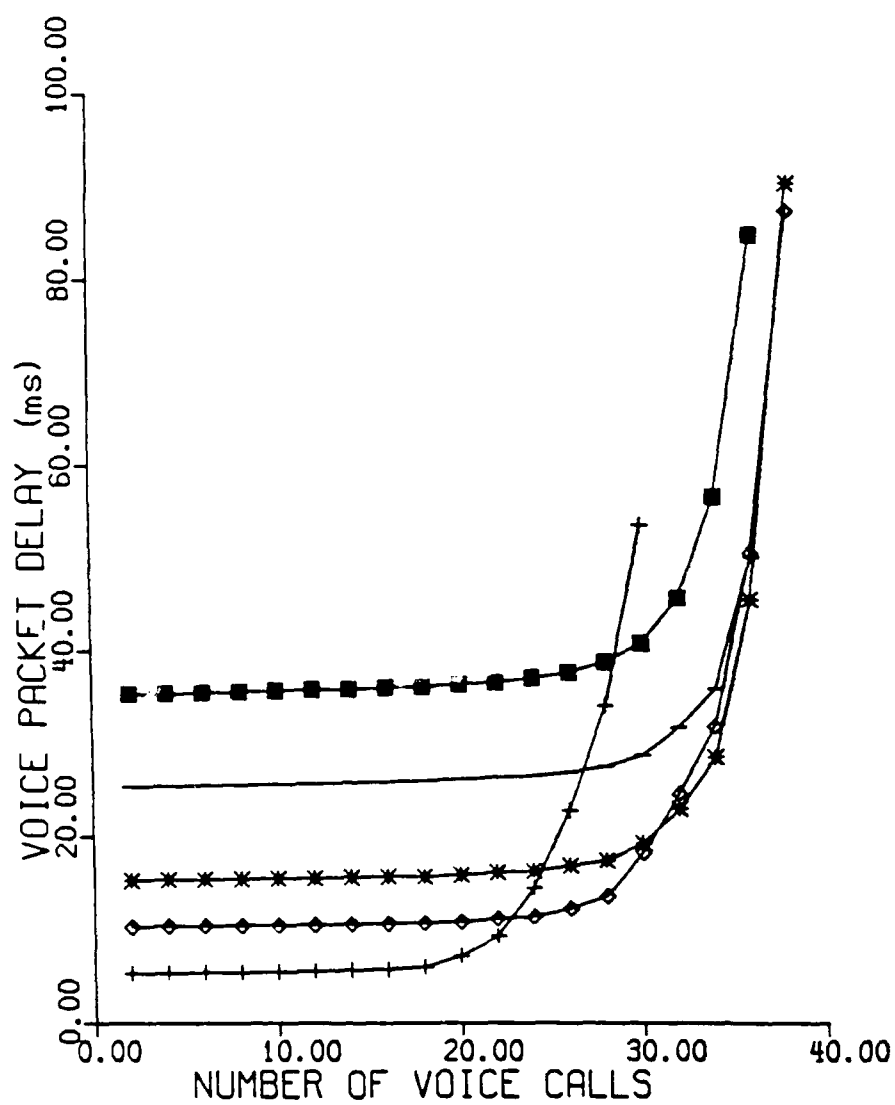


Figure 3. Voice packet transmission delay with hybrid protocol.

(+: $x_v = 320$ bits; ◇: $x_v = 640$ bits; *: $x_v = 960$ bits;
 -: $x_v = 1600$ bits; ■: $x_v = 2240$ bits)

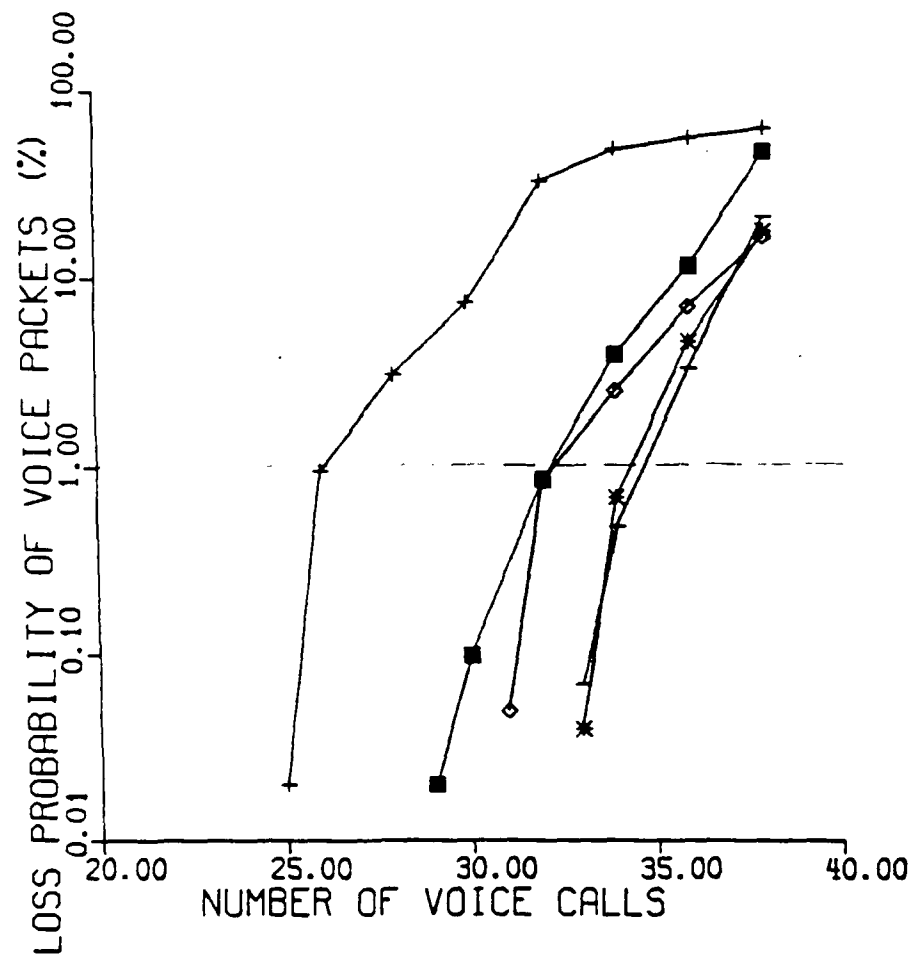


Figure 4. Voice packet loss probability
with hybrid protocol.

(
 $+$: $x_v = 320$ bits; $\diamond$: $x_v = 640$ bits; $*$: $x_v = 960$ bits;
 $-$: $x_v = 1600$ bits; $\blacksquare$: $x_v = 2240$ bits; $\tau = 150$ ms)

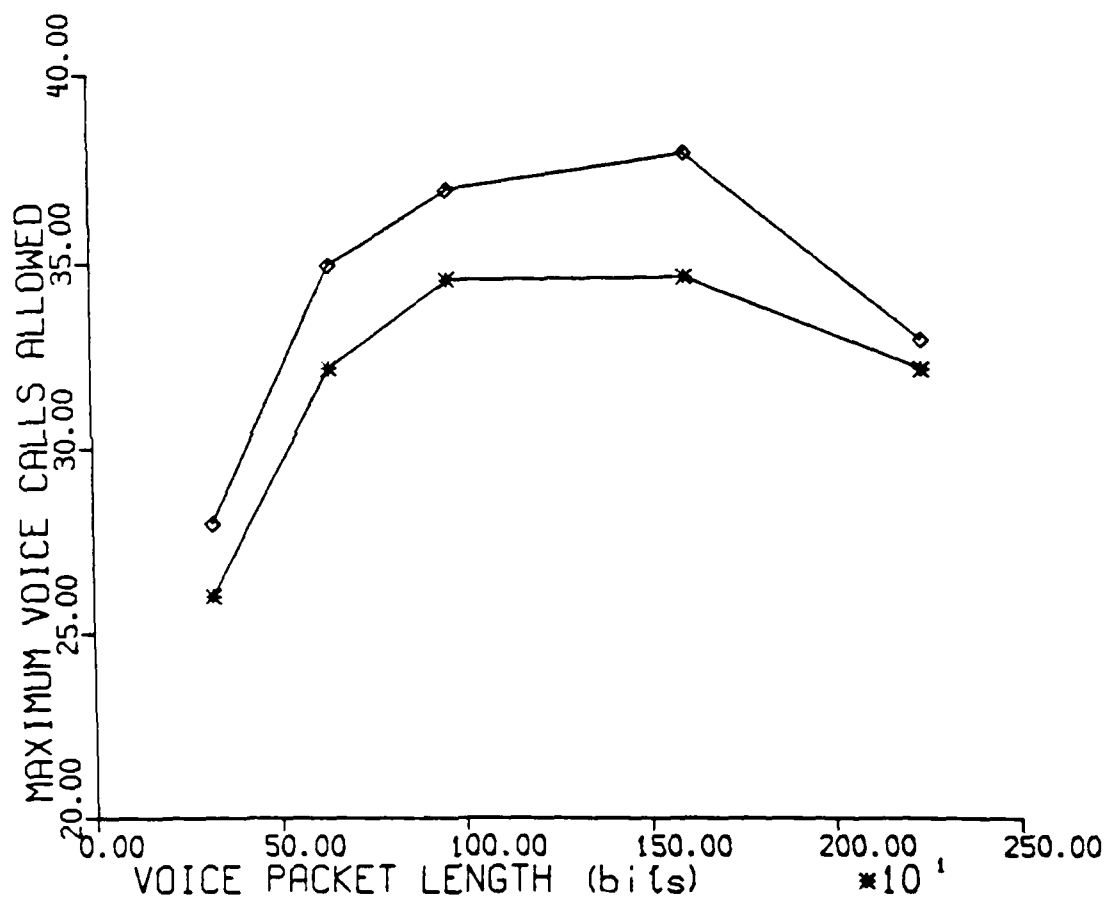


Figure 5. Maximum number of voice calls allowed on a network with hybrid protocol.

(* : $\tau = 150$ ms; $\diamond$: $\tau = 200$ ms)

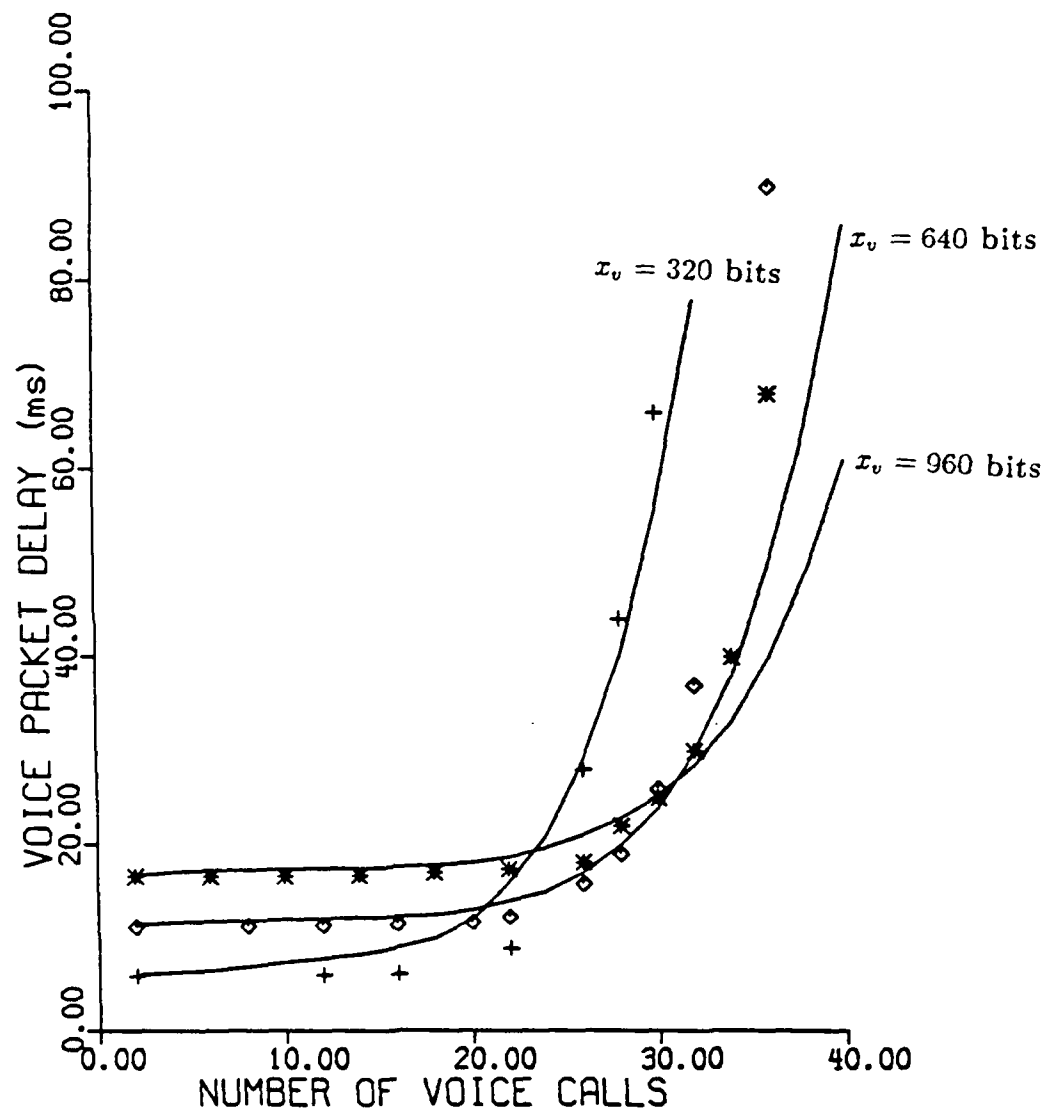


Figure 6. Comparison of analytical and simulation results for distributed token ring protocol.

- (analytical results: solid lines;
simulation results: +: $x_v = 320$ bits; o: $x_v = 640$ bits; *: $x_v = 960$ bits)

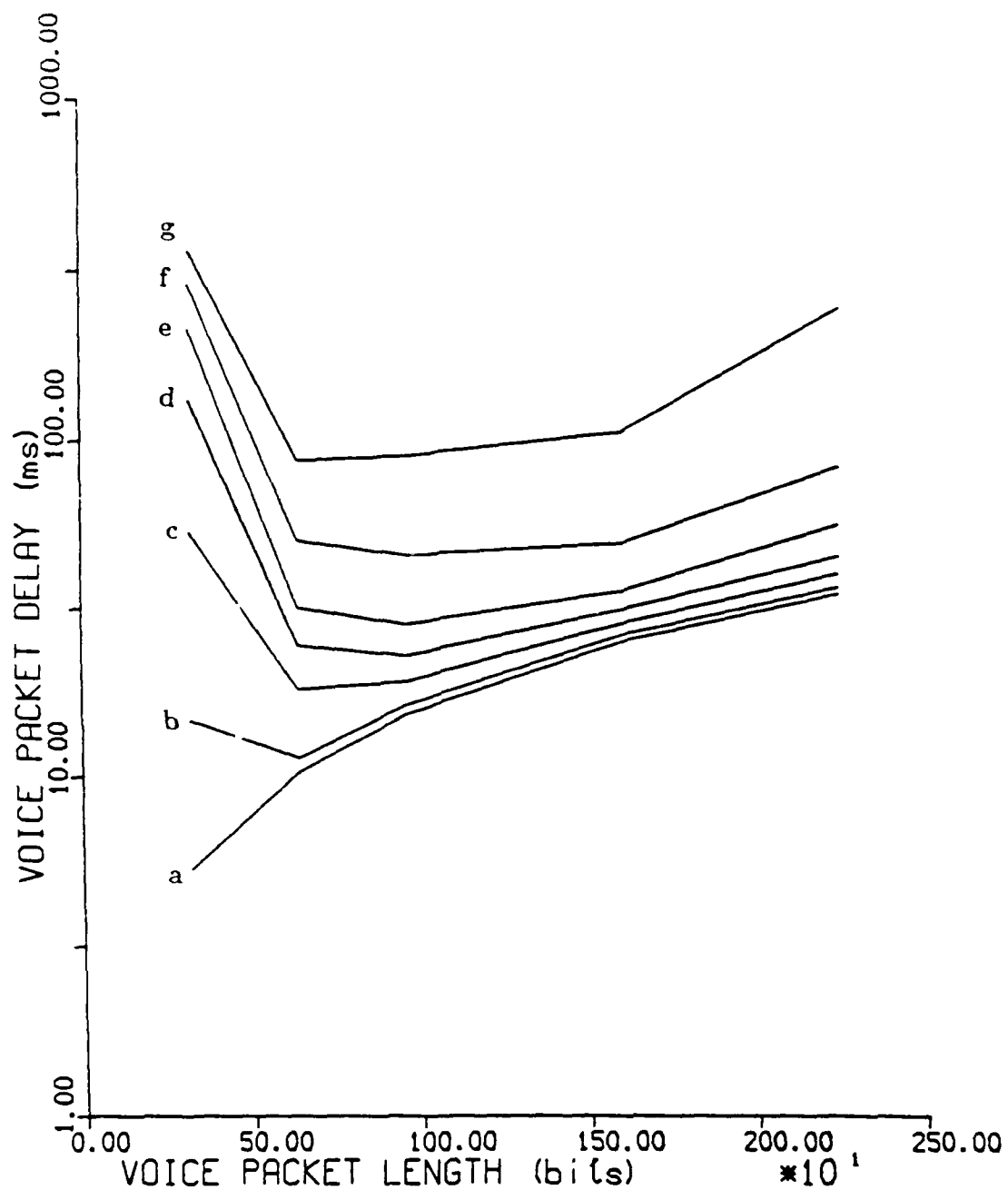


Figure 7a. Voice packet transmission delay as a function of packet length (hybrid).

(a: $M_v = 4$; b: $M_v = 24$; c: $M_v = 30$; d: $M_v = 32$;
e: $M_v = 34$; f: $M_v = 36$; g: $M_v = 38$)

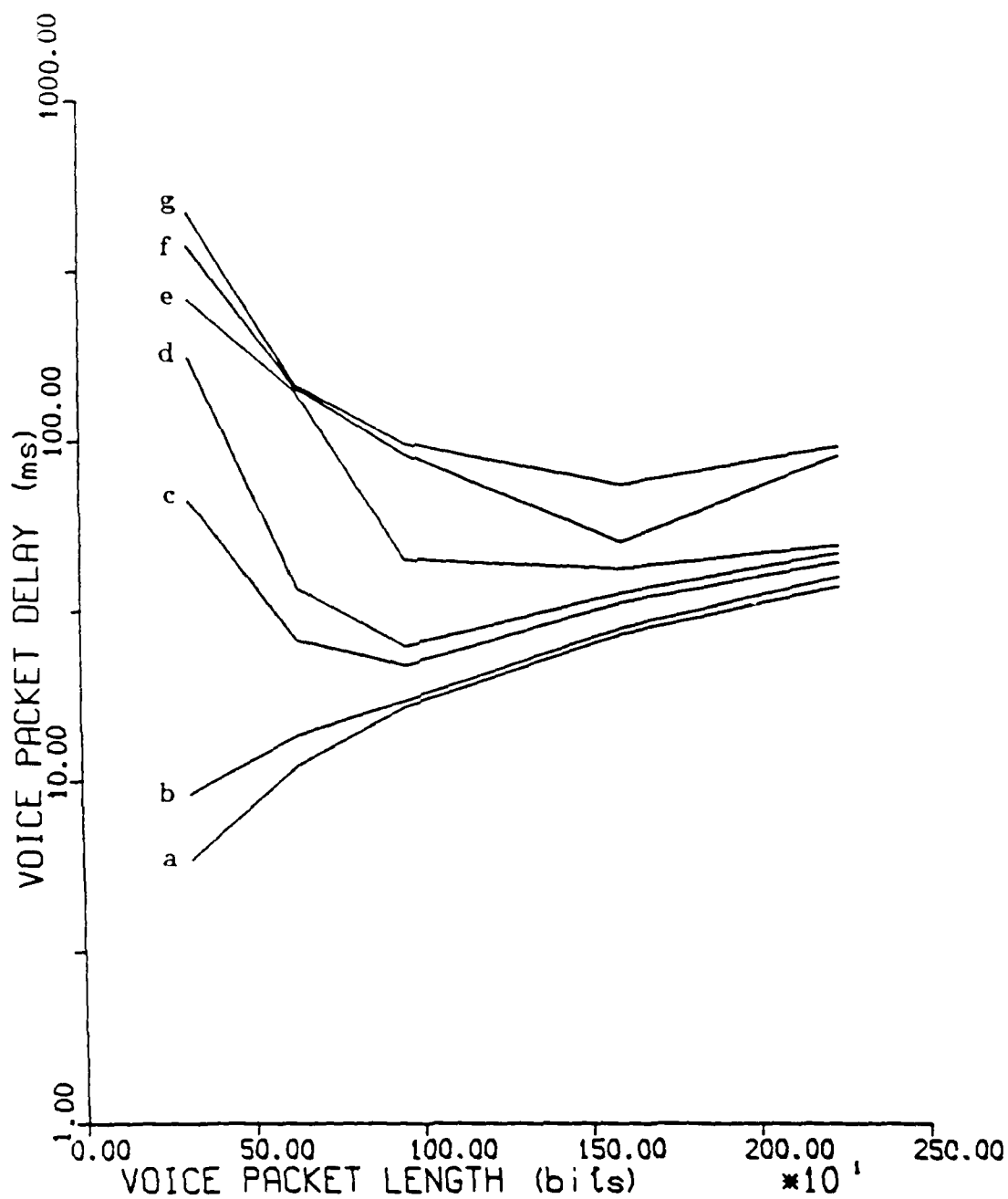


Figure 7b. Voice packet transmission delay as a function of packet length (distributed token).

(a: $M_v = 4$; b: $M_v = 24$; c: $M_v = 30$; d: $M_v = 32$;
e: $M_v = 34$; f: $M_v = 36$; g: $M_v = 38$)

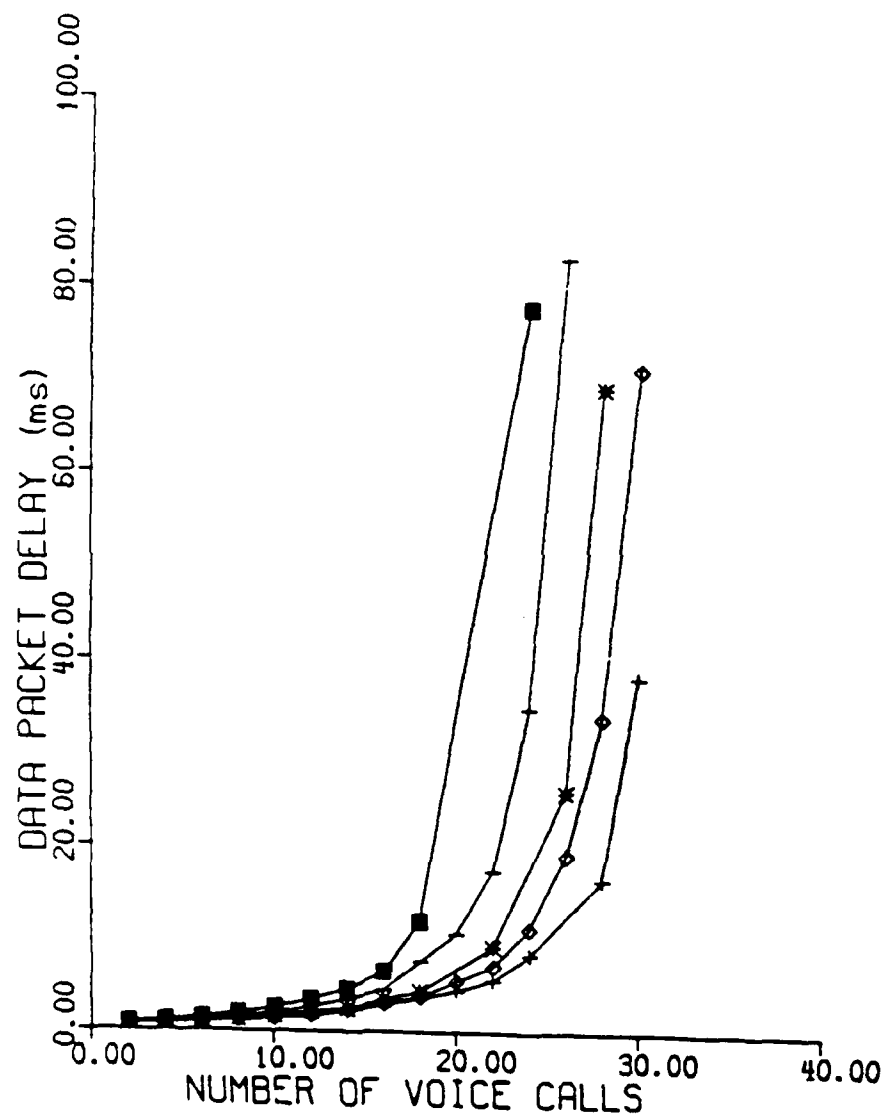


Figure 8a. Data packet transmission delay (hybrid).

(
 -: $x_v = 320$ bits; $\diamond$: $x_v = 640$ bits; *: $x_v = 960$ bits;
 -: $x_v = 1600$ bits; ■: $x_v = 2240$ bits)

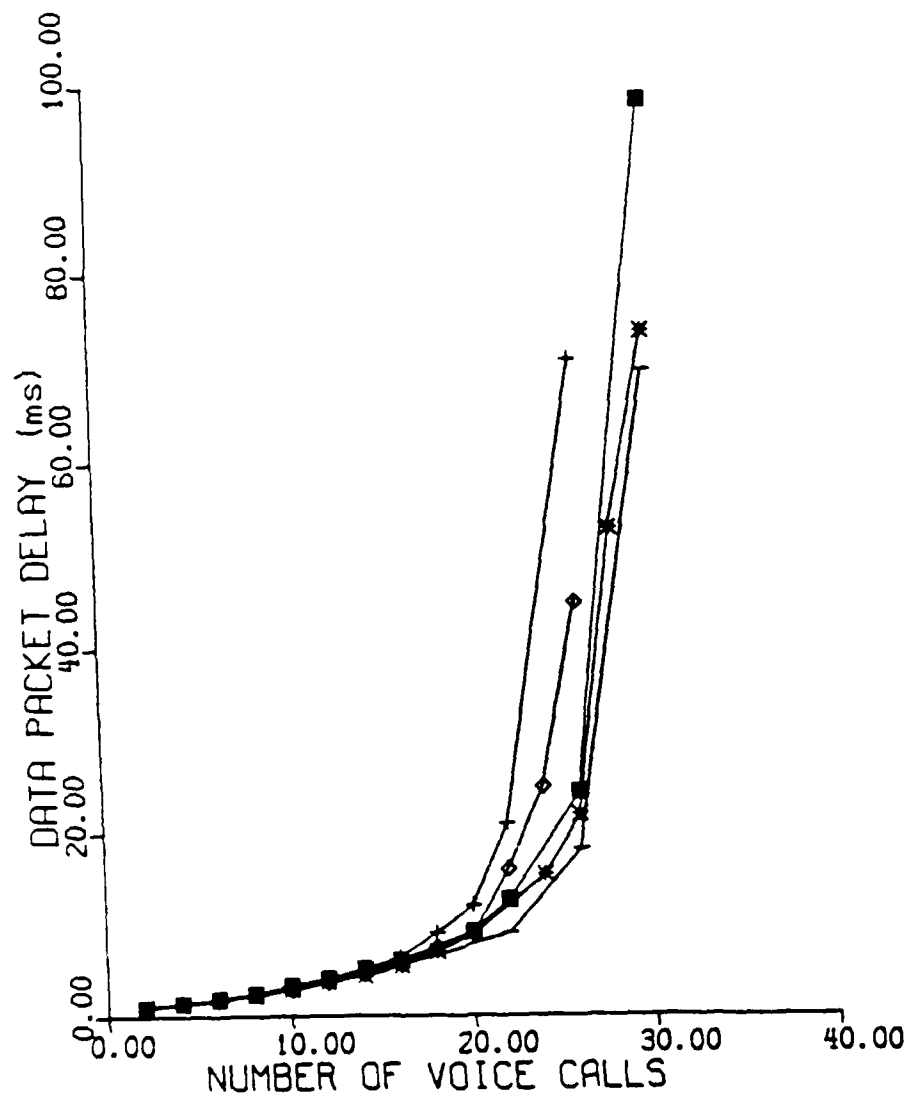


Figure 8b. Data packet transmission delay (distributed token).

(+: $x_v = 320$ bits; ◇: $x_v = 640$ bits; *: $x_v = 960$ bits;
 -: $x_v = 1600$ bits; ■: $x_v = 2240$ bits)

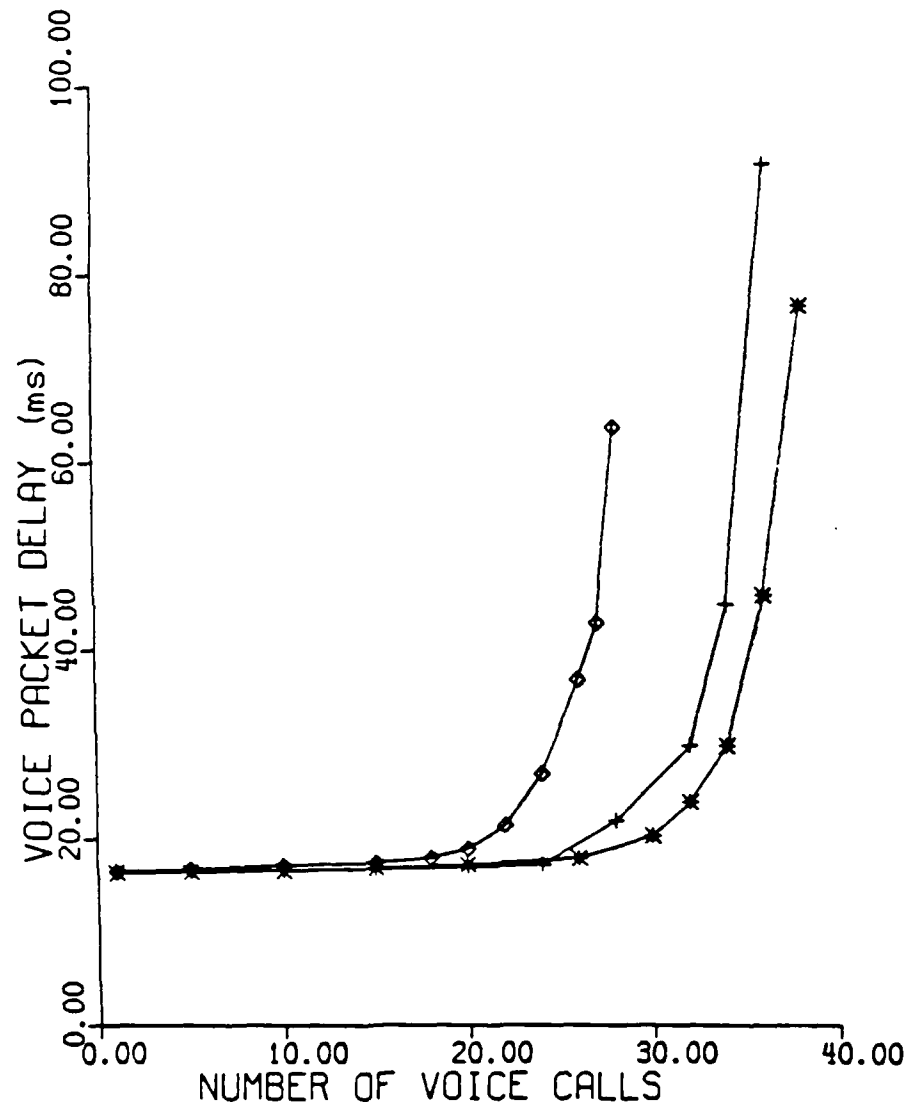


Figure 9. Performance comparison of different protocols.

($x_v = 960$ bits; $\diamond$: conventional token protocol;
 *: hybrid protocol; +: distributed token protocol)

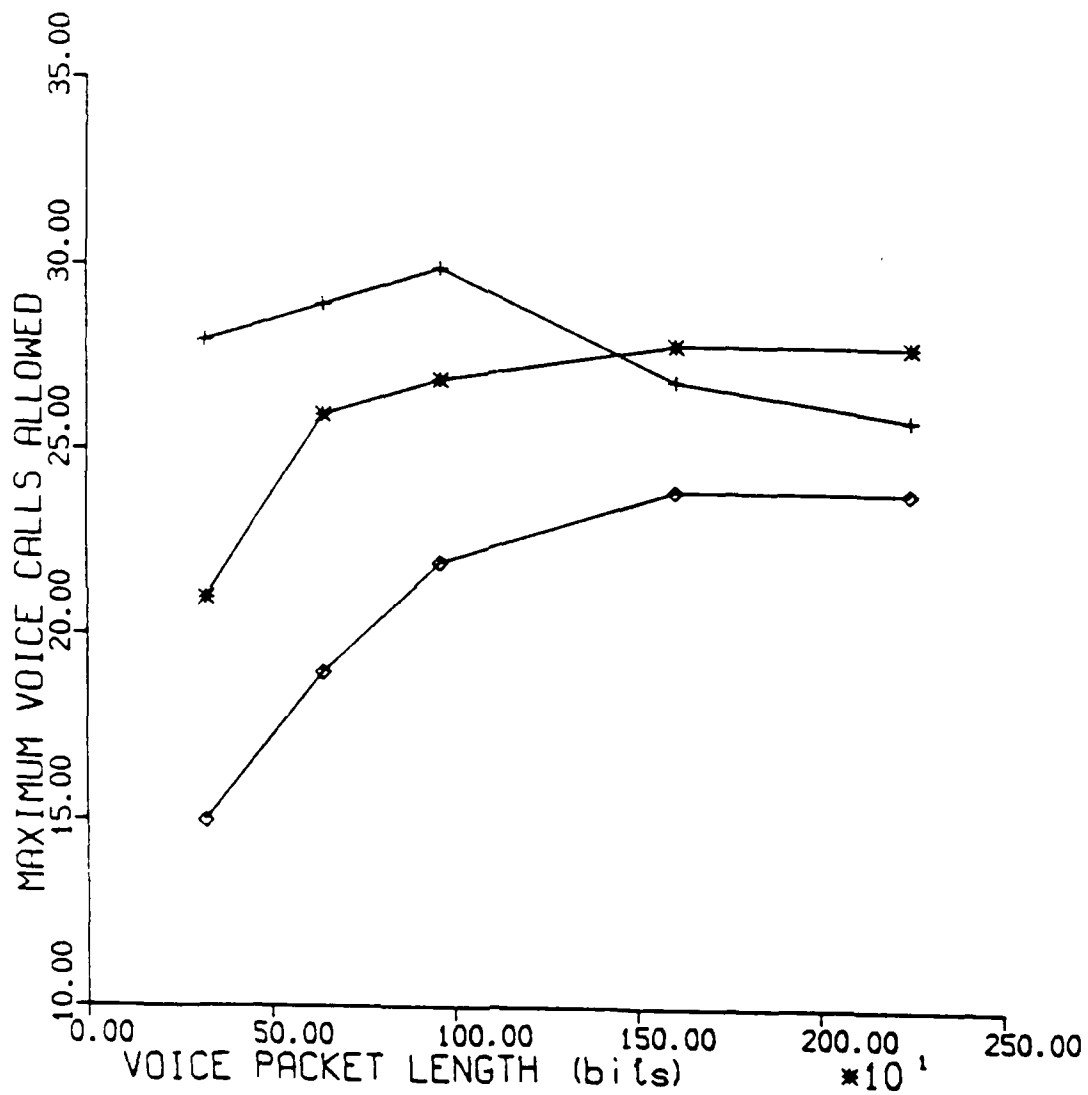


Figure 10. Performance comparison of different protocols.

($\tau = 100$ ms; ◇: conventional token protocol;
 *: hybrid protocol; +: distributed token protocol)