DESIGN CONSIDERATIONS IN PACKET MOBILE

RADIO DATA NETWORKS

Final Report to the

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ABSTRACT

Pressure of demand on the UHF radio spectrum for mobile telephony and digitally controlled dispatch systems calls for the development of efficient channel utilization schemes. This report examines alternative shared channel access procedures for packet voice/data systems in the 800 MHz band. Savings of close to 50% relative to conventional techniques are possible. Selected uplink and downlink strategies are analyzed to determine the relationship of talkspurt length, bandwidth, and number of active users to blocking probability, delay, and channel utilization. Related experimental work is presented which examines the effect of packet length and packet loss rate on received voice intelligibility.

Contents

1. Introduction

1.1 Mobile Data/Voice Systems

Considerable demand is being placed on the UHF radio spectrum owing to increased use of mobile telephony and radio dispatch systems. At present, all large scale systems use a common circuit concept in which mobile users are given a radio frequency circuit when they initiate or receive a call. When voice and data are both transmitted in a mobile system, a separate group of channels is normally reserved for each. This arrangement can lead to poor use of available capacity as loads vary. Recent advances in speech processing and digital communications allow speech transmission in packet form. This technique ensures an inherent compatability with future digital telephone systems, and a relatively easy integration of voice and data transmission services without duplication of system resources.

When speech and data are transmitted on the same channel as packets, advantage can be taken of the bursty nature of each of these sources. Speech studies show that speech occurs in talkspurts with typical durations in the regions 0.5-2.0 seconds. Talkspurts occupy about 40% of the available time, with the remaining 60% being silent gaps. With packetized voice, it is thus possible to institute an extended version of time assignment speech interpolation (TASI), whereby the silent gaps in one call are filled with a mixture of speech and data packets from other calls. Properly designed, a system of this type provides extremely efficient channel utilization with almost complete transparency to the user.

The intent of this project is to examine feasible configurations for packet voice/data radio systems in the 800 MHz band. Work carried out during the contract period, 1 August 1980-31 March 1981, has been concentrated in these areas:

(a) specification of alternative access procedures for shared control of mobile communication channels;

(b) analysis of selected alternatives to relate talkspurt length, bandwidth, and number of active users to blocking probability, delay, and channel utilization;

(c) experimental work to determine the effect of packet loss on voice intelligibility for various packet lengths.

1.2 Related Research and Development

The area of packetized voice transmission over general data networks has received increasing attention over the past few years. The main focus of the research in this area has been on evaluating alternative switching strategies for the integration of voice and data transmissions over the same network.

A survey of the advantages and costs of packet-switched voice and data networks has been compiled in [1]. Three fundamental problems have been addressed in the survey: the economics of integrating applications by users who lease tarrif communications lines, comparison of alternative switching technologies for integrating voice and data, and cost-effectiveness of alternative voice digitization rates and strategies. The survey concluded that the packet switching technology is substantially more cost-effective for serving large classes of voice and data requirements than the other alternatives examined. The study also concluded that the relative and absolute cost savings achieved by packet switching increase as the voice digitization rate increases.

A study of multiple user variable rate coding for packet transmission systems has been reported recently in [2]. The study applied variable rate coding to multiple users and demonstrated experimentally the potential gains for the case of 12 shared conversations on two-party telephone. The study also presented a practical method for implementing a variable rate ADPCM system for multiple user applications. A comprehensive study of packet voice communications models has been presented in [3]. A model for voice traffic is first developed, followed by a discussion of performance criteria. The speech traffic is shown to have a more regular and predictable arrival pattern than data. On the other hand, the network performance criteria for speech require regularity and consistency with respect to network delays and packet losses. The study also exposes the difficulty associated with multilink hopping of speech traffic.

A novel approach for transmitting packetized voice over multi-hop networks is reported in [4]. The approach, known as Packetized Virtual Circuit (PVC), has been developed at the Lincoln Laboratory of MIT. It handles voice and data in an essentially uniform fashion. Short fixed length packets with abbreviated headers are transmitted along routes established when virtual connections are set up between subscribers. The assignment of connection to links is done in such a way as to keep the probability of internal queue overflows to values which are small enough so that such overflows may be handled on an exception basis. This mechanism is augmented by placing limits on peek and average data rates at the periphery to establish a statistical flow control policy.

The above discussion gives a representative sample of the substantial body of research in the area of packetized speech in general data networks. Only recently, however, has some research effort been directed to the application of packetized speech to mobile radio networks. The prime motivating factor in this direction is the need is maximize the utilization of the scarce spectrum resource.

The topic of transmitting digitized speech over mobile radio channel has been examined recently by Ellershaw [5] and DaSilva [6].

Ellershaw [5] considered two time division multiplexing schemes in which speech is recorded for a short time, digitized and transmitted in a packet form. In the first scheme, packets are sent regularly from each active radio telephone resulting in a pure time division multiplexed system. In the second scheme packets are generated only when speech occurs and each packet is transmitted when ready. Here use is made of the natural breaks in speech to interpolate more simultaneous conversations onto the radio channel. The second scheme is more complex than the first but provides twice the number of telephone conversations in the same radio bandwidth.

DaSilva [6] developed two models for multiple-access land mobile radio systems caarrying digitized speech and packetized data. In the first model, it is shown that a more efficient use of the radio spectrum is possible by taking advantage of the statistical properties of voice and the excellent contention characteristics of the non-persistent carrier sense multiple access protocol. The second model is concerned with the study of the performance of an integrated voice and data mobile system. By giving a higher priority to the voice traffic which could be carried in analog or digital form, the model takes advantage of the natural silence gaps in a two-way voice conversation to accommodate a large number of data traffic users. An expression for the total average data packet delay is derived and the results indicate that this delay could be kept within acceptable bounds.

1.3 Report Outline

Chapter 2 introduces the basic two-hop transmission model in which mobile units communicate with each other through a fixed base station. It is evident that access procedures must be devised for both uplink and downlink channels. The alternative procedures are introduced briefly, and a qualitative comparison of their performance is made in terms of channel efficiency, delay, and complexity.

An analysis of a random access uplink model is presented in Chapter 3. One of the essential parameters of the uplink model is the size of channel group to which a caller is assigned. At one extreme, the pool of all channels could be regarded as commonly available to each user on demand, so that the channel pool consists of one group. At the other extreme, a specific channel could be allocated to each of n users, so that n groups are formed. It is hypothesized that some intermediate condition, in which N_g channels are assigned to each group, may be a more effective allocation of communications resources. The relationship between group size, delay, blocking probability, and traffic intensity is examined in Chapter 3 through analysis and simulation.

Chapter 4 considers downlink transmission in a similar fashion, with particular reference to the effect of signalling overhead.

Chapter 5 turns to the relationship between packet loss and speech intelligibility, the latter being measured by subjective testing. Finally, Chapter 6 draws conclusions from the results which have been obtained during the course of this work.

2. Mobile Unit Packet Radio Model

2.1 Introduction

This chapter examines the structure of the model assumed for mobile packet radio communication. To take advantage of the bursty nature of speech and data, channels should be shared between users. There are several methods by which this can be carried out, each one representing a particular tradeoff among the factors of efficiency, delay, and equipment complexity. The principal alternatives are discussed in section 2.3, and a simplified evaluation is presented in section 2.4. Section 2.5 summarizes results.

2.2 Network Model

The basic network model is shown in Figure 2.1. Since users cannot rely on line-of-sight transmission from sender to receiver, all calls must be relayed by a base station. It is assumed that N_u uplink channels and N_d downlink channels are available. If N_u is large, these channels, may be divided into groups of channels, with N_g channels per group.

Users may be either active or inactive. An active user is one who has established communication (i.e., a call) with another user. Note that the user is considered active for the call duration, including pause time between talkspurts. An inactive user achieves active status through a request to

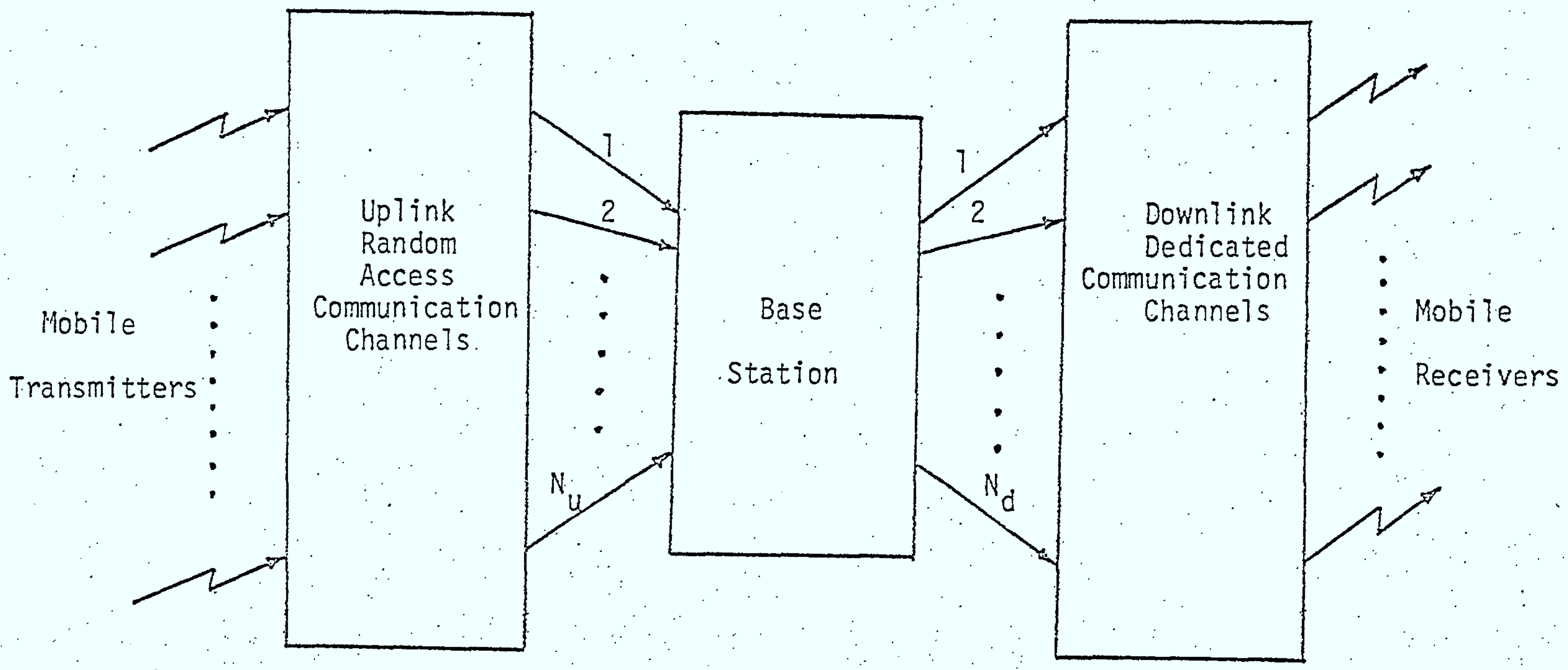

Figure 2.1 Packet Speech Network

the Call Establishment Protocol (CEP) described in section 3.2. Once a call is established, an uplink channel is assigned as required for the duration of each talkspurt. Talkspurt voice signals are digitized at 16 kb/s and formed into packets. These are placed in buffers at the transmitter sending channel acquisition, and then sent to the base station. There the packets may be queued again if it is necessary to assign a downlink channel; if the latter is dedicated, of course, no delay need be incurred.

2.3 Communications Access Strategies

The simplest channel allocation scheme is that in which a specific uplink channel and a separate downlink channel are dedicated to each pair of active users for the duration of their call. A total of n active users would then required n channels. This arrangement minimizes both equipment and protocol complexity as well as end-to-end delay, but makes poor use of channel capacity.

Each speaker on average generates talkspurts approximately 40% of the time. If it is assumed that only one member of a user pair speaks at a time, then n users would require in theory as few as 0.4n channels between them. Of course, such an allocation would yield poor performance because of delay and blocking. Moreover shared channels require additional communications capacity for the overhead associated with dynamic allocation.

In the following sections a number of channel allocation strategies will be outlined and subjected to a simplified first order analysis. The intent is to determine their approximate capacity requirements and (qualitateively) the resultant characteristics of delay and system complexity.

Available channel capacity must be divided both in bandwidth and in time. Suppose that voice signals are digitized at a rate of bits per second and that a total channel capacity of Nα bits per second is available. This may be regarded, at one extreme, as a single shared high speed channel which transmits "compressed" talkspurts at N times the normal rate. Alternatively it may be regarded as N channels, each capable of carrying one real time voice signal (in packet form) at any instant. In this report, the main emphasis is placed on the second would lead to significant loss of capacity through collisions if random access methods were used. Moreover, the time required for signal synchronization when a new transmission begins is relatively insensitive to signalling rate. The effective transmission rate of a wideband channel is reduced considerably if frequent resynchronization is necessary. On the downlink channel, with only one transmitter, these factors would not be present. It is therefore reasonable to consider wideband downlink transmission as one alternative.

Another consideration is the method of uplink channel allocation. If N_u uplink channels exist for a set of n active users, the alternatives are:

channel dedication;

frequency division multiple access (FDMA);

time division multiple access (TDMA);

demand assignment with reservations;

random access.

Fixed FDMA takes the form of channel dedication, while synchronous TDMA is inefficient with bursty traffic. The last two methods are, in effect, adaptive versions of FDMA and TDMA, since they allocate capacity and time on a demand basis. Because of the real time transmission requirements, a reservation system has not been considered. Emphasis will be placed in this report on random access methods for uplink channel communication.

Downlink communications alternatives are:

channel dedication;

single wideband channel;

dynamic channel allocation.

It has been observed previously that channel dedication is relatively simple to implement but inefficient in channel utilization. The second alternative, a single wideband channel to which all receivers are tuned, would transmit voice packets of all calls, identified by destination. Each talkspurt packet sequence would be processed and transformed to an audio signal by the appropriate receiver. Such a scheme should be reasonably effective with a fixed user population, but introduces some additional delay owing to channel switching.

2.4 Evaluation

This section examines a highly simplified model of the channel allocation alternatives as a first approximation. It is assumed that there are n active users. An active sender is assumed to generate talkspurts during 40% of the available time, with an equivalent of 10% of additional overhead to allow for channel acquisition time, packet headers, etc.

2.4.1 Uplink Transmission

a) Dedicated uplink channel (DUC): n users require n dedicated uplink channels.

b) Random access, sequential search (RASS): Base station transmits a narrowband tone on a downlink channel to indicate that a specific corresponding uplink channel is busy. Transmitter searches downlink channels sequentially until absence of busy tone is detected; n users share 0.5n uplink channels.

c) Random access, parallel search (RAPS): Base station uses a dedicated downlink channel to indicate uplink channel availability through a channel status signal. Channel availability is detected directly by the transmitter, which monitors the signalling channel. n users share (1+0.5n) channels.

2.4.2 Downlink Transmission

(a) Dedicated downlink channel (DDC): n users require n dedicated downlink channels.

(b) Dynamic downlink allocation (DDA): All receivers in quiescent state tune to a signalling channel. Base station notifies individual receivers of incoming calls, and directs them to tune to an available channle. n users share (1+0.5n) downlink channels.

(c) 'Quasi-dynamic' allocation (QDA): One downlink channel is allocated to each two receivers, A and B. If a message appears for A, receiver B continues to listen passively. If a message appears for receiver B during transmission of the talkspurt to A, an interrupt packet is generated directing receiver B to another channel. The latter may be a quiescent channel to which other receivers are tuned, or it may be one of a pool of additional 'overflow' channels designated for this purpose. If c overflow channels are maintained (c=0,1,2,3...), then n users share (c+0.5n) downlink channels.

(d) Wideband downlink transmission (WDT): All downlink transmissions a are sent at high speed over a single wideband channel to which all receivers are tuned. Talkspurt packets are extracted from the message stream by appropriate receivers, buffered, and reconstructed at a reduced clock rate as normal speech output. n users share 0.5n downlink channels.

2.5 Summary

Three uplink access strategies and four downlink ones have been presented. The resultant characteristics are shown in Tables 2.1 and 2.2. It is evident that, combining the two hops, n users require at least 0.5n channels, but not more than n channels in total, providing that only one party talks at a time. Those methods yielding greatest channel efficiency also lead to relatively complex equipment and a measure of delay in transmission. An important parameter not present in these tables is the probability of blocking. In the next sections, quantitative models will be examined to yield more accurate estimates of delay and blocking probability for selected access methods.

Table 2.1

Characteristics of Uplink Access Strategies

UPLINK: n USERS

Technique	No. of Channels	Channel Efficiency	Delay	Complexity
DUC	n	Poor	Negligible	Low
RASS	0.5n	Good	Med/large;variable	Moderate
RAPS	1+0.5n	Fair/Good	Med; fixed	Moderate

Table 2.2

Characteristics of Downlink Transmission Strategies

DOWNLINK: n USERS

Technique	No. of Channels	Channel Efficiency	Delay	Complexity
DDC	n	Poor	Negligible	Low
DDA	1+0.5n	Fair/Good	Medium; fixed	Moderate
QDA	c+0.5n	Fair/Good	Medium; fixed	Moderate
WDT	0.5n	Good	Small; Variable	Moderate

3. Random Access Uplink Model

3.1 Introduction

This chapter examines the random access serial search method for uplink transmission. Section 3.2 describes the connection establishment protocol, while section 3.3 outlines the uplink random access procedure. A general congestion model, applicable to the RASS technique, is analyzed in section 3.4. Predicted performance results are presented in section 3.5, and corresponding simulated results are given in section 3.6. Section 3.7 is a discussion and summary.

3.2 Connection Establishment Protocol

Before one mobile unit can initiate communication with another, it must make a call request through a dedicated control channel to the Connection Establishment Protocol (CEP) supported by the base station. Messages transmitted through the control channel are of short duration, and a slotted ALOHA random access method is used for CEP access. Upon receipt of a call establishment request, the CEP attempts a connection between source and destination. If this is successful, the two units become active, and can communication using the assigned talkspurt access protocol (e.g., RASS/DDA).

The pool of N_u uplink channels may be subdivided by the

CEP into multiple groups of N_g channels each. A similar grouping is possible, if desired, for downlink channels. Upon receiving a call request, channel groups are assigned to the call pair (the two mobile units associated with the call). The group assigned is the one currently serving the smallest number of pairs, so that channel group utilization is kept approximately uniform. Within the downlink group, a nominal "listening" channel is assigned to each unit; as explained in chapter 4, this is the channel to which the receiver tunes at the completion of a talkspurt reception, while awaiting a further talkspurt.

Once channel groups have been assigned, they remain unchanged for the call duration. However, specific channels are allocated only for the duration of a talkspurt. At the end of any talkspurt the transmitter (uplink) or receiver (downlink) relinquishes the channel so that it may be used by another unit.

3.3 Uplink Access Procedure

With the RASS access technique, no additional signalling channel is used for dynamic channel acquisition. Instead, the presence of a narrow band tone at one edge of each downlink passband is used to signal a busy state of a specific corresponding uplink channel. The assumption here is that the number of downlink channels is not less than the number of uplink ones.

When a talkspurt is generated, therefore, the transmitting unit's receiver begins searching for a free channel indication. The search begins at channel i=1, the downlink channel corresponding to the uplink one used for the previous talkspurt transmission. If a busy tone is received, the next channel (i+1) is sensed, and the process is repeated until a free channel is found or until k_{max} attempts have been made, whichever occurs first. Each step of the process involves a delay of t_c seconds. Figure 3.1 shows the random access method. The first time axis depicts the contents of a speech source. The second shows the speech detector response delayed by t_s, the speech detection time. Encoding and packetization of voice will start after t_s (this part of the talkspurt is already lost during voice detection). The creation of the first packet triggers the search for a free channel within the channel group. This process starts t_c seconds before the first packet is created. Thus the transmission could start immediately after the creation of the first packet if a free channel is found. A header containing address information is added to each packet prior to transmission. The end of a talkspurt is detected after a delay t_e, following which uplink channel i is relinquished. At this moment it becomes

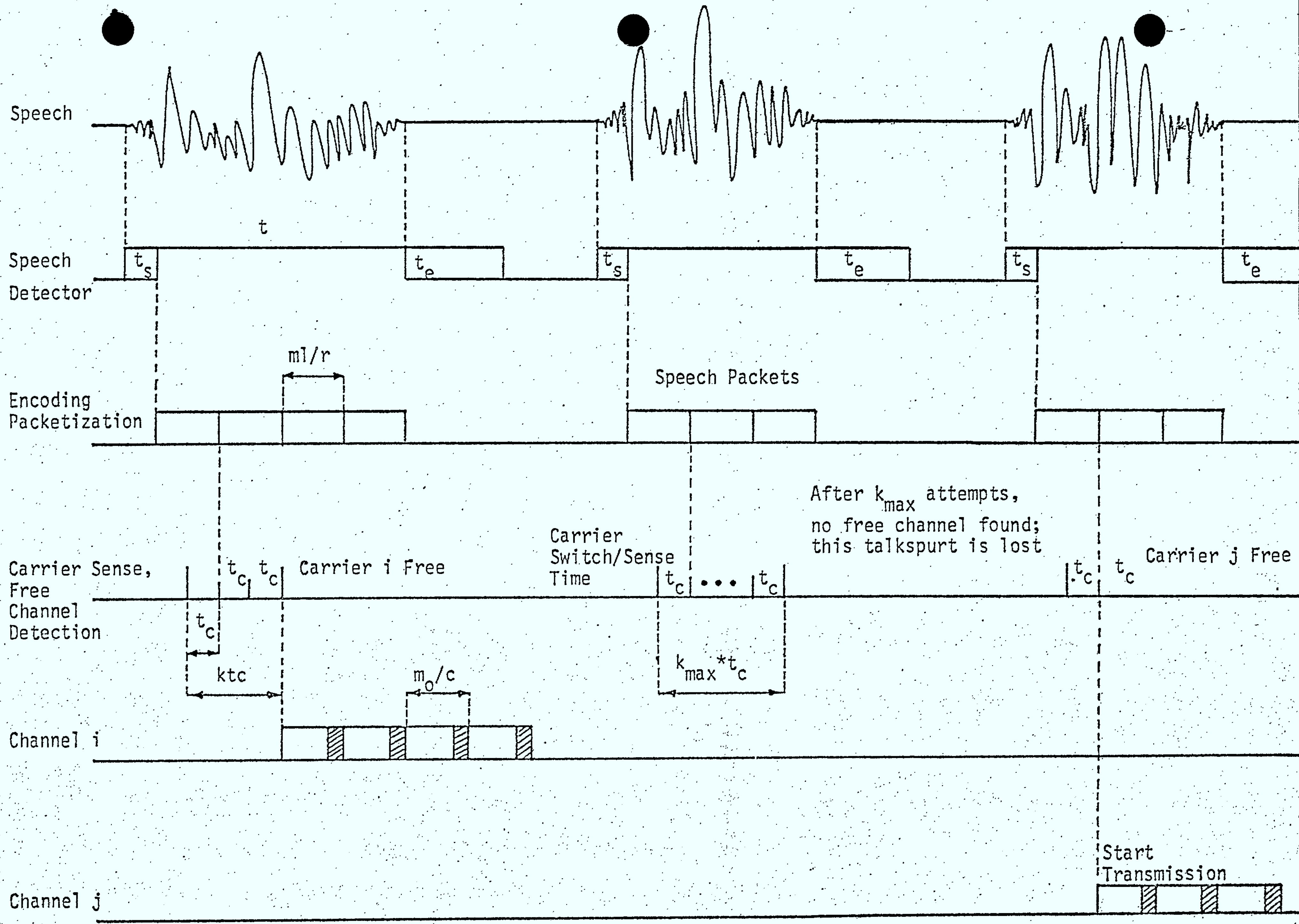

Figure 3.1

Packet Speech Model

available for the transmission of another user's talkspurt.

3.4 Random Access Congestion Model

The purpose of this section is to develop a model relating the delay encountered in search for an uplink transmission channel to the total talkspurt traffic intensity. Talkspurts are assumed to arrive in a Poisson distribution with average arrival rate λ_u per second per active channel group user. Length of talkspurts is exponentially distributed with mean t seconds. With n active users sharing a group of N channels, the group traffic intensity is therefore

$$\rho = \frac{n\lambda_u t_u}{N_g} \tag{3.1}$$

Consider a search policy in which a free channel is sought by carrier sense testing of successive channels in the group. When a free channel is discovered, packet transmission begins (owing to finite propagation time and carrier sense time, the possibility of collision still exists). If no free channel is found after k_{max} channels have been searched, the transmission channel is blocked; it is emptied, and new incoming speech packets are stored in an alternate buffer. The search then begins again. If this second search is unsuccessful in k_{max} attempts, the user is blocked, and an indication to that effect is given to him.

If carrier sense time is t_c seconds, then a maximum of

$k_{max} t_c$ seconds of speech can be lost through buffer blocking with interruption to the conversation. Since t_c is expected to be about 30 ms, and speech dropouts should not exceed 250 ms, it is reasonable to choose $k_{max}=8$ with this search policy. To avoid correlation between users' searches, it is desirable that each start on a randomly chosen channel within the group. This effect can be achieved by starting the search on the channel which was used for the previous talkspurt transmission.

When the search begins, it is assumed that v of the available N_g channels are already occupied; assuming independence, the probability of a given channel being occupied is thus v/N_g. Suppose that k is the number of searches for a free channel which occur up to and including the moment of channel acquisition or abandonment of search. We observe that

$$p(k) = \sum_{v=0}^{N_g} p(k|v)\, p(v) \tag{3.2}$$

Consider first the determination of p(v) in equation (3.2). This function is the probability that v servers are occupied in an $M/M/N_g/N_g$ queue. Erlang's loss formula gives

$$p(v) = \frac{(N_g\rho)^v / v!}{\sum_{i=0}^{N_g} (N_g\rho)^i / i!} \tag{3.3}$$

where ρ is given by equation (3.1).

The second term to be considered is the conditional probability $p(k/v)$. Observe that for $v \geq k$,

$$p(1|v) = \frac{N_g - V}{N_g} \qquad , \qquad p(2|v) = \frac{v}{N_g} \frac{N_g - v}{N_g - 1}$$

$$p(3|v) = \frac{v}{N_g} \frac{v-1}{N_g - 1} \frac{N_g - 1}{N_g - 2} \qquad \text{etc.}$$

while for $v=k-1$

$$p(k|v) = \frac{v}{N_g} \cdot \frac{v-1}{N_g - 1} \cdot \frac{v-2}{N_g - 2} \cdots \frac{1}{N_g - v + 1}$$

For $v<k-1$, $p(k/v)=0$.

Thus, in general,

$$\begin{aligned} p(k|v) &= \frac{v!}{(v-k+1)!} \frac{(N_g - k)!}{N_g!} (N_g - v) \ , && k \leq v \leq N_g \\ &= \frac{v!(N_g - v)!}{N_g!} \ , && v = k - 1 \\ &= 0 \ , && \text{otherwise} \end{aligned} \tag{3.4}$$

Now, substitution of (3.4) and (3.3) into (3..2) yields

$$p(k) = \sum_{v=0}^{N_g} p(k|v)\ p(v) = \sum_{v=k-1}^{N_g} p(k|v)\ p(v)$$

$$= p(k|k-1)\ p(k-1) + \sum_{v=k}^{N_g} p(k|v)\ p(v)$$

$$p(k) = [N_g! \sum_{i=0}^{N_g} (N_g\rho)^{i}/i!]^{-1} \ [(N_g\rho)^{k-1} \ (N_g - k+1)!$$

$$+ (N_g - k)! \sum_{v=k}^{N_g} \frac{(N_g\rho)^{v} \ (N_g - v)}{(v-k+1)!} \ ] \tag{3.5}$$

3.5 System Performance

The probability of channel blocking P_B has been calculated for different channel groups N_g and different arriving traffic levels ρ. The maximum number of channels searched k_{max} is kept constant at δ . Figure 3.2 shows that for traffic levels below 0.6 erlangs, P_B decreases as N_g increases and shows asymptotic behaviour for higher values of N_g. However, for traffic levels over 0.6 erlangs P_B increases with N_g to reach a maximum value then decreases again as N_g increases. Such maximum is a function of traffic level and channel group. This is depicted by the dashed line on Figure 3.2. Table 3.1 gives values of the maximum probability of channel blocking P_{Bmax} and the corresponding channel group N_g for different traffic levels. It must be noticed that the first two values in table (marked by an asterisk) are calculated with the assumption that the maximum number of channels searched is reduced to 6. Thus their values must be relatively higher than the true values.

Figure 3.3 shows the expected number of channel searches in a successful transmission $E(k/k \leq k_{max})$.

3.6 Simulation Results

A SLAM combined network-discrete event model is used to simulate the Random-Access Uplink Model (Figure 3.4).

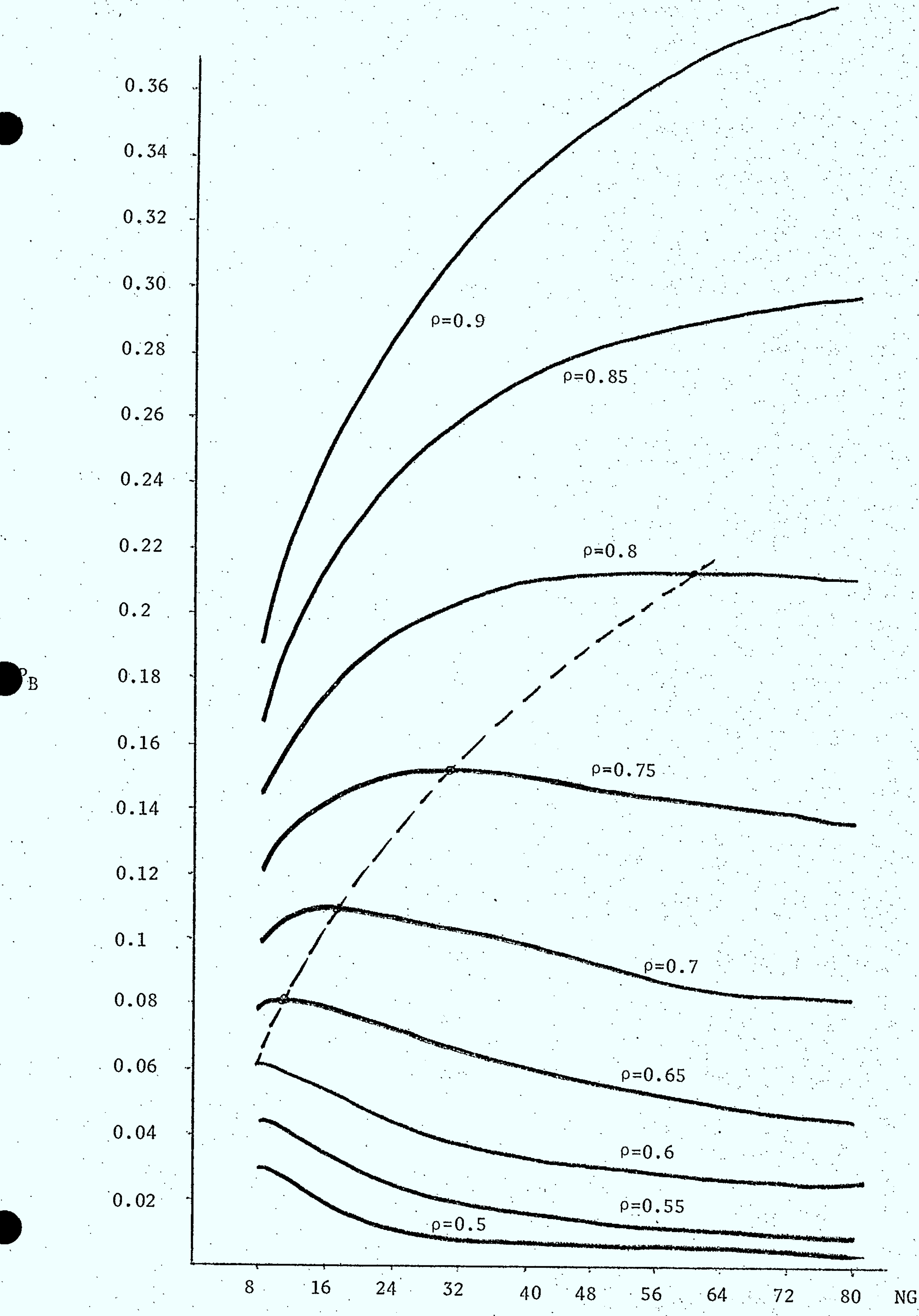

Figure 3.2 <u>Channel Blocking Probability</u>

Table 3.1

ρ	P_{Bmax}	N_g
0.6	0.07328	7*
0.625	0.08299	7*
0.65	0.08116	11
0.675	0.09412	14
0.7	0.10960	18
0.725	0.12824	23
0.75	0.15085	32
0.775	0.17844	43
0.8	0.21230	61
0.825	0.25375	$\geq$80

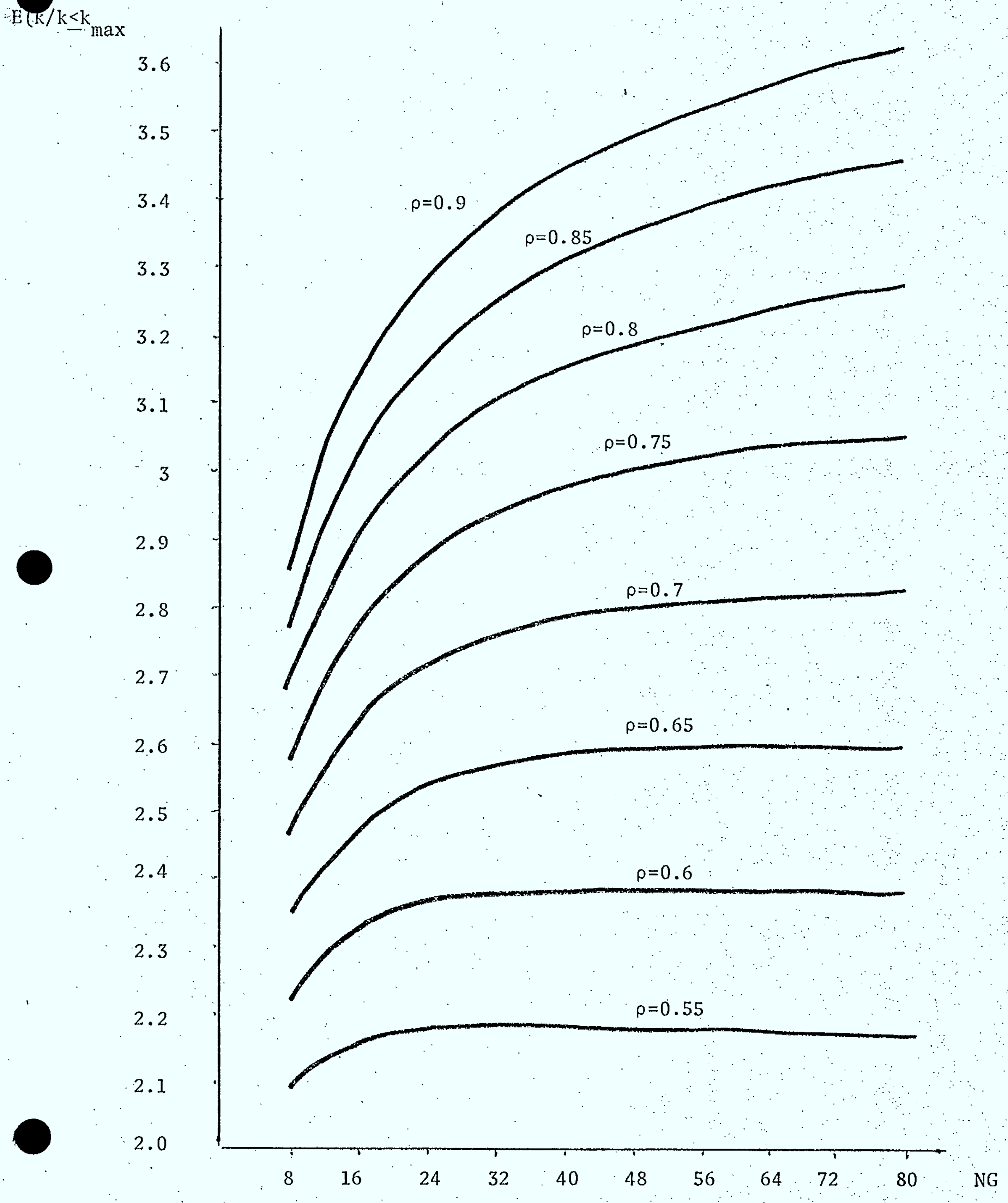

Figure 3.3 Expected Number of Channel Searches

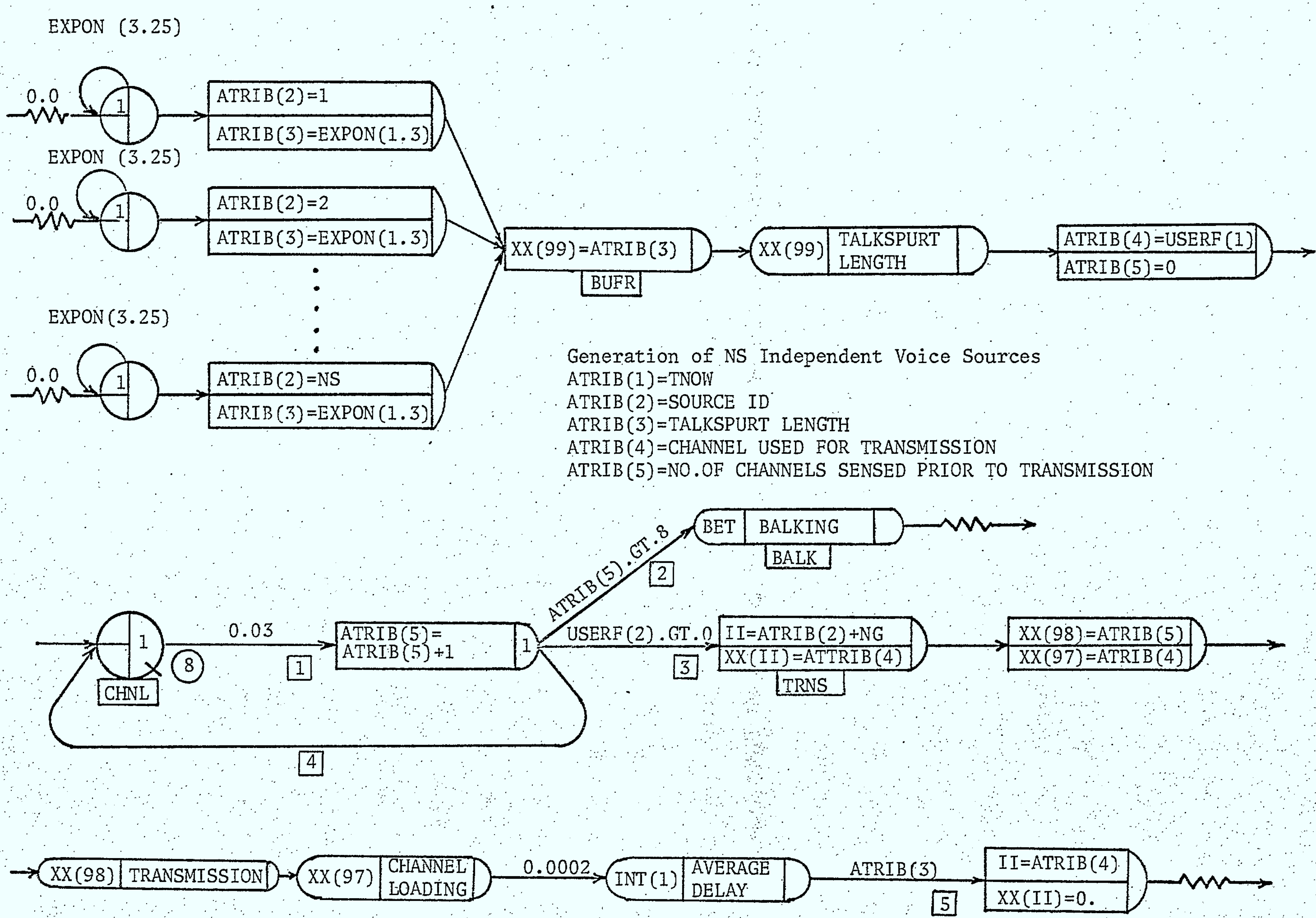

Figure 3.3(a) Random Access Uplink Model

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      FUNCTION USERF(IFN)
      COMMON/SCOM1/ATRIB(100),DD(100),DDL(100),DTNOW,II,MFA,MSTOP,NCLNR
     1,NCRDR,NPRNT,NNRUN,NNSET,NTAPE,SS(100),SSL(100),TNET,TNOW,XX(100)
      GO TO (1,2),IFN
C
C   XX(1) TO XX(NG) ARE RESERVED FOR CHANNEL OCCUPANCY STATE
C   XX(41) AND MP ARE RESERVED FOR CHANNEL USED
C
C   ASSIGNING CHANNEL USED IN THE PREVIOUS TALKSPURT
  1   JR=40+IFIX(ATRIB(2))
      USERF=XX(JR)
      RETURN
C
C   SEARCHING FOR A FREE CHANNEL
  2   J=IFIX(ATRIB(4))
      KTS=IFIX(XX(J))
      IF(KTS.EQ.0) GO TO 10
C   SCHEDULE SEARCH FOR NEXT CHANNEL
      ATRIB(4)=ATRTB(4)+1.
      IF(ATRIB(4).GT.20.)ATRIB(4)=1.
      USERF=0
      RETURN
C   CHANNEL J IS FREE FOR TRANSMISSION
  10  XX(J)=1.
      USERF=J
      RETURN
      END
```

Figure 3.4(b) Channel Search Function

Talkspurts are generated from independent voice sources with exponential distribution and averages of 1.3 sec. and 1.95 sec. for talkspurts and silent gap length respectively. After a talkspurt is generated, the channel sensing process starts. At 30 ms per search a maximum of eight searches are allowed. Search is carried out successively and starts from the previous channel sensed by the same subscriber (voice source). If a free channel is sensed the talkspurt will be transmitted (in packetized form) otherwise it will balk. Statistical information is collected through the network for channel blocking, loading, talkspurt delay, etc. Simulation runs are performed each for a period of 1000 sec. All statistical arrays including the file statistics are cleared after 100 sec. to insure that initial conditions do not bias the results seriously.

A typical simulation run output is shown in Figure 3.5-3.7. Figure 3.5 shows a typical statistical summary report. Probabilities of k searches where $k=1,\ldots,k_{max}$ for a successful transmission is given in Figure 3.6. The distribution of the transmitted traffic on the different channels is shown in Figure 3.7.

Statistical results of the simulation runs are summarized in Table 2 for different traffic levels. Figure 3.8 shows the distribution of the probability distribution of

Figure 3.5 S L A M S U M M A R Y R E P O R T

SIMULATION PROJECT MOBILE NETWORK BY S.E. AIDAROUS

DATE 2/16/1981 RUN NUMBER 1 OF 1

CURRENT TIME .1000E+04
STATISTICAL ARRAYS CLEARED AT TIME .1000E+03

STATISTICS FOR VARIABLES BASED ON OBSERVATION

	MEAN VALUE	STANDARD DEVIATION	COEFF. OF VARIATION	MINIMUM VALUE	MAXIMUM VALUE	NUMBER OF OBSERVATIONS
GAP LENGTH	.3973E+01	.3600E+01	.9063E+00	.9125E-02	.1695E+02	226
GAP LENGTH	.3842E+01	.3526E+01	.9178E+00	.1651E-01	.1685E+02	233
GAP LENGTH	.4006E+01	.3997E+01	.9979E+00	.1653E-01	.2226E+02	225
TALKSPURT LENGTH	.1307E+01	.1324E+01	.1013E+01	.2384E-03	.1007E+02	2771
BALKING	.1062E+02	.1998E+02	.1881E+01	.1081E-02	.1006E+03	85
TRANSMISSION	.2164E+01	.1495E+01	.6907E+00	.1000E+01	.8000E+01	2685
CHANNEL LOADING	.4503E+01	.2294E+01	.5094E+00	.1000E+01	.8000E+01	2685
AVERAGE DELAY	.6510E-01	.4484E-01	.6887E+00	.3018E-01	.2402E+00	2685

STATISTICS FOR TIME-PERSISTENT VARIABLES

	MEAN VALUE	STANDARD DEVIATION	MINIMUM VALUE	MAXIMUM VALUE	TIME INTERVAL	CURRENT VALUE
NO. OF SEARCHS	.2231E+01	.1562E+01	.1000E+01	.8000E+01	.9000E+03	.1000E+01
CHNNEL 1 STATE	.4927E+00	.4999E+00	.0000E+00	.1000E+01	.9000E+03	.1000E+01
CHNNEL 2 STATE	.4877E+00	.4998E+00	.0000E+00	.1000E+01	.9000E+03	.1000E+01
CHNNEL 3 STATE	.5084E+00	.4999E+00	.0000E+00	.1000E+01	.9000E+03	.0000E+00
CHNNEL 4 STATE	.5036E+00	.5000E+00	.0000E+00	.1000E+01	.9000E+03	.0000E+00
CHNNEL 5 STATE	.4672E+00	.4989E+00	.0000E+00	.1000E+01	.9000E+03	.0000E+00
CHNNEL 6 STATE	.4614E+00	.4985E+00	.0000E+00	.1000E+01	.9000E+03	.0000E+00
CHNNEL 7 STATE	.4809E+00	.4999E+00	.0000E+00	.1000E+01	.9000E+03	.0000E+00
CHNNEL 8 STATE	.4912E+00	.4999E+00	.0000E+00	.1000E+01	.9000E+03	.1000E+01

HISTOGRAM NUMBER 3

Figure 3.6   TRANSMISSION

```
 OBSV    RELA    CUML      UPPER
 FREQ    FREQ    FREQ   CELL LIMIT   0         20        40        60        80      100
                                     +    +    +    +    +    +    +    +    +    +    +
    0    .000    .000    .0000E+00   +                                                 +
    0    .000    .000    .5000E+00   +                                                 +
 2138    .408    .408    .1000E+01   +********************                             +
    0    .000    .408    .1500E+01   +                   C                             +
 1304    .249    .657    .2000E+01   +************                    C                +
    0    .000    .657    .2500E+01   +                                C                +
  698    .133    .790    .3000E+01   +*******                               C          +
    0    .000    .790    .3500E+01   +                                      C          +
  459    .088    .878    .4000E+01   +****                                       C     +
    0    .000    .878    .4500E+01   +                                           C     +
  268    .051    .929    .5000E+01   +***                                          C   +
    0    .000    .929    .5500E+01   +                                             C   +
  189    .036    .965    .6000E+01   +**                                             C +
    0    .000    .965    .6500E+01   +                                               C +
  106    .020    .985    .7000E+01   +*                                               C+
    0    .000    .985    .7500E+01   +                                                C+
   79    .015   1.000    .8000E+01   +*                                                C
    0    .000   1.000    .8500E+01   +                                                 C
    0    .000   1.000    .9000E+01   +                                                 C
    0    .000   1.000    .9500E+01   +                                                 C
    0    .000   1.000    .1000E+02   +                                                 C
    0    .000   1.000       INF      +                                                 C
  ---                                +    +    +    +    +    +    +    +    +    +    +
 5241                                0         20        40        60        80      100
```

HISTOGRAM NUMBER 4

Figure 3.7 CHANNEL LOADING

OBSV FREQ	RELA FREQ	CUML FREQ	UPPER CELL LIMIT
654	.125	.125	.1000E+01
0	.000	.125	.1500E+01
655	.125	.250	.2000E+01
0	.000	.250	.2500E+01
639	.122	.372	.3000E+01
0	.000	.372	.3500E+01
669	.128	.499	.4000E+01
0	.000	.499	.4500E+01
622	.119	.618	.5000E+01
0	.000	.618	.5500E+01
669	.128	.746	.6000E+01
0	.000	.746	.6500E+01
666	.127	.873	.7000E+01
0	.000	.873	.7500E+01
667	.127	1.000	.8000E+01
0	.000	1.000	.8500E+01
0	.000	1.000	.9000E+01
0	.000	1.000	INF

5241			

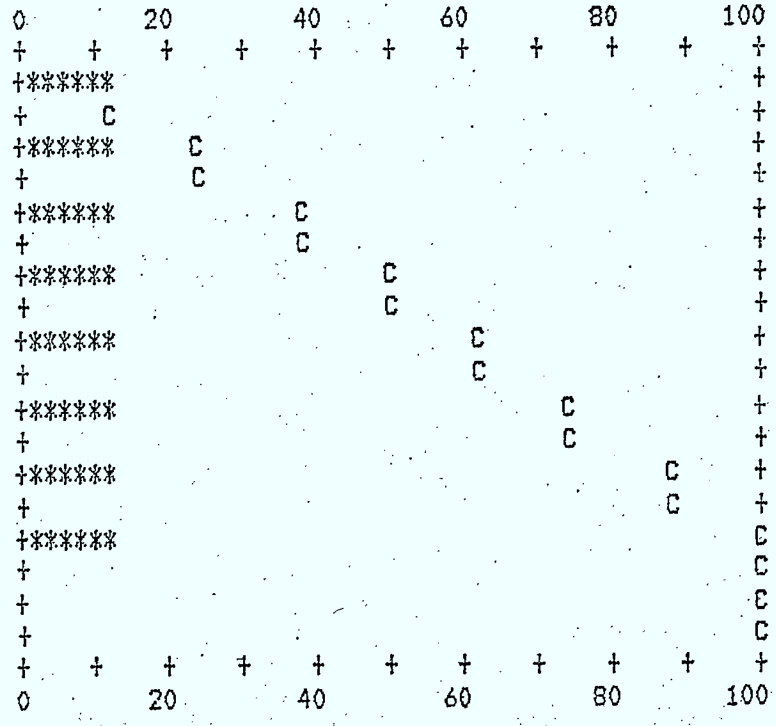


Table 3.2

$\rho = 0.5$

N_g	8	12	16	20	24
P_B	0.03	0.0173	0.02	2.0215	0.22
$E(k/k \leq k_{max})$	2.164	2.135	2.25	2.205	2.239
av. delay m.s.	65.1	64.23	67.88	66.29	67.35

$\rho = 0.6$

N_g	8	12	16	20	24
P_B	0.0562	0.0577	0.0503	0.054	0.472
$E(k/k \leq k_{max})$	2.342	2.468	2.514	2.464	2.47
av. delay m.s.	71.88	79.75	75.68	74.1	74.2

$\rho = 0.7$

N_g	8	12	16	20	24
P_B	0.0948	0.0815	0.0834	0.0851	0.086
$E(k/k/k_{max})$	2.655	2.804	2.785	2.771	2.78
av. delay m.s.	79.83	84.3	83.74	83.32	83.4

HISTOGRAM NUMBER 6

Figure 3.8(a) TRANSMISSION $\rho=0.5$ Ng=8

```
  OBSV    RELA    CUML       UPPER
  FREQ    FREQ    FREQ   CELL LIMIT    0         20        40        60        80        100
                                       +    +    +    +    +    +    +    +    +    +    +
     0    .000    .000    .0000E+00    +                                                 +
     0    .000    .000    .5000E+00    +                                                 +
  1215    .453    .453    .1000E+01    +**********************                           +
     0    .000    .453    .1500E+01    +                      C                          +
   686    .256    .708    .2000E+01    +*************                     C              +
     0    .000    .708    .2500E+01    +                                  C              +
   347    .129    .837    .3000E+01    +******                                   C       +
     0    .000    .837    .3500E+01    +                                         C       +
   202    .075    .913    .4000E+01    +****                                         C   +
     0    .000    .913    .4500E+01    +                                             C   +
   113    .042    .955    .5000E+01    +**                                             C +
     0    .000    .955    .5500E+01    +                                               C +
    63    .023    .978    .6000E+01    +*                                               C+
     0    .000    .978    .6500E+01    +                                                C+
    41    .015    .993    .7000E+01    +*                                                C
     0    .000    .993    .7500E+01    +                                                 C
    18    .007   1.000    .8000E+01    +                                                 C
     0    .000   1.000    .8500E+01    +                                                 C
     0    .000   1.000    .9000E+01    +                                                 C
     0    .000   1.000    .9500E+01    +                                                 C
     0    .000   1.000    .1000E+02    +                                                 C
     0    .000   1.000          INF    +                                                 C
  ----                                 +    +    +    +    +    +    +    +    +    +    +
  2685                                 0         20        40        60        80        100
```

HISTOGRAM NUMBER 3

Figure 3.8(b) TRANSMISSION ρ=0.6 Ng=8

```
 OBSV    RELA    CUML      UPPER
 FREQ    FREQ    FREQ    CELL LIMIT    0        20        40        60        80       100
                                       +    +    +    +    +    +    +    +    +    +    +
    0    .000    .000    .0000E+00     +                                                +
    0    .000    .000    .5000E+00     +                                                +
 1299    .416    .416    .1000E+01     +********************                            +
    0    .000    .416    .1500E+01     +                   C                            +
  758    .243    .659    .2000E+01     +***********                  C                  +
    0    .000    .659    .2500E+01     +                             C                  +
  416    .133    .792    .3000E+01     +*******                            C            +
    0    .000    .792    .3500E+01     +                                   C            +
  258    .083    .874    .4000E+01     +****                                   C        +
    0    .000    .874    .4500E+01     +                                       C        +
  162    .052    .926    .5000E+01     +***                                      C      +
    0    .000    .926    .5500E+01     +                                         C      +
  110    .035    .961    .6000E+01     +**                                         C    +
    0    .000    .961    .6500E+01     +                                           C    +
   67    .021    .983    .7000E+01     +*                                           C+
    0    .000    .983    .7500E+01     +                                            C+
   54    .017   1.000    .8000E+01     +*                                            C
    0    .000   1.000    .8500E+01     +                                             C
    0    .000   1.000    .9000E+01     +                                             C
    0    .000   1.000    .9500E+01     +                                             C
    0    .000   1.000    .1000E+02     +                                             C
    0    .000   1.000       INF        +                                             C
 ----                                  +    +    +    +    +    +    +    +    +    +    +
 3124                                  0        20        40        60        80       100
```

HISTOGRAM NUMBER 6

Figure 3.8(c) TRANSMISSION $\rho=0.7$ Ng=8

```
OBSV    RELA    CUML        UPPER
FREQ    FREQ    FREQ     CELL LIMIT     0        20        40        60        80       100
                                        +    +    +    +    +    +    +    +    +    +    +
   0    .000    .000    .0000E+00       +                                                 +
   0    .000    .000    .5000E+00       +                                                 +
1279    .363    .363    .1000E+01       +******************                               +
   0    .000    .363    .1500E+01       +                 C                               +
 812    .231    .594    .2000E+01       +************                 C                   +
   0    .000    .594    .2500E+01       +                             C                   +
 527    .150    .743    .3000E+01       +*******                             C            +
   0    .000    .743    .3500E+01       +                                    C            +
 304    .086    .829    .4000E+01       +****                                    C        +
   0    .000    .829    .4500E+01       +                                        C        +
 243    .069    .898    .5000E+01       +***                                         C    +
   0    .000    .898    .5500E+01       +                                            C    +
 154    .044    .942    .6000E+01       +**                                            C  +
   0    .000    .942    .6500E+01       +                                              C  +
 118    .033    .976    .7000E+01       +**                                              C+
   0    .000    .976    .7500E+01       +                                                C+
  86    .024   1.000    .8000E+01       +*                                                C
   0    .000   1.000    .8500E+01       +                                                 C
   0    .000   1.000    .9000E+01       +                                                 C
   0    .000   1.000    .9500E+01       +                                                 C
   0    .000   1.000    .1000E+02       +                                                 C
   0    .000   1.000       INF          +                                                 C
 ---                                    +    +    +    +    +    +    +    +    +    +    +
3523                                    0        20        40        60        80       100
```

k searches before a successful transmission.

3.7 Summary

This chapter has examined the performance of the random access serial search technique for uplink transmission. Of particular interest is the effect on system performance of the group size N_g and the traffic intensity ρ. As expected, both the probability of blocking and the expected number of search attempts increase for increasing traffic intensity. The effect of N_g, however, is much more complex, and not entirely as expected. In Figure 3.2 it can be seen that for low traffic intensity (ρ>.6) and increase in group size tends to decrease the probability of blocking, as one might expect. As the traffic intensity increases, it can be seen that a maximum value of blocking probability exists, such that either an increase or a decrease in group size at this point has the effect of decreasing blocking probability. For practical operational reasons group sizes below eight have not been considered; the maximum group size used was 80. Within these limits another interesting feature is evident. For values of ρ>0.7 minimum blocking ability is obtained with a large group size. For traffic values above this level, however, it appears that a small group size reduces the probability of blocking. This result is quite unexpected, and should be the subject of further investigation.

It can be seen in Figure 3.4 that large group sizes tend to increase the expected delay. This trend is particularly noticeable for high traffic intensity. The additional delay, when combined with a decreased probability of blocking, as is the case with low traffic intensity, is to be expected. The decrease in blocking probability causes an increase in total throughput, which reasonably should give rise to additional delay. At higher traffic intensities, however, additional delay is present in combination with increased blocking probability. This result seems to run counter to normal experience in which resource pooling yields an increase in efficiency.

On the basis of these preliminary results, it would appear advantageous to place uplink transmitters in large groups under conditions of low traffic intensity, but to reform them into smaller groups of 8 each when traffic intensity increases beyond 0.7.

4. Random Access Downlink Model

4.1 Introduction

Dynamic downlink allocation (DDA) and quasi-dyanmic allocation (QDA) are considered in this chapter. Each of these has been introduced briefly in chapter 2. Section 4.2 outlines the protocol used in somewhat more detail. The appropriate analytic model here is simply an $M/M/N_d/N_d$ queue in which a population of n active users shares a group of N_d downlink channels available at the base station. The resulting Erlang blocking model is presented in section 4.3, while system performance is considered in section 4.4. Section 4.5 summarizes results.

4.2 Downlink Protocol

For each source/receiver pair in a call, talkspurts are generated by the source and transmitted over the uplink leg, as described in chapter 3. Assume that there are n sources, and hence n "active" receivers. A total of N_d downlink channels is allocated, where $N_d \leq n$. The downlink assignment strategy is designed to share the pool of N_d channels amongst n users to achieve high channel utilization, reasonably low delay, acceptable blocking probability, and modest signalling overhead.

With dynamic downlink allocation (DDA), one specific channel is set aside for signalling. All receivers not currently receiving a talkspurt tune to the signalling channel to await the arrival of a message (talkspurt). Message arrivals are announced in the form of signal packets which specify destination unit, together with the channel over which the message will be sent. Upon detection of a signalling packet, the appropriate receiver tunes to the indicated channel. This operation requires a time t_W, typically about 30 milliseconds; the base station withholds transmission for time t_W, and then begins sending the talkspurt. Upon completion of the transmission the receiver switches again to the signalling channel.

Quasi-dynamic allocation (QDA) differs in that no signalling channel is used. Each receiver is assigned to a specific channel when not receiving a talkspurt, i.e., in the "listening" state. Since $N_d < n$ in nearly all cases, there will be more than one receiver awaiting a message on each downlink channel. Consider the case of two receivers, A and B, tuned to an idle channel. A talkspurt arriving at the base station for unit B is immediately transmitted as a packet sequence. Suppose now that a message arrives for A during transmission of B's talkspurt. In such a case, an interrupt packet is generated to alert A to the incoming talkspurt, and to allocate an alternate channel to which A must switch (taking time t_W) to receive its message. On completion of A's message, receiver A reverts to the listening state on its nominal channel.

4.3 Traffic Model

As previously mentioned, the traffic model is an Erlang B queue. A total of n users share N_d downlink channels (here N_d does not include possible signalling channels). It is assumed that the downlink talkspurt arrival rate λ_d is equal to the uplink arrival rate λ_d. This assumption is slightly conservative, since in fact $\lambda_d < \lambda_d$ because of blocking in the uplink channel. The downlink traffic intensity ρ_d is given by

$$\rho_d = \frac{\beta n \lambda_d t_u}{Nd} \tag{4.1}$$

where t_u is mean talkspurt length, and β is a factor which takes account of signalling overhead. In DDA channels signalling is relegated to a separate dedicated channel; the value of β reflects both this additional channel and the time lost in channel switching. Each time a talkspurt of (mean) length t_u is sent, a receiver channel switch is required, taking t_w seconds. In addition, an extra signalling channel is used in conjunction with N_d active downlink channels. Thus for DDA,

$$\beta_1 = \left(1 + \frac{1}{Nd}\right)\left(1 + \frac{t_w}{t_u + t_w}\right) \tag{4.2}$$

For QDA, no additional overhead signalling is necessary when a talkspurt arrives for a user whose assigned downlink channel is idle. The probability that any source in a source/destination pair is idle is $(1-\lambda_d t_u)$. The mean number of active receivers per downlink channel is n/N_d. Given that no message is being sent for a particular receiver the probability that all other $(\frac{n}{N_d} - 1)$ receivers sharing the channel are also idle is

$$p_c = (1-\lambda_d t_u)^{(\frac{n}{N_d} - 1)} \quad , n > N_d \tag{4.3}$$

if statistical independence of sources is assumed. If the channel is not free when a talkspurt arrives for a waiting receiver, then a signalling packet of length t_z must be inserted into the packet stream to signal a channel switch, the latter requiring a further t_w seconds. Transmission of a talkspurt of length t_u in this case entails occupation of the channel for an additional time $t_w + t_z$. Thus for QDA, the overhead factor β_2 is given by

$$\beta_2 = 1 + \frac{(1-\lambda_d t_u)^{(\frac{n}{Nd} - 1)} (t_w + t_z)}{t_u + t_w + t_z} \tag{4.4}$$

Downlink blocking probability is given by the Erlang B equation

$$p_B = \frac{(\beta_n \lambda_d t_u)^{N_d} / N_d!}{\sum_{i=o}^{N_d} (\beta n \lambda d t_u)^i / i!} \tag{4.5}$$

where $\beta = \beta_1$ (DDA) or $\beta = \beta_2$ (QDA).

4.4 System Performance

Comparison of equations (4.2) and (4.4) yields an estimate of the relative efficiency of DDA and QDA in terms of channel usage. Consider the following typical values for an operational system:

$\lambda_d = 0.32 \text{ sec}^{-1}$

$t_u = 1.25$ sec

$t_w = 30$ msec

$t_z = 5$ msec

With these values, $\beta_1 \geq 1.0234$ from (4.2), while $\beta_2 \leq 1.0272$ from (4.4). In particular, suppose that twenty active users are generating calls on twelve downlink channels ($n=20$, $N_d=12$). Then $\beta_1 = 1.109$, while $\beta_2 = 1.019$. Quasi-dynamic allocation using no downlink signalling channel is in this case significantly more efficient than dynamic downlink allocation.

Figure 4.1 shows the relationship between downlink blocking probability, number of downlink channels, and number of active talkspurt generators for DDA and QDA.

4.5 Summary

This chapter has examined two variants of random access downlink channel allocation. It has been shown that good channel efficiency can be achieved by use of quasi-dynamic

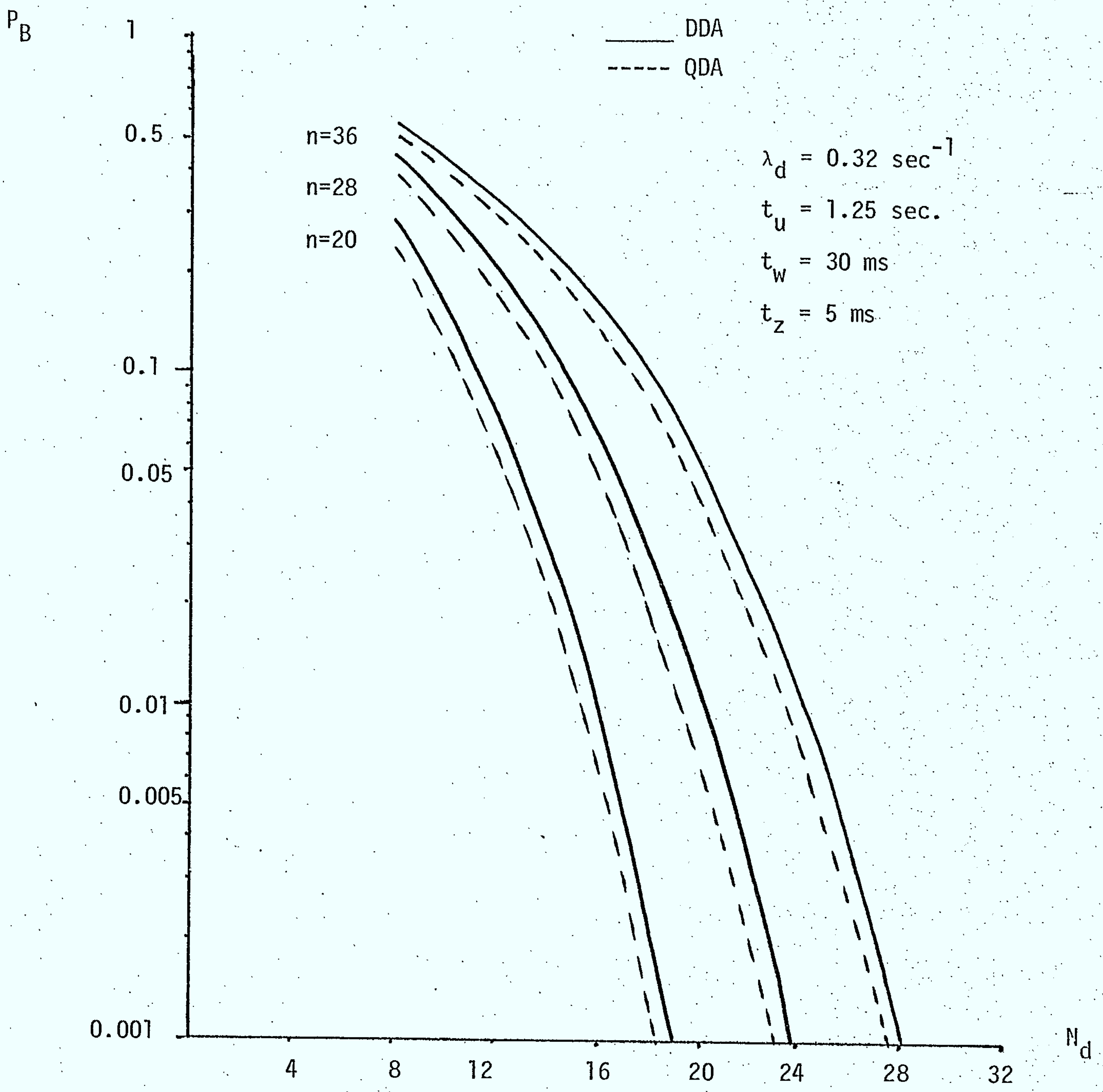

P_B
1
0.5
0.1
0.05
0.01
0.005
0.001
DDA
QDA
n=36
n=28
n=20
λ_d = 0.32 sec^{-1}
t_u = 1.25 sec.
t_w = 30 ms
t_z = 5 ms
4
8
12
16
20
24
28
32
N_d

Figure 4.1

allocation. With this technique each receiver is assigned a nominal channel upon which to listen for transmissions. If the channel is free when the message arrives, it is passed on without delay. However, since more than one receiver is assigned to each channel, the nominal channel may be in use by another receiver when a message for the waiting receiver arrives at the base station. If this happens, an interrupt packet is generated which directs the receiver to tune to an idle channel to receive its message.

Analysis shows that the use of this technique yields low signalling overhead. Providing the channel pool is large, the blocking probability can also be kept to an acceptably low level.

5. Transmission Effects

5.1 Introduction

Little is known about impairments produced in digital processing of speech signals. The need of subjective testing arises from the lack of accurate objective measures that reflect the human perception [7].

Subjective testing is controversial. Should simple grading, unsatisfactory to high quality in 5 steps, or multidimensional analysis be used? What form should the test take, e.g., alphanumeric characters, words, text, carefully assembled sentences, natural dialogue, etc.? However, what is even more in dispute is relating subjective testing results to objective measurements.

Subjective tests have been performed [8,9] to investigate perceptual effects of digital encoding of speech while describing the quality of different encoders. Subjects listened to different lists of consonant-vowel-constant syllables processed by all of the digital circuits and identify the initial consonant. In each listening session, subjects rated circuit quality on a 5 point category scale. Influence on intelligibility and subjective quality of three distortions: bandwidth reduction, peak clipping, amplitude quantization was investigated.

The goal of this experiment is to determine the effects of packet size and loss rates on digital voice intelligibility. A subject will listen to different sets of words, alphanumeric codes, sentences and short texts transmitted under different conditions. The quality of the transmitted voice is evaluated using a 5 point scale: high, good, fair, poor, unsatisfactory for sentences and short texts. The subject is asked also to print transmitted words and alphanumeric codes in order to get some measure of the recognition accuracy.

5.2 Experimental System

Voice coding/decoding is performed by a hybrid codec chip using the Continuous Variable Slope Delta Modulation (CVSD) technique at a rate of 16 kps. The encoder samples the input analog voice signal and generates a bit for each period. The bit level depends on whether the sampled analog signal is greater or smaller than the approximate analog signal of the previous sample generated by the local decoder. The decoder includes an integrator whose signal is increased or decreased by a variable step size in response to each bit.

Figure 5.1 shows the general configuration of the system. The CVSD encoder binary output is processed using an Intel 80/20 single board microcomputer (see Figure 5.2),

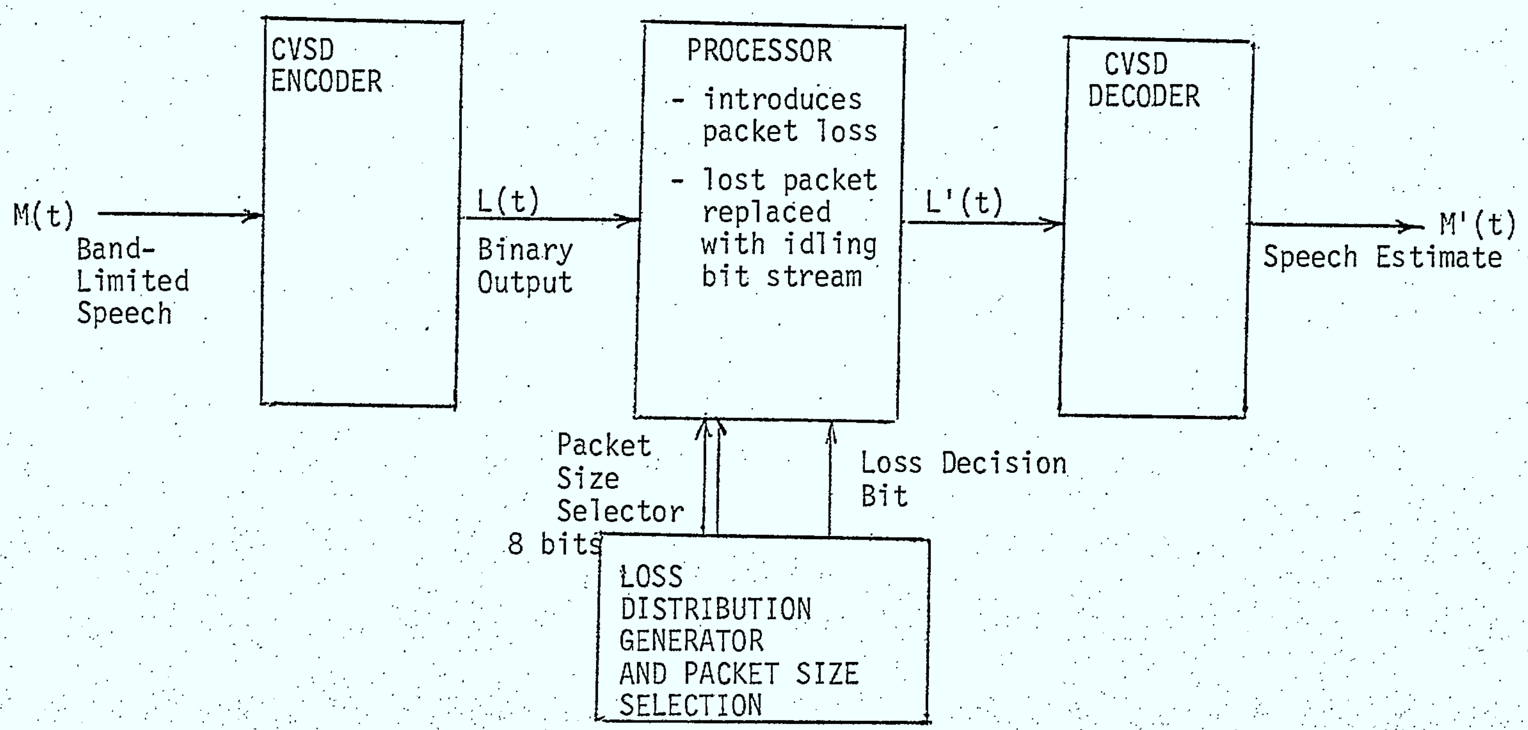


Figure 5.1 Test System Configuration

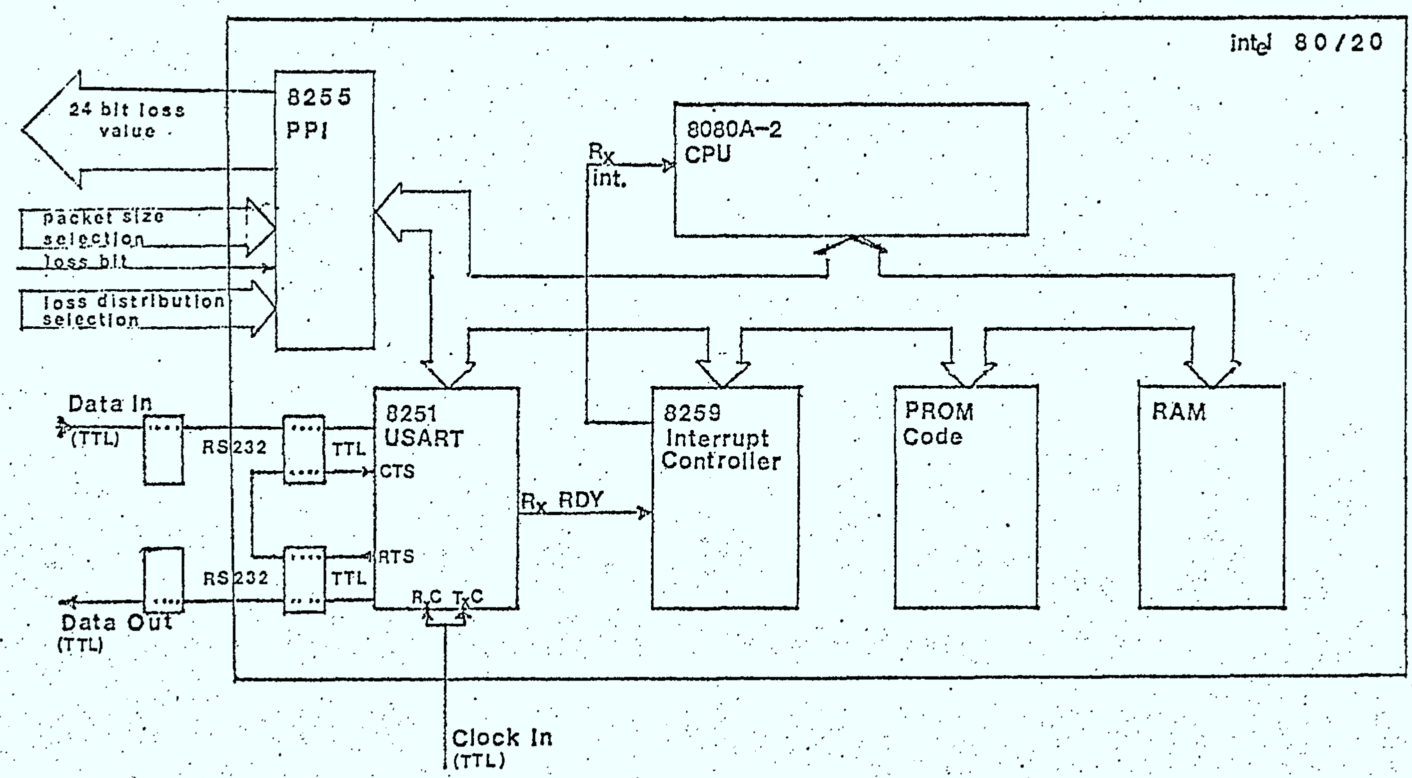


Figure 5.2 **Processor Configuration**

that represents the channel between the transmitter and receiver, before reconstructed by the CVSD decoder. The bit stream is grouped into packets of a preselected size and handling of such packets depends on an external loss bit (see Figure 5.3).

Packet size can be varied from 1 to 256 bytes. Packet loss decision bit is generated from a 24-bit psuedo-random generator and a comparator. Packet % loss can be adjusted to values from 0 to 31% with a step of 1%. Figures 5.4 to 5.6 show the packet size/loss hardware configuration.

The loss distribution hardware was tested using a stream of 1000 samples. First, the packet size was fixed to one byte/packet and different % loss was selected. Table 5.1 shows the different settings and the corresponding measured loss.

Table 5.1

Loss Setting	0	10	20	30	40	50	60
Measured Value	0	11.5	20.6	30	41.1	50.4	60.2

When a fixed loss of 50% is selected and the packet length is changed, Table 5.2 shows that the alteration of the packet size does not affect the loss distribution.

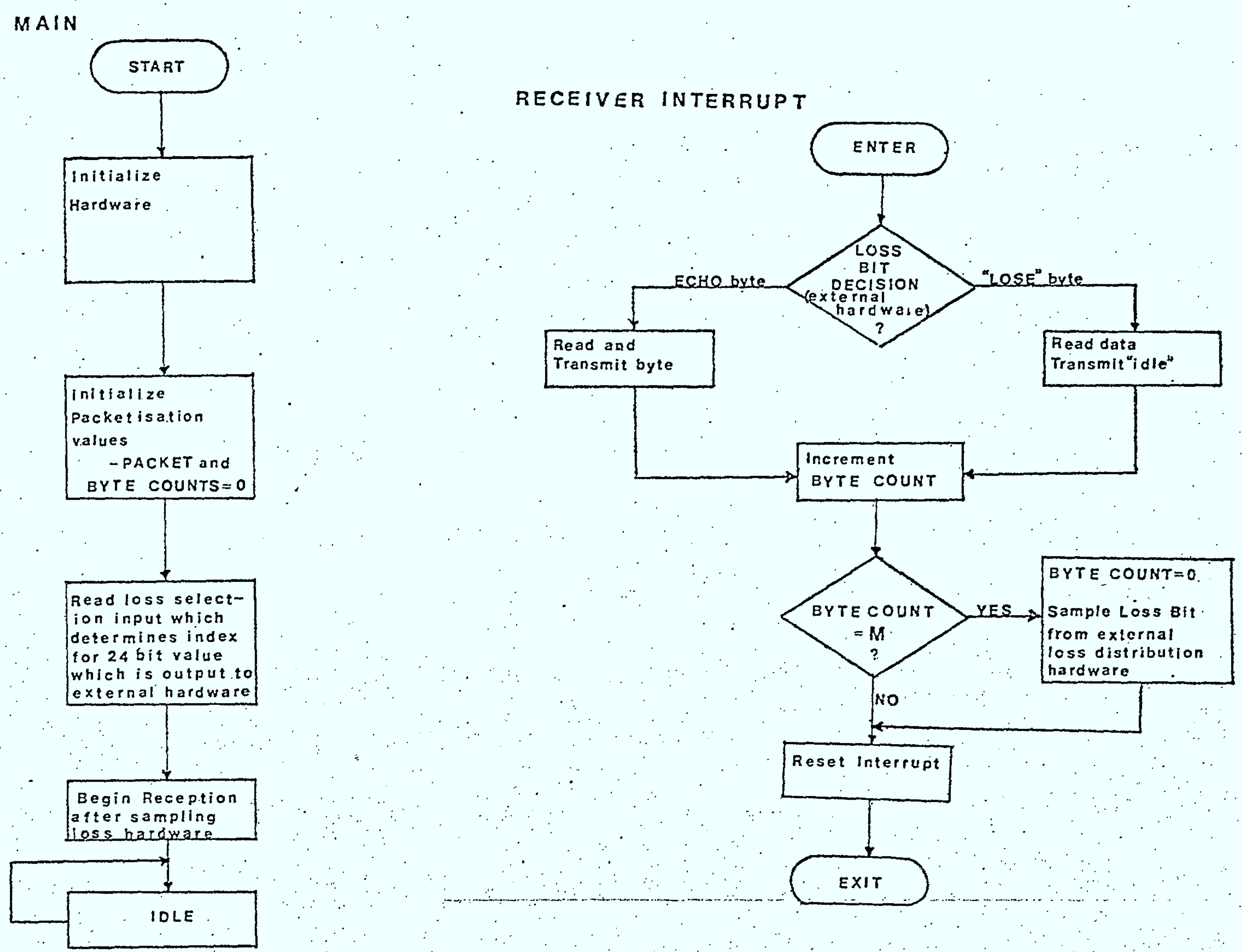


Figure 5.3 Program Flowchart

Figure 5.4 Packet Size/Loss Analysis Hardware

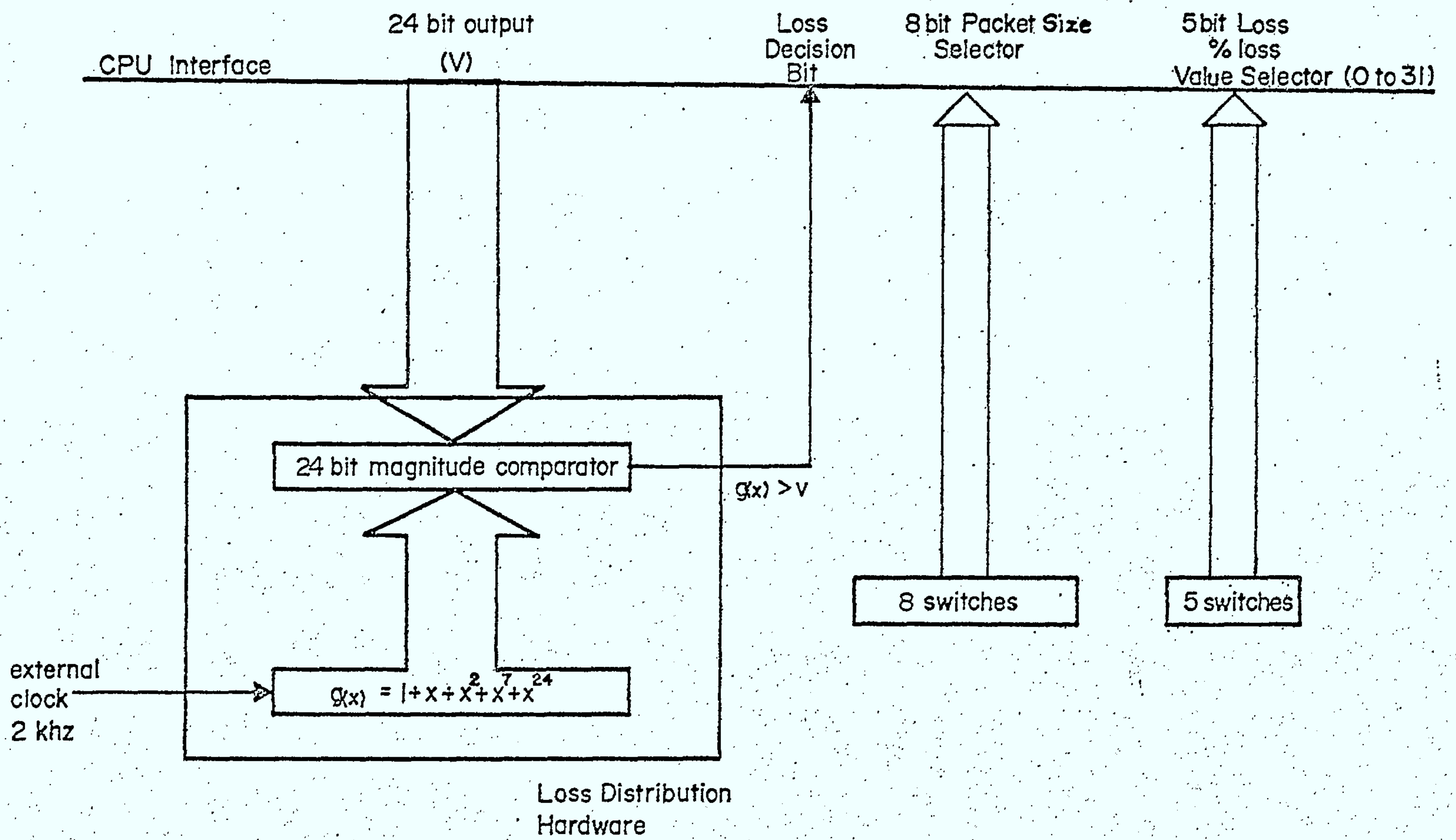
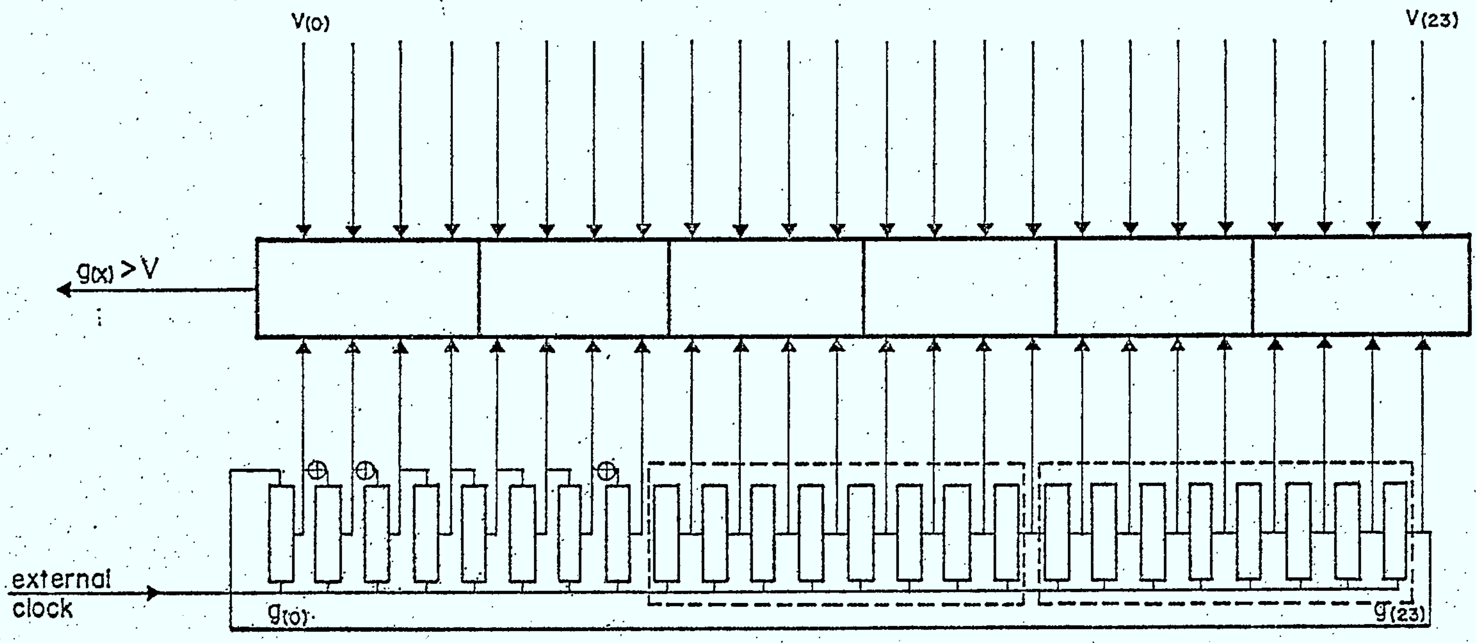


Figure 5.5 <u>Loss Distribution Hardware</u>



24 bits from CPU (V)

6 4bit comparators

$g(x) = 1 + x + x^2 + x^7 + x^{24}$

8 D-flip-flops

2 8bit shift registers

3 XOR gates

Figure 5.6 (a) Pseudo-Random Circuit $g_{(x)} = 1 + x + x^2 + x^7 + x^{24}$

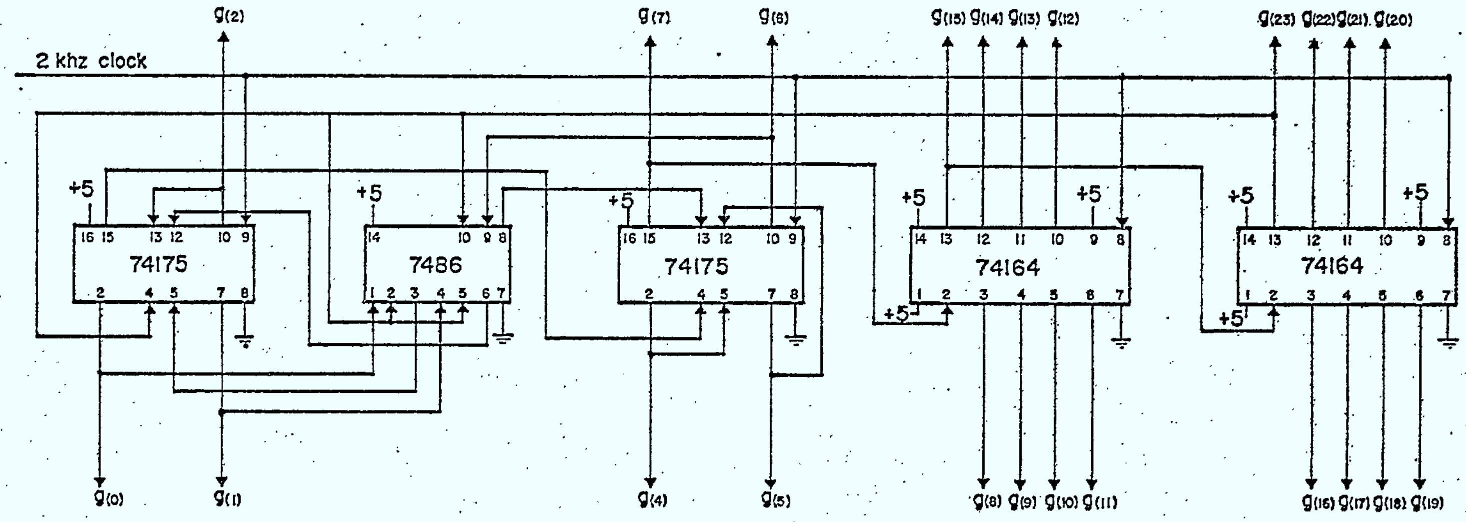


Figure 5.6 (b) Loss Distribution Comparison $g_{(x)} > V_{(x)}$

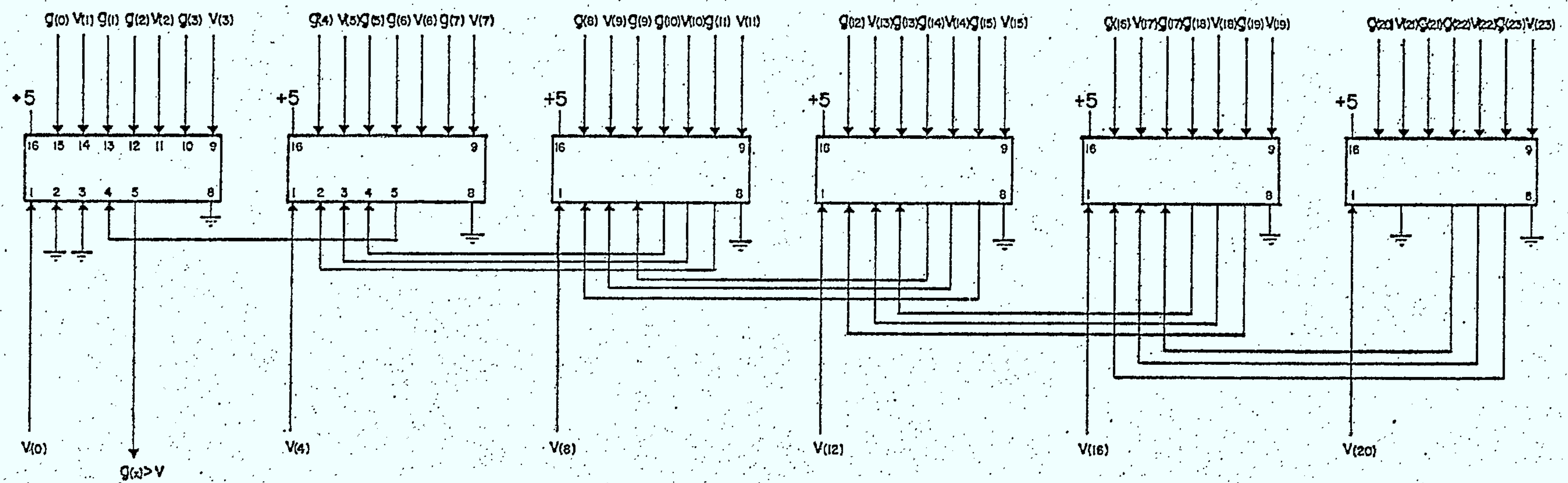


Table 5.2

Packet Size (bytes)	2	4	8	16	32	64	128	256
Measured Loss	49.8	49.1	51.6	50.5	48	51.1	50.4	50.3

5.3 Speech Intelligibility Tests

Two main tests have been performed to "measure" digital voice intelligibility corresponding to different packet loss rates and sizes.

In the first test the attributes considered are full context comprehension and individual word intelligibility. The following ratings are considered:

A full content easily followed

B full content generally followed

C full content spotily followed

D full content not followed

3 individual words easily followed

2 individual words generally followed

1 missed many individual words

0 missed most or all words

Table 5.3 shows the results of a sample of tests using these ratings. Input to the system is through a microphone and output is through a speaker.

Table 5.3

bytes /packet \ % loss	0	4	8	12	16	20	24	28
1	A3	A3	A2	B2	B2	C2	D1	D0
2	A3	A3	B2	B2	C2	C1	C1	D1
4	A3	B3	B2	C1	C1	C1	D1	D0
8	A3	B2	C2	C1	D1	D0		
16	A3	B2	C2	D1	D0			
17	A3	B2	C2	C1	D0			
32	A3	C1	C1	C1	C0	D0		
33	A3	B2	C1	C0	D0			
34	A3	C1	C1	C0	D0			
64	A3	B2	C1	C1	D0			
65	A3	B2	C1	C0	D0			
66	A3	B2	C1	C0	D0			
67	A3	B2	C1	C0	C0	D0		
128	A3	B2	C2	C1	C1	D1	D0	
129	A3	B2	B2	C1	C1	D0		
130	A3	B2	C2	C1	C0	D0		
131	A3	B2	B2	C1	C1	D0		
254	A3	A2	B2	C1	C1	C1	D0	
255	A3	A3	B2	C1	C1	C1	D0	
256	A3	A3	B2	C1	C1	D0		

In the second test the cassette tape recorder is used as voice source and voice is transmitted through a microphone to the encoder. Two tapes are used, each side has a sample of words, alphanumeric codes, sentences and a short text. Another tape is used for demonstration to show the reference quality of the transmitted voice and to familiarize the subjects with the system. The contents of each tape were registered by different speakers and in different order. The vocabulary chosen (see Appendix) is a combination of monosyllables as well as multisyllable words. Furthermore, some fairly close sounding words were included. Texts and alphanumeric codes were chosen representing different subjects and disciplines. As such, the vocabulary has a moderate degree of difficulty.

The vocabulary was read in a normal environment using a commercial quality microphone. No attempt has been made either to eliminate sources of ambient noise or to compensate for acoustic variations due to different speakers.

Each subject uses an evaluation sheet (see Figure 5.7) for each tape side and evaluation will be as follows:

a) Text and sentence quality are evaluated using a five point scale:

high quality

Figure 5.7 Evaluation Sheet

Name: Tape:

Date: Side:

		High	Good	Fair	Poor	Unsatisfactory
Paragraph*	1					
	2					
	3					
Sentences*	1					
	2					
	3					
	4					
	5					
	6					
	7					
	8					
	9					
	10					
	11					

Words+	Alphanumeric+

*Place a mark on the corresponding quality of the transmitted voice.

\+ Print the transmitted word or alphanumeric code.

good quality

fair quality

poor quality

unsatisfactory

If the subject finds that a five point scale is too coarse, he may place marks next to two categories, producing ratings such as "fair-good" or "poor-fair".

b) Words and alphanumeric codes are to be printed in order.

Words, alphanumeric codes and sentences are registered with 2 second pause periods between them. In addition, a five second pause is left between groups of five words or alphanumeric codes to facilitate printing. The subject is using an earphone attached to both ears and listens to the four tapes in one session.

Table 5.4 shows the results of the intelligibility test. The numerical rating scale was derived by assigning scores 9, 7, 5, 3, 1 to the five categories high to unsatisfactory. In addition the percentage of correct recognition of words and alphanumeric codes is given in Table 5.5.

Table 5.4

% loss bytes/packet	0	2	4	8	12	16	20	24	28
2	9	8	7	7	5	5	3	1	1
4	9	8	8	7	5	3	1	1	1
8	9	7	7	5	3	1	1		
16	9	7	7	5	5	3	3	1	1
32	9	9	7	5	5	5	3	3	1
64	9	9	8	7	7	5	5	3	3
128	9	9	9	7	7	7	5	3	3
256	9	9	9	8	7	7	5	5	3

Table 5.5

bytes/packet % loss	2	4	8	16	32	64	124	256
2	0.79	0.77	0.76	0.83	0.9	0.96	0.98	0.98
4	0.88	0.86	0.77	0.8	0.85	0.87	0.92	0.98

5.4 Discussion of Results

The results show nonlinear variations in the relations between voice intelligibility and packet size/loss tuples. This indicates that for a certain packet percent loss there exists an optimum packet size that produces a maximum level of intelligibility.

The characteristics of the codec chip used explains why the intelligibility scores exhibit significant variations w.r.t. packet size/loss values. When a loss occurs continuity is lost in the data stream and the waveform reconstruction loses step with encoder. Once the transmission is resumed then synchronization is re-established but a shift in the waveform is resulted. Such shift is affected by the packet length.

Results suggest the reassessment of the questions of the appropriate number of speakers and subjects, the basis for their selection, the testing equipment, and the testing environment.

6. Conclusions

Developments in VLSI have made the widespread use of digitally controlled radio dispatch systems technically feasible and economically attractive. The resulting pressure of demand on a limited spectrum requires that it be used as efficiently as possible. One approach to this goal is the combination of both speech and data over a common channel. A second involves the investigation of channel access methods. This report considers such access methods and their application to bursty data systems typical of speech and real time database query systems.

The nature of mobile data communications is such that transmission from source to receiver takes place over two hops. The central node is a fixed base station which controls data/voice traffic and acts as a repeater. Over both the uplink and downlink paths, user pairs share a channel pool. A given channel is acquired by a transmitter only during actual transmission of a talkspurt or data burst; upon completion of this event, the channel is relinquished, to be made available to another user.

A variety of channel access methods has been outlined, and it has been shown that the number of channels necessary for n users (who sustain 0.5n two-way conversations) can approach 0.5n as a lower limit. This number yields a 50%

saving over the n channels necessary if no channel sharing takes place. Such a saving is obtained at the price of added complexity and signalling overhead, together with queueing delay and a non-zero probability of blocking.

Pooled channels may be placed in groups, with N_g channels per group. An important question is that of determining the most suitble value for N_g. On the uplink path, it has been shown that, under light traffic conditions, a large value of N_g yields lower blocking probability but longer delay than a small N_g. This result appears reasonable; the greater blocking associated with small N_g reduces throughput, so that delay is reduced for those messages which are accepted. At higher traffic levels (ρ>0.65), however, a surprising phenomenon is observed. Fairly small groups (N_g=8) yield superior performance, both in terms of blocking probability and delay, to that of a single large pool of channels. Further investigation appears necessary to determine whether this effect is a peculiarity of the analytic model used, or is indeed a genuine characteristic of the random access sequential search procedure.

No such phenomenon is present in the downlink channel, where best performance is obtained with large channel groups.

It has been shown that the use of a separate signalling channel may reduce delay, but will decrease the efficiency of channel usage. Because a separate signalling channel always requires downlink receiver channel switching, it appears to be less efficient, even for many downlink channels, than the use of the quasi-dynamic allocation scheme. The latter avoids the use of a signalling channel by the assignment of a nominal "listening" channel to each receiver; receivers are directed to an alternate quiescent channel if the nominal one is busy at the time of a message arrival.

Speech packets are not subject to acknowledgement or retransmission, owing to the requirement for real time processing and transmission. UHF mobile radio transmission is subject to rapid fading, however. It follows that a significant voice packet loss rate must be expected. To estimate the effect of loss rate on intelligibility, recorded voice samples were packetized and subjected to controlled random loss. The results, evaluated by a variety of subjects, indicate that the optimum packet length is a function of loss rate. It is suspected that this phenomenon may be in part an artefact associated with the particular codec chip used for the experiments. Further testing with one or more different signal coding schemes is recommended.

The research carried out in this project has established that considerable bandwidth savings can be effected in mobile data/voice communication by means of suitable channel sharing. The resulting levels of delay and blocking probability appear to be acceptable. Initial experiments indicate that packet length can be optimized to maintain adequate intelligibility in a rapidly fading communications environment. Further simulation and voice testing is recommended prior to a complete system emulation.

References

[1] H. Frank and I. Gitman, "Economic analysis of integrated voice and data networks", Proc. IEEE. 1978, 66, pp.1549-1570.

[2] R.V. Cox and R. Crochiere, "Multiple User Variable Rate Coding for TASI and Packet Transmission Systems", IEEE Transactons on Communications, Vol.COM-28, No.3, pp.344-344, March 1980.

[3] D. Minoli, "Issues in packet voice communication", Proc. IEEE, Vol.126, No.8, pp.729-740, August 1979.

[4] J. W. Forgie and A. G. Nemeth, "An efficient packtized voice/data networks using statistical flow control", ICC 1977, pp.38.2-44 to 38.2-48.

[5] J. C. Ellershaw, "A Radio Telephone System Using Packet Radio", Proceedings of the IREE, pp.16-24, March 1979.

[6] J. S. DaSilva, "Performance Analysis of Land Mobile Packet System", Ph.D. dissertation, Department of Systems and Computer Engineering, Carleton University, Ottawa, July 1980.

[7] D. J. Goodman, J. S. Goodman and M. Chan, "Intelligibility and Ratings of Digitally Coded Speech", IEEE Trans. Acoustics, Speech and Processing, Vol.ASSP-26, No.5, October 1978, pp.403-409.

[8] B. G. Haskell and R. Steele, "Audio and Video Bit-Rate Reduction", Proc. IEEE, Vol.69, No.2, February 1981, pp.252-262.

[9] C. P. Smith, "Talker variance and phonetic feature variance in diagnostic intelligibility scores for digital voice communications processors", IEEE ICASSP 79, Washington, D.C., April 1975.

APPENDIX

Tape Contents

Contents of Demonstration Tape

Computer security is initially concerned with determining and implementing cost-effective counter-measures to make a system secure against the many threats which can occur. It is concerned, therefore, with reducing the frequency with which any threat (other than trivial threats) is expected to occur and/or reducing the impact upon, or the degree of disruption to, the correct functioning of the system which the occurence of the threat causes. Secondly, it is concerned with what has to be done when the normal mode of operation of the computer system is disrupted, whatever the elapsed time of the period of disruption may be. It is concerned, therefore, with contingency planning, ie with the preparation and execution of plans for the introduction of a standby mode of operation, and with the preparation and execution of recovery plans for the restoration, subsequently, of the normal mode of operation. Finally, computer security is concerned with the auditing of the system during both normal and standby modes of operation.

Contents of Tape 1 Side A

Before addressing this issue, however, it's well to examine the impact of the new technologies in more detail, for there's much more involved here than just the development of a better typing machine. Perhaps the most far-reaching implications stem from the union of computers and telecommunications. It's the prospect of computers talking to each other - or, more to the point, humans talking to each other through computers - that has created all the excitement, in both the positive and the negative sense.

We are only beginning to come to grips with the communications potential of computers; indeed, CNCP's Glen Woodin notes that, as yet, many people who use word processors are completely unaware of their communications features. But, inexorably, today's office - with a typewriter here, a telephone there, a photocopier down the hall, a filing cabinet around the corner - is vanishing and the "integrated" office is emerging in its place.

This is not the right definition.
Is there a draft in here?
Please speak up when addressed.
This could go on all day.
He put the cat out with the garbage.
The doctor has her own practice in town.
The day before yesterday was just as sunny.
Can Johnny come out and play?
We dine at eight, dress is optional.
I can see myself in this shine.
Now you may relax as the worst is over.

Yoga	Many	Factory	Period	Khomainy	Last
Enormous	On	University	Complexity	Grasp	Suddenly
Junior	World	Order	Impact	Zenith	Helical
Nutcrackers	Mad	Enough	Long	Human	Indian
Zeller	Few	Music	Voice	Kettle	Between
Joint	Queen	Screen	Productivity	Telephone	Becomes
National	As	Different	Adequate	Umbrella	Crystal
Tour	Weeping	Does	Grid	Yellow	Quantity
Theatre	Zig Zag	Vessel	Xerox	Cable	Butterfly
Examination	Mud	Diet	Road	Guitar	Zodiac

TUV 573	881 04 39	Interdata 7 32	12:00 noon	100 m.grams.
VAX 11 780	K2C 5M3	ME 351	RSX 11M	15°F.
567 2253	3.5 Meters	2nd Floor	Room C460	60 Cycles/sec.
72 cm. Mercury	18 Minutes	VRO 45	1.2 Amp.	GV 848.4 U6
VHF	April 4th	3 Lines	CONTAC 12	J580 938 TC

Contents of Tape 2 Side B

We have finally tracked down your records.

They should have tried it on the rats first.

I have a headache.

The rabbit has died.

It is over there on the far wall.

The bread will not rise if the yeast is cold.

The paper was folded in the middle of a story.

When crossing the street you should look both ways.

There was a fault in the design.

Wherever you look there is always some beauty to see.

It is a serious offence to make a false return.

Over the past several years the student movement in Canada has been focusing its energy on the fight against government restraint and the effects that this policy has had on access to Canadian colleges and universities. The fight has not been easy nor has it brought dramatic results. Yet despite the problems of uniting against the educational policy of 11 different governments, students have been able to press the governments and win concessions.

Through the 1970's students have become increasingly more effective in exposing the effects of government policy in the area of post-secondary education and bring the debate more into the public eye. Students have won changes to the student aid plans in the provinces and at the federal level. We have been able to win temporary freezes in tuition fee increases and wage increases in provincial government employment programs.

PDP 11	7.5 lbs	CBF FM 100.9	LADA 1300	K1S 5B6
AF 305	XYB 146	224 79 38	11:15 AM	325 Miles
64 Miles/h.	220 volts	SAFT 93	L Shape	September 2nd
14 57	176 cm.	Extension 359	8 Tracks	110 A.C.
PS 65K10	DL1	CP Air	U.C.L.A.	KE 204.0 0503

Typewriter	Youth	Polluted	Ziegler	Kill	Back
Revolutionary	Following	Technology	New	Two	Guard
Society	Useless	Much	Jaguar	Throughout	Lead
Information	Kaiser	Distances	Old	Fast	Chain
Depending	Queue	Will	Home	Onion	Data
Lion	Nation	An	Jacket	Ever	Simple
Pantomine	Distortion	Identity	Economy	Privacy	Viking
After	Heavy	Broadcasting	Young	Cold	Interface
Scheme	Wagon	Mozart	Vapour	Government	Ugly
Quality	Zero	X Rays	Insect	Eleplant	Nancy

Only the grey ape would behave as they do.

He has not eaten.

They are very like me.

If you fall, simply swim back.

Choose the day for your first try very carefully.

Safety is the major factor to consider.

And darkness was upon the face of the deep.

Why are we here?

Some trees grew in the centre of the clearing.

It seemed just another morning.

I have been here just three weeks.

As computers become better understood and more economical, every day brings new applications. Many of these new applications involve both storing information and simultaneous use by several individuals. The key concern in this paper is multiple use. For those applications in which all users should not have identical authority, some scheme is needed to ensure that the computer system implements the desired authority structure.

For example, in an airline seat reservation system, a reservation agent might have authority to make reservations and to cancel reservations for people whose names he can supply. A flight boarding agent might have the additional authority to print out the list of all passengers who hold reservations on the flights for which he is responsible. The airline might wish to withhold from the reservation agent the authority to print out a list of reservations, so as to be sure that a request for a passenger list from a law enforcement agency is reviewed by the correct level of management.

RSY 329	1120 Guilford	T 1600	8:30 PM	K2C 3L6
QA 76.9 A25	1500 R P M	MEK 6800 D2	46 Minutes	GPSS/360
A4123 68434	12 Volts DC	450 Mhz.	Datsun 210	1300 CC
231 5578	900 McLeod	75 Km./h.	150 Dollars	16 K bits/sec.
23rd of June	8 channels	HF 307	19 81	CP 6

Threshold	Law	Highway	Structure	File	Meadows
Narrow	Optical	Jacqueline	Ambulance	Parklane	Hills
Refinery	Psychiatric	Preempt	Crossing	Breakdown	Queueing
Newspaper	Facsimile	Warehouse	Unified	Lattice	Histogram
Initialize	Event	Grouping	Shark	Emergency	Resource
Heading	Company	Trouble	Distribution	Priority	Conditioning
Drive In	View	Insurance	Switchover	Downtown	Movie
Triangular	Woods	Statistical	Jockeying	Freeing	Detect
Attributes	Kodak	Acid	Ziess	Buffer	Yes
Opera	Gold	Quebec	Kidney	X Axis	Usable

Contents of Tape 2 Side B

"Women are lucky today! They have the opportunity to choose between being housewives, mothers, or career women. Finally, they have achieved what they always wanted: their liberation!"

This is a statement which is innocently made by both men and women quite frequently. Thus, most of us come to the conclusion that "all's well, ends well."

The problem is that all is not well, and all has not ended well. No doubt, we have gone over the first hurdle and we have made some progress. But progress has also been slow and difficult. Most of our achievements have been in solemnly proclaimed conventions rather than observable changes. "Equal pay for equal work, irrespective of sex" is merely one of the cynically non-observable conventional examples.

Do you need a set of knives?

I think he wants to be friends.

Aren't you going to phone the airport?

This is to the best of my knowledge and belief.

Do you want a receipt?

All the clocks have stopped.

Are you sure that you are comfortable?

This is my first blind date.

I see you've fixed the drip.

He had to build it by hand.

It will go away if you don't bother it.

BHO 440	AG 105 W2	CP 5	80 CENTS	75 CH 1050
UHF	IBM 11 30	17 54	120 Grams	TV 10
941 3624	Boeing 707	April 27	K2K 2A6	COS 310
Zenith 37670	RCA 20"	57 Km.	FIAT 128	-32° C.
112 Kent St.	FORTRAN 77	Size 42	ME 375 K	6:45 am

Scarf	Zurich	Undergraduate	River	Keep	People
Olympic	Change	Frequency	Needle	Inventions	Quarter
Effect	Book	Links	Together	Italic	Over
Maritime	Form	Hot	Quantities	Open	Cost
Justice	Picture	Miracle	Great	Normal	In
Dictionary	King	Culture	Women	Lemonade	Ball
Talent	Growth	White	At	Have	Trend
Yougart	Diffusion	Voluntary	Ecology	Yankee	Fuel
Automation	Zahia	Without	Savoy	Utilities	X Y Plane
Frost	Dragon	Honey	Magic	Education	Organ