# IEEE
# INFOCOM '85

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# PROCEEDINGS

# IEEE INFOCOM 85

MARCH 26-28, 1985
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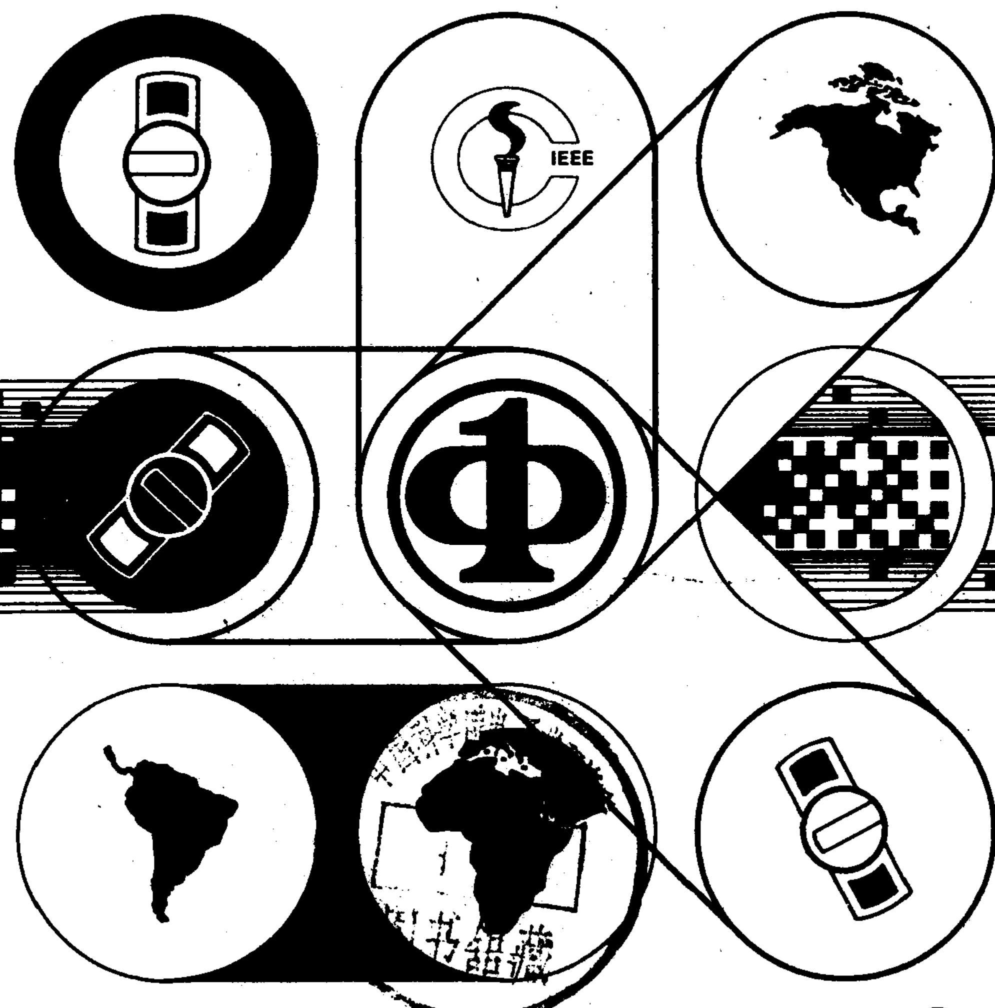
WASHINGTON, D.C.



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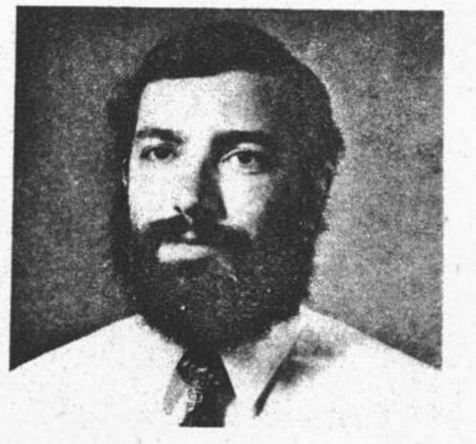
# General Chairman's Message



Roughly five years ago, the technical committees of the IEEE Communications Society and the IEEE Computer Society began discussions on a proposal for a jointly sponsored annual conference, international in scope and dedicated to computer communications. Considering the number of conferences available to attend, there was a serious question as to whether we ought to devote our attention to such a conference. However, the interdependability of the futures of computers and communications and the rapid convergence of both to a single technology led the committees to fully support the concept.

We have now experienced three highly successful conferences in the INFOCOM series. Each of these conferences has been a special experience for me. The technical programs have been uniformly strong and attendance has been good, but perhaps the real measure of a successful conference is the quality of the conferees. In this last dimension, I am convinced that INFOCOM is the leader.

This year there are four preconference tutorials, ranging in scope from broad to highly specialized. The keynote session features an address, "The Information Age and Other Coming Attractions," by Robert W. Lucky, a key contributor in the field. Adequate breaks are provided between refereed paper sessions, of which there are seventeen, to provide time for meaningful discussions while ideas are fresh. Eight panel sessions provide the conferees an opportunity for real-time discussion. In addition, the early evening social hours allow conferees an opportunity to meet on a less formal basis.

Organization of INFOCOM 85 has been a major undertaking to which many people have contributed and deserve thanks. I especially thank Tom Stack, his committee, and his referees for putting together a fine technical program; Terry Contreras for her outstanding performance in local arrangements; Celia Desmond for the special care she took in publicizing the conference; Michael Hluchyj for his conscientious care of financial matters; and Jeff Wong for his efforts to enlist patrons. I am indebted to Harry Hayman, Gerrie Katz, and the Computer Society staff for their exceptional assistance, cooperation, support, and dedication, which have been key factors in the success of IEEE INFOCOM.

Welcome to INFOCOM 85 and thank you for attending. I hope that you have a most pleasant and productive experience.

John N. Daigle
**General Chairman**

## Program Chairman's Message

The rate of change we are now experiencing in the evolution of digital communications rivals that of digital computing. The dynamics of these technologies challenge us all. This year's INFOCOM presents papers and panels that address the confluence of these turbulently changing and converging technologies.

After the keynote address from Dr. Robert W. Lucky to all attendees, our three-day conference is divided into three tracks (A, B, and C) of eight to nine sessions each. The A track presents computer/communications functional or performance results in the areas of voice/data integration, distributed data processing, gateway architecture, network security, network control, protocol design, packet radio, and mobile radio. The B track presents a two-session update on the ARPANET followed by primarily quantitative delay/throughput or reliability results for RING, CSMA and other LAN networks, channel interference, teletext performance, and multi-hop message segmentation. The C track is a panels track on circuit and packet switching advances, the increasing role of experimentation in LAN research, military networks, metropolitan networks, network management, and the role of consortia in advancing computer and communications technology.

I hope you will enjoy the program we have assembled for you. I thank our program committee and the reviewers who helped to select the many fine papers we have included in these proceedings.

Tom Stack
**Program Chairman**



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*(Organizer: C. Walker, Foundation for Advanced Research)*

**Session 9A: Mobile Radio**



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# Keynote Session

**Welcome**
John N. Daigle, *University of Rochester*

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**Keynote**
"The Information Age and Other Coming Attractions"

Robert W. Lucky, *AT&T Bell Laboratories*

Session 1A

# Voice and Data Integration

# END-TO-END PACKET LOSS FRACTIONS IN A TANDEM PACKET VOICE NETWORK

Youngho Lim

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40 Sylvan Road, Waltham, MA 02254 USA

## ABSTRACT

An important performance measure of the packet voice network is the fraction of speech packets lost during the transmission between the origin and the destination nodes in the network. Weinstein [Ref. 1] derived the upper bound for the speech packet loss fraction through a single packet switching node under the assumption that the packet queueing time is limited to a single packet frame. In this paper, we derive the upper bound for the end-to-end speech packet loss fraction passing through a series of tandem packet switches, under the assumption that the packet queueing time at each switch is limited to a single packet frame. The network under consideration consists of a series of packet switches and the common destination at the end. A group of voice calls is supported by each switch and each switch multiplexes speech packets generated from these calls as well as those received from the previous switch to the common destination. It is shown that the end-to-end packet loss fraction for a particular group of calls can be expressed as a unit value subtracted by the ratio of statistical expectations of two random variables. The paper then presents the recursive computing algorithm for determining the distribution of these random variables. An example computes the end-to-end packet loss fraction for each group of calls in a tandem packet voice network consisting of five switches and links.

## I. INTRODUCTION

A packetized speech multiplexer can support the number of callers whose combined maximum bandwidth is larger than the transmission capacity of an output link. Utilizing Speech Activity Detection (SAD), it sends speech packets only when callers are active, that is, in *talkspurts*. Since a typical voice conversation consists of about 35% to 50% talkspurts and of 50% to 65% silent periods, depending on a specific SAD scheme, the introduction of packetized speech multiplexing technique can improve the bandwidth utilization up to 2 to 2.8 times compared to the traditional circuit-switching method without Time Assignment Speech Interpolation (TASI). A packetized speech multiplexer differs from a circuit-switched TASI system in the following two ways.

1. A packetized speech multiplexer has a buffer, and speech packets can be queued rather than clipped out.
2. The TASI-like advantage of bandwidth utilization is naturally extended to a network environment such that all transmission links in the network can be shared simultaneously by numbers of callers whose combined maximum bandwidths are larger than the capacity of links.

Due to the statistical fluctuation of speech activity, however, there can be periods of time when more speech packets are generated than the capacity of the output link from the switch. In this case, excessive packets should be buffered until bandwidth becomes available. Since voice traffic needs a real-time delivery, there should be a limit on the speech packet queueing delay. Under the assumption that the maximum queueing delay is limited to a single packet frame (a fixed length of time interval between packet generations), Weinstein [Ref. 1] derived the packet loss fraction through a single packet switching node. This can be considered the upper bound for the packet loss fraction because the fraction would decrease if the system allows longer queueing delays. There is a tradeoff between the packet loss fraction and the averge queueing delay.

In this paper, we derive the upper bound for an end-to-end speech packet loss fraction passing through a series of tandem packet switches under the assumption that the packet-queueing



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delay at each switch is limited to a single packet frame. The network under consideration consists of a series of packet switches and the common destination at the end. A group of voice calls is directly supported by each switch and each switch multiplexes speech packets generated from these calls, as well as those received from the previous switch, to the common destination.

In Section II, we describe the network operation and define a set of random variables representing the stationary behavior of the numbers of speech packets originated from different groups of calls at different switches in the network.

In Section III, the end-to-end packet loss fraction is derived for each group of calls in the network. It is expressed as a unit value subtracted by the ratio of statistical expectations of two random variables defined in Section II. The recursive computing algorithm for determining the distributions of those random variables is presented.

In Section IV, following the computing algorithm presented in Section III, we compute the end-to-end packet loss fractions for five different groups of calls in a tandem packet voice network consisting of five switches and links.

Section V concludes the paper and suggests future research problems.

## II. MODEL

Consider a series of packet switches tandemly connected to a common destination at one end. Figure 1 shows the case when there are s switches supporting groups of $n_i$ calls, respectively.

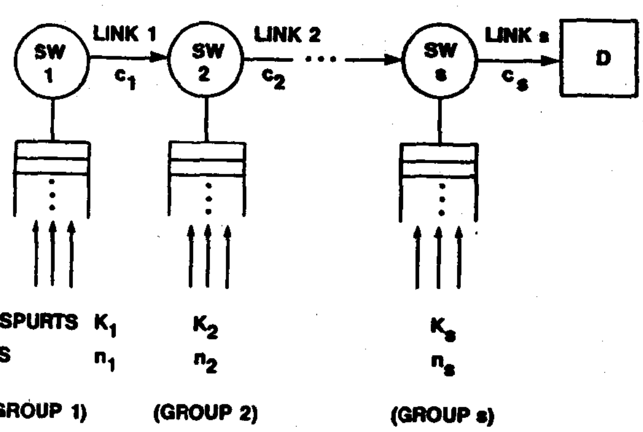


Figure 1. Tandem packet voice network.

A total of s transmission links connects them to the common destination D. It is assumed that each call in the network generates speech packets of a fixed size at every T seconds (packet frame) when it is in talkspurts. No packets are generated when it is in silent periods. Furthermore, considering the fact that T is much shorter than the average talkspurt duration, we assume that all calls in the network generate packets at every T seconds according to a synchronous clock. We can ignore propagation and nodal processing delays in the network as long as we are concerned with the packet loss fractions. This is because these delays are again much shorter than the average talkspurt duration.

The above assumptions in effect imply that a speech packet passes through all switches and links to the destination in a single packet frame if it is not lost during the transmission. Each link in the network has a transmission capacity of $c_i$ packets per frame $i = 1, 2, \ldots, s$. Switch i multiplexes speech packets generated from group i calls directly supported by it as well as those transmitted from the previous switch $i-1$. If it receives a total of more than $c_i$ packets at any single frame, it randomly discards excessive packets.

A statistical fluctuation of speech activity of voice calls can be modeled by an alternating renewal process of talkspurts and silent periods with mean durations $1/\alpha$ and $1/\beta$, respectively [Ref. 2]. We define

$$p = \frac{1/\alpha}{1/\alpha + 1/\beta} = \frac{\beta}{\alpha + \beta}$$

to be the voice call activity level. We can assume that both $1/\alpha$ and $1/\beta$ are integer multiples of T, because T is much shorter than either of them. In this way, the packet frame can, in effect, be considered as a time measuring unit. For the rest of this paper, we use the word *frame* to refer to a *frame duration*.

It should be noted that under a set of assumptions (such as the fixed frame size, the fixed queueing delay equivalent to a frame size, and no propagation or nodal processing delays), the various aspects of the traffic behavior in the network at a random frame can be represented by deterministic random variables rather than stochastic processes, if the states of speech activities of all calls in the network are given. In the rest of this paper, we assume that the speech activities of all calls in the network reached their equilibrium states.

We define the following random variables representing the various aspects of the traffic behavior in the network in equilibrium.

$K_i$ = the total number of speech packets generated by $n_i$ calls in group i at a random frame. $(0 \leq K_i \leq n_i,\ 1 \leq i \leq s)$



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$$K = \sum_{i=1}^{s} K_i$$

$T_j$ = the total number of input packets to switch j at a random frame.

$M_j$ = the total number of output packets from switch j at a random frame.
$(0 \le M_j \le c_j,\ 1 \le j \le s)$

$M_{ij}$ = the total number of output packets from switch j which are originated from group i calls at a random frame.
$(0 \le M_{ii} \le K_i,\ 0 \le M_{ij} \le M_{ij-1},\ i \le j \le s)$

$$\overline{M_{ij}} = M_j - M_{ij} = \sum_{\substack{1 \le k \le j \\ k \ne i}} M_{kj}$$

It is well known from the renewal theory [Ref. 3] that $K_i$ has a binomial distribution with parameters $n_i$ and p, respectively. That is,

$$P(K_i = k_i) = \binom{n_i}{k_i} p^{k_i} (1-p)^{n_i - k_i} \quad ,$$

$k_i = 0, 1, 2, \ldots, n_i$ , and

$$E(K_i) = n_i p \quad .$$

Since all $K_i$, i = 1, 2, ..., s are independent, K also has a binomial distribution with parameters

$$n = \sum_{i=1}^{s} n_i \quad \text{and p, respectively.}$$

One can also see the following recursive relationships among the random variables defined above.

$M_{j-1}$ = truncation of $T_{j-1}$ onto the range $[0, c_{j-1}]$.

$$T_j = M_{j-1} + K_j \quad , \qquad j = 1, 2, \ldots, s \quad .$$

where

$M_0 = 0$ with probability 1.

The distributions of $M_{ij}$ and $\overline{M_{ij}}$, where $j = i, i+1, \ldots, s$, and $i = 1, 2, \ldots, s$, will be discussed in Section III.

## III. END-TO-END PACKET LOSS FRACTIONS

The end-to-end speech packet loss fraction of voice calls in group i is defined to be the long-run ratio of the total number of packets lost at all intermediate switches i, i+1, ..., s during the transmission to the total number of packets generated by them. The packet loss fraction of voice calls in group i at switch j $(i \le j \le s)$ is defined to be the long-run ratio of the total number of packets discarded at switch j to the total number of packets generated by them. We define $\phi_i$ to be the end-to-end packet loss fraction and $\phi_{ij}$ to be the packet loss fraction of group i calls at switch j. They should satisfy the following relationship.

$$\phi_i = \sum_{j=1}^{s} \phi_{ij} \tag{1}$$

From the stationarity of the network, it can be seen that the above definitions $\phi_i$ and $\phi_{ij}$ = i, i+1, ..., s, and i = 1, 2, ..., s, are equivalent to the following ratios of statistical expectations of random variables defined in Section II.

$$\phi_i = \frac{E(K_i - M_{is})}{E(K_i)} = 1 - \frac{E(M_{is})}{n_i p} \tag{2}$$

$$\left.\begin{aligned} \phi_{ii} &= \frac{E(K_i - M_{ii})}{E(K_i)} = 1 - \frac{E(M_{ii})}{n_i p} \\ \phi_{ij} &= \frac{E(M_{ij-1} - M_{ij})}{E(K_i)} = \frac{E(M_{ij-1}) - E(M_{ij})}{n_i p} \end{aligned}\right\} \tag{3}$$

$(K_i - M_{is})$ represents the difference between the number of speech packets generated by group i calls and those that finally reach the destination. This is the number of speech packets lost during the transmission. Likewise, $(M_{ij-1} - M_{ij})$ represents the difference between the number of speech packets received by switch j and those transmitted from switch j. This is the number of speech packets lost at switch j. The ratios of the statistical expectations of these two random variables to $E(K_i) = n_i p$ then would be the end-to-end packet loss fraction and the packet loss fraction at switch j, respectively, of group i calls. It can be seen that the right-hand sides of Eqs. (2) and (3) satisfy Eq. (1).

We define the total packet loss fraction in the network to be

$$\phi = \frac{\sum_{i=1}^{s} n_i \phi_i}{\sum_{i=1}^{s} n_i} = \frac{\sum_{i=1}^{s} n_i \phi_i}{n} \quad . \tag{4}$$

Substituting the right-hand sides of Eqs. (2) and (3) into that of Eq. (4), we can obtain

$$\phi = 1 - \frac{E(M_s)}{np} \quad . \tag{5}$$



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